

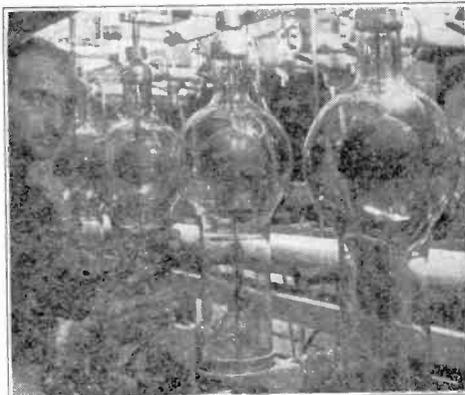
# INTRODUCING YOU TO RADIO

## Your Future in Radio

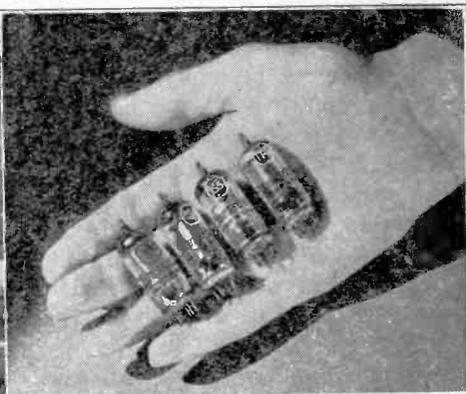
**D**ECIDING to study the N.R.I. Radio Course is one of the wisest decisions of your life. Why? Because this radio Course has been planned and written from beginning to end especially for men who must do their studying at home, usually after their

regular day's work. You will learn radio with the aid of short-cut teaching techniques—not ordinary classroom methods, but specially-developed, thoroughly tested home-study methods. You will obtain a complete basic radio training without wasting your time in the study of unnecessary material.

diately usable, because in the very first lessons you begin preparing for practical everyday radio jobs. The amount of education you now have is of less importance than an unswerving ambition to succeed. The first lesson starts right at the begin-



Courtesy Westinghouse Elec. & Mfg. Co.



Courtesy Emerson Radio & Phonograph Corp.

In this very first lesson of your N.R.I. Course, you will learn the fundamental manner in which all radio tubes operate. This means that after mastering this first lesson, you will know how electrons flow through the giant high-voltage radio tubes (at the left) of station KDKA's transmitter, and you will be just as familiar with the behavior of electrons in the four tiny radio receiver tubes shown at the right.

regular day's work. You will learn radio with the aid of short-cut teaching techniques—not ordinary classroom methods, but specially-developed, thoroughly tested home-study methods. You will obtain a complete basic radio training without wasting your time in the study of unnecessary material.

You're going to get real results from this Course sooner than you expect. The fundamental radio ideas and basic testing methods which you master one after another are almost imme-

ning of radio knowledge, and prepares you for the more advanced second lesson. Each new technical or special radio word, each symbol, and each abbreviation is explained the first time it appears.

Each lesson is full of dollars-and-cents information which is vital to real mastery of radio. By the time you finish the last lesson, you just can't help but be worth a whole lot more money to the world than you are today.

*The surest way not to fail is to determine to succeed.* Make a sincere vow right now that you are going to follow up your hunch about radio—that you are going to finish the N.R.I. Course and succeed in radio. Give this world-famous Course just a fifty-fifty chance, and it'll help you just as it has helped so many thousands of other men. Stick to your vow, and you won't have to worry any more about your future.

If you have faith in yourself, if you sincerely believe that you have the ability to master radio, and never waver from that belief, your battle is half won already! The N.R.I. Course will do the rest if you give it a chance to work for you. We have actual records to prove that it has brought real success to thousands of other men who had the will-power to finish it. Surely you consider yourself just as good in every respect as these men. In fact, you are probably a lot more ambitious, a lot more in earnest than some of them. If you are as good as these men, you can succeed, too.

About a hundred years ago, Abraham Lincoln said: "*I will study and prepare myself, and some day I know my chance will come.*" These words are even more true today than they were in Lincoln's time, because there are more opportunities now and they come much faster.

In radio, for instance, there are hundreds of different job opportunities in just that one branch known as entertainment broadcasting, which deals with the sending and receiving of programs through space. You will learn about some of these jobs in the very next section of this lesson.

As an N.R.I. graduate, you will be in a position to choose your opportunity, instead of waiting for it to come to you. You can decide for yourself whether you want an operating job,

whether you want to do radio receiver servicing, or whether you want to play a part in building the great transmitting stations or the millions of receivers which serve this fast-growing broadcasting industry.

Furthermore, as the recently-developed frequency modulation system of broadcasting spreads throughout the country, the existing job opportunities will increase correspondingly in number. In addition to all this, commercial television is now a reality, and will be seeking trained men in increasingly greater numbers each year. Truly, the career of radio offers a promising future to men like you!

But a still more amazing fact about a radio career is that there is an even greater variety of opportunities for you outside of the entertainment broadcasting field. In short-wave radiotelephone and radiotelegraph communication systems on land, on the sea, and in the air are thousands of fascinating good-pay jobs for men who have the complete basic radio training that you are now obtaining.

In addition to all this, there are still more job opportunities for you in fields beyond the sending and receiving of messages or programs by radio.

You may some day find yourself in a theater, installing, adjusting and repairing the electric eyes and amplifier systems which make modern movies talk.

You may go to ball parks and conventions, to take charge of public address equipment which boosts man's feeble voice as much as a million times.

You may hike through canyons and foothills, using modern radio and electrical prospecting equipment to locate new oil fields and new deposits of precious minerals.

Wherever there is equipment which uses radio tubes, radio parts, radio

principles and radio servicing methods, there also are opportunities for men with the thorough radio training you will soon have.

You are going to see great things happen in radio. You are going to see breath-taking new developments which far outshadow even the recent miracles of color television and noise-free broadcasting. You will be prepared for the today undreamed-of jobs which are created by new developments, because your N.R.I. training is built upon a sound foundation of fundamental radio and electrical facts.

In a way, you are to be envied, because as a graduate of this N.R.I. Course you will *start* your career in radio with knowledge which it has taken others almost a lifetime to acquire. Yes, one of the great things about the N.R.I. method of home training is that it gives you in a very few months even more up-to-date knowledge and information than many "old timers" have when they retire from radio!

Now you're ready to GO—ready to start your study of radio.



### Radio Maps Will Guide You

At regular intervals in the N.R.I. Course, you will be given an outline of what comes next. Each outline is really a "radio map" of part of the Course, because it shows how each of the study subjects fits into your complete radio training. You will use these outlines as guides for getting

your radio training as fast as possible, just as you would use a road map to guide you to a distant town over the shortest possible route.

**Map of a Radio System.** In this lesson, you start right in to learn about electricity, magnetism, radio tubes, and transformers—things you will work with on radio jobs every day. These subjects are so important that they deserve a *radio map* all their own, just for this lesson.

This first radio map takes you on a quick imaginary trip through the complete broadcasting system pictured in Fig. 1. You find out where radio tubes and transformers are used in a modern radio system, and discover that electricity and magnetism have highly important effects in each one of the six parts of the system. Also, you will begin learning about important good-pay radio jobs for which you prepare yourself.

Once you have become familiar with the practical uses for electricity, magnetism, tubes and transformers, you will be eager to tackle the rest of this lesson and learn more about each one of these subjects.

**1. Microphone in Studio.** Our flying trip through a radio system starts in the broadcasting studio, where the sounds which make up a radio program are produced. Sounds are mighty important to a radio man, because he has to make sure that the sounds which come out of loudspeakers in homes are as nearly as possible like the sounds which enter the microphone at the studio.

Without the aid of radio, ordinary sound waves can travel only a few hundred feet at the most. But radio changes feet into miles, and allows you to hear sounds which are hundreds or even thousands of miles away without using wires. Not many years ago this would have been called magic, but

today people think nothing about hearing sounds which are halfway around the world. All this is possible simply because radio men know how to put electricity and magnetism to work!

The microphone is really the doorway of our broadcasting system. Sound waves enter it through a wire screen, and make a thin piece of metal move back and forth or *vibrate*. In the wires of this microphone are millions and millions of tiny particles called *electrons*. Magnetism gets into action and makes these electrons move back and forth in step with the vibrations of the thin metal piece.

Thus, the microphone converts sound waves into a varying electron movement. This electrical signal is called an audio signal and is actually *sound in an electrical form*. In other words, audio signals are *electrical equivalents of sounds*. (The word "audio" comes from a Latin word meaning "to hear," and is used by radio men to describe an electrical signal having characteristics of sound.)

The microphone feeds its audio signal into a *cable*, which consists of two insulated wires surrounded by braided metal which is covered with rubber.

**2. Control Room.** The audio signal which is produced by the microphone travels through the microphone cable to an *audio amplifier* in the control room. This audio amplifier consists of radio tubes, transformers and other parts mounted inside a cabinet and connected together in such a way that when a weak signal is fed in, the amplifier produces a new signal which is much stronger but still has *the same characteristics* as the weak signal.

An audio amplifier is thus a signal-booster which takes in the weak audio signal of the microphone, and puts out a new signal which is hundreds of times stronger—strong enough for the

long trip over telephone lines to the transmitter.

To be technically correct, however, we must point out that signal-boosting is accomplished by radio tubes which are capable of making *more electrons* move in step with the original sound. Furthermore, it is *magnetism* which does the work of transferring the signal from one tube to the next through transformers.

The man who adjusts the knobs at the front of the audio amplifier while watching the radio artists through a large plate-glass window and listening to their program with headphones is the *monitor operator*. Chatting with him for a few minutes, we learn that some announcers talk too loud for radio purposes, while others barely speak above a whisper. All speakers must sound just about the same in loudness to radio listeners in homes, so this monitor operator turns up the volume (makes the audio amplifier do more boosting) for a weak-voiced speaker, and cuts down the volume when any one starts shouting in front of the microphone.

Excessively loud sounds can overload a transmitter and cause it to go "off the air" for a few moments. To avoid this the monitor operator must turn down his volume control *before* a loud musical passage, a cough, or any other loud sound. Some operators become so skilled at figuring out when loud sounds will occur that they are jokingly called mind-readers!

**3. Transmitter.** Leaving the control room, we follow our radio program (now in the form of a strengthened audio signal) over a telephone line to the transmitter building which is usually located a few miles outside the city.

Once inside the transmitter, we find that our audio signal gets another

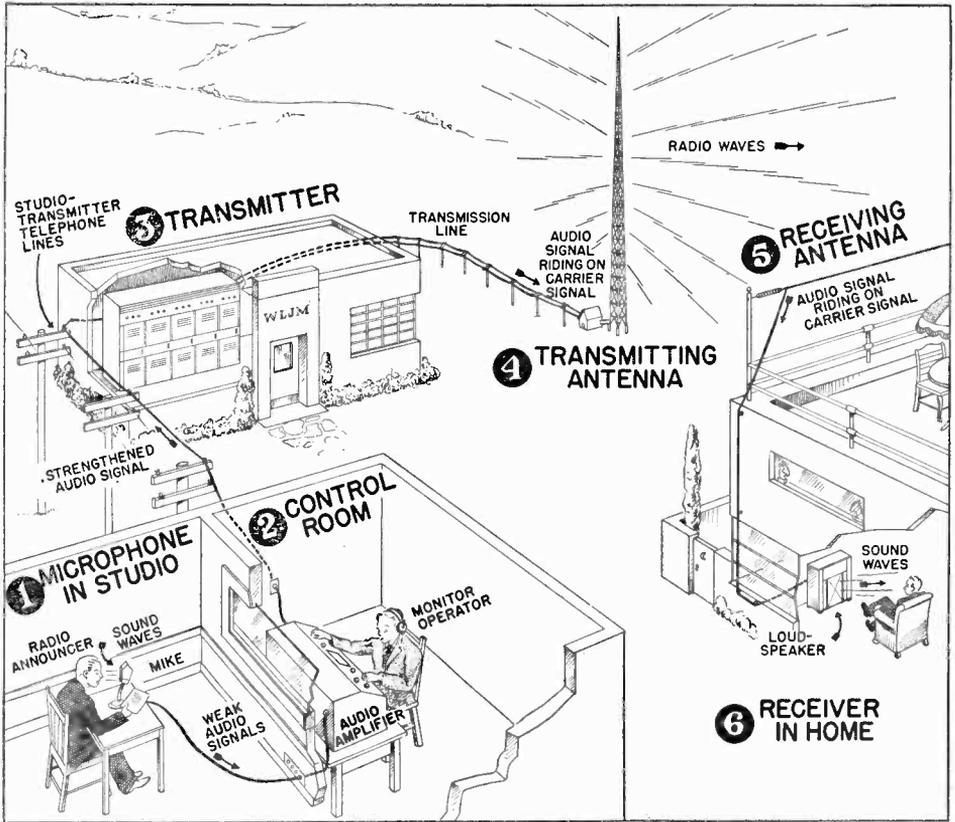


FIG. 1. This first radio map shows the six important sections of a radio broadcasting system. Start with the studio microphone at 1 and trace the signals until you reach the receiver loudspeaker at 6.

tremendous boost in strength from tubes and transformers. But no matter how much we boosted this audio signal, we wouldn't be able to make it travel more than a few hundred feet through space at the most.

We have to carry our audio signal through space on another signal known as the *carrier signal*. This carrier signal is the true *radio signal*.

Each station is assigned a carrier signal of its own by the Government. The carrier signal is produced by more radio tube circuits, and is combined with the audio signal in a radio tube circuit designed especially for "mixing" purposes.

The audio signal, riding on its own

radio carrier, is given a final boost in strength by huge power tubes, and is then fed to the transmission line which serves the transmitting antenna. There's plenty of power behind our signal now—enough to fry to a crisp any unlucky mouse or grasshopper who gets across the high-voltage tube terminals.

The radio operator on duty at the transmitter must see to it that the radio signal which the transmitter is sending out into space has the same form at all times as the original sounds in the studio.

In addition, the operator must check meter readings in the transmitter regularly and try to anticipate possible

breakdowns of tubes and parts. He can then replace weakened parts after the station shuts down for the day.

The goal of an operator is to keep his station on the air every second of the scheduled operating time. This is not as difficult a job as it seems, for an operator who fully understands the simple fundamental characteristics of each radio part and circuit in the transmitter rapidly acquires an uncanny ability to tell when a particular part is failing and in need of attention or replacement.

**4. Transmitting Antenna.** Coming out of the transmitter building, we hike along the transmission line to a little house just under the transmitting antenna tower. Technically speaking, we should call this the *tuning house*, but since you plan to be a practical radio man, you might as well get accustomed to hearing it called the "*dog house*." Every transmitter has its dog house in the shadow of the antenna tower, and nearly every radio operator has at one time wisecracked about "leading a dog's life" when he had to go out to the dog house in stormy weather to read the meters.

The dog house contains radio parts which can be adjusted so as to "tune" the tower to the assigned carrier signal of the station. The antenna then gets all the signal power which the transmitter can feed to it through the transmission line. The meters in the dog house tell "how the station's doin'," and must be read at regular intervals each day.

From the tuning house, our radio signal goes right up the giant antenna tower, to the very top. The signal still consists of electrons moving back and forth, back and forth, the entire length of the tower, and these electron movements through a tower in the sky produce the radio waves which travel off into space away from the tower.

Our radio program is now being carried out through space in all directions by radio waves.

You will learn in the next lesson that a radio wave is just a combination of electric and magnetic effects, varying in strength from instant to instant just as the pressure of the original sound wave at the studio is varying.

**5. Receiving Antenna.** Through the sky we trace the radio waves to the receiving antenna at some far-off home. As these waves move past the receiving antenna wire, the millions of electrons which are *always* in a metal wire start moving back and forth in exactly the same way as did the electrons in the transmitting antenna tower. The receiving antenna thus changes our radio waves back into an audio signal riding on a carrier signal. It is like the signal we had at the output of the transmitter, except that the signal is millions of times weaker due to its trip through space.

**6. Receiver in Home.** We follow our signal down the antenna lead-in wire, and we arrive at the radio receiver in the home.

We find that the first few sections of the receiver build up the strength of this weak signal, step by step. The carrier signal, no longer needed, is next removed from the audio signal by a section called the *detector*. After this, the audio signal is boosted in strength by more radio tube circuits in the *audio amplifier* section. This section feeds our signal into the very last transformer in the receiver, called the *output transformer*.

Finally, we come to the all-important loudspeaker, which is connected to the output transformer. The loudspeaker uses both electricity and magnetism to change the greatly strengthened audio signal back into sounds.

Radio servicemen are highly important in connection with this last section of our broadcasting system, for these men are needed to keep the millions of receivers in this country in operating condition. They have to

locate and replace defective parts, and make correct adjustments of receivers whenever necessary, so the sounds coming out of the loudspeakers will be as nearly as possible like the original sounds in the studio.

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## Electricity and Magnetism

Even in a brief study of a radio system, we cannot help but marvel at the important parts which electricity and magnetism play in the operation of practically every radio device. But what is electricity? What is magnetism? If you learn the answers to these two questions once and for all now, *you will have the foundation for everything else you will study and work with in your radio career.*

Our first question is about electricity. But don't expect to find a long and complicated explanation of electricity—don't even expect to find the rest of this lesson devoted to this one subject. This is a *practical* home study Course, so it gives you only those electrical facts *which practical radio men need to know.* Once you master these few facts about electricity—once you realize that all forms of electricity are due either to electrons in motion or electrons at rest—you'll be ready to make electricity work for you in radio equipment.

### What Electricity Is

Everything on this earth is made up from tiny particles, so small they cannot be seen even with the most powerful microscope. These particles are attracted to each other with great force in the case of solid materials like iron or copper, with less force in the case of liquids like water or oil, and with the least force in the case of gases like oxygen or hydrogen.

Scientists have identified several different kinds of these particles, but the only one in which radio men are interested is the electron. It has been proved beyond all doubt that electrons exist everywhere—in the air, in water, in every single object in the world!

During normal conditions, each radio part, each wire, each object on the earth has its own definite number of electrons. The exact number runs way up into the billions even for small objects, but always depends upon the size and nature of the object. Whenever this normal condition exists for some object, that object has no electricity, and is said to be *neutral* or *uncharged.*

In metals like copper, silver or aluminum, some of the electrons are continually wandering around inside the metal itself. These electrons are known as *free electrons*, because they are free to move. The rest of the electrons always stay in the same locations; in fact, if we could discover how to move them, we would be able to change one metal to another—we would realize the age-old dream of changing lead into gold.

We can secure electricity either by taking some of these free electrons away from an object or by adding more free electrons to the object, because we then upset the normal balanced condition. This brings us to an important basic definition:

*Whenever an object has more or fewer free electrons than normal, electricity exists in the object.*

**Sources of Electricity.** But where can we get extra free electrons? This is an easy question to answer—get the extra free electrons from a battery, a power line, a generator, or any other device *which possesses the ability to supply free electrons*. Examples of a few common sources of electricity are pictured in Fig. 2.

Since radio men always work with free electrons (never with the fixed electrons), we can just say *electrons* from now on, when we mean *free electrons*. Remember—whenever you read about electrons in radio magazines or N.R.I. lessons, think of *free electrons*.



STORAGE BATTERY



RADIO BATTERY



POWER LINE OUTLET



WIND-DRIVEN GENERATOR



FLASHLIGHT CELL

FIG. 2. Examples of sources of electricity used by radio men to make electrons move in radio circuits. Storage batteries are used with radio receivers in automobiles, ships, airplanes, and farm homes. Radio batteries are used chiefly in portable receivers. Power line outlets serve by far the greatest portion of the radio receivers in this country. Flashlight cells are used in small pocket-size portable receivers. Wind-driven generators keep radio storage batteries charged in farm homes.

**Law of Electric Charges.** Technically speaking, a law is simply a statement which always holds true. There are only a few of these laws which are important to a practical radio man. Once you understand these laws, you can use them to figure out how radio parts and circuits will behave under various conditions.

The first law which we take up is one which tells how electrons will behave when near other electrons or when near *electric charges*. An electric charge is simply a quantity of electricity.

There are two kinds of electric charges, quite unlike, and for convenience they are called *positive charges* and *negative charges*.

The electron is a *negative charge*, and is also the smallest amount of electricity which can exist by itself.

An object or terminal has a *negative charge* whenever it has *more* electrons than normal. An object or terminal has a *positive charge* whenever it has *fewer* electrons than normal.

Two electric charges cannot exist side by side without trying to move each other. Nature has its own law which tells which way they will move. If the two charges are *alike* (if both are *negative* or if both are *positive*), they will *repel* each other, or tend to push each other away. If the two charges are *unlike* (one is *negative* and the other is *positive*), they will *attract* each other and tend to move together.

Let's repeat this law now in three-word statements which will be easy for you to remember:

#### LAW OF ELECTRIC CHARGES

*Like charges repel.*

*Unlike charges attract.*

Once you learn this simple law, you will have the true basic idea of

how all radio tubes work. You will find that this law helps you to understand the use of test instruments, and aids you in many other ways when you are working on a radio receiver, transmitter or other piece of radio apparatus.

Here is a helpful suggestion: Write this law on a scrap of paper now, and place this paper where you will see it once in a while during the next few days.

The battery terminal which has fewer electrons than normal is called the *positive terminal* of the battery. On diagrams and on batteries, we use a plus sign (+) to indicate that this terminal is positive.

Nature always tries to restore things to normal. That's the reason why the electrons on the negative terminal want to get over to the positive terminal of the battery. Yes, there's just one place those electrons



Sixty billion electrons side by side would be needed to form a line across the dot in the letter "i" shown above.

A free electron can travel through metal as easily as an ant can travel through a sponge full of holes.

Sponges look as solid as rocks when you're standing a hundred feet away from them, and . . . . .

A piece of metal would look as full of holes as a sponge if our eyes could see its actual metal structure.

**Battery Terminals.** An ordinary battery, such as a dry cell or an automobile storage battery, always has *two* terminals, one having more electrons than normal and the other having less electrons than normal. (Electrons, remember, are negatively charged particles.) The battery itself produces this condition by a chemical action which need not be studied here.

The battery terminal which has more electrons (more negatively charged particles) than normal is called the *negative terminal* of the battery. On radio diagrams, as well as on actual batteries, we use a minus sign (—) to indicate that this terminal is negative.

on the negative terminal want to go—to the positive terminal of *their own* battery.

When we connect a wire between the two terminals of a battery, we secure a *complete circuit*. Here is what happens. The instant the circuit is completed by touching the wire to the — terminal, the extra electrons on this terminal tend to rush into the wire because they "sense" a clear path ahead to the + terminal. These fast-moving electrons from the battery bump into free electrons in the wire, and push the free electrons forward toward the positive terminal. In no time at all, every electron gets bumped.

All along the wire, electrons begin pushing other electrons toward the + terminal. Each time an electron is pushed into the + terminal, another electron enters the wire from the - terminal.

Suppose that we connect a length of wire to the positive terminal of an ordinary storage battery, as shown in Fig. 3A. Our wire contains great numbers of free electrons which are ready to "go places," but nothing can happen yet. Our wire is merely an extension of the + terminal.

Suppose, however, that we connect the other end of the wire to the negative terminal of the storage battery, as shown in Fig. 3B. We get action immediately. The wire gets warmer and warmer, a sure sign that electrons are moving.

Of course, a radio man would warningly point out that we are "shorting" the battery with the wire, but we can do this for a *short* time without seriously draining a good storage battery.

Which way are the electrons moving through our wire? Well, if the negative terminal wants to get rid of electrons because it has too many, and the positive terminal wants more electrons to make up its shortage, you should be able to figure out the answer yourself.

Yes, electrons move *from the negative terminal to the positive terminal* through the wire, just as is shown by the arrows in Fig. 3B. This agrees with the law of electric charges, which says that electrons will be attracted by a positively charged terminal. Thus, we've made use of this law already!

The movement of electrons through a complete electrical circuit can be compared to the movement of marbles through the length of tubing shown in Fig. 4. Each time you push a marble into one end of a filled tube, an en-

tirely different marble pops out of the tube at the other end.

As soon as electrons move from the - terminal into the wire, the battery forces more electrons to the - terminal.

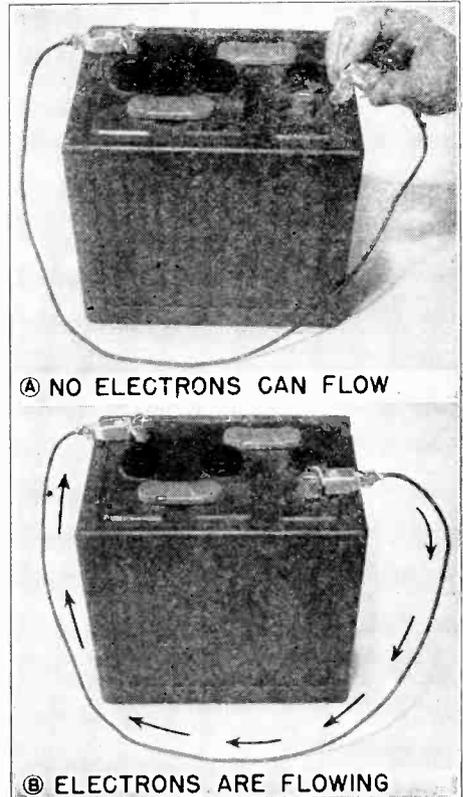


FIG. 3. Nothing happens when a wire is connected to only one terminal of a battery. There must be a complete path for electrons between the two battery terminals in order for electrons to flow. CAUTION: Never "short" a battery needlessly with a wire.

nal. This keeps up as long as we leave the circuit complete or until we have used up all the power stored in the battery. It is because a battery can supply electrons in this way that we consider batteries as *sources of electricity*.

The movement of electrons through a complete circuit is called an *electric current* or simply a *current*. When the source of electricity forces elec-

trons through a circuit in the same direction at all times, the resulting flow of electrons is known as a *direct current*, abbreviated as *d.c.* The more electrons we have moving past a given point in the wire each second, the greater is the value of direct current flowing through the wire.

**Speed of Electricity.** Whenever a wire is connected to the two terminals of a battery so as to give a complete circuit, the free electrons in the wire begin moving toward the positive terminal. These free electrons get into motion almost instantly, hence we say that the *effect* of electricity travels almost instantly from one point to another. (The actual speed of this electrical effect has been measured by scientists and found to be about 186,000 miles per second, which is the same as the speed of light.) The first effect of electricity travels at this speed regardless of battery strength.

Although the effect of electricity travels so rapidly during the first instant when a connection is made, the free electrons move slowly through the wire (only a fraction of an inch per second) once they are in motion. It is just as if we had a long line of soldiers, each standing with his bayonet pressed against the man ahead. When the man at the rear of the line moves forward, the entire line must jump into motion almost instantly so no one will be stabbed in the back. Once in motion, however, the soldiers march at a slow and steady speed corresponding to the slow movement of electrons through a wire.

The instant we complete the circuit, the free electrons in the wire start moving, and continue moving at the slow, steady rate determined by the strength of the battery. If we increase the strength of our battery more electrons will move out of the negative terminal each second and hence the

*current* will increase. If we reduce the strength of the battery, the *current* will decrease. In radio, we often adjust the strength of each battery so exactly the desired number of electrons will be moving through each part in the circuit per second.

There is a need for a convenient way in which to specify the strength of a battery or other source of electricity. Radio men use the term *volts* for this purpose. Thus, you will often hear radio men speak of a 6-volt storage battery, a 115-volt power line,

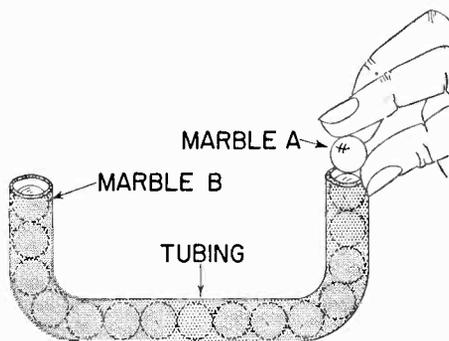


FIG. 4. When you push marble A into one end of this marble-filled tubing, marble B will pop out of the other end. Although marble B will move instantly, it will take some time before enough marbles can be pushed in to make marble A come out at the other end. Continuous electron flow through a wire occurs in much the same way.

a 45-volt B battery or a  $1\frac{1}{2}$ -volt A battery. In later lessons you'll learn a lot more about what volts mean in connection with these *voltage sources*. For the time being, just remember that *volts* tell you *how strong* a source of electricity is.

A voltage which sends a direct current through a circuit is known as a *direct current voltage*, usually abbreviated as *d.c. voltage*. All batteries produce *d.c. voltages*. An automobile generator is also a *d.c. voltage source*.

**Polarity.** When giving the strength of a *d.c. voltage source*, it is often

Desirable to tell which terminal is positive or which is negative. We are then telling what the *polarity* (pronounced po-LAIR-i-tee) of the voltage source is. Thus, when a radio man says, "this wire of your auto radio should be connected to the + terminal of the car battery," he is telling what the *polarity* of the connections to the voltage source should be.

**Conductors.** The wire used for connecting parts together in radio circuits is made from copper. The wire used in making radio coils, transformers, loudspeakers and many other parts is likewise copper. The reason this metal is used so much in radio equipment is that it has large numbers of free electrons.

Silver, aluminum and brass are other metals which have plenty of free electrons. These metals and copper are known as *conductors*, because they pass or conduct electricity. A good conductor is said to have *low resistance*, because it offers little opposition or *resistance* to the flow of electrons.

**Insulators.** To make electrons work for us, we must make correct paths for them by using conductors like copper wire, and we must prevent these paths from touching undesirable paths. In other words, we must separate or *insulate* our desired wire paths from all other paths, so that electrons will go through every part in the path instead of taking short-cuts through other parts.

A wire can be insulated by covering it with an *insulator*, a material which offers a great deal of opposition to movements of electrons.

Silk, cotton, enamel, rubber, paper and glass are good insulators, because they are materials which have very few free electrons.

Free electrons have just as difficult a time in moving through an insulator

as soldiers would have in moving through a barbed wire entanglement. Just as barbed wire is used to limit the movement of soldiers, so are insulators needed in radio circuits *to make electrons take the correct paths through radio parts and wires*. In other words, insulators offer high *resistance* to the flow of electrons.

An insulator can become a conductor if the voltage is made high enough to cause actual destruction of the material. This breakdown is accompanied by an electric arc or spark which burns the insulating material, thereby providing a conducting *path* for electrons.

This is exactly what happens when the insulation in a radio part breaks down; the radio man simply says the part is *shorted*, but actually the insulation fails due to the heat of the spark, and electrons flow through the resulting charred hole in the insulation. The action is much like that of a tank traveling through a barbed wire entanglement; once the tank has forced its way through, soldiers have no difficulty in following.

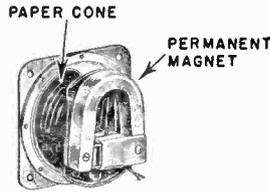
### Radio Uses for Magnetism

Without magnetism, *there would be no radio industry at all*, for magnetism and electricity working together give us *radio waves* to carry messages, music and programs through space.

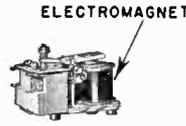
There are also quite a few radio parts which could not work without magnetism. Both *magnetic loudspeakers* and modern *permanent magnet dynamic loudspeakers* use powerful *permanent magnets* to help make the large paper cone move and produce sound. Each *headphone* unit uses a tiny but powerful permanent magnet to help make the round metal disc move and produce sound.



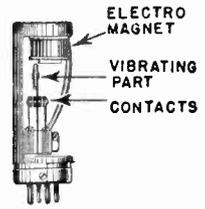
HEADPHONES



MAGNETIC LOUDSPEAKER



RELAY



VIBRATOR

Radio parts which use permanent magnets or electromagnets.

An *electrodynamic loudspeaker* has an *electromagnet*, a device which acts as a magnet only when current is actually flowing through the magnet coil.

Electromagnets are used in some television picture tubes to make a beam of electrons sweep back and forth in the correct manner for "painting" a picture or pattern on a screen.

You will also find electromagnets in auto radio *vibrators*, in the *relays* which do automatic switching work in radio transmitters, and in many special radio parts, which are taken up in later lessons.

Radio transformers depend upon magnetism for the transfer of signals and electric power from one circuit to another. The silent magnetic action in a transformer is just as important in a radio system as the humming sound of a vibrator, the chatter of a loudspeaker, or the clicking of a relay.

Since magnetism is so important to a radio man, let us pry into its secrets now, to see what it really is and how it can work in so many different ways in radio parts.

**Permanent Magnets.** The ancient Greeks discovered, near the city of Magnesia in what is now Turkey, a mineral which possessed the unusual ability to attract small pieces of iron. The Greeks called this material *magnes*, and from this term comes the word *magnet* which we know today.

Early experimenters found that they could make needles and other pieces of hard steel act like magnets simply by rubbing these steel objects on a piece of this magnetic ore. These *magnetized* objects were called *permanent magnets*, because the magnetic characteristics seemed to be permanent.

When a magnetized steel needle is placed on a cork which is floating in water so it can turn easily, the needle will always line up in a direction corresponding closely to north and south. This magnetic experiment is illustrated in Fig. 5. It led to the first practical use for magnets, in the compasses of

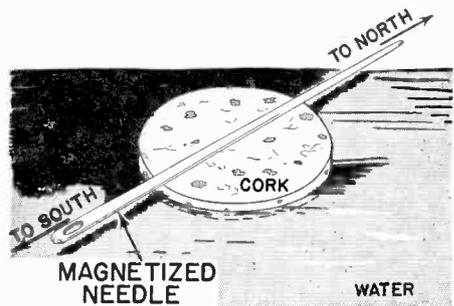


FIG. 5. If a magnetized steel needle is placed on a floating flat piece of cork, the needle will line up in the north and south directions like a compass, because the needle lines up with the magnetic field of the earth. In radio, we produce magnetic fields much stronger than those of the earth, and use these fields to make iron or steel objects move or line up in desired ways.

early sea voyagers and travelers. For practical reasons, the needle in a compass is mounted on a delicately pivoted bearing which turns just as easily as the cork in water.

**Poles of a Magnet.** The ends of a permanent magnet are called *poles*, probably because these ends point toward the *poles* of the earth when a magnet is pivoted. The magnet pole which points north is called the *north*

net is brought near the *S* pole of the compass needle. Our conclusion is *that magnet poles which are alike repel each other.*

If the *S* pole of a bar magnet is brought near the *N* pole of a compass, a strong attracting action is noticed, just as is indicated in Fig. 6B. Thus, *unlike magnet poles attract each other.*

This law of magnetism is fixed by nature, and always holds true. It is

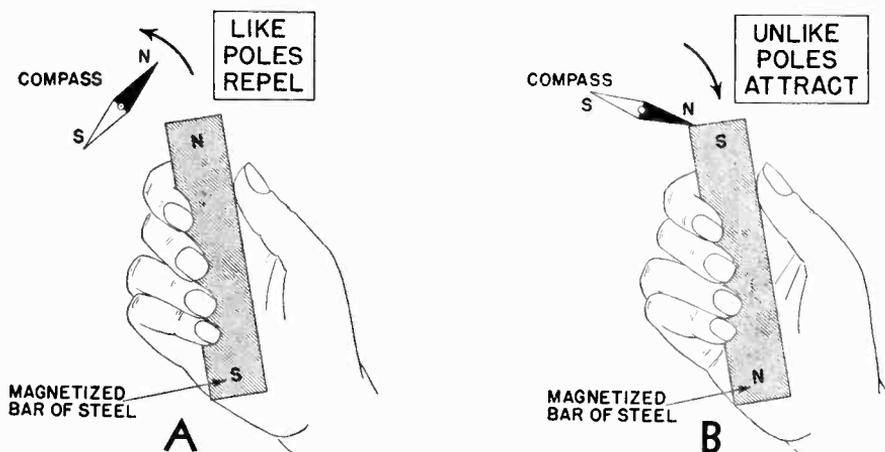


FIG. 6. A simple method of demonstrating the law of magnetism. This law applies to the operation of loudspeakers, headphones, relays, vibrators, motors and many other radio parts.

*pole* of the magnet. The magnet pole which points south is called the *south pole*. On diagrams of magnets, the north pole is usually marked *N*, and the south pole is usually marked *S*.

One law which is particularly useful to a radio man is the law which tells how magnets will behave. It can be demonstrated with a compass needle and a bar magnet (a magnetized bar of steel) in the following manner:

If the north (*N*) pole of the bar magnet is held near the north pole of the compass, just as is shown in Fig. 6A, a repelling action is noticed because the compass is itself a small bar magnet. This same repelling action occurs when the *S* pole of the bar mag-

net is brought near the *S* pole of the compass needle. Our conclusion is *that magnet poles which are alike repel each other.*

#### LAW OF MAGNETISM

*Like magnetic poles repel.*

*Unlike magnetic poles attract.*

Again you may want to write this law on a piece of paper so you can look at it often.

**Magnetic Lines of Force.** If a thin sheet of cardboard is placed over a bar magnet, and iron filings are sprinkled evenly over the cardboard, the iron filings will arrange themselves in definite lines, as shown in Fig. 7A. If the experiment is repeated when the north poles of two bar magnets are end

to end, the pattern shown in Fig. 7B is obtained. If the experiment is repeated once more, with opposite poles of the two bar magnets near each other, the pattern shown in Fig. 7C is obtained.

These experiments show definitely that there is some kind of force acting through the space surrounding a magnet. The scientist Faraday studied magnets for a long time, and finally decided that these forces must act

along invisible but definite lines or paths in space. He said that these paths were the same as those traced out by the iron filings, and gave the name "*magnetic lines of force*" to the natural forces which a magnetic pole produces in space. Radio men still use this name today, and also say that all the magnetic lines of force around a given magnet make up the *magnetic field* of the magnet.

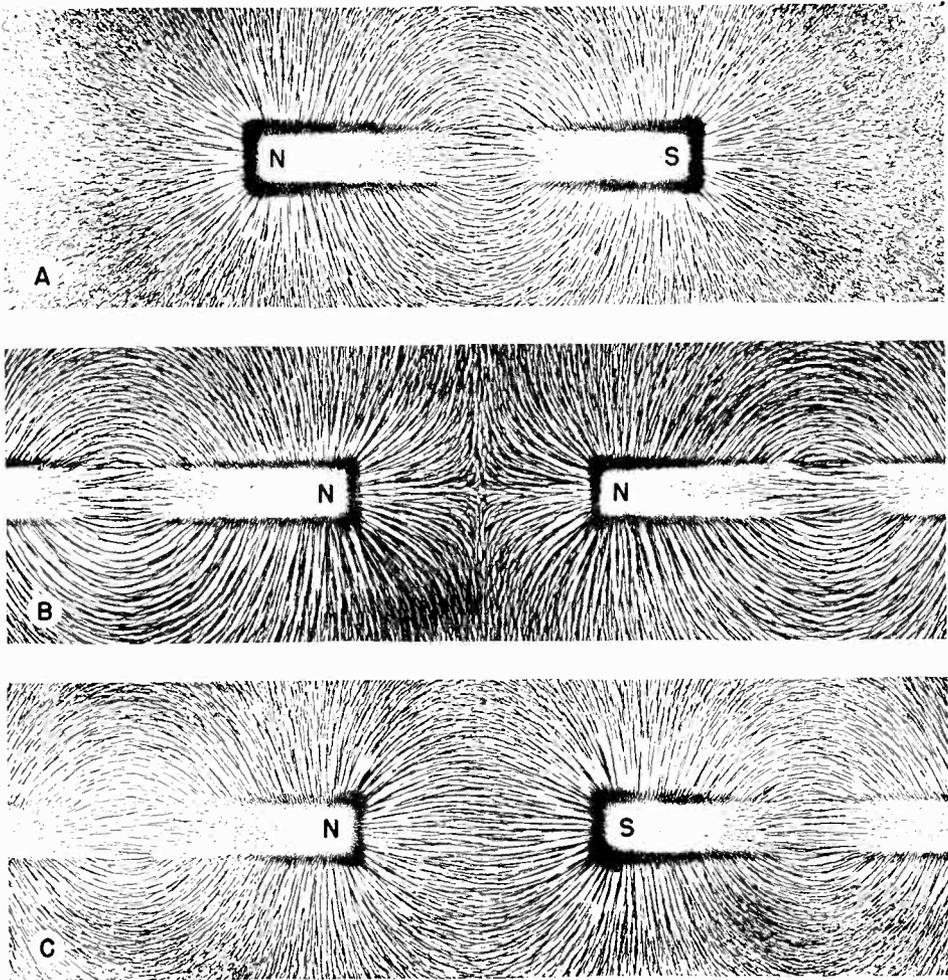


FIG. 7. Here are some of the results which can be obtained by placing sheets of cardboard over permanent bar magnets, then sprinkling iron filings over the sheets. Magnetic lines of force are traced out by the filings almost magically.

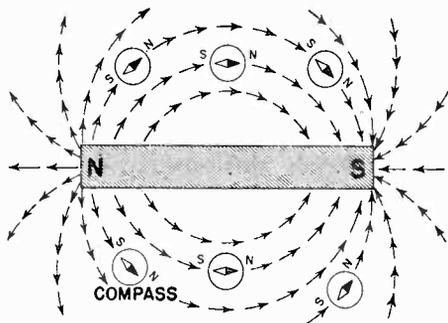


FIG. 8. A small compass can be used to trace magnetic lines of force near a permanent magnet.

When a small compass (which always has an N pole and an S pole) is moved to various positions in the vicinity of a bar magnet, as indicated in Fig. 8, the compass needle will line up with the magnetic lines of force at each position, and will thus trace out the magnetic lines of force in the vicinity of the bar magnet. Compare Fig. 8 with Fig. 7A, and you will see that the compass and iron filing experiments check perfectly.

Magnetic lines of force are considered to have a *direction* as well as a position in space. This direction is always that in which the N pole of a compass will point. Thus, the arrows in Fig. 8 show that magnetic lines of force come out of the N pole of a magnet, and enter the S pole. This rule is worth repeating here for reference purposes, even though you do not have to memorize it.

Magnetic lines of force come out of the N pole of a magnet, and go into the S pole. The N pole of a compass (painted a dark color) always points in the direction of magnetic lines of force.

**Electromagnets.** In 1820, the scientist Oersted discovered that when an electric current is flowing through a wire, the current produces *magnetic* effects in the space around the wire. When he moved a compass around a

current-carrying wire, the compass needle definitely traced magnetic rings (circular magnetic lines of force) as shown in Fig. 9A. (Only two rings are shown in Fig. 9A, but actually there are many more magnetic rings surrounding a current-carrying wire, and these rings are of many different sizes.)

This experiment with a compass and a current-carrying wire proves that *electrons in motion produce magnetic effects* similar to those of a bar magnet.

The direction of the magnetic lines of force around a current-carrying wire depends upon the direction in which electrons are moving through the wire, and can be determined almost instantly with the Left-Hand Rule given here and illustrated in Fig. 9B. This rule is well worth remembering.

#### LEFT-HAND RULE FOR A CURRENT-CARRYING WIRE

Imagine you are grasping the wire with your left hand in such a way that your thumb points in the direction of electron flow. Your fingers, curled around the wire, will then be pointing in the direction of the magnetic lines of force.

The magnetic lines of force which surround just a single current-carrying

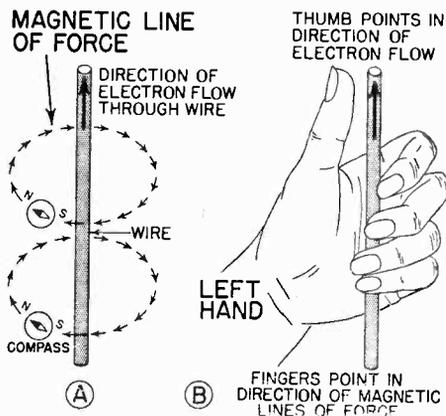


FIG. 9. Whenever electrons flow through a wire, magnetic lines of force surround the wire and have the direction shown here.

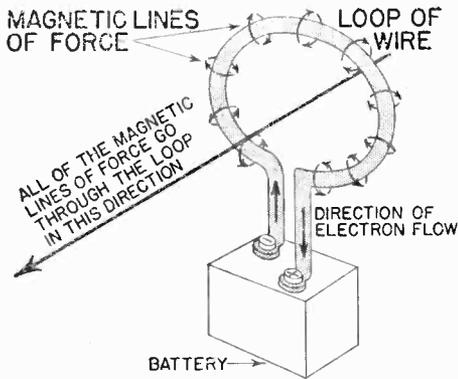


FIG. 10. When we send electrons through a loop of wire, the magnetic lines of force all go through the center of the wire loop in the same direction.

wire are weak and do not have much practical use, but we can concentrate these lines of force and make them more useful by bending the wire into a loop or single-turn coil, as shown in Fig. 10. Now all the circular magnetic rings pass through the center of the coil in the same direction and reinforce each other.

If we add more turns of wire to this coil, as in Fig. 11, the magnetic field inside the coil becomes much stronger even though the same current is still flowing through the wire. For this reason, radio coils usually have a great many turns of wire.

Whenever a coil like that shown in Fig. 11 is carrying a direct current, this coil acts exactly like a permanent magnet. If we trace out the magnetic lines of force with a small compass, we find that they all go into the right-hand end of the coil and come out of the left-hand end. The right-hand end of the coil corresponds to the S pole of the permanent magnet shown in Fig. 8, so we call this end the S pole of the coil. Similarly, we call the left-hand end of the coil the N pole.

The more current we send through a given coil, the greater will be the

strength of its magnetic field. This is logical, for more current means more electrons moving through the wire, and each additional electron in motion adds its share of magnetic rings to those passing through the center of the coil.

When a piece of iron or soft steel is placed in the center of a current-carrying coil, the magnetic field which passes through the coil becomes much stronger, because soft steel is a much better path for magnetic lines of force than air. Since the magnetic field exists only when electric current is flowing, coils with iron cores like this are called *electromagnets*.

The magnetic effect of an electromagnet is only temporary, for it disappears almost as soon as the current flow through the coil is stopped. This characteristic makes electromagnets more desirable than permanent magnets for many radio parts.

Any one who wants to demonstrate how an electromagnet picks up iron objects like tacks can make one from ordinary materials. Simply wind about 25 turns of insulated copper wire around a large nail, then connect the ends of the wire to the two terminals

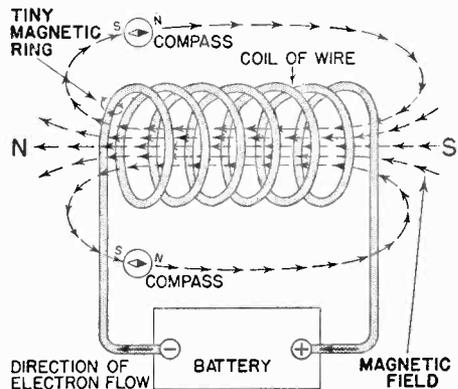
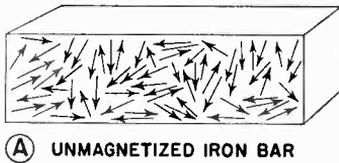


FIG. 11. The more turns of wire we have in a coil, the stronger will be the magnetic field when electrons flow through the coil.

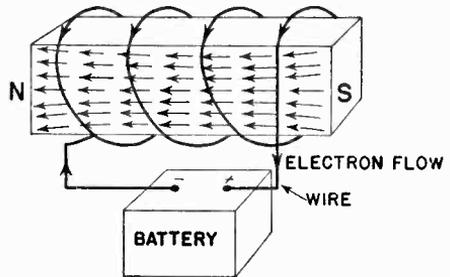
of a flashlight cell and bring the nail near the tacks. (Leaving this electromagnet connected longer than a minute or two will drain the battery.)

The strength of an electromagnet having a particular size of iron core depends chiefly upon two things, the *number of turns* of wire used in the coil and the *amount of current* flowing through the coil. Increasing either one will make the electromagnet stronger. Reducing the current or

If we place an iron bar in a strong magnetic field such as that existing inside a current-carrying coil, the tiny magnets within the bar will tend to line up with the magnetic lines of force of the coil. When this occurs, the entire bar becomes one large and powerful magnet, and we have the electromagnet illustrated in Fig. 12B. When the magnetizing force is removed by stopping the current or taking the iron bar out of the coil, practically all of



(A) UNMAGNETIZED IRON BAR



(B) IRON BAR MAGNETIZED BY CURRENT THROUGH COIL

FIG. 12. Magnetic story of a bar of iron. The small arrows represent invisibly tiny magnets inside the bar.

removing some of the turns of wire will make the electromagnet weaker.

The magnet in every electrodynamic loudspeaker is an electromagnet. The coil of this electromagnet is known as the *field coil* of the loudspeaker, because it produces the *magnetic field* required to operate the loudspeaker. An electrodynamic loudspeaker will work only when current is flowing through the field coil, so this type of loudspeaker will always have two extra wires for this field current.

**Explanation of Magnetism.** What makes it possible for metals like iron to become magnets? The answer which scientists have for this question is that each invisible tiny particle of iron is a tiny permanent magnet. Ordinarily these tiny magnets are arranged "helter-skelter" in iron, as indicated in Fig. 12A, so that their magnetic fields act in all directions and cancel each other.

the magnets return to their original helter-skelter arrangement.

If a bar of hard steel is used instead of iron, however, the magnets remain in their new lined-up position even after the magnetizing force is removed, and we have a permanent magnet. The more tiny magnets we can line up in one direction, the stronger is the permanent magnet.

Iron is the most useful magnetic metal. Nickel and cobalt (a silver-white metal resembling nickel) have weak magnetic characteristics by themselves, but when one or both of these pure metals are mixed with other metals in definite ways to give magnetic alloys, the results are even better than are possible with steel. One example is *alnico*, a combination of aluminum, nickel and cobalt which is now widely used for permanent magnets in loudspeakers.

# Radio Tubes

Practically every piece of radio equipment being made today contains at least one radio tube, so there should be no question whatsoever regarding the importance of learning about tubes as soon as possible in your course. You already know where tubes are used in broadcasting systems—you know the vital facts about the electrons which make radio tubes possible—so now you're fully prepared to learn what radio tubes are like and how they work.

**First Amplifier Tube.** The invention by Lee DeForest in 1904 of the "audion" radio tube shown in Fig. 13 did more than anything else to make possible radio as we know it today. With tubes like this, it was possible for the first time to *amplify* a weak radio signal and thereby build up its strength hundreds or even thousands of times.

This first radio amplifier tube is remarkably simple, as you can see for yourself by examining the sketch in Fig. 13. There are only three main parts or *electrodes* (pronounced *ee-LEK-troads*). Each electrode has its own name.

The *filament* of this tube is just like the filament of an ordinary electric lamp. When current is sent through this filament, it gets hot. On the surface of this filament is a chemical coating which actually gives off or *emits* electrons when heated. A filament which supplies free electrons in this manner is also serving as the *cathode* electrode.

The rectangular plate of metal in this tube is called the *plate*. When a voltage source is connected between the plate and the cathode in such a way that the positive terminal goes to

the plate, the plate *attracts* the electrons which are emitted by the cathode. Under this condition, the plate electrode is said to be positive with respect to the cathode.

The *grid* is the important electrode introduced by DeForest; it is between the filament and the plate, in a posi-

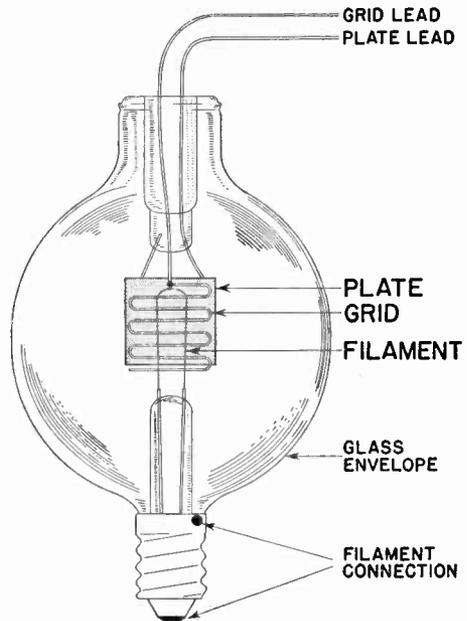


FIG. 13. DeForest "audion" radio tube, sketched from a tube in the N.R.I. collection of history-making tubes. If you ever visit N.R.I., be sure to look for this audion tube in the display rack on the third floor.

tion where it can be made to have a powerful control over the number of electrons which get to the plate.

Many additional parts have been added to this first DeForest "audion" tube and many improvements have been made, but modern radio tubes still use the same control grid idea.

**Modern Tube Construction.** A cut-away diagram showing the con-

struction of a typical modern radio tube is shown in Fig. 14.

In the center of the tube is the *filament*, which becomes red-hot and supplies heat.

Surrounding the filament is a cylindrical or tubular-shaped electrode called the *cathode*, which emits electrons when heated by the filament.

The remaining two electrodes encircle the cathode. The outermost is a metal cylinder which attracts the electrons and is still called the plate even though it has no resemblance to the flat metal plates of early radio tubes. Between the cathode and the plate is a spiral wire arrangement called the *grid*, which is used to control the number of electrons moving from the cathode to the plate.

The electrodes of a vacuum tube are enclosed either in a metal or glass housing called the *envelope*, from which all air has been pumped out during manufacture so as to give a vacuum, necessary for maximum electron emission.

The tube electrodes are connected to prongs on the tube base. This base fits a socket which provides individual contacts for each electrode.

The filament is considered an electrode of a radio tube only when, in addition to its job of providing heat, it has the extra job of giving off electrons. Modern 1.4-volt battery-operated tubes and the old DeForest audion are examples in which the filament also serves as the cathode.

The least number of electrodes a tube can have is two: 1. The *cathode*, which emits (gives off) the electrons; 2. The *plate*, which attracts the emitted electrons. There is no definite limit to the maximum number of electrodes, but radio receiving tubes usually have somewhere between two and six electrodes. A tube can have two or more different types of grids,

but for the present we will consider only tubes having one grid.

The tube in Fig. 14, which we are using as an example, has three electrodes—a cathode, a plate, and a grid. For this reason, it is known as a triode (pronounced "*TRY-oad*"). Let us examine the important parts of this vacuum tube individually now, and see what voltage each part requires.

**Filament.** The filament in a tube like that in Fig. 14 is a fine hairpin-shaped wire, with heavier wire leads going from the ends of the filament to the two filament prongs on the tube base. The fine filament wire becomes red-hot when filament current is sent through it by a battery or other voltage source. This filament merely provides heat, so it cannot be called a cathode.

The filament voltage source is commonly designated by the letter "A." Thus, either an A battery or some other A voltage supply can be used to send current through the filament. In those battery receivers where the polarity of the filament voltage is important, the positive terminal is identified as "A+," and the negative terminal is marked "A—."

**Cathode.** Surrounding the filament of our tube is a metal sleeve about  $\frac{1}{16}$  inch in diameter, serving as the *cathode*. This cathode sleeve is coated with special chemicals, which have the ability to give off large quantities of free electrons when they are heated by the red-hot filament. Remember that the cathode is *the electrode which emits electrons*.

**Plate.** The metal cylinder farthest away from the cathode is the *plate*, sometimes known as the *anode*. *The plate in a vacuum tube attracts the electrons which are emitted by the cathode*, because the plate voltage source makes the plate highly positive with respect to the cathode. This is

a practical example of the Law of Electric Charges; the electrons are negative and the plate is positive, so we have *unlike* charges which *attract* each other. The plate is firmly mounted and can't move toward the electrons, so the electrons move toward the plate.

**Plate Current.** When a vacuum tube is properly connected to its voltage sources, the electrons emitted by the cathode will be attracted by the plate. This electron flow (current) through the tube from cathode to plate is commonly called the *plate current* of the tube. The more positive the plate voltage is with respect to the cathode, the *higher* will be the plate current.

The plate voltage source is commonly designated by the letter "B". Thus, either a B battery or some other B voltage supply must be used to provide the required high positive voltage for the plate. The positive terminal of the B voltage supply is connected to the plate, and the negative terminal is connected to the cathode.

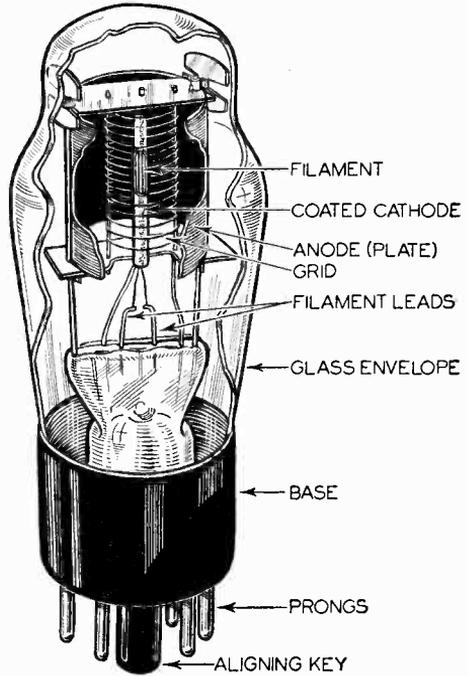
Radio men use the notation "B+" to identify the positive terminal of the plate voltage source and the plate *lead* to which this terminal should be connected. (A *lead*, pronounced "lead," is a wire used for connecting purposes.) Likewise, "B—" is used for the negative terminal of the plate voltage source and the cathode lead to which this negative terminal should be connected.

**The Grid.** A radio tube must have one additional electrode besides the plate and the cathode before it can act as an *amplifier* for electrical signals. This additional electrode is the *grid*; it is usually in the form of a coil of fine wire mounted between the cathode and the plate, with considerable space between the turns of wire, as shown in Fig. 14. The name "grid"

comes from the fact that in early radio tubes, this in-between electrode was a grid-shaped wire (see Fig. 13).

When there is no electrical charge (no voltage) on the grid, electrons can readily pass between the grid wires when traveling from the cathode to the plate.

When a charge is placed upon the



Courtesy National Union Radio Corp.

FIG. 14. Cut-away sketch showing the construction of a typical radio tube. Part of the cylindrical plate electrode is cut away to show the grid and cathode, and part of the cathode is cut away to show the filament wires.

grid of a vacuum tube, however, the grid controls the number of electrons which move from the cathode to the plate. This charge on the grid is commonly called the *grid bias voltage*.

If a positive charge is placed upon the grid, the grid will attract electrons and speed them up, so that more electrons pass through the grid wires and reach the plate. If a negative charge

is placed on the grid, it will repel electrons and reduce the number which reach the plate. This is simply an application of the fundamental electrical law that *like charges repel and unlike charges attract*.

In radio work, the grid usually is made *negative* with respect to the cathode, because a negative grid bias voltage permits more nearly perfect amplification of signals.

When a negative charge is placed on the grid, so it is more negative than the cathode, the electrons on the grid repel the free electrons coming from the cathode, and actually prevent some of these free electrons from passing through the grid wires. *The greater the negative charge on the grid of a radio tube, the less is the electron flow through the tube.*

We can place a negative charge on the grid of a vacuum tube simply by connecting the negative terminal of a voltage source to the grid, and connecting the positive terminal of the voltage source to the cathode. The grid is then *negative* with respect to the cathode.

The grid voltage source is commonly designated by the letter "C," with the terminals being marked "C+" and "C—" respectively. Thus, a C battery can be used to make the grid negative with respect to the cathode. Radio men say then that the grid has a *negative grid bias voltage* or a *negative C bias voltage*.

**Operating Voltages.** We have thus seen that a modern three-electrode (triode) radio tube requires certain definite voltages. These are known as *operating voltages*, and can be summarized as follows:

1. The *filament voltage*, required to send current through the filament for heating purposes.

2. The *plate voltage*, required to make the plate highly positive with

respect to the cathode, so the plate will attract the electrons emitted by the cathode.

3. The *grid bias voltage*, required to make the grid negative with respect to the cathode, so as to permit more nearly perfect amplification of signals.

**More Tube Facts.** We often deal with very weak signals, which must be boosted in strength (in voltage) without changing their nature. A radio tube is one device which can do this job successfully (a transformer is another).

With a definite grid bias voltage, a definite plate current flows through the tube. When we apply a weak signal to the grid of a radio tube along with the grid bias voltage, this weak signal (still varying in strength from instant to instant) makes the grid voltage change rapidly above and below its definite grid bias value. As a result, the plate current of the tube varies above and below its original value in exactly the same way that the weak signal is varying. This varying plate current gives us an entirely new signal, much stronger but still having the same nature as the weak signal. This is how it is possible for a radio tube to *amplify* weak signals.

In Great Britain, radio tubes are called *valves*. This is quite an appropriate name, because the grid does act as a sort of valve when it increases or decreases the stream of electrons flowing from the cathode to the plate.

Now what is the practical use for all this information about radio tubes? Here's the answer—it prepares you to understand the next lesson, which goes right ahead to give you a great deal of interesting and practical information about radio tubes at work.

**Vacuum Tube Symbols.** It is inconvenient to draw a picture or sketch

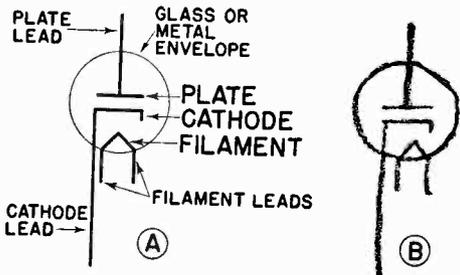


FIG. 15. Symbol for a two-electrode radio vacuum tube, as you might see it in textbooks, magazines and service manuals (A), and as you and other radio men might draw it free-hand for practical use (B).

of a vacuum tube each time we wish to show it in a circuit diagram. For this reason, radio men use simple designs or *symbols* to represent vacuum tubes as well as other radio parts. Symbols are a "shorthand" system of showing radio parts in simple form.

The symbol for a two-electrode vacuum tube having a heater-type cathode (a cathode heated by a filament) is shown in Fig. 15A. The filament is shown as a V-shaped wire projecting inside the circle which represents the glass or metal envelope of the tube. The cathode is shown as a heavy line tending to surround the filament, just as it does in an actual tube. The plate is another heavy, straight line, parallel to the cathode and some distance away from it. Thus, you can see that each line on a symbol tells its own story.

Although tube symbols are carefully drawn with compass and ruler for diagrams which appear in textbooks and magazines, the radio man can make equally understandable symbols free-hand. For example, if a practical radio man were to draw the tube symbol of Fig. 15A while explaining some circuit to a radio-minded friend, the symbol would probably look like that shown in Fig. 15B. Try sketching

this symbol roughly yourself a few times.

The symbol for a three-electrode vacuum tube having a heater-type cathode is shown in Fig. 16A. The grid is shown on this symbol as a dash-dash line located between the cathode and the plate. When drawn free-hand, this symbol will appear somewhat as in Fig. 16B. Try sketching this roughly a few times, pronouncing to yourself the names *filament*, *cathode*, *grid* and *plate* as you draw them in.

**Battery Connections.** A three-electrode radio tube requires three separate voltages. When these are provided by batteries as they are in modern portable receivers, we need an A battery for the filament, a B battery for the plate, and usually a separate C battery for the grid. These batteries all deliver the same kind of electricity (d.c. voltage), but the batteries vary in size because different voltages are required and because some parts of the tube draw more current than others.

The actual battery connections for a radio tube might be as shown in Fig. 17A. The triode tube is shown in symbol form because a picture of a tube would not tell you how the connections were made to the electrodes.

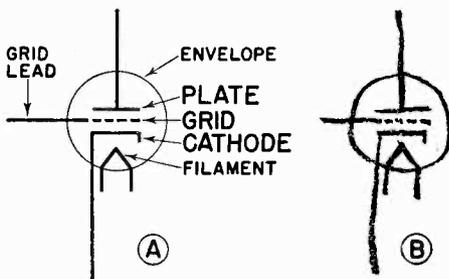


FIG. 16. Symbol for a three-electrode radio tube. Practice drawing it free-hand, as at B.

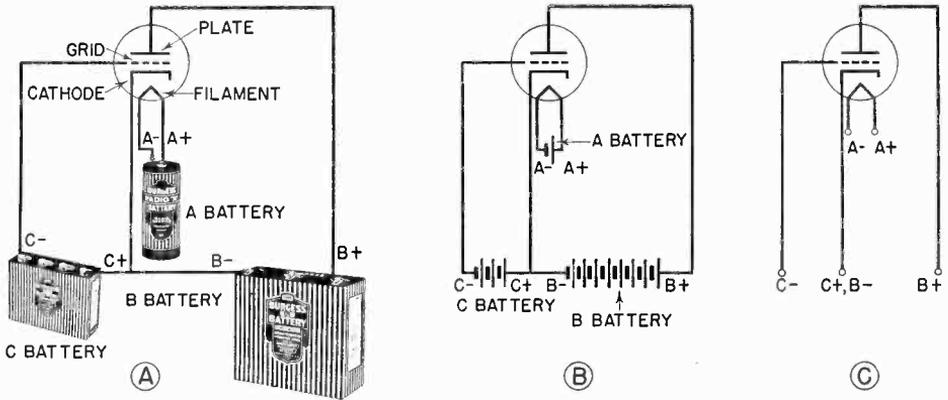


FIG. 17. These three types of diagrams each contain exactly the same amount of useful information about a vacuum tube circuit. Obviously, the diagram at C is the simplest to draw; for this reason, you will encounter diagrams like it most often in your radio studies and radio work. There is no need to make fancy diagrams like that at A when a simple diagram will serve just as well.

As you can see, the A battery in Fig. 17A is connected to the two filament leads. The B battery is connected between the plate and the cathode, so as to make the plate *positive* with respect to the cathode. The C battery is connected between the grid and the cathode, so as to make the grid *negative* with respect to the cathode. Thus, both the + terminal of the C battery and the - terminal of the B battery are connected to the cathode. Each battery acts only in its own circuit, however, because the electrons which leave the - terminal of a battery always go to the + terminal of that same battery.

There is no need to draw a pictorial sketch of each battery, however. We can tell just as much with the battery symbols shown in Fig. 17B. Each pair of long and short lines represents one cell or unit. The short line is always the - terminal of that cell, and the long line is +.

The A battery in the circuit of Fig. 17A has only one cell, so we show only one long and one short line for it. (Of course, in other tube circuits, the A, B and C batteries may have entirely different numbers of cells.)

The C battery which we are using as an example has three cells, so we use three pairs of lines side by side to represent it in Fig. 17B. Notice how there is a long line at the + end of the C battery, and a short line at the - end.

The B battery in this example actually has thirty cells. It isn't necessary to draw in thirty pairs of lines on our diagram, however. We just let the B battery symbol have a few more pairs of lines than the other batteries have, and specify the size of the B battery in some other way (such as by writing its voltage on the diagram.)

Even battery symbols are unnecessary and often omitted. It is entirely sufficient to label the circuit terminals which should go to voltage sources, as shown in Fig. 17C. Furthermore, even the lettering A- and A+ on the filament leads is often omitted when a tube has both filament and cathode, for then the filament connections can be interchanged without affecting performance. Just remember that the notations A-, A+, B-, B+, C-, and C+ on radio circuit diagrams indicate connections which are to be

made to A, B and C voltage sources.

To secure practice in making battery symbols, copy the diagram in Fig. 17B several times freehand. Don't worry about making the short lines thicker than the long lines, and don't worry

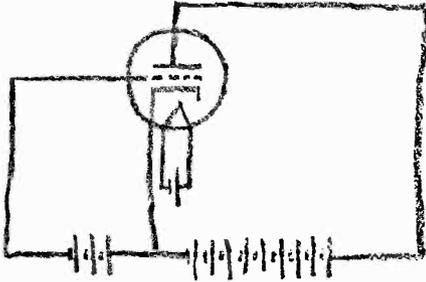


FIG. 18. This free-hand pencil sketch is typical of diagrams made by radio men.

if your symbols look uneven. The sample sketch shown in Fig. 18 is certainly no marvel of perfection, yet it would be perfectly clear and acceptable to any radio man.

**How Tubes Amplify.** Both the plate voltage and the grid voltage affect the amount of electron flow through a three-electrode vacuum tube. Since the grid is closer than the plate to the cathode, the grid has much more effect than the plate upon the electrons which are coming off from

the cathode. This means that a small variation in the grid voltage will change the electron flow through the tube far more than will that same variation in the plate voltage.

For instance, a 1-volt change in the grid voltage might have *ten times greater effect upon electron flow* than a 1-volt change in plate voltage would have. In other words, the radio tube in this example makes a 1-volt radio signal in the grid circuit have the same effect on electron flow as a 10-volt signal would have in the plate circuit. We then say that the tube is able to *amplify* a signal ten times.

Suppose we connected a microphone to the grid of a tube along with the C battery, as shown in Fig. 19. The audio signal voltage of the microphone will now alternately increase and decrease the total voltage which is acting between the grid and cathode. As a result, the electron flow from cathode to plate (the plate current) will vary in exactly the same manner as the microphone output voltage. Now if we place a suitable radio part in the plate circuit, we will find that the audio signal voltage across this radio part is many times stronger than

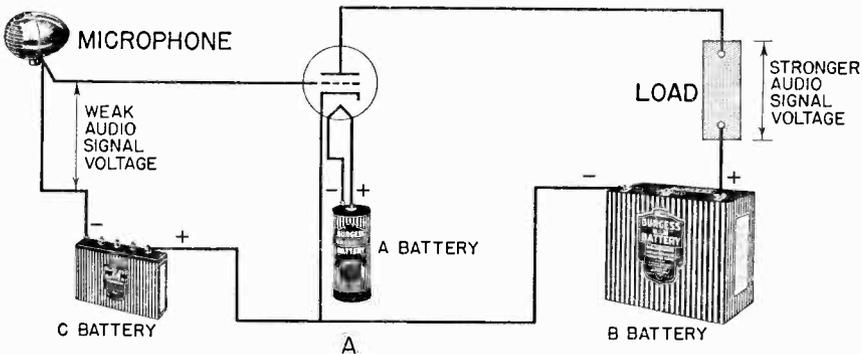


FIG. 19. Elementary diagram showing how a three-electrode radio tube can amplify the weak audio signal voltage of a microphone. Actual microphone connections would be quite different.

the microphone voltage. In other words, *we can use a three-electrode tube to amplify the microphone signal.*

It is not ordinarily practical, however, to connect a microphone directly to a radio tube; the connection is

usually made through a simple device known as a *transformer*. Since a transformer is another highly important radio device, it will be worthwhile to consider briefly here its general characteristics.

## Transformers

**What Is a Transformer?** A transformer in its simplest form is nothing more than two coils of wire mounted close to each other, with no wire connections whatsoever between the two coils. The coil across which we apply our original signal voltage (the *input*

tails of its construction appear in Fig. 20B.

If there is only insulating material like paper or fiber inside the coils (no iron), we have what is known as an *air-core transformer* or a *radio frequency transformer*.

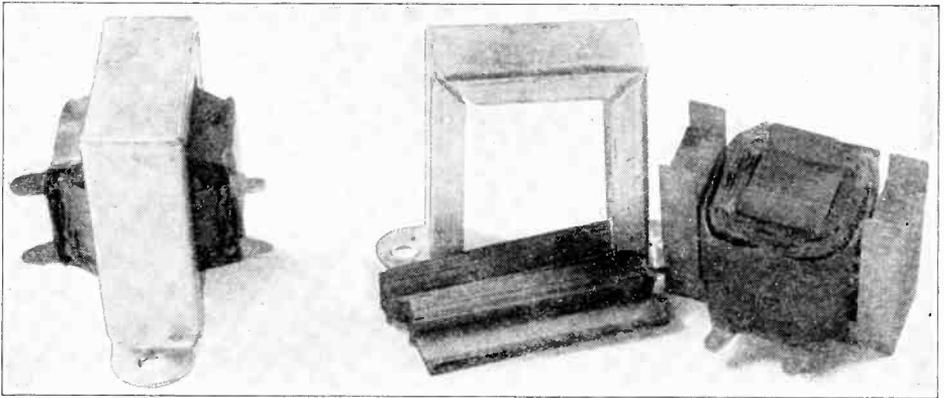


FIG. 20A. Typical iron-core transformer used in radio circuits.

FIG. 20B. Iron-core transformer partly disassembled, to show how the coils fit over the center leg of the iron core. Note that the core is built up from many thin sheets of iron.

signal voltage) is called the *primary coil* or *primary winding*. The coil from which we take the new signal voltage (the *output* signal voltage) is called the *secondary coil* or *secondary winding*.

The material on which the coils of a transformer are wound is known as the *coil form*. When the coils are wound one over the other on a paper form inside of which is a *core* of thin iron sheets, we have an *iron-core transformer*. A typical transformer of this type is shown in Fig. 20A, and de-

**Iron-Core Transformer.** When a current is sent through the primary coil of an iron-core transformer, the iron will become magnetized. If the electric current is varying, as it is in the case of an audio signal, the number of magnetic lines of force flowing through the iron core will vary in an exactly corresponding manner.

Since the iron core passes through both the primary and secondary coils of the transformer, the varying magnetic lines of force are also passing through the secondary coil. These

changing magnetic lines of force cause the electrons in the secondary coil to move just like the electrons in the primary coil are moving, and the result is that a signal voltage is *induced* (created) in the secondary coil.

Notice the use of magnetism; the magnetic lines of force flow through the iron core and make it possible to transfer an electrical signal from one coil to another in an iron-core transformer. This is such an important action to a radio man that it is worth repeating here in a form which will be easy for you to remember.

**Whenever we change the number of magnetic lines of force which pass through a coil, a voltage is induced in the coil.**

If the secondary winding has the same number of turns of wire as the primary, the transformer merely transfers the signal without changing its voltage. For this condition, we use the symbol shown in Fig. 21A, where the two windings are drawn to have the same number of loops. The letter *P* identifies the primary winding, while *S* identifies the secondary winding.

The straight lines between the coil symbols indicate that the transformer has an *iron* core. The frechand way of drawing the iron-core transformer symbol of Fig. 21A is given in Fig. 21B; practice drawing this several times right now, if you have a pencil handy.

If the secondary has more turns of wire than the primary, the voltage induced in the secondary will be higher than the voltage fed to the primary, and we say that the voltage has been *stepped up* by the transformer. The symbol in Fig 21C indicates this by having more loops or curls in the secondary. The microphone transformer in the control room of a radio station is one example of a *step-up transformer*.

If the secondary has fewer turns than the primary, the secondary voltage will be less than the primary voltage, and we say that the transformer has *stepped down* the voltage. The symbol in Fig. 21D is used to show this condition. The output transformer in a radio receiver (the one usually mounted right on the loudspeaker) is an example of a *step-down transformer*.

When radio men draw transformer symbols, however, they sometimes do not bother to show which winding has

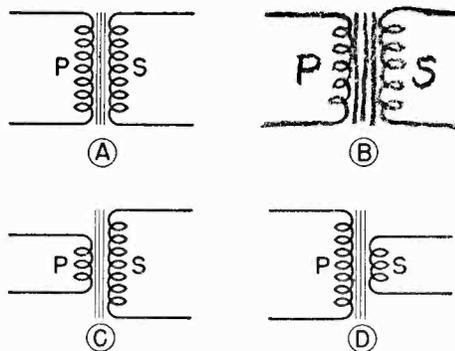


FIG. 21. Symbols for iron-core transformers.

more turns. In many cases, they're interested only in showing that there is a transformer in the circuit. As you acquire radio knowledge and experience, you will gradually become able to tell which winding should have more turns, regardless of how the symbol happens to be drawn.

### Insist Upon Understanding

Now that you have completed your study of this lesson in the logical manner set forth in the Study Schedule on the inside front cover, won't you pause for a moment? Think over what you've learned about radio in just this one lesson.

Already you have a clear idea of what the main parts in a broadcasting

system are. You really do know a great deal about electrons now. The word magnetism means a whole lot more to you today than it did last month. Did you ever expect that you would learn so much about tubes and transformers in the very first lesson?

Yes, already you are well started on a clear, interesting and satisfying Course in radio. You will agree that there has been nothing difficult or "over your head" in this first lesson.

The wonderful thing about the N.R.I. Course is that the remaining lessons will be just as interesting as the first, even though you keep learning more and more from each one. Your knowledge builds up by leaps and bounds once you master the fundamental laws and facts given in the early lessons.

A worth-while practical radio education is worth every bit of the time and effort you give to secure it. Every time you tackle a new idea *and master it*, you achieve still another advance in skill and experience.

In this Course, *you* will supply the ambition, but we're right behind you, ready to give you help whenever it is needed. *You* will do the thinking and studying, but we're eager to help you—we're willing to explain anything in a different way whenever you desire. You know when you understand an explanation and when you don't. Make sure always that you do understand.

Do your very best to master each idea yourself, but don't ever expect that you'll be able to understand everything without some help. Sooner or later, you will come to something which just doesn't seem to make sense

to you. Whenever this happens, just remember that we are prepared and eager to give you personal instruction and special explanations, if you'll tell us exactly what your difficulties are. You have a Technical Consultation Service blank for this purpose. We'll send you another consultation blank with our reply. Don't be afraid to use the blank you have when necessary, for we don't want any missing stones in the foundation you're building for a successful radio career.

### Looking Ahead

In this lesson, you visited the six main parts of a radio system, and secured a "bird's-eye view" of how radio programs and messages are broadcast and received. Before studying radio parts and circuits in earnest, it is natural for you to expect a more complete picture—a real insight into the radio work you hope to do. This desire will be satisfied in the very next lesson.

You will learn exactly *how* sound is picked up by the microphone in a broadcasting studio and converted into an electrical signal, then boosted in strength by amplifiers and given a ride through space on a radio wave. Following the radio wave again to a far-off receiver, you will study the fascinating radio tube circuits which convert the radio signals back into sound again. After this little journey, you will be able to recognize many radio parts and tell what their jobs are in a radio system.

Turn back to the inside front cover now, to see what comes next in your Study Schedule.

**HOW RADIO PROGRAMS  
ARE SENT FROM  
THE STUDIO TO YOUR HOME**

2FR-5

**NATIONAL RADIO INSTITUTE**

ESTABLISHED 1914

**WASHINGTON, D. C.**



# STUDY SCHEDULE NO. 2

By dividing your study into the steps given below, you can master this part of your N.R.I. Course quickly and thoroughly. Check off each step when you finish it.

- 1. Read:—How This Lesson Will Help You.....Page 1
- 2. Microphone in Studio.....Pages 2-8  
Read once at an ordinary rate, then re-read slowly so you clearly understand what sound is, how it is produced, how it travels, what frequencies it includes, and how a microphone changes sound into audio signals. Answer Lesson Questions 1, 2 and 3 on page 29 now.
- 3. Control Room.....Pages 8-13  
Read the first time at your usual reading speed, then re-read slowly to secure a clear understanding of what an audio signal is and how the strength of a weak audio signal is boosted by the audio amplifier in the control room. Answer Lesson Question 4 now.
- 4. Transmitter.....Pages 13-18  
This information which is included to complete your picture of a radio system. Let your own interests determine how much time to spend on this section; one thorough reading will ordinarily be sufficient. Answer Lesson Question 5 now.
- 5. Transmitting Antenna.....Pages 18-20  
Here is more information which you need to read only once. You should, however, get at least a general idea of what radio waves are. Answer Lesson Question 6 now.
- 6. Receiving Antenna.....Page 20  
You can master this short section in a short time and get a satisfactory understanding of how the receiving antenna feeds signal currents down to the receiver. Answer Lesson Question 7 now.
- 7. Radio Receiver.....Pages 21-26  
This is a highly important section both for servicemen and communications men, so read it slowly at least three times. A clear general idea of how a simple tuned radio frequency receiver changes antenna signals into sounds will help you to master later lessons, because basic receiver circuits are to be found in all types of radio equipment. Answer Questions 8, 9 and 10 now.
- 8. A Review.....Pages 27-28  
Read the review slowly several times to refresh your memory on the most important points covered in this lesson. A single reading will be enough for the "Looking Ahead" section, which is a short discussion of the practical radio information you are going to get from the next lesson and how much you have already obtained.
- 9. Mail Your Answers for Lesson 2FR-5 to N. R. I. for Grading.
- 10. How To Use Radio Tools.....RSM Booklet No. 2  
This is another of the series of Booklets that are to give you practical, how-to-do-it training in fixing radios. Read this Booklet now, then keep it handy for future reference. The "shop training" you get from these RSM Booklets is going to prove valuable to you throughout your service career, so you will want to read and review these Booklets many times. NOTE: Some of the Study Schedules in future Lessons may refer to "Job Sheets" instead of RSM Booklets. Read the correspondingly numbered RSM Booklet in such cases. These Booklets are an improved and enlarged version of the older Job Sheets, and are replacing them.
- 11. Start Studying the Next Lesson.

# HOW RADIO PROGRAMS ARE SENT FROM THE STUDIO TO YOUR HOME

## How This Lesson Will Help You

**L**EARNING radio is much like building a skyscraper. Once the steel framework of a skyscraper is in place, the various floors of a skyscraper

This lesson takes you through the same broadcasting system you studied in the first lesson. Now, however, you will go through more slowly, so



Courtesy RCA Mfg. Co., Inc.



Courtesy General Electric Co.

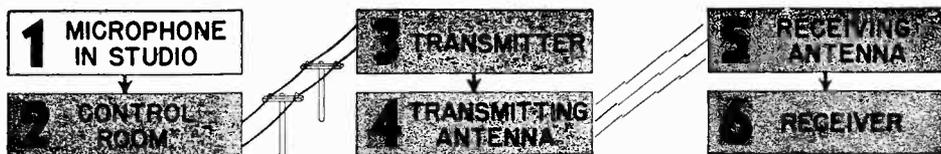
A radio program starts at a microphone which is usually located in a broadcasting studio. On special occasions, however, this microphone may be set up at any point to which man can travel—above, under or on the face of the earth. From this microphone, the program travels over a long and fascinating path to the loudspeakers of countless radio receivers. Somewhere along this path, there is a good-pay radio job awaiting completion of your training.

can be finished in any desired order. Likewise, once you become thoroughly familiar with the various radio parts and circuits which make up a complete modern broadcasting system, you will be in a position to study these radio parts and circuits faster.

Yes, our experience gained from many years of teaching radio has proved this to be true—that radio is *easiest to master* when you have a clear idea of how each study subject fits into the complete picture of radio.

you have time to get acquainted with more of the *important* parts.

This detailed study of a radio broadcasting system will be your framework for the entire N.R.I. Course, regardless of what radio work you eventually choose. Even if you plan to follow radio receiver servicing or some other branch of radio dealing chiefly with receivers, a broad, general understanding of how radio programs are put on the air is essential to the completeness of your training.



**1 MICROPHONE IN STUDIO.** In radio, we start with sounds at the microphone, and must end up with reproductions of these same sounds at the loudspeaker. It is only logical, then, that you find out what sounds are like before you learn how microphones convert sounds into audio signals.

**What Is Sound?** When you strike a piano key, a small hammer swings against a wire inside the piano and makes that wire *vibrate*. If you touch the wire lightly, you will be able to *feel* these vibrations. Your ear can *hear* the vibrating wire even though many feet away, however, so these vibrations must travel through the air.

When you blow a whistle, the air inside and outside the whistle *vibrates*. Raindrops falling into a lake start many small *vibrations* both in the water and the surrounding air. The vocal cords in your throat *vibrate* when you speak or sing.

Since all these different vibrating materials produce sounds which can be heard, we can simply say that sound is a *vibration of any material at a rate which can be heard by human ears*. This vibrating material can be a *solid* object like wire, a *liquid* like water, or a *gas* like air.

**How Sound Travels.** Sounds produced by vibrating bodies must usually travel some distance before they reach human ears. Sound can travel through *any gases, any liquids, and any solid objects* which are elastic enough to vibrate. In other words, *sound can travel through anything which possesses the ability to vibrate*. Sound

cannot travel through a vacuum, because there is nothing in a vacuum which can vibrate. This and other interesting characteristics of sound are illustrated in Fig. 1 (on next page).

Most of the sounds which we hear travel through air, which is a mixture of invisible gases and dust particles. Of particular interest to us as radio men is the way in which a loudspeaker sends sound through the air.

**How a Loudspeaker Produces Sound.** In an ordinary radio loudspeaker, we have a large paper cone which vibrates back and forth when the radio set is operating. This vibrating cone alternately pushes and pulls on nearby particles of air, thereby setting these air particles into vibration.

When the loudspeaker cone pushes air *forward*, it *compresses* the air particles, making them bump into other air particles which are farther away. These bump into still more particles, thereby giving a region of higher air pressure than normal. One of these higher-pressure regions begins traveling away from the loudspeaker each time the cone pushes forward.

When the cone moves backward, it leaves more room directly in front of itself for air particles, thus reducing the air pressure and creating a *partial vacuum* (a vacuum is a space having no air, and hence no pressure). Nearby air particles rush into this lower-pressure area, leaving a partial vacuum where they were. This lower-pressure region likewise travels outward in all directions from the cone of the loudspeaker.

Thus, a lower-pressure region begins traveling away from the loudspeaker each time the cone moves *backward*. This process repeats itself each time the cone moves forward and backward, which happens many times each second. The resulting regions of air pressure higher and lower than normal, traveling away from a sound source, are known as *sound waves*.

One thing we should definitely understand about sound is this: The vibrating particle of air which bumps into your eardrum and makes you hear the voice of a radio announcer did *not* travel all the way from the radio loudspeaker to you.

A sound wave made up of vibrating air particles can be compared to a wave moving across a lake on a windy day. This traveling wave is just a slow up-and-down vibration of water.

The water particles stay in practically the same locations all the time, and only the vibrating or up-and-down action of water travels across the lake with the wind. Likewise, air particles in a sound wave merely move back and forth without getting very far. Each particle transfers its back-and-forth movement to the next particle, and these back-and-forth movements form *traveling* regions of higher and lower air pressure than normal.

**Speed of Sound.** Sound waves travel through air at a speed of approximately 1089 feet per second. This is slow in comparison to the 186,000-mile-per-second speed of radio waves, so you may be hearing the words of a radio announcer coming from your loudspeaker a split second before those sound waves reach persons at the rear in the studio audience.



FIG. 1A. The radio scientist says **YES**, because he knows shattered pieces of coconut will vibrate at a rate which can be heard by human ears. The medical scientist says **NO**, because he defines sound as a vibration acting on and audible to human ears, and there are none on this uninhabited South Sea isle near the Equator.

FIG. 1B. No matter how violent an explosion of the moon might be, neither you nor any other person on this earth could hear the sound of the explosion. The reason is simply that sound travels only through materials which can vibrate, and there is nothing in the etherial space between us and the moon—just vacuum.

FIG. 1C. When a large drum is struck vigorously, the stretched diaphragm of the drum vibrates in and out. This action alternately increases and decreases the air pressure in the vicinity of the drum, creating a sound wave which travels away from the drum. Your hand definitely *feels* the varying air pressure of this sound wave.

FIG. 1D. If some husky youngster should blow a large whistle or horn close to your ear, you would definitely agree that sound can be *painful* to the eardrum if sufficiently loud. This is why artillery men must stuff cotton into their ears or use special ear plugs when firing big guns during land warfare or naval maneuvers.

**How We Hear Sound.** The outer passage of the human ear is closed by a thin membrane called the *eardrum*. When the eardrum is set into vibration by sound waves which travel through air and enter the ear, the vibration is transferred through a series of small bones to a fluid in the inner ear, and through this fluid to the sensitive fibers of the hearing nerves. These nerves convey to the brain the sensation of sound.

**Wavelengths and Cycles.** If your sense of touch were sufficiently delicate, you could stand in the path of a sound wave and feel every variation above or below normal air pressure, just as is illustrated in Fig. 1C. You would feel an *increase* in the pressure of the air, followed by a return to normal, a *decrease* from normal air pressure, and a return to normal again. This would repeat itself as long as the sound was being produced. Each complete *increase* and *decrease* in air pressure, corresponding to one complete vibration back and forth of the sound source, is known as a *cycle* of the sound wave (pronounced *SIGH-kull*).

If you could in some way stop a sound wave and make its variations in air pressure visible, you could take a yardstick and measure the distance between two adjoining higher-pressure regions of the sound wave. Your result would be the length of the wave, or the *wavelength* of that particular sound. Wavelengths of common sounds range from about one inch for the chirp of a cricket to about fifty feet for the booming sound of a bass drum.

**Frequency of Sound.** The number of complete back-and-forth vibrations which a sound source makes in one second of time is called the *frequency* of a sound. If you could count the number of vibrations the cone of a loudspeaker makes in one second, your

result would be the *frequency* of the sound in *cycles per second*.

The lowest sound frequency which a person can hear is about *20 cycles per second*; the highest is about *20,000 cycles per second*. The higher the frequency, the more high-pitched or shrill the sound is to our ears.

The approximate range of frequencies which we can hear is thus *between 20 cycles and 20,000 cycles*; this range is called the *audible range*. Any frequency in this range is an *audio frequency*, abbreviated *a.f.* Of course, these figures represent the extreme limits of hearing, and the average person will have a somewhat narrower hearing range. Older persons, especially, have difficulty in hearing the higher audio frequencies.

Notice that we omitted the phrase "per second" after the frequency values in the previous paragraph. Radio men often do this, and no one is confused, simply because the only interval of time ever used with frequency in radio work is one second. Just remember that whenever you read "20 cycles" or any other value in cycles, the meaning is "cycles per second," unless some other meaning is definitely indicated.

#### **Radio Program Frequencies.**

Most of the radio broadcasting systems in this country are designed to handle audio frequencies from about 80 cycles to about 5000 cycles. A few high-fidelity broadcasting stations handle frequencies up to about 10,000 cycles, while the new frequency modulation broadcasting systems (developed by Major E. H. Armstrong, and described later in the N.R.I. Course) handle audio frequencies from about 30 cycles to about 15,000 cycles. Increased frequency range makes programs seem more natural when heard with frequency modulation receivers.

**Audio Reminders.** You will encounter the terms *audio* and *a.f.* hundreds of times throughout this course. Just remember that they both mean the same in common usage, and both refer to *frequencies which can be heard*.

For example, take the audio signal in a receiver. It is strictly an electrical signal, and cannot be heard. Because it is an *audio signal*, however, we know it has the same frequency (the same number of cycles per second) as the original sound. By feeding this audio signal into a loudspeaker, we can make the nearby air vibrate at exactly this same frequency, and we can hear the resulting sound waves.

### The Microphone

As you have already learned, the purpose of a microphone in a radio system is to *change sound waves into audio signals*. The microphone does this job in two steps: 1. It changes the sound waves into corresponding back-and-forth movements or *vibrations* of a flexible part; 2. It changes these vibrations into *audio signals*.

**How Sound Waves Move a Diaphragm.** Every microphone has one flexible part which vibrates whenever sound waves hit it. This part is called the *diaphragm*, and is usually a thin, light-weight metal disc which is loosely mounted so it can be pushed back and forth by sound waves.

With the aid of a pencil, a length of paper tape, a telephone mouthpiece and a few small parts arranged as shown in Fig. 2A, we can demonstrate how the diaphragm of a microphone acts when a sound wave comes along. (In this diagram, half of the mouthpiece is cut away to show the diaphragm.) The diaphragm and pencil are joined by a link and lever in such a way that the pencil moves to the

left when the diaphragm is *pushed* by the increased-pressure part of a sound wave, and the pencil moves to the *right* when the diaphragm is *pulled forward* by the reduced-pressure part of a sound wave.

If we let the point of the pencil rest on a length of paper tape, and pull this tape upward at a uniform speed while the sound wave of a pure whistled note is making the diaphragm vibrate, the pencil will trace

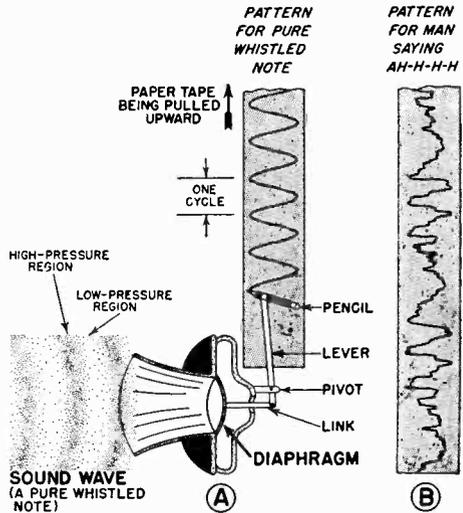


FIG. 2. The simple diaphragm and lever mechanism shown here will trace on paper tape the manner in which a microphone diaphragm is moved back and forth by a sound wave.

on the paper a smooth wavy line like that appearing on the tape in Fig. 2A.

If the sound wave is produced by a man saying "ah-h-h-h," however, the pencil will trace on the paper a jagged pattern like that shown in Fig. 2B. Each other type of sound will give a different pattern on the paper tape, and each pattern will represent the way in which the diaphragm and the source of sound are vibrating.

The one important thing for you to remember is that whenever sound waves hit the diaphragm of a micro-

phone, they make this diaphragm vibrate in the same way that the original source of sound is vibrating.

**How a Diaphragm Produces Audio Signals.** Although all microphones start out by converting sound waves into vibrations of a diaphragm, many different types of microphones are used for changing diaphragm movements into audio signals. The important types in use today are the dynamic microphone, crystal microphone and velocity microphone. Condenser microphones and carbon microphones were once popular, but are now gradually fading out of the radio picture.

Let us assume that the microphone being used in the studio we are inspecting is a *dynamic microphone*. This is today one of the most widely used of all microphones, so an understanding of how it works will be sufficient until you come to the N.R.I. lesson which covers the other types of microphones.

The important features of a dynamic microphone are shown in a simplified manner in Fig. 3. Sound waves enter the microphone through the holes in front, and make the thin circular metal diaphragm vibrate in the same way as the original sound source is vibrating.

Attached to the back of the diaphragm is a paper-thin bakelite coil form about half an inch in diameter. On this cylindrical form is a small coil of wire called the *voice coil*, made from insulated copper wire almost as fine as human hair. The voice coil has two flexible wire leads (pronounced *leads*) which connect to the two wires of the *microphone cable*. The moving system of this microphone (the diaphragm and voice coil) is so light in weight that it vibrates readily even for faint sounds.

Surrounding the voice coil in Fig. 3 are the poles of a powerful *permanent magnet*, arranged in such a way that magnetic lines of force pass through this tiny coil at all times. Yes, that's all there is to a dynamic microphone—just a coil of wire mounted on a diaphragm, surrounded by a magnet, and connected to the two wires of the microphone cable.

When the diaphragm of a dynamic microphone is set into vibration by a sound wave, the voice coil moves in

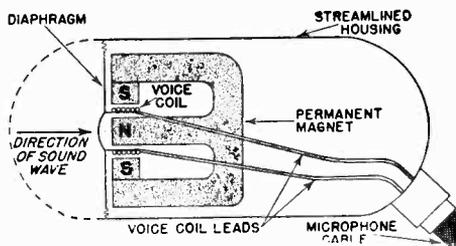


FIG. 3. Cross-section diagram of a typical dynamic microphone, showing the essential features which would be visible if we sliced vertically through the center of the microphone. When the voice coil of a dynamic microphone is moved in and out by sound waves, the number of magnetic lines of force which pass through the coil is changed, and therefore an audio voltage is induced in the coil. (The actual microphone appears in Fig. 6.)

and out through the magnetic lines of force which exist between the poles of the permanent magnet. This movement of the voice coil changes the number of magnetic lines of force which pass through the coil, thereby inducing (producing) an *audio voltage* in the coil.

When the voice coil moves *into* the magnet, the voltage acts in one direction; when the coil moves *out* of the magnet, the voltage acts in the opposite direction. The voltage which the microphone puts out (called the *microphone output voltage*) thus reverses its polarity (direction) from instant to instant. A voltage which regularly reverses or *alternates* its polarity in

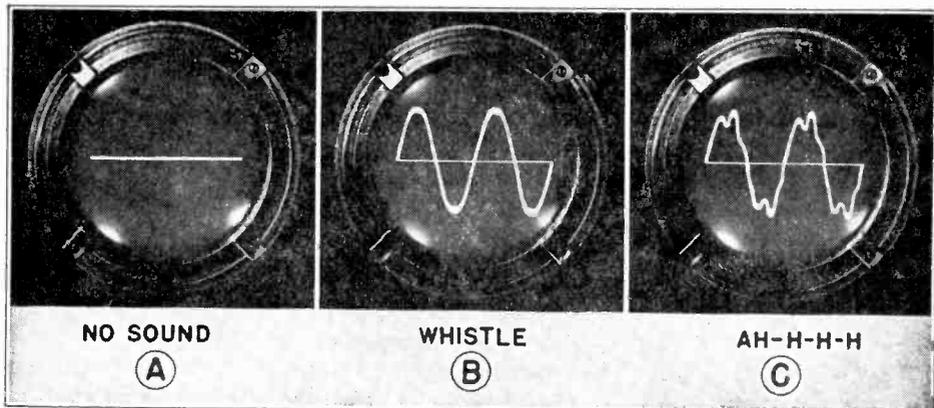


FIG. 4. Patterns like these, traced on the screen of a cathode ray oscilloscope by a flying beam of electrons, show exactly how an audio voltage is varying. The weird, bright green fluorescent glow of the lines in these patterns is truly fascinating. (A fluorescent screen is one which glows when hit by electrons.) By using an oscilloscope to show how invisible voltages and currents are acting, radio engineers can pry into the secrets of radio circuits and actually see how adjustments or new parts affect performance.

this manner is known as an *alternating current voltage* (abbreviated *a.c. voltage*).

The a.c. voltage produced by a microphone has exactly the same *frequency* as the original sound, because the voltage reverses in polarity each time the original sound source reverses its direction of motion. For this reason, the a.c. voltage which is generated in the voice coil of a dynamic microphone is generally called the audio voltage or *a.f. voltage*. Sometimes this audio voltage is simply called the *audio signal* or *a.f. signal*.

The *louder the sound*, the greater is the movement of the voice coil and the *stronger* is the audio voltage produced by the microphone.

**Audio Voltage Patterns.** When radio men want to know how a particular audio signal voltage is varying, they sometimes use an electrical instrument called the *cathode ray oscilloscope* (pronounced *ah-SILL-oh-SKOPE*). In this instrument, the varying voltage is made to move a beam of electrons up and down across the white screen at the large end of a

cathode ray tube. A bright greenish-white glow appears wherever the electron beam hits the enameled coating which forms this *fluorescent screen* (pronounced *FLEW-oh-RESS-sent*). A fluorescent screen glows *while* being hit by electrons.

When the audio signal voltage applied to an oscilloscope is zero (or when there is no sound), the electron sweeps horizontally from side to side continually, and "paints" on the screen the horizontal line shown in Fig. 4A.

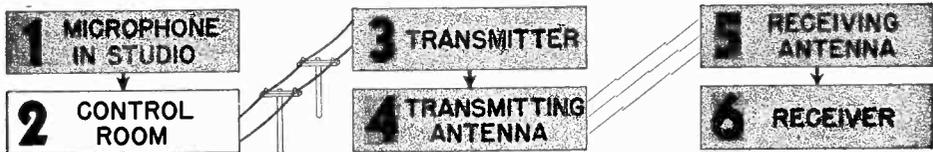
When we use an oscilloscope to look at the audio voltage produced by a pure whistled note, the audio voltage pattern shown in Fig. 4B appears on the screen. Compare this with the pattern produced by the moving pencil in Fig. 2A, and you will see that this audio voltage varies exactly in step with the movements of the microphone diaphragm. The horizontal line still shows faintly here, for the electron beam returns along this line while tracing the pattern over and over again. Each time the pattern crosses the horizontal line, the voltage

drops to zero and reverses its polarity.

When we apply to an oscilloscope the audio voltage produced by a microphone while a man is saying "ah-h-h-h," the audio voltage pattern shown in Fig. 4C appears on the screen. Comparing it with Fig. 2B, we find again that the a.f. voltage of

a microphone varies in step with the diaphragm movements.

The audio voltage produced by a microphone makes electrons move back and forth through the microphone wires when a complete circuit is provided for the microphone by apparatus in the control room.



**2 CONTROL ROOM.** The audio signal voltage delivered by a microphone is so weak that it must be boosted a great deal before it can be sent over telephone lines to the transmitter. This building up of the signal voltage is done in the *control room*, under the watchful eye of the *monitor operator*.

The control room of a typical radio station is pictured in Fig. 5. Let's imagine that we have just stepped into this room, and have asked the monitor operator to tell us about his work. His answer might run something like this:

"The audio signal travels through the microphone cable to this sound-proofed wall in front of me. The sig-



*Courtesy Western Electric Co.*

**FIG. 5.** The monitor man in the control room watches the performers in the studio through a large glass window as he "rides gain." The small microphone at the left of the control unit is used principally for intercommunication between the monitor room and studio prior to a broadcast or during rehearsals, for the wall between the two rooms is sound-proof.

nal is brought through the wall by means of a plug and outlet device on each side, after which it travels through another cable to this audio amplifier unit on my desk.

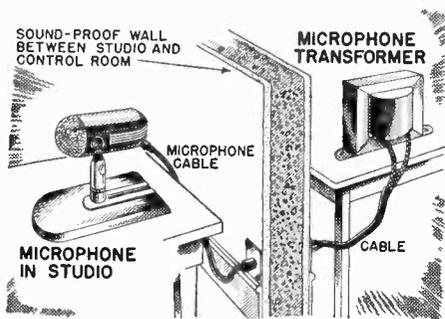


FIG. 6. The first circuit in our studio-to-home radio system extends from the microphone in the studio to a microphone transformer in the audio amplifier which is located on the control desk in the control room.

"Of course, there are other microphones in the studio, each feeding into my amplifier over a similar cable, plug and outlet path. Switches allow me to change instantly from one microphone to another during a broadcast, or use any combination of microphones when the program includes a dramatic production or music by a large orchestra.

"This heavy dual plate-glass window, built into the wall between the control room and the studio, blocks all sounds but allows me to watch the artists and the announcer while I 'ride gain' on the program. This is my most important duty; to 'ride gain' means to adjust the volume control here on the panel continually to compensate for excessive changes in the loudness of the program.

"For example, if I notice that the announcer is gradually moving away from the microphone, I gradually advance this volume control so the listeners cannot notice any change in the loudness of their program. If this meter in front of me indicates that the

program is getting too loud, I immediately cut down the volume, to prevent overloading of the transmitter circuits. Overloading destroys the naturalness of the sound signal, and can even put the transmitter 'off the air' for a few moments because of the operation of automatic circuit-breaking switches which protect tubes and parts from burning out during overloads.

"But let's follow the audio signal as it enters this amplifier. The first thing it encounters is an iron-core transformer known as the *microphone transformer*. Notice that the two microphone wires go to the two terminals on the primary winding on this transformer (see Fig. 6); this gives us a complete closed circuit over which electrons can travel.

"Here's the way I like to imagine that electrons are moving through the microphone circuit. I think of a neck-



Using a cathode ray oscilloscope to see how the a.f. signal voltage in a radio set is varying.

lace of balls strung end to end around the entire circuit, as on this diagram (see Fig. 7). The lever of our microphone diaphragm mechanism (the same mechanism as in Fig. 2) is attached to one point on the necklace.

“Now, when a sound wave pushes down on the diaphragm, it makes the necklace end of this lever move upward, and the whole string of balls has to travel in the direction indicated by the arrows in Fig. 7.

“When the sound wave has pushed the diaphragm as far down as it can, both the diaphragm, the lever, and the necklace stop moving. Then, as the diaphragm moves backward due to its own springiness and to the pulling effect of the following lower-pressure

start up the car gradually as you leave one point, you travel at maximum speed when half-way between the two points, and you start slowing down before you arrive at the other point so as not to go past it.

“The actual electrons which make up an alternating current or signal current in a radio circuit move back and forth just like this imaginary necklace of balls. Of course, a piece of wire has millions of times as many electrons as there are balls in the

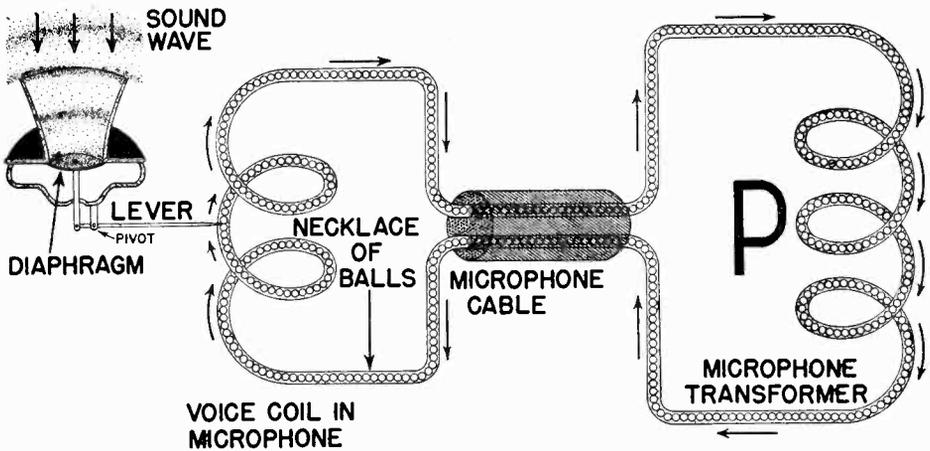


FIG. 7. To picture the way in which electrons move back and forth in a wire to make up an audio signal, think of a necklace of balls which is being moved back and forth through a circuit.

region in the sound wave, the lever starts moving downward, and the entire necklace moves in the opposite direction to that shown by the arrows in Fig. 7.

“We thus have our necklace starting up gradually in one direction, increasing rapidly in speed, slowing down again as the diaphragm reaches the limit of its travel, stopping as the diaphragm stops, then reversing in direction and again repeating this increase and decrease in speed.

“You can compare the speed of this necklace to that of an automobile which you are driving back and forth continually between two points. You

necklace shown in Fig. 7, and there are no strings connecting the electrons together. But—whenever the electrons in a wire are acted on by an a.c. voltage, the electrons just move back and forth without getting anywhere, like the necklace of balls. That’s all any audio signal *current* is—just a back-and-forth movement of electrons through the wires and parts of a circuit. The lower the frequency, the fewer back-and-forth movements there are per second.

“Here is a simplified circuit diagram which represents this first stage or section of our radio system (see Fig. 8 now). The audio signal current

which flows through primary winding  $P$  of the microphone transformer causes an audio signal voltage to be induced in secondary winding  $S$  of this transformer. This audio voltage is several times stronger than the microphone voltage, because the microphone transformer secondary winding has more turns of wire than the primary.

"Notice that the stronger audio signal voltage which exists in secondary  $S$  acts upon the grid and cathode of the first a.f. amplifier tube, along with the d.c. voltage of the C battery which

the time. Each circuit is so planned, however, that the motion of the electrons in one circuit controls the motion of the electrons in the next circuit in the desired manner.

"To make this perfectly clear, let me redraw the circuit of Fig. 8 in such a way that each complete path for electrons is separate from the others (see Fig. 9).

"The electrons which travel back and forth in the microphone circuit cannot get into the grid circuit . . . cannot get into any other tube circuit

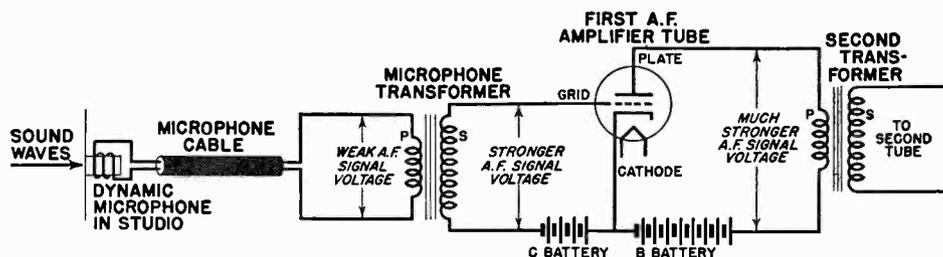


FIG. 8. Simplified circuit diagram showing the first part of the path taken by signals in traveling from the studio to your home. Only the most important parts of the circuit are shown here, the complete circuit will be taken up in later lessons.

is in this same circuit. This first tube and its B battery together have amplifying ability, so a still stronger a.f. voltage is delivered to the primary of the second transformer in our circuit.

"A microphone transformer and an amplifier stage acting together may amplify an a.f. signal voltage several hundred times, but even this is not enough. We must have many stages in this amplifier, with many tubes and transformers. Each stage steps up the signal voltage some more. Finally, we secure a strong enough voltage to send our signal over the telephone lines to the transmitter.

"There is just one more thing I must make clear before you leave the control room. Electrons do *not* move from one circuit to another in a radio system. The electrons in motion in a circuit remain in that circuit all

. . . and obviously cannot ever get into the transmitting antenna. Only magnetism, acting in the microphone transformer, links the microphone circuit with the grid circuit. It is magnetism which induces in our grid circuit the audio signal voltage which acts on the grid and cathode of the tube.

"You might at first think that the C battery would produce a continuous movement of electrons in one direction through the first grid circuit in Fig. 9, but a little study will show why this does not happen.

"First of all, we learned in the previous lesson that the *negative* terminal of a battery tries to send electrons through the circuit to the *positive* battery terminal. With the C battery connected as shown, its negative terminal will try to make electrons flow from the grid to the cathode through

the tube, so they will get to the + terminal of the C battery. No electrons can flow this way, however, because the grid does not emit any electrons. Electrons emitted from a heated cathode are the only ones which can bridge across the gaps between the electrodes of ordinary radio tubes.

"This is an idea well worth remembering, so let's repeat it in simpler form: When the grid of a vacuum tube is *negative*, there can be *no continuous flow* of electrons in the grid circuit.

"The electrons in the plate circuit are influenced by the grid circuit electrons, but there is no movement of electrons between the two circuits through the tube. We might think of these grid circuit electrons as traffic officers stationed on the grid wires of the tube, regulating the electron traffic which passes from cathode to plate in the plate circuit.

"Thus, you can see how a few electrons at the grid of a tube can control a much greater number of electrons in the plate circuit. In other words, a weak audio signal at the grid gives us a much stronger audio signal in the plate circuit. The stronger signal will behave exactly like the weak signal—that is, will have the same frequency.

"The B battery in the plate circuit

is connected to make the plate *positive* with respect to the electron-emitting cathode, so the plate attracts electrons and we have a continuous electron flow from the cathode to the plate and around the complete plate circuit—a *d.c. plate current*.

"We thus have both a d.c. plate current and an audio signal flowing through the plate circuit at the same time. There is nothing unusual about this; the audio signal voltage in the grid circuit simply makes the d.c. plate current fluctuate above and below its steady value, and these *momentary changes* in the amount of electron flow (alternate increases and decreases) form the audio signal in the plate circuit.

"The B battery provides the additional power needed to make the audio signal in the plate circuit stronger than the signal in the grid circuit. In fact, no tube can generate electrical power; it merely utilizes the power from a voltage source in such a way as to amplify or strengthen a signal.

"It is this audio signal flowing through the primary of the second transformer (Fig. 9) which induces an audio signal voltage in the secondary of the transformer (in the second grid circuit).

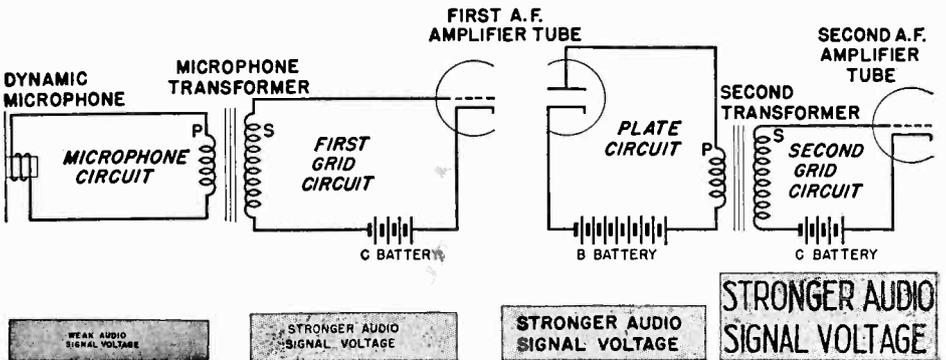
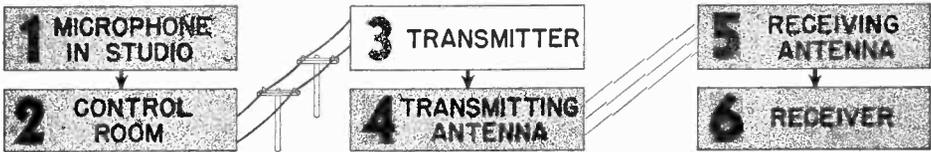


FIG. 9. This diagram shows that electrons in the microphone circuit cannot possibly travel through the other circuits of a radio system. It is the *effects* of these electron movements which are transferred from circuit to circuit to give increasingly stronger signals.

“Each other circuit in this audio amplifier likewise has its own set of electrons and its own audio signal. Tubes and transformers make elec-

trons in one circuit act upon electrons in the next circuit, and thus we secure the desired increasingly stronger audio signal in each succeeding circuit.”



**3 TRANSMITTER.** We leave the monitor operator in his control room, and follow the audio signal through the telephone lines which run to the transmitter.

By the time the audio signal has arrived at the transmitter, its voltage (strength) has dropped considerably due to the long trip through the telephone line. This is why the audio signal is sent through several more voltage-boosting a.f. amplifier stages after it reaches the transmitter.

**R.F. Carrier Signal.** No matter how strong we make the audio signal voltage, however, it will not produce radio waves when fed into the transmitting antenna. The only thing that will produce radio waves which will “go places” is a high-frequency signal. This required higher frequency is known as a *radio frequency*, and is abbreviated *r.f.* A radio frequency signal is usually higher than 100,000 cycles per second, whereas, you will remember, an audio frequency signal is rarely higher than 15,000 cycles.

A radio frequency signal can be used to *carry* an audio signal through space. When used in this way, the r.f. signal is called the *r.f. carrier signal*. Each radio station in the United States is assigned a definite carrier frequency value by the Federal Communications Commission.

**Crystal Oscillator.** The r.f. carrier signal for a modern broadcast

transmitter is produced by a special radio vacuum tube circuit known as a *crystal oscillator*. This circuit contains three important radio parts which we have not yet taken up—a crystal, a variable tuning condenser and an r.f. transformer. It is the crystal which produces the r.f. carrier signal, so we will first see what the crystal is like.

**The Crystal.** A modern radio transmitter is kept on its assigned carrier frequency by a carefully ground piece of natural quartz crystal about one inch square, which is known to radio men as a *crystal*. One of these finished crystals is shown in Fig. 10, alongside a block of natural quartz from which radio crystals of



Courtesy General Electric Co.

FIG. 10. A quartz crystal slab which is ready for use in a broadcast transmitter is here held by tweezers alongside a piece of quartz as found in its natural state.

this type are cut. Natural quartz is mined in South America, in Brazil.

**Frequency of a Crystal.** When a piece of quartz crystal is jarred or tapped, it will vibrate like a strip of stiff spring steel. Each crystal has its own natural frequency of vibration, which depends upon the thickness of the crystal. For example, a crystal one-fifth of an inch thick has a natural vibration frequency of about 600,000 cycles.

A vibrating crystal has another peculiar characteristic—it produces an a.c. voltage between its faces whenever it vibrates. This a.c. voltage has the same frequency as that at which the crystal is vibrating. Still more interesting is the fact that we can reverse this characteristic—we can make the crystal vibrate by connecting it to an a.c. voltage source.

**Crystal Oscillator Circuit.** A simplified version of a crystal oscillator circuit is shown in Fig. 11. The crystal is connected between the grid and cathode of the tube, while the B battery and a tuning circuit are connected between the plate and cathode.

The coil in the tuning circuit is the primary winding of an r.f. transformer, which consists simply of two coils of wire wound side by side or one over the other on a length of paper or fiber tubing called a coil form. There is no iron core inside the coil form.

The variable condenser consists of two sets of metal plates, insulated from each other. One set of plates is fixed. The other set is mounted on a shaft which turns or rotates in such a way that its plates can be meshed any desired amount with the fixed plates. (Two sets of plates are meshed when they fit between each other without touching.)

When the crystal oscillator circuit of Fig. 11 is operating properly, the crystal is vibrating at its natural fre-

quency. Since this is generally higher than 100,000 cycles, it is a *radio frequency*. The resulting weak r.f. voltage produced by the vibrating crystal acts between the grid and cathode of the tube. The tube and its B battery boost the strength of this r.f. signal, just as in an audio amplifier stage, so we have a stronger r.f. signal voltage in the plate circuit.

Some of the r.f. signal voltage in the

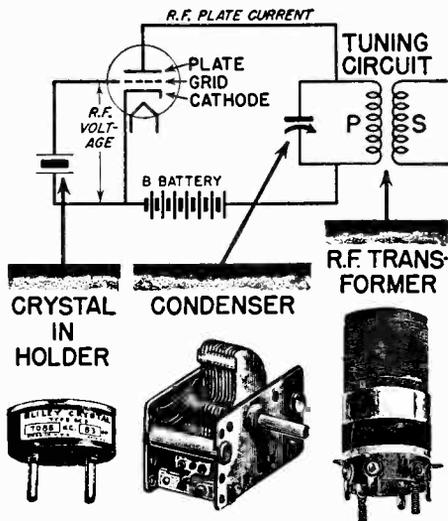


FIG. 11. Simplified diagram of a crystal oscillator circuit, with illustrations of the three parts you have not yet studied. Only the more important parts are shown, because the other parts will be taken up in later lessons. Thus, by-pass condensers are omitted, and no C supply is shown. The condenser is a tuning condenser.

plate circuit feeds back through the tube and gives the crystal a little "push" during each cycle of vibration. A small part of the power obtained from the B battery thus serves to keep the crystal vibrating. The explanation of how the signal feeds back comes later in your Course.

By rotating the shaft of the variable condenser, the tuning circuit can be adjusted for best possible operation of the crystal oscillator. Maximum r.f.

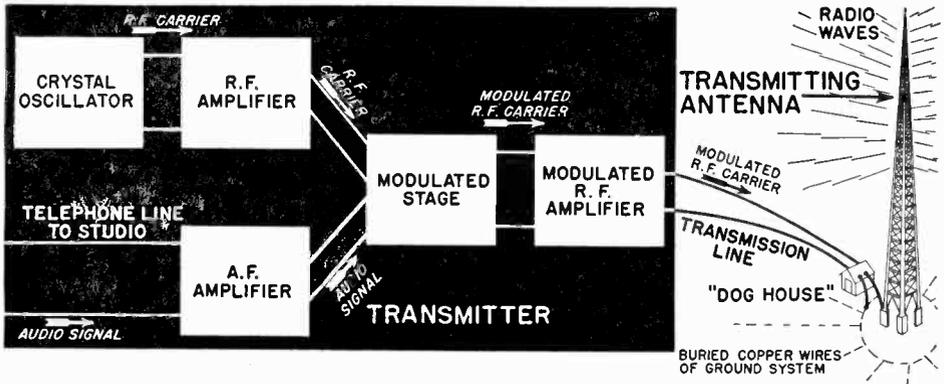


FIG. 12. All the main sections of a broadcast transmitter are shown here, along with the basic connections to the transmitting antenna.

current then flows through the coil of the tuning circuit, and this current induces a maximum r.f. voltage in the secondary of the r.f. transformer, for transfer to the next section of the transmitter. This r.f. voltage is still much too weak for broadcast purposes, however, so this r.f. carrier signal must be sent through a number of r.f. amplifier stages to build up its strength.

**R. F. Amplifier.** At this point, a "picture" of the main sections of a complete transmitter will help you to understand how the audio signal and the r.f. carrier get together. This picture is given in the block diagram in Fig. 12.

You will recognize the crystal oscillator and the a.f. amplifier immediately, for you have already studied these sections. The *modulated stage*

is the highly important one which combines the audio signal and the r.f. carrier signal to produce the modulated r.f. carrier signal. The *r.f. amplifier*, located between the crystal oscillator and the modulated stage, boosts the r.f. output voltage of the crystal oscillator. The *modulated r.f. amplifier* is the most interesting of all sections, for it contains huge air or water-cooled output tubes which strengthen the modulated r.f. carrier signal sufficiently so it can be fed over the transmission line and through the *dog house* to the huge steel transmitting antenna tower.

Even in a moderate-size broadcasting station, one or more vacuum tube stages are required in the r.f. amplifier. These are all essentially alike, so only the first r.f. amplifier stage is shown in Fig. 13. Its operation is the same

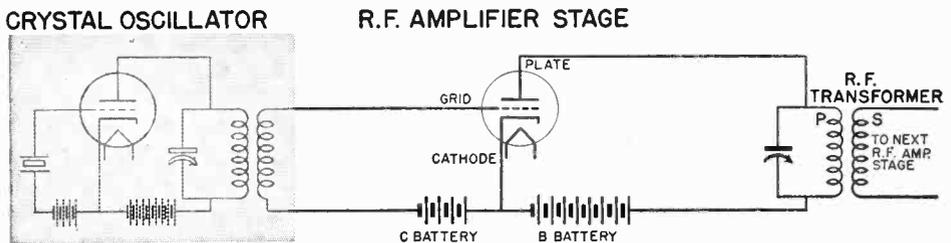


FIG. 13. Simplified circuit diagram of the first r.f. amplifier stage in a transmitter, with its connections to the crystal oscillator.



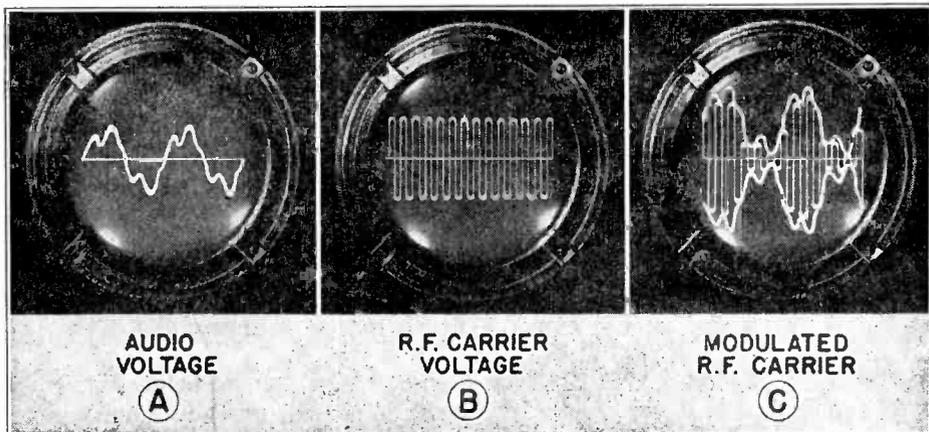


FIG. 15. These three patterns give the complete story of the modulated stage in a transmitter, as it would appear on the screen of a cathode ray oscilloscope.

In Fig. 15B is the pattern for the r.f. carrier voltage which is also fed into the modulated stage. Its loops are close together because, as you will remember, this r.f. voltage reverses its polarity many more times per second than does an audio voltage.

The most interesting pattern of all is Fig. 15C, which shows how the *modulated* r.f. carrier voltage, which we obtain from the modulated stage, is varying in strength from instant to instant. The farther the pattern swings above and below the horizontal line at a particular instant, the greater is the strength of the signal at that instant. The pattern of our audio voltage is clearly recognizable both above and below the horizontal zero-voltage line, proving definitely that our radio program is still present at the output of the modulated stage. Thus do electrons in a cathode ray oscilloscope tube make it possible for us to find out what electrons in other circuits are doing.

The modulated r.f. plate current flows through primary *P* of the second r.f. transformer in Fig. 14, and induces in secondary *S* the desired *modulated r.f. voltage*. At last we have an audio

signal "riding" on an r.f. carrier—the required combination for producing radio waves which will carry radio programs for long distances through space.

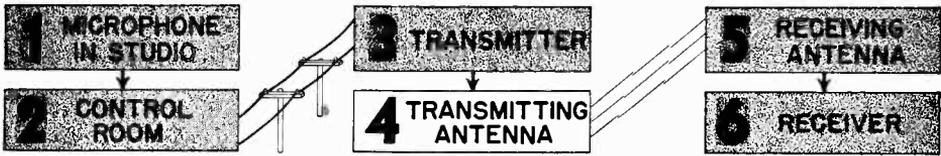
In low and medium-power transmitters, the modulated r.f. carrier signal is often fed directly to the transmitting antenna without additional amplification.

In large transmitters, however, the modulated signal is usually fed into a *modulated r.f. amplifier*. This contains one or more amplifier stages much like the r.f. amplifier stage you have already studied, except that now we are using much larger tubes.

Our signal acquires more power (more strength) each time it passes through an amplifier. In a large broadcast transmitter, the huge vacuum tubes in the modulated r.f. amplifier stage handle so much signal power and take so much power from the B voltage source that the tubes must be cooled either by blower fans or by a circulating water system. This is particularly true of the final tubes in the transmitter; these must handle the government-assigned output power

of the transmitter, which for most U. S. broadcast stations is some power value between 250 watts and 50,000 watts. (A *watt* is a unit used for

measuring quantities of electric power. The antenna of a 100-watt station gets the same quantity of power as does a 100-watt lamp bulb.)



#### 4 TRANSMITTING ANTENNA.

The modulated r.f. carrier signal at the output of a radio transmitter is fed to the transmitting antenna tower over a special two-wire transmission line which runs through the tuning house or "doghouse," essentially as was illustrated in Fig. 12. One of the transmission line wires goes to the base of the huge steel antenna tower. This tower is insulated (electrically separated) from the earth by means of insulators. The other transmission line wire goes to a network of buried copper wires which radiate outward for hundreds of feet from the tower like the spokes of a wagon wheel. These wires serve as the *ground* connection.

The transmitting antenna tower and the nearby ground area together act like a tuning circuit containing a coil and condenser. This means that maximum signal current will flow through the antenna tower when the tower is tuned exactly to the r.f. carrier frequency of our transmitter.

An antenna can be tuned to a definite frequency simply by adjusting the height of the antenna tower. Engineers figure out the height which will approximately give the desired tuning, then build the tower and provide at its top a telescoping metal pipe or mast which can be adjusted to increase or decrease the height of the antenna.

The tuning house or doghouse may contain an r.f. transformer with a con-

denser placed across each of its coils so as to provide two additional tuning circuits. These insure that the antenna will receive the maximum possible amount of signal power from the transmission line.

Remember, our modulated r.f. carrier signal in the transmitting antenna is simply a combination of our desired audio signal and its r.f. carrier.

**Producing Radio Waves.** In the previous lesson, you learned that whenever electrons *move* through a wire, they produce a *magnetic field* around the wire.

In addition to this, electrons have their own *electric fields*, consisting of *electric lines of force* which represent the *directions* in which electrons will repel or attract other charged objects.

When the electron flow through a transmitting antenna changes in amount or direction, both the electric and magnetic fields of the electrons will travel out in all directions from the tower at the speed of light. These moving electric and magnetic fields make up the *electromagnetic waves* which are known as *radio waves*.

During intervals of silence when no audio signal is produced by the microphone in the studio, the only signal which reaches the transmitting antenna is the r.f. carrier alone. We call it the *unmodulated* r.f. carrier signal, because it has no modulation (no audio signal riding on the r.f. carrier).

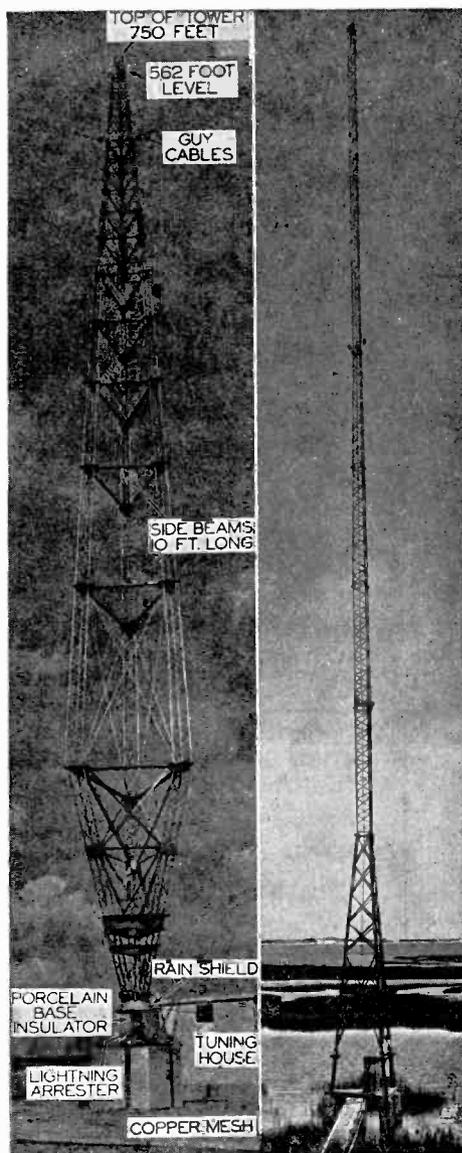
When an unmodulated r.f. carrier signal is being supplied to a transmitting antenna by a broadcast transmitter, the electron flow in the antenna tower is changing in direction and amount many hundred thousand times per second, essentially as indicated by the r.f. carrier voltage pattern in Fig. 15B. As a result, electric and magnetic fields which also vary in much this same way are being produced continuously by the transmitting antenna. These electric and magnetic fields travel off into space as radio waves.

Here is the important fact to realize—only the magnetic and electric fields leave the transmitting antenna. The electrons stay right in the antenna tower at all times.

When the microphone is picking up sound waves and producing audio signals, the r.f. carrier signal is *modulated* with an audio signal. The maximum strength of the electric and magnetic fields then varies from instant to instant much as in Fig. 15C, because the electron flow in the antenna tower is now varying in strength in this manner. A radio wave therefore varies in strength from instant to instant in such a way that it still carries the same radio program we traced through the transmitter stages.

When radio waves leave a transmitting antenna tower, they travel up into the sky as *sky waves*, and along the ground as *ground waves*. Those waves which go up into the sky are not lost, however; about 50 miles above the earth there are layers of electrons and gas particles which can bend or reflect radio waves back to the earth.

All long-distance radio reception, in short-wave bands as well as the broadcast band, is by means of sky waves. Ground waves give the best reception, but have a range of only about 100



Courtesy Chicago Tribune & WGN.

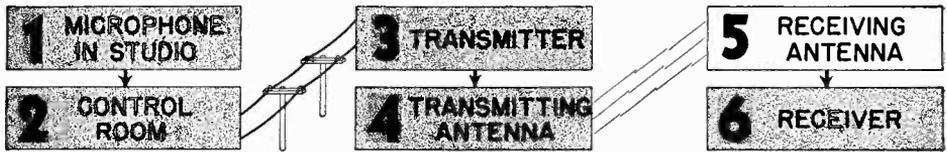
Courtesy Station WFOY.

Left: Famous 750-foot high steel transmitting antenna of radio station WGN in Chicago. A single porcelain insulator supports the 65-ton weight of the tower.

Right: Self-supporting 200-foot high steel tower of station WFOY, located in a salt-water marsh near St. Augustine, Florida. The dog house (tuning house) is directly under the tower, with a catwalk running from it to the transmitter.

miles at the most. Transmitting antennas for broadcast and police radio stations are designed to send out a high proportion of ground waves, while antennas for short-wave stations are designed to send out chiefly sky waves.

This completes the first chapter of our story of radio; in the next chapter, we discover how the receiving antenna and receiver reverse the entire transmitting process and give us sound waves again.



**5 RECEIVING ANTENNA.** When a radio wave arrives at a receiving antenna, the electric and magnetic fields of this radio wave make the free electrons in the antenna wire move back and forth *along the wire* at exactly the same frequency as that of the r.f. carrier in the transmitter. One electron transfers its motion to the

next through collision, so the electrons in the receiving antenna move back and forth in the same way as the electrons in the transmitting antenna. The result is a modulated r.f. current in the receiving antenna, like the current in the transmitting antenna but much weaker. The farther the radio waves have to travel, the weaker is the receiving antenna current.

The circuit for this modulated r.f. current is from the antenna down to the receiver, through the first r.f. coil, and then to the ground, as shown in Fig. 16.

The farther the receiving antenna is from the transmitting antenna, the weaker will be the radio wave by the time it reaches the receiving antenna, and the weaker will be the resulting modulated r.f. current in the antenna.

Each receiving antenna has a natural frequency of its own, at which the antenna current is the strongest for a given radio wave. Changing the length of the antenna changes its "best-results" frequency.

For ordinary broadcast band reception of entertainment programs, we pay little or no attention to the length of a receiving antenna. It is only when best possible reception of distant short-wave stations is desired that we begin measuring antenna lengths.

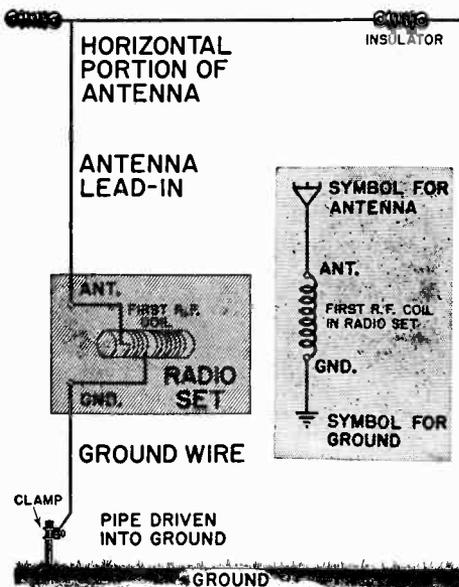
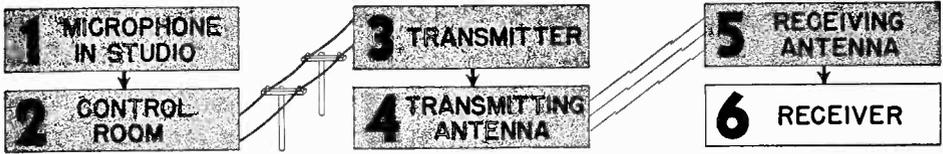


FIG. 16. Simple receiving antenna system, showing path from antenna to ground for modulated r.f. currents set up in the antenna by radio waves. Symbols used by radio men for the antenna and ground are shown at the right.



**6 RECEIVER.** Two types of receivers are in general use today. The simpler of these is the *tuned radio frequency receiver*, commonly referred to as a *t.r.f. receiver*. The other type is the *superheterodyne receiver*, which is now the leader because its special circuits give superior results. Naturally you will prefer to take up first the simpler of these receivers, so we will consider only t.r.f. receivers now.

As the diagram in Fig. 17 indicates, a t.r.f. receiver has only four sections, having the following purposes:

1. The *r.f. amplifier*, which builds up the strength of the signal from a desired station, and also keeps out undesired signals.
2. The *detector*, which separates the audio signal from the r.f. carrier.
3. The *audio amplifier*, which builds up the strength of the audio signal.
4. The *loudspeaker*, which converts the audio signal into sound.

**R.F. Amplifier.** If a radio man glances at the simplified circuit in

Fig. 17 he says immediately that our t.r.f. receiver has two r.f. amplifier stages, with three tuning circuits. The two r.f. amplifier stages boost the strength of the desired modulated r.f. carrier signal a tremendous amount. The three tuning circuits keep back all other modulated r.f. carrier signals which may be present in the receiving antenna, and also do their share toward boosting signal strength. (Signals from hundreds of different stations, some weak and some strong, may be present in a receiving antenna at any one time.)

The three tuning circuits in Fig. 17 are  $L_1-C_1$ ,  $L_2-C_2$  and  $L_3-C_3$ . (In radio, we use the letter *C* to represent a condenser, and the letter *L* to represent a coil. When there is more than one coil or more than one condenser,

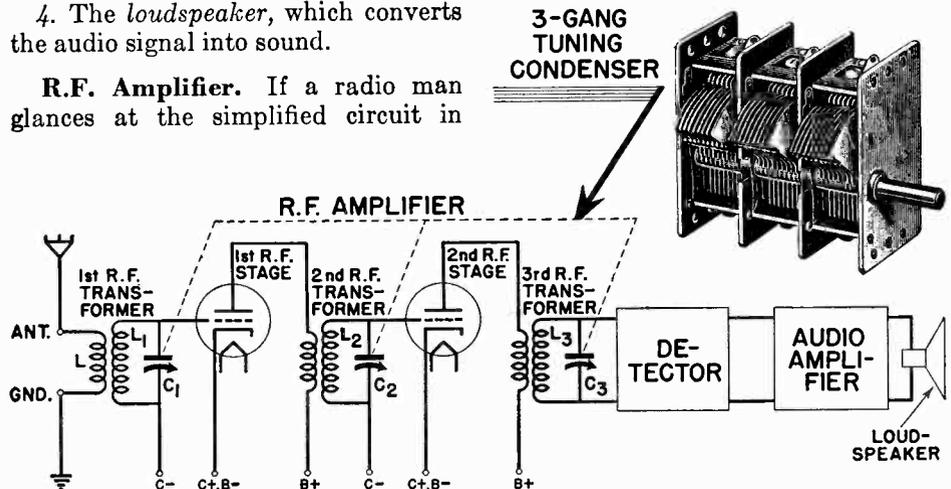
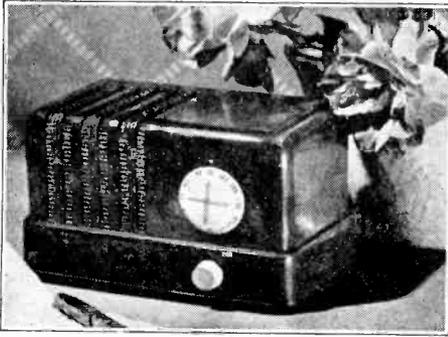


FIG. 17. Circuit arrangement of a typical t.r.f. receiver, with the two-stage r.f. amplifier circuit shown in simplified form. All d.c. circuits are completed through the power supply, which is not shown. All paths for signal currents are completed through condensers and other parts, which have been omitted here to simplify the diagram.



Courtesy General Electric Co.

Typical tuned radio frequency (t.r.f.) receiver. This set has five tubes, and uses circuits very similar to those which serve as examples in this lesson. Stations are tuned with the large dial-knob, while the set is turned off or on and the volume is adjusted with the smaller knob.

numbers are used after the letters to distinguish between the similar parts.) Each tuning circuit contains a coil and a condenser. The condensers are shown separately in the diagram in Fig. 17, but in the actual receiver these three condensers are combined into a single unit like that shown at the upper right in the diagram. The dotted lines connecting together the three tuning condenser symbols indicate that these condensers are on the same shaft and tuned by the same knob.

When we tune in a program with the tuning knob, we are actually adjusting all three condensers at the same time so they will tune all of the tuning circuits automatically to the carrier frequency we desire to receive, and at the same time reject the undesired signals.

In the early days of radio, single-control tuning was unknown. Each tuning circuit had its own separate tuning condenser. To tune in a station, it was necessary to adjust all of the tuning knobs, one at a time. Some sets had as many as five tuning condensers, each with its own knob, so it

was quite a feat to tune in a station in those days.

Getting back to our own t.r.f. circuit in Fig. 17, the antenna signal current flowing through coil  $L$  induces a signal voltage in the first tuning circuit,  $L_1-C_1$ . The tuning circuit transfers this weak voltage to the grid and cathode of the first r.f. tube (the C battery, not shown, completes the circuit from  $C-$  in the tuning circuit to  $C+$  at the cathode).

As a result of the amplifying ability of the tube and its B voltage source, the modulated r.f. signal current flowing in the plate circuit of the first r.f. stage is much stronger than the antenna signal current. In Fig. 18 is a

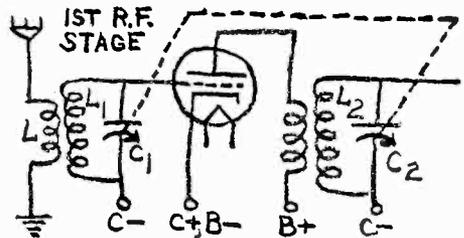


FIG. 18. The first r.f. stage of Fig. 17 is redrawn here as a guide for you to follow in preparing the rough sketch called for in Question 9. Your sketch can be even rougher than this; just be sure it shows the circuit correctly.

rough sketch showing this first r.f. stage as it might be drawn by a radio man.

The second r.f. amplifier stage and the two remaining tuning circuits provide still more boosting of our signal, so that the signal voltage in tuning circuit  $L_3-C_3$  is strong enough to be handled by the detector.

For the present, it is sufficient to understand that the r.f. amplifier in a t.r.f. receiver accepts and *steps up the voltage of a desired radio signal*, and also *keeps out undesired signals*. By using a sufficient number of tuned stages, complete rejection of undesired stations can be obtained.

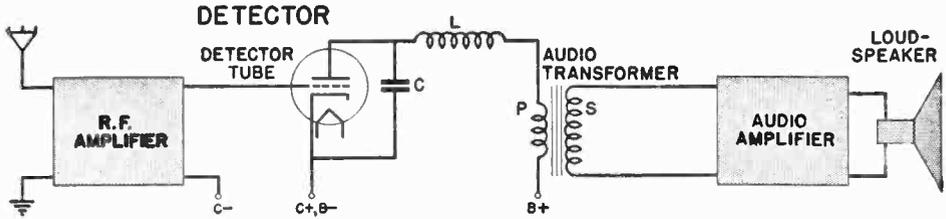


FIG. 19. Simplified detector circuit of a t.r.f. receiver.

**The Detector.** Now that the modulated r.f. signal has been built up sufficiently in strength, the r.f. carrier signal is no longer needed; in fact, we must get rid of the r.f. carrier now, so it will not interfere with the operation of the remaining stages. We separate the audio signal from its r.f. carrier with a vacuum tube operated in a special manner in the *detector* stage. A simplified version of one common type of detector circuit is shown in Fig. 19.

The modulated r.f. current which we feed into the detector is an alternating current, and its pattern on the screen of a cathode ray oscilloscope would be exactly the same as the modulated r.f. carrier pattern shown in Fig. 15C for the transmitting antenna.

The detector tube allows the input

signal voltage to act on the grid in one direction only. This has the effect of cutting off one entire half of this signal pattern, because it allows only half of each alternating current cycle to affect the plate current of the detector tube.

The signal pattern at the output of the detector tube would therefore be as shown in Fig. 20A. This pattern represents a d.c. voltage which is always acting in the same direction, but is varying in value from zero to a maximum at an r.f. rate (many hundred thousand times each second). The maximum value at each instant is determined by the *audio signal*.

The job of getting rid of the r.f. variations in the detector output signal is taken care of by two simple radio parts, condenser *C* and coil *L*.

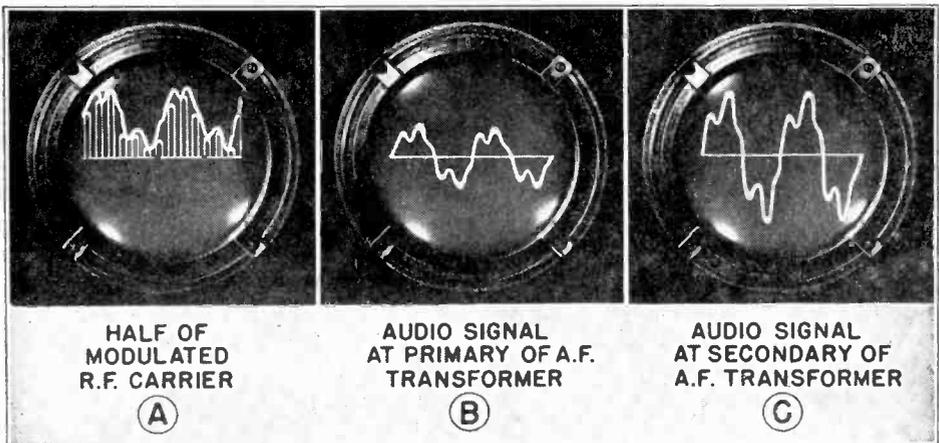


FIG. 20. Here is the story of how the audio signal is separated from its carrier in the detector stage of a t.r.f. receiver, as presented by patterns on the screen of a cathode ray oscilloscope.

The values (electrical characteristics) of these two parts are so chosen that condenser *C* lets r.f. signals pass through but blocks audio signals. Coil *L*, on the other hand, lets the audio signal pass through but blocks the r.f. signals. Together these two parts force r.f. signals to take the path through *C* from the plate to the cathode of the detector tube, and this is the end of the r.f. signal so far as our receiver is concerned.

The signal current which gets through coil *L* and also flows through the primary of the audio transformer in Fig. 19 is the *audio signal* we want. Its pattern is shown in Fig. 20B, and is simply the pattern of Fig. 20A with the vertical lines of the r.f. signal removed.

The audio transformer has more turns on its secondary than on its primary, and consequently it steps up the audio signal as well as transfers it to the next section, the audio amplifier. The pattern of our strengthened audio signal at the secondary of the audio transformer is shown in Fig. 20C. Notice that this strengthened signal swings much farther above and below the horizontal zero line than does the signal in Fig. 20B; this indicates the increased *strength* of the signal.

We thus secure from our detector circuit an audio signal like the original signal produced by the microphone. The one important fact to remember is that the detector in a radio receiver *separates the audio signal from its r.f. carrier*.

**The Audio Amplifier.** The audio output signal of the detector stage in a receiver is strong enough to operate headphones, but is usually too weak for loudspeaker operation.

We build up the strength of this audio signal in exactly the same way that we build up the strength of the

microphone output signal in the transmitter, by means of audio **amplifier** stages using vacuum tubes.

A t.r.f. receiver may have one, **two** or even more a.f. amplifier stages; the more stages there are, the greater will be the audio signal power fed to the loudspeaker and the louder will be the maximum volume obtainable from the receiver. The last audio amplifier stage is often called the *output stage*.

The iron-core transformer which transfers the audio signal from the output stage to the loudspeaker is called the *output transformer*. The connections are shown in Fig. 21.

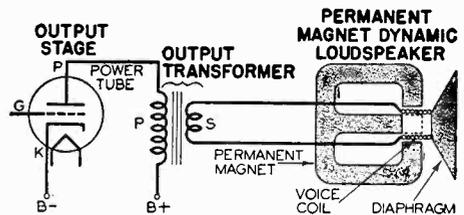


FIG. 21. Here is a simplified version of the very last stage in a radio receiver. It connects the power tube (the last tube in the audio amplifier) to the loudspeaker.

The output transformer is usually mounted directly on the loudspeaker, simply because this is a convenient position, but you will occasionally encounter sets which have the output transformer on the chassis.

For the present, all you need to know about the output transformer is that it supplies the correct signal voltage to the loudspeaker without too greatly affecting the operation of the output stage. Later, you will learn more about output transformers—why they are needed, why they are step-down transformers, and how they can affect the tone quality and volume of a receiver.

**The Loudspeaker.** The newest type of loudspeaker, known as a *permanent magnet* (p.m.) *dynamic loud-*

*speaker*, consists essentially of three parts: a powerful *permanent magnet*, a paper *cone*, and a *voice coil*. The voice coil is mounted at the center of the paper cone and is located between the poles of the permanent magnet, as indicated at the right in Fig. 21.

When audio frequency current is sent through the voice coil of a p.m. dynamic loudspeaker, a magnetic field is produced by the coil. This magnetic field reacts with the magnetic field of the permanent magnet, with the result that the voice coil moves either in or out. Since the diaphragm is attached to the coil, the diaphragm moves also.

The movements of the voice coil correspond exactly to the variations in the audio current passing through the coil, and consequently the diaphragm vibrates at audio frequencies and produces sound waves.

The sound waves produced by the loudspeaker are the same kind as those which acted on the microphone in the radio studio. They vary in the same manner and affect our ears in the same way, so we hear at the receiver the very same sounds which are in the studio. No wonder people say it is the *magic* of radio which transfers sounds thousands of miles in a split second!

A p.m. dynamic loudspeaker is similar to a dynamic microphone. In fact, in most intercommunication systems employed in factories and stores, a special loudspeaker of this type also serves as a microphone. The talk-listen switch automatically connects the loudspeaker to the input of an audio amplifier for microphone operation, and to the output of the audio amplifier for loudspeaker operation.

The loudspeaker is the only part in a radio receiver which moves continually, and even its motion cannot be seen because it is so fast. In fact, all the signals in a radio system

travel faster than a bolt of lightning.

Even though you can't see radio receiver parts move like parts in a locomotive or a threshing machine, you will soon be able to visualize plenty of action in radio circuits. As you become more familiar with the tubes, transformers, resistors, condensers and other parts used in radio, you will find yourself thinking of electrical actions in these parts—actions more exciting and more rapid than in any man-made engine. Radio will become *alive*, and you will develop an *enthusiasm* which makes your radio job a pleasure rather than a chore.

**Volume Control.** The average radio receiver, when tuned to a local station, is capable of producing far more volume than is required by listeners in homes. Furthermore, people like to vary the loudness of a radio program to suit the occasion; they want much more volume for a news broadcast than for background music which accompanies the conversational chatter of a dinner party. For these reasons, every radio receiver has a device which serves as the volume control. It is adjusted by a knob or lever on the front panel of the receiver.

When you turn down the volume of a radio receiver by rotating the volume control knob counter-clockwise, you are simply reducing the strength (voltage) of the signal which is being transferred from one point to another in the receiver. When you advance the volume control all the way, for reception of a distant or foreign station, you are simply using the full signal-boosting ability of the receiver.

**On-Off Switch.** Another control which you will find in every radio receiver is an *on-off switch* or *power switch*. This switch turns off the receiver by disconnecting it from the

power line or other source of power. Assuming that our t.r.f. receiver operates from an a.c. power line, this switch would be located on the front panel of the receiver and would be connected into one of the leads of the receiver power cord.

Oftentimes the volume control knob also operates the on-off switch. Turning this knob a slight amount clockwise clicks on the switch, and further turning increases the volume.

**Power Pack.** An a.c. power line provides only an a.c. voltage, but the tube electrodes in a receiver require d.c. voltages like those obtainable from batteries. In order to change the a.c. line voltage into the various d.c. and lower a.c. voltage values required by the tubes, a special vacuum tube stage known as the *power pack* is needed in every receiver which operates from an a.c. power line.

Although receivers may have different types of power packs or may operate directly from batteries, you will find that all receivers—aircraft sets, auto radios, portable receivers, and others—have essentially the same signal circuits. This is indeed fortunate, for it means that you need to study only a limited number of signal circuits.

The vacuum tube used in a power pack is known as a *rectifier tube*. Its job is to convert the a.c. voltage into a *pulsating* d.c. voltage, which always acts in the same direction even though it rises and falls in value. Condensers and a coil acting together in a *filter circuit* smooth out the variations in this pulsating d.c. voltage, so as to give the *pure* (unvarying) d.c. voltage required by the tubes.

The other important duty of the power pack is to provide the correct low a.c. voltage for each tube filament. In your very next lesson, you will study several ways in which this is done.

#### **Avoiding Distortion of Sound.**

The sound as reproduced by the loudspeaker of a radio receiver will be identical with the original sound picked up by the microphone in the broadcasting studio *only* if our audio signal varies in exactly the same way at all points on the long path between the studio and the home. One improper adjustment or one defect anywhere along this path can destroy the faithfulness of reproduction of the sound, making repairs necessary. This unnaturalness of a radio program coming from a loudspeaker is called *distortion*.

**Your Duty.** As a radio serviceman, it would be your duty to maintain the receiver end of this radio broadcasting chain in perfect condition. As a communications expert, you would be entrusted with the care of the transmitter end.

The only part of the radio signal path which is completely out of the hands of radio men is the space between the transmitting and receiving antennas. Here nature produces electrical storms which cause static interference. In addition, the reflecting layers high in the stratosphere shift up or down from time to time and cause fading of radio signals. But even these natural enemies of radio are gradually being conquered by radio engineers, with improved circuits, improved equipment, and a greater knowledge of radio.

# A Review

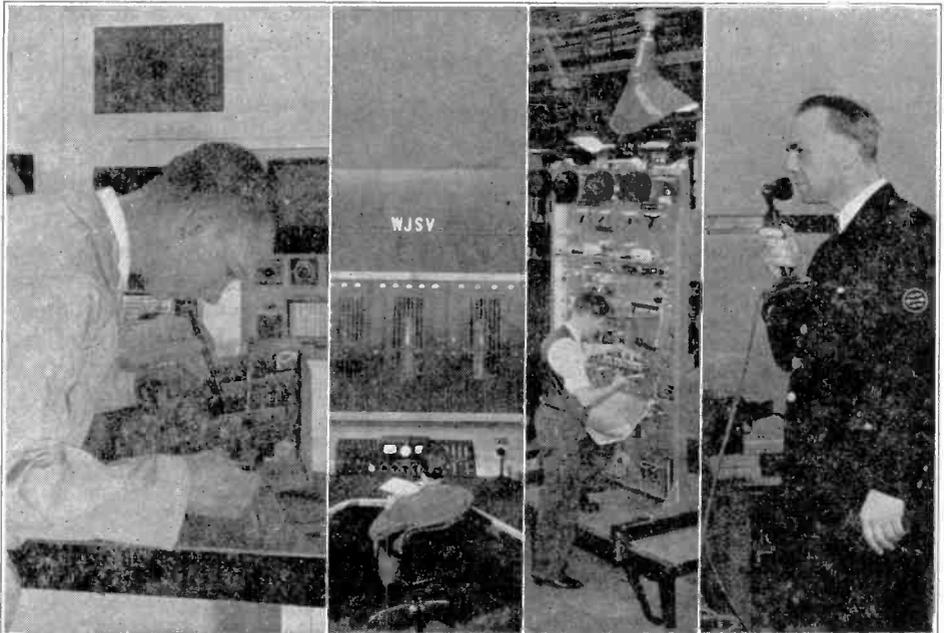
A brief review of the process of sending and receiving a radio broadcast will tie together in your mind the various steps involved in sending a sound radio program from the studio to your home.

Starting at the studio, the performers produce sound waves. These are picked up by the microphone and converted into an audio signal having the same characteristics as the sound waves. This audio signal travels through the microphone cable to the control room, where it is amplified many times by vacuum tube amplifier stages.

The amplified audio signal is then fed over telephone lines to the transmitter proper, which may be many miles from the control room and

studio. The audio signal undergoes additional amplification at the transmitter before it is fed into the modulated stage. A radio frequency carrier voltage, produced by a crystal oscillator and amplified by various r.f. amplifier stages, is also fed into the modulated stage.

The modulated r.f. carrier signal of the modulated stage is amplified further by r.f. amplifier stages, and the resulting strong modulated r.f. carrier signal is then fed to the transmitting antenna. The electrons vibrating in the transmitting antenna under the influence of this modulated r.f. carrier current produce vibrating electric and magnetic fields which travel out through space as radio waves.



**RADIO RECEIVER  
SERVICEMAN**

**BROADCAST  
OPERATOR**

**FACTORY  
TECHNICIAN**

**AIRCRAFT RADIO  
OPERATOR**

Here are four important radio jobs for which this lesson gives essential background information.

At a receiver location, the free electrons in the receiving antenna wire are set into vibration at the frequency of the radio waves. The result is a very weak modulated r.f. carrier current in the receiving antenna. This is amplified thousands of times by the r.f. amplifier stages in the receiver, after which the signal is sent into the detector.

The detector stage separates the audio signal from its r.f. carrier, and feeds only the audio signal into the audio amplifier. The audio amplifier in turn delivers a strong audio frequency current to the loudspeaker; this current causes the loudspeaker diaphragm to push and pull against the surrounding air, creating sound waves which are a true reproduction of the sound produced by the performers in the broadcasting studio.

### Looking Ahead

You have now obtained enough knowledge of radio to give you the necessary foundation for one of the most fascinating careers in the world—RADIO. Best of all, you didn't have to wade through fifteen or twenty lessons to get this information. You secured your foundation knowledge from just two lessons, because the N.R.I. Course gives you only *what you really need*.

You have now passed a real milestone, and you are certainly entitled to a satisfying feeling of accomplish-

ment. At the same time, you have prepared yourself for the next milestone, which is a fascinating study of individual radio parts, circuits and meters. Here you will take up circuit details which were purposely saved for this next lesson, and study many other interesting arrangements of radio parts which are sometimes used in place of the circuits already studied.

In the next lesson you get a wealth of practical information about meters, which are the most valuable tools of all for radio men. You will learn exactly how meters are used in real radio circuits, and how they tell you what the invisible electrons and radio signals are doing in your circuits. The ease with which you master this next lesson will be additional proof of how much you have learned in such a short time.

You will also learn about practical battery connections such as are used in portable receivers, and about different ways of changing voltages from one value to another.

With the confidence and "feel" of radio you have already developed, you will be anxious to start the next lesson as quickly as possible. Therefore, turn back to the first page now and take care of those last few steps in the Study Schedule, so you can start on the next lesson while the important facts of the first two lessons are still fresh in your mind.

# Lesson Questions

**Be sure to number your Answer Sheet 2FR-5.**

**Place your Student Number on every Answer Sheet.**

**Send in your answers for this lesson immediately after you finish them.** Doing this insures that the graded answers will reach you while the subject matter is still fresh in your mind, and you will get the greatest possible benefit from our speedy personal grading service. **Never hold up a set of lesson answers.**

1. What is the approximate range of frequencies (the lowest and highest) which can be heard by human ears?
2. What is the important purpose of a microphone?
3. Write on your answer sheet the missing words which will correctly complete this statement:

When the voice coil of a dynamic microphone moves in and out between the poles of the permanent magnet, an audio voltage is induced in the coil because the number of \_\_\_\_\_ which pass through the coil is being changed.

4. Do the electrons in the voice coil of a dynamic microphone ever get into the transmitting antenna?
5. Name the two important signals which are combined in the modulated stage of a broadcast transmitter to produce the modulated r.f. carrier signal.
6. What *two kinds of fields* make up the radio waves (electromagnetic waves) which travel away from a transmitting antenna in all directions?
7. Is the modulated r.f. current in the receiving antenna **STRONGER**, **WEAKER**, or the **SAME IN STRENGTH** as the modulated r.f. current in the transmitting antenna? (*Only one of these three possible answers is correct. Write down on your answer sheet the one which you think is correct.*)
8. What are the *two* purposes of the r.f. amplifier section of a t.r.f. receiver?
9. Make a rough pencil or pen sketch of the simplified circuit given in Fig. 18 for the first r.f. stage of a t.r.f. receiver. . (*Any convenient size will do.*)
10. What stage in a t.r.f. receiver separates the audio signal from its r.f. carrier signal?

## YOU'VE GOT COURAGE

An eight-year-old boy was discussing arithmetic with a friend. Said Johnny: "Teacher's gonna start us on *subtraction* tomorrow; wish I could stay home." But Harry laughed outright at Johnny's fears, and said: "Aw, I've had that and it's easy once you get started; what I'm worrying about is *multiplication!*"

It is natural even for grown men to feel like these boys—to fear most the things about which they know the least.

Did you know that some of the world's best speakers are always afraid when they get up to make a speech before a strange audience? Their *courage* gets them started, and in no time at all their fear changes to a confident enthusiasm which makes their talk a big success.

You've got courage! Use your courage to overcome normal fears, to carry you into each new radio subject and carry you over each difficulty. In no time at all, you will be looking forward to new subjects with intense interest—you will be *eager* to tackle new problems. Remember that each *conquered* difficulty brings you *one step closer to your goal of SUCCESS IN RADIO!*

J. E. SMITH.

# **SIMPLE RADIO CIRCUITS AND METERS**

**3FR-2**

**NATIONAL RADIO INSTITUTE**

ESTABLISHED 1914

**WASHINGTON, D. C.**



# STUDY SCHEDULE NO. 3

By dividing your study into the steps given below, you can master this part of your N.R.I. Course quickly and thoroughly. Check off each step when you finish it.

**How to Study Each Step.** Unless otherwise instructed, first read the pages specified for the step at your usual reading speed. Next, reread them slowly enough so you understand every paragraph. Go over the material as often and as slowly as you find helpful, endeavoring always to *understand* as well as remember. Finish up with one quick reading to tie the ideas together in your mind, then write down roughly on scrap paper your answers to the questions specified for that step. After finishing the lesson, copy all your answers neatly and in order on an Answer Sheet.

- 1. **Meters Tell What Circuits Are Doing.....Pages 1-2**  
Read this introductory section slowly just once. You will find out why meters are studied along with simple circuits. A few of the interesting ways in which you will use meters in your radio work are described.
- 2. **What Radio Men Measure.....Pages 3-5**  
In this section, important facts about voltages, currents and resistances which you measure in radio work are given in a straightforward, easy-to-remember manner. Answer Lesson Questions 1, 2 and 3.
- 3. **How Voltage and Resistance Affect Current in Simple Circuits..Pages 6-10**  
Here is information which you will use in practically every radio job, so give it all you've got! Answer Lesson Questions 4 and 5.
- 4. **Load Connections for Radio Circuits.....Pages 10-13**  
You'll learn how loads can be connected either in series or in parallel to a voltage source, and you'll find out how a voltage-dropping resistor uses up some of the source voltage. Answer Lesson Questions 6, 7 and 8.
- 5. **What Makes Meters Work.....Pages 14-21**  
Meters will mean a lot to you after you read this clear and interesting explanation of how the basic meter works, and how it is used with a few simple additional parts to give us all of the meters ordinarily used by radio men. Answer Lesson Question 9.
- 6. **Voltage Sources for Radio Circuits.....Pages 21-28**  
An interesting section dealing with storage batteries, dry cells, B batteries, and other voltage sources used in radio circuits. Answer Lesson Question 10, then read "Looking Ahead."
- 7. **Mail Your Answers for Lesson 3FR-2 to N.R.I. for Grading**
- 8. **Equipment Used by Servicemen.....RSM Booklet No. 3**  
This is another of the series of Booklets that are to give you practical, how-to-do-it training in fixing radios. Read this Booklet now, then keep it handy for future reference. The "shop training" you get from these RSM Booklets is going to prove valuable to you throughout your service career, so you will want to read and review these Booklets many times. NOTE: Some of the Study Schedules in future Lessons may refer to "Job Sheets" instead of RSM Booklets. Read the correspondingly numbered RSM Booklet in such cases. These Booklets are an improved and enlarged version of the older Job Sheets, and are replacing them.
- 9. **Start Studying the Next Lesson.**

# SIMPLE RADIO CIRCUITS AND METERS

## Meters Tell What Circuits Are Doing

**R**ADIO receivers, transmitters, public address systems and photo-electric controls are all alike in the sense that each one is made up of various combinations of *simple circuits*. Once you are familiar with the few simple circuits which are used over and over again in radio apparatus, you will have a good practical start toward understanding the complete circuits of receivers and other equipment.

This third lesson of the N.R.I. Course is a particularly important part of your training, because it gives the practical facts you want to know about these simple circuits in order to become a capable and well-trained radio man.

In addition, this third lesson explains in detail how different kinds of meters are used to find out what is going on in simple circuits. Since meters are the radio man's most valuable tools, and since this is definitely a practical training course in radio, it is logical that you study meters along with circuits.

As you study this lesson, you are going to be pleasantly surprised at how much the previous two lessons are already helping you. You have really acquired a wealth of information about radio, and you will already be using this knowledge.

To illustrate the importance of knowing how to use meters, we will now consider a few of the ways in which meters are used by radio servicemen and radio operators on actual jobs.

### How a Serviceman Uses Meters.

Let's look into the future for a few minutes now, and imagine that you are the owner of a prosperous radio servicing business on the main street of your home town. A customer brings in his table model radio receiver, say-



Serviceman using a d.c. voltmeter to check the plate voltages of tubes in a defective receiver. A few simple measurements with meters are often enough to locate a defective part.

ing: *"It went completely dead last night, right in the middle of my favorite program. This never happened before!"*

Taking the receiver to the test bench, you transfer the tubes one by one from the receiver to your tube tester, an instrument having a meter as its most important part. For every tube, the meter pointer swings to the green area marked *GOOD TUBE*.

The tubes are okay, so now you begin using your knowledge of radio fundamentals and servicing techniques. With the receiver turned on, you momentarily remove each tube in turn or touch its top cap. The resulting clicks or thumps in the loudspeaker for some but not all tubes tell you which stage of the receiver is at fault.

Now you reach for another instrument (*a voltmeter*) and measure the voltage at each electrode of the tube in this stage.

"Ah ha! No plate voltage. Something wrong here!", you exclaim in triumph. You have traced the trouble to one simple circuit, and a few quick measurements reveal the breakdown part which made the set go dead.

Your meters thus locate for you in a few minutes a part which is defective even though it looks perfectly good. After the meters have done their work, it takes just a few minutes longer to remove the defective part, put in a brand-new replacement part, then give the receiver a final complete check-up before handing it back to your customer and collecting your money.

Watching you at work in that imaginary radio shop of the future is a great deal like watching a doctor meet his daily assortment of patients. Just as the doctor uses thermometers, X-ray machines, stethoscopes and other instruments to supplement his medical knowledge in analyzing and treating human ailments, so do you use meters of various types to supplement your radio circuit knowledge in locating the causes of receiver ailments.

**How a Radio Operator Uses Meters.** The scene changes. Imagine that you are a radio operator temporarily assigned to the evening shift at a powerful broadcasting station.

From your control desk in the center of the transmitter room, you can see dozens of meters, arranged in rows on the metal panels of the transmitter. Each meter has its own particular story to tell you about one of the simple circuits in that transmitter.

Let's say it is 10 o'clock, with three more hours to keep the station on the air, when the pointer of one meter begins slowly creeping downward. The meter is telling you that one of the tubes is starting to fail.

You get a new tube and place it on your desk, ready for speedy replacement if necessary. But you don't put in the new tube now, because that would mean shutting off the transmitter. From experience, you know the tube will probably last quite a few hours yet anyway.

Fortunately, the tube does last. The announcer signs off for the night, you open up the master switches and disconnect all power from the transmitter, then you pull out the failing tube and put in the new tube.

Clearly, meters help you to keep a transmitter *on the air*. Without meters, you wouldn't know anything was wrong until a tube actually failed, and then would lose many precious minutes in finding out which tube had failed.

In addition to helping you locate trouble, the meters in a transmitter tell you when each circuit is correctly adjusted so the transmitter and the transmitting antenna will be operating at maximum efficiency. Even the monitor operator in the control room relies on a meter to aid in adjusting the volume control.



# What Radio Men Measure

There are only three *values* which radio men ordinarily need to measure in a radio circuit: *voltage*, *current* and *resistance*.

You already have a general idea of what *d.c.* and *a.c. voltages* are like, and are also somewhat familiar with both *alternating current* and *direct current*, so we need only review these subjects briefly before taking up the new subject of electrical *resistance*.

A great many technicians think of voltages, currents and resistances as *things which can be measured with meters*. If you mention 45 volts to them, they immediately think of a meter with its pointer at 45 volts. Practical slants like this are emphasized throughout your N.R.I. Course, along with essential technical information.

## Voltage

The electrical "pressure" which is capable of setting electrons into motion in a circuit is called *voltage*. Any device which produces a voltage is a *voltage source*. Storage batteries, dry cells, and generators are all voltage sources.

The stronger the voltage, the greater the effect it has on electrons. The term *volt* is used by radio and electrical men as a convenient unit for specifying the *strength* of a voltage source. For example, a 6-volt storage battery is stronger or *higher in voltage* than a 2-volt storage battery.

Voltages are measured with meters called *voltmeters*. Though scientists have elaborate ways of specifying how much a volt is, a radio man is satisfied to rely upon his voltmeter to tell how many volts a particular voltage source has.

**D.C. Voltage.** A voltage which

always moves electrons in the same direction is a *d.c. voltage*. It always has the same polarity. (One terminal is always + and the other always -).

The *d.c.* voltage sources most often used in radio are dry batteries, storage batteries, *d.c.* generators, and power packs. (A power pack converts *a.c.* voltage to *d.c.* voltage.)

**A.C. Voltage.** That type of voltage which reverses its polarity regularly, sending electrons through the circuit alternately in one direction and then in the opposite direction, is known as an *a.c. voltage*. The electrons do useful work, however, even though they just move back and forth many times each second without getting anywhere.

*A.C.* generators, ranging in size from the huge units in power plants down to small units driven by gas engines, are *a.c.* voltage sources. In addition, every radio signal source produces an *a.c.* voltage. Microphones, oscillator circuits, transformers and receiving antennas are thus all *a.c.* voltage sources.

**How A.C. and D.C. Voltages Act Together.** Here is a fact which is sometimes hard to believe: Both *a.c.* and *d.c.* voltages can exist in the same circuit.

When there are *a.c.* voltages in the same circuit with a *d.c.* voltage, *each voltage will act independently* just as if the others didn't exist. By using the proper meters, you can even measure these voltages independently.

Every amplifier stage is an example, for you have in the grid circuit a *d.c.* voltage (the C bias voltage), and an *a.c.* signal voltage provided by the signal source.

**Pulsating D.C. Voltage.** When a *d.c.* voltage is varying regularly in

value above and below a fixed value, without reversing its polarity, we have a *pulsating d.c. voltage*. It is simply a d.c. voltage combined with a smaller a.c. voltage, and these two voltages can be considered independently just as if they were separate voltages.

The most familiar example of a pulsating d.c. voltage is that produced by the rectifier tube in the power pack of a radio receiver. Remember, a pulsating d.c. voltage always has the same polarity, but changes in strength or value continuously.

**Other Names for Voltage.** You will sometimes find the terms *potential* and *electromotive force* (abbreviated *e.m.f.*) used in place of voltage. Potential means essentially the same as voltage, while e.m.f. is used only for voltages produced by true basic voltage sources like batteries or generators.

### Current

The electron movement produced by a voltage is called a *current*. More generally speaking, any flow of electricity is a current.

The more electrons we have moving past a given point per second, the *stronger* is the current. The *ampere* (pronounced *AM-peer*) is the basic unit used for specifying the *strength* of a current. For instance, an automobile storage battery supplies about 5 amperes of current to an auto radio receiver, about 200 amperes to the starter, and about 7 amperes to each headlight bulb.

In radio work, we deal so much with currents smaller than one ampere that we frequently use a smaller unit called the *milliampere* to specify the strength of a current. A milliampere is *one-thousandth of one ampere*, so 1,000 milliamperes is equal to one ampere.

Current values are measured with meters called *ammeters* or *milliam-*

*meters*. Ammeters indicate the current value in *amperes*. Milliammeters indicate the current value in *milliamperes*.

Since current-measuring meters give current values in amperes or thousandths of an ampere, there is no need for you to know how many electrons per second are involved in a current flow of 1 ampere. The actual figure is unimaginably large, and is presented here only as an oddity of science: *It takes 6.3 million million million electrons moving past a given point in one second to give a current of 1 ampere.*

**Direct Current.** A current which consists of electrons flowing always in the same direction is a *direct current*. It is produced by a d.c. voltage.

With direct current, the direction in which electrons are moving is of importance only when we need to figure out the correct polarity for connections to batteries, meters or radio parts. When direction is important, we speak of *electron flow* and specify a direction; when direction does not matter, we just speak of *current*.

Both batteries and radio tubes can serve as "signposts" in circuits, just as is shown in Fig. 1. These parts tell us which way electrons are moving. Thus, electrons always move away from the *negative* battery terminal,

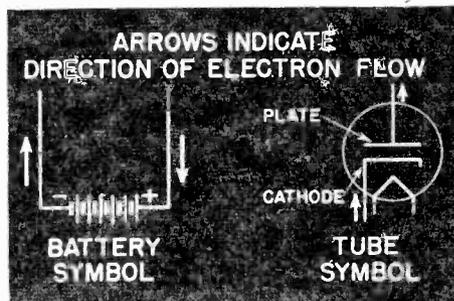


FIG. 1. Batteries and tubes are the signposts of radio circuits, for they tell you the directions of electron flow in d.c. circuits.

and move through the circuit towards the *positive* terminal of the battery.

In a tube circuit, the electron flow is always in such a direction that the electrons flow from the cathode to the plate in the tube.

**Alternating Current.** A current which consists of electrons which regularly reverse their direction of motion is an *alternating current*.

**Pulsating Direct Current.** A current which changes regularly in strength but still consists of electrons flowing always in the same direction is a *pulsating direct current*. It is produced by a pulsating d.c. voltage, which is really a combination of a d.c. voltage and an a.c. voltage.

### Resistance

The opposition which a wire, radio part or circuit offers to the flow of electrons is called *resistance*.

Everything has resistance, but metals like copper and aluminum have very low resistance. These metals offer little opposition to electron flow, and are therefore considered to be good paths for electrons and good conductors of electricity. Copper is the metal most frequently used when resistance should be very low, such as in the connecting wires of radio circuits. The resistance of a copper connecting wire is so low that it is usually neglected by radio men, and the wire is assumed to have *zero resistance*.

Materials like carbon and nichrome have quite a high resistance, and are used in radio circuits when it is necessary to reduce the strength of a current to a desired value.

**The Ohm.** The unit used for expressing the *amount* of resistance is the *ohm* (pronounced like "ome" in "home," and named after the scientist George Simon Ohm). Radio men use

meters called *ohmmeters* to tell how many *ohms* of resistance a part has.

**Resistors.** When we want to reduce the amount of direct current flowing through a circuit, we connect into the circuit a radio part called a *resistor*. (*Resistance* is the actual opposition to electron flow, while a *resistor* is a part which has *resistance* or opposition to electron flow.)

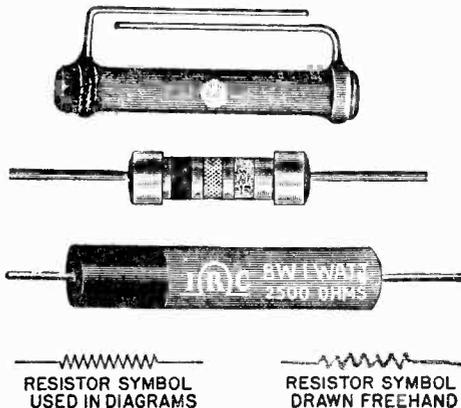


FIG. 2. Here are some illustrations of typical radio resistors. The symbol used to represent resistors is shown in two forms—as it appears in printed diagrams, and as it would be roughly sketched by a radio man.

Two types of resistors are widely used in radio apparatus. *Carbon-type resistors* are made from a mixture of powdered carbon and a special cement. *Wire-wound resistors* are usually made from nichrome wire.

Examples of resistors you will work with are shown in Fig. 2, along with the symbol used on diagrams to represent a resistor. Practice drawing this zig-zag symbol which says to radio men "*Here's a resistor.*"

The fifth lesson in your Course is devoted entirely to resistors, so we won't say much more about them now. For this lesson, all you need to remember is that a resistor is a radio part which *limits current*.

# How Voltage and Resistance Affect Current in Simple Circuits

Every complete circuit has at least two parts, a *voltage source* and a *load*. The source and load can be connected together either by wires or radio parts.

The two things which affect the amount of *current* flowing in a complete circuit are the *voltage* of the source and the *resistance* of the load and other parts in the circuit. With the aid of meters, we can find out exactly what happens to the current when changes are made in voltage or resistance.

Continuing with our practice of talking in the radio man's language, we shall use throughout this lesson two terms which describe the two basic ways of connecting radio parts together in circuits. Here are the meanings of these terms:

**In Series.** When radio parts are connected together end to end, so that electrons flow through one part after

another, we say that they are *in series*.

**In Parallel.** When radio parts are connected all to the same two terminals, so the electron flow divides among the parts, we say they are *in parallel*.

## How Voltage Affects Current

Suppose you have a simple circuit consisting of a  $1\frac{1}{2}$ -volt dry cell as voltage source and a 500-ohm resistor as load. You want to find out what happens to the *current* in the circuit when you increase the *voltage* by adding more dry cells *in series*.

Let us assume that for the current measurement you choose a milliammeter which reads from 0 to 10 milliamperes (abbreviated 0-10 ma.), and for the voltage measurement you choose a voltmeter which reads from 0 to 5 volts.

**One Cell.** You connect the volt-

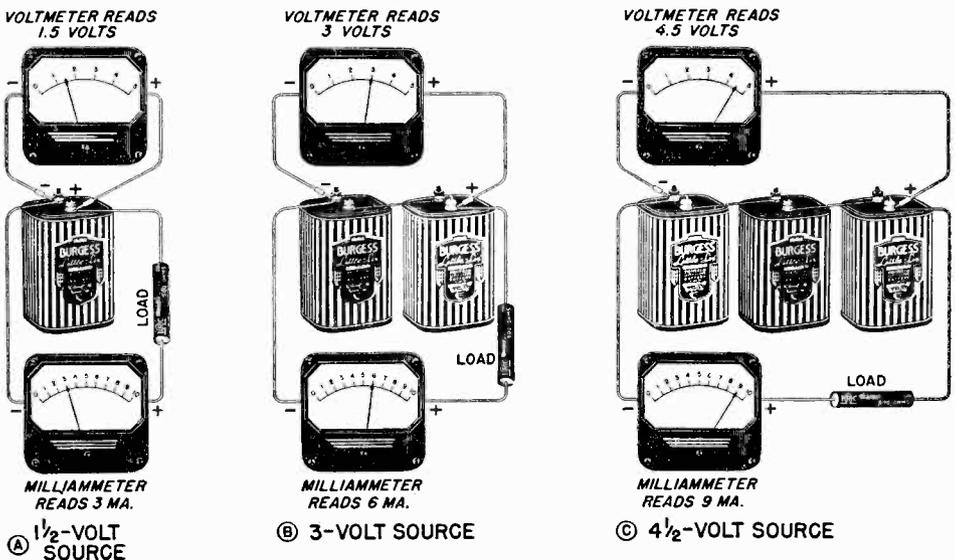


FIG. 3. These simple circuits with meters show you how voltage affects the current in a circuit.

SOURCE VOLTAGE	CURRENT
1½ VOLTS	3 MILLIAMPERES
3 VOLTS	6 MILLIAMPERES
4½ VOLTS	9 MILLIAMPERES

FIG. 4A. This table gives the results of the measurements pictured in Fig. 3.

meter across the two terminals of the cell which serves as the voltage source, and connect the milliammeter *in series* with the load resistor, as shown in Fig. 3A. The voltmeter reads 1½ volts and the milliammeter reads 3 milliamperes (usually abbreviated as 3 ma.), so you record these values on a piece of paper, somewhat as shown in Fig. 4A.

**Two Cells.** The next question is —what will happen to the current if you double the voltage? To find out, you add another cell in series to your circuit, as shown in Fig. 3B, and read the meters again. The voltmeter reads 3 volts (twice the first voltage) and the milliammeter reads 6 ma. (twice the first current), so you record this set of values also in Fig. 4A.

**Three Cells.** Suppose you are still in an experimental mood, and add another dry cell to your circuit, as in Fig. 3C. Now the voltmeter reads 4½ volts and the milliammeter reads 9 ma., so these values are recorded in Fig. 4A.

**Conclusion.** Now, what do these results mean? Well, by carefully studying and comparing the values in Fig. 4A, you could figure out for yourself the conclusion that *current increases when you increase the voltage.*

**Graph Diagrams.** It is not always easy to make correct conclusions just by studying a list of measured values, however. This is why radio men so often use a special diagram called a *graph* (pronounced *GRAFF*), which gives an easily-understood picture of the story hidden in a series of measured values.

**Bar Graphs.** For example, we could take the values obtained in our little experiment and use them as the basis for a bar graph like that shown in Fig. 4B. One glance at this graph is almost enough to tell you that the current goes up when the voltage is increased.

**Radio Graphs.** The bar graph has

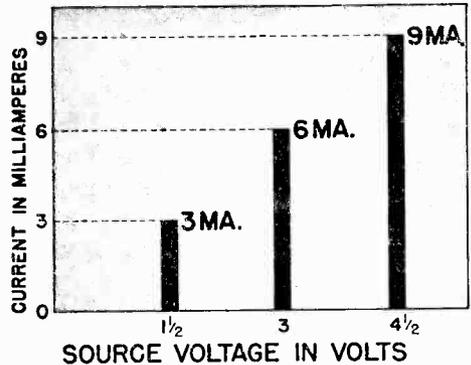


FIG. 4B. This graph presents the results of the Fig. 3 measurements in a manner more quickly understood than the table in Fig. 4A. You will often find bar graphs like this in newspapers, where they are used to show such things as the number of homes built each month of the year.

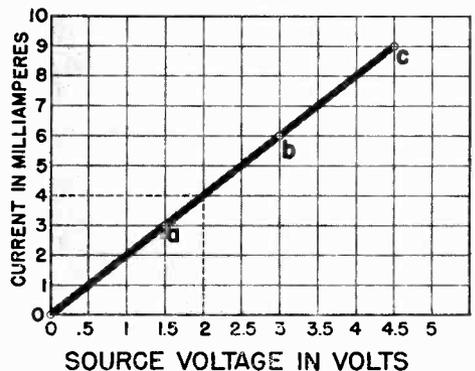


FIG. 4C. Here's the way a radio man would present the results of his measurements for the circuits of Fig. 3. This type of graph is easily drawn, and has the added advantage that it tells what the current will be for other values of voltage besides those measured.

one drawback for the radio man—it does not tell anything about values in between those actually measured. The radio man almost always uses a graph like that shown in Fig. 4C, because he can determine from this graph what the current will be for *any* value of voltage between 0 and 4.5 volts. Here is how this graph would be made up from the values in Fig. 4A.

**Making a Graph.** Starting with the first set of values in Fig. 4A, you locate 1.5 volts on the voltage scale at the bottom, and note the vertical (up and down) line which goes through this value. Next, you locate 3 on the current scale at the left and note the horizontal line which goes straight across the graph from this value. Move to the right along this line until you come to the vertical line going down to 1.5 volts, and at the crossing draw a dot or small circle, just as at *a* in Fig. 4C. This first point on your graph corresponds to the top of the shortest bar in Fig. 4B.

In the same way, you go up from 3 volts and to the right from 6 ma. until you reach point *b* on your graph and place a circle there. Finally, place a circle at *c* for 4.5 volts and 9 ma.

A bit of reasoning tells you that if the voltage is zero, no electrons will be flowing through the circuit, and the current will consequently be zero, so you place a small circle at *0* also.

Placing a ruler on your graph, you see that all four circles (*0*, *a*, *b* and *c*) are on a straight line. Therefore, as the final step in making this radio graph, you connect the four small circles together with a straight line.

**Radio Curves.** Whenever radio men connect together a series of points on a graph like this, they call the resulting line a *curve*. In fact, most of the graphs encountered in radio are

actually curves because the points are not on a straight line.

**What a Graph Tells.** A radio graph like that in Fig. 4C gives even more information than the original set of values. For instance, if you wanted to know what the current would be for a voltage of 2 volts in the circuit of Fig. 3, you would simply trace up from 2 volts till you hit the curve, then trace left to the current scale and read 4 ma. as the current. Try this yourself for 2.5 volts. If you get 5 ma. as the current for 2.5 volts, you have the right idea.

**Remember This.** The one important idea for you to get from this entire experiment is that *when you increase the voltage, the current increases too*. This basic rule works backwards just as well—when you *decrease* the voltage, the current goes *down*.

#### How Resistance Affects Current

**Another Experiment.** Now, let us take the very same circuit shown in Fig. 3C, and see what happens to the current in the circuit when we increase the resistance.

**One Meter is Enough.** The only meter we need for this next experiment is a milliammeter, because we will leave the voltage source value fixed at  $4\frac{1}{2}$  volts. Our new circuit arrangement is shown in Fig. 5A.

The milliammeter can be anywhere in the circuit, because *the same amount of current flows through every part of a complete series circuit like this*.

**One Resistor.** We start with one 500-ohm resistor, and get the same 9-ma. current as for the circuit of Fig. 3C.

**Two Resistors.** How can we increase the resistance in this circuit? Well, the simplest way is by adding more 500-ohm resistors. If we con-

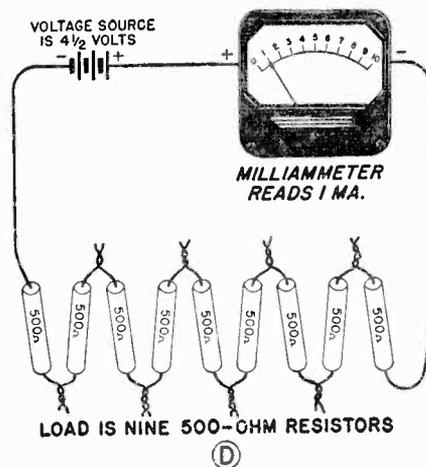
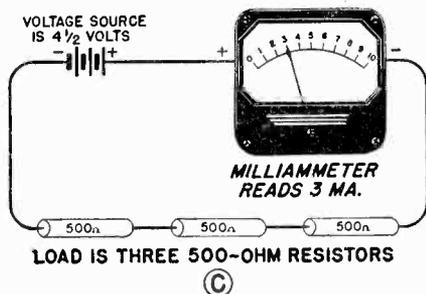
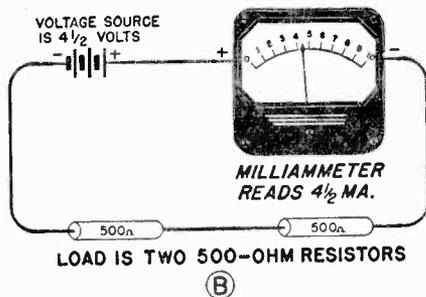
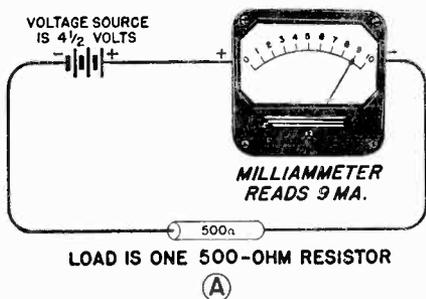


FIG. 5. These experimental circuits illustrate the basic fact that current GOES DOWN when you INCREASE the resistance in a circuit by adding resistors in series. If you start at circuit D and work back through C and B to A, these circuits will also illustrate the reverse condition, that current WILL INCREASE when you REDUCE the resistance in a complete circuit.

nect them *in series*, their resistance values will *add*.

Thus, two resistors in series as in Fig. 5B give us a total circuit resistance of  $500 + 500$ , or 1000 ohms. We now have twice as much resistance in the circuit as before, and the milliammeter now reads  $4\frac{1}{2}$  ma. This, incidentally, is exactly one-half of the reading we obtained in Fig. 5A.

**Three Resistors.** We continue increasing the resistance in the circuit, and make another measurement with three 500-ohm resistors in series to give a total of 1500 ohms. The milliammeter reads 3 ma. now, which is less than for either of the two previous measurements.

**Nine Resistors.** Finally, we take nine 500-ohm resistors in all, and con-

nect them in series as shown in Fig. 5D. Adding the values of these resistors together gives us now a total circuit resistance of 4500 ohms. With this resistance, the milliammeter reads 1 ma., which is still lower than before.

**The Reverse Situation.** If we start with high resistance in our circuit, as in Fig. 5D, we will have an initial current of 1 ma. Now if we *decrease* the amount of resistance in our complete circuit by removing six resistors, the current will *increase* to 3 ma., as in Fig. 5C. Decreasing the resistance still more, as in Figs. 5B and 5A, makes the current increase more yet.

**Conclusion.** The important facts to remember are that circuit current *decreases* when we *increase* the circuit

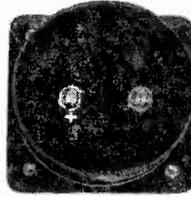
resistance, and circuit current *increases* when we *decrease* the circuit resistance.

### A. C. Measurements

Measurements in a.c. circuits are made in essentially the same way as d.c. measurements. Special a.c. meters must be used for a.c. measurements, and polarity of connections does not matter, but a.c. meters measure voltage and current just like d.c. meters. You're interested in actual values—

volts, amperes or milliamperes—and that's what a.c. meters show.

The amount of current flowing in an a.c. circuit depends upon the source *voltage* and the circuit *resistance*, just as it did for d.c. circuits.



Just two terminals—that's all you'll find at the back of practically any radio meter. On d.c. meters, the positive terminal will usually be marked +. The other terminal, not marked, is negative (-).

## Load Connections for Radio Circuits

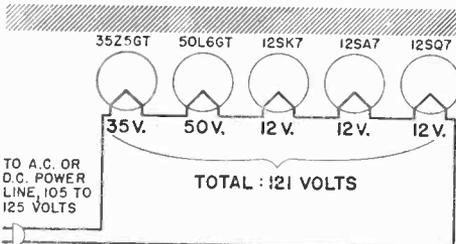
**Radio Loads.** In radio, loads are grouped together quite frequently. For example, a radio receiver may have from 4 to 10 or even more tubes, with each tube filament acting as a load and hence requiring a voltage. Instead of using a separate filament voltage source for each tube, we can connect the filaments together either in series or in parallel and use one common

voltage source (or two at the most) for all of the filaments.

In a series connection of loads, the same current flows through all of the loads, one after another. In a parallel connection of loads, a different current flows through each load. With this distinction between series and parallel connections in mind, let us consider examples of these connections in radio receivers.

**Loads in Series.** When loads are connected together *in series*, the same current flows through all the loads. The total available voltage is divided among the loads, however, in proportion to the resistance of each load. The *sum* of the individual load voltages is then equal to the total voltage available for all the loads. Let us see how this applies to such well-known loads as tube filaments.

**Filaments in Series.** The filaments of radio tubes can be connected together *in series* if all of the filaments have the *same current rating* and the available voltage is *high enough* to take care of *all the required filament voltages added together*. The filament



**FILAMENTS IN SERIES (UNIVERSAL A.C.-D.C. SET)**



**FIG. 6.** Practical example of loads connected in series. This example is taken from an actual receiver, and the type numbers of the tubes used in the receiver are given above the tube symbols. Notice how the first two numerals in the type number of a tube tell its filament voltage. This holds true only for newer tubes, however.

circuit arrangement shown in Fig. 6 for a 5-tube universal a.c.-d.c. receiver is a practical example of this. All the tubes have the same filament current ratings, and the power line provides a voltage which is just about equal to the sum of the required filament voltages. (A tube will operate satisfactorily even if the filament voltage is a bit higher or lower than the rated value.)

**Loads in Parallel.** When loads are connected together in parallel, all loads get the same voltage. The current flowing through each of the loads will depend upon the resistance of that load. Again we have tube filaments as examples, so let us consider the conditions under which these familiar loads can be connected in parallel.

**Filaments in Parallel.** The chief requirement for connecting the filaments of radio tubes together *in parallel* is that all of the tubes must have *the same filament voltage rating*. The available source must, of course, provide this voltage value, and in addition must be able to furnish a current

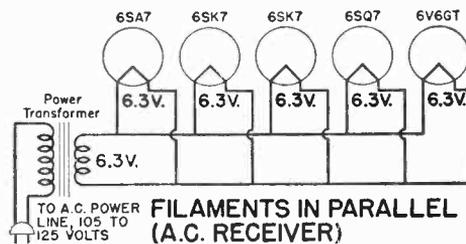


FIG. 7. Practical example of loads connected in parallel. The tube numbers are those of an actual receiver using this connection, so you are already studying real receiver circuits.

equal to the sum of all the rated filament currents.

The arrangement in Fig. 7 is a practical illustration of filaments connected in parallel. The five tubes each require a filament voltage of 6.3 volts. On the power transformer is a secondary winding which provides 6.3 volts,



Courtesy Zenith Radio Corp.

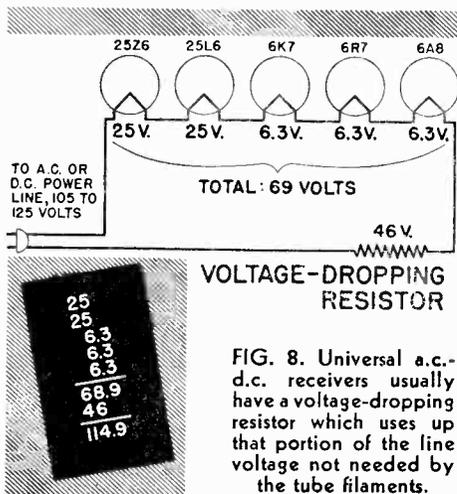
Before each man on this final inspection line of a huge radio factory are meters which tell whether or not each receiver performs as it should. No matter what radio factory you visit, you will see hundreds of meters in use, telling radio men what is going on in radio circuits.

and all five filaments are connected in parallel to this winding.

### Voltage-Dropping Resistors

**Too Much Voltage.** Very often in radio, we have a voltage source which provides a certain definite voltage value, and we have a load which requires a lower voltage value. In other words, the load requires only a portion of the source voltage, and there are no other loads which can use up the remainder of the voltage.

**Getting Rid of Voltage.** We take care of a surplus-voltage condition in radio work simply by inserting in the circuit a *resistor* which will use the undesired part of the source voltage. This resistor uses up surplus voltage in much the same way that automobile brakes use up part of the forward-



VOLTAGE-DROPPING RESISTOR

FIG. 8. Universal a.-d.c. receivers usually have a voltage-dropping resistor which uses up that portion of the line voltage not needed by the tube filaments.

NOTE: When a radio man encounters a number like 68.9, he calls it 69 because that is sufficiently accurate for practical radio purposes.

acting force of an automobile when we want to slow down.

**Here's an Example.** A practical example would be a universal a.c.-d.c. receiver having a tube line-up like that shown in Fig. 8. Here the line voltage provides somewhere between 105 and 125 volts, but the tube filament voltage requirements only add up to about 69 volts. Assuming for convenience a line voltage of 115 volts, this leaves 46 volts to be

used up somewhere in the circuit.

We use up this unwanted 46 volts with a resistor of the correct value, inserted in series with the filaments just as shown in Fig. 8. We can insert this resistor anywhere in the circuit, because the same current flows through all parts of this series circuit.

**What's Its Name?** When a resistor is used in the manner just described, the resistor is called a *voltage-dropping resistor*. In effect, it *drops* the source voltage to the correct value for the load.

**Voltage Rule.** This example illustrates one of the basic rules in radio, that in any one circuit you have to use up *all* of the source voltage. If you don't need it all in the load, you have to insert resistance. Otherwise there'll be too much voltage for the load and something will burn out.

**Voltage Drops.** A voltage drop is simply *some portion of a source voltage*, appearing across a load, resistor or other radio part. Whenever electrons flow through some part, a *voltage drop exists across that part*.

When there are no voltage-dropping resistors in a circuit and there is only one load, the entire source voltage will act on the load. The voltage drop across the load will in this case be ex-

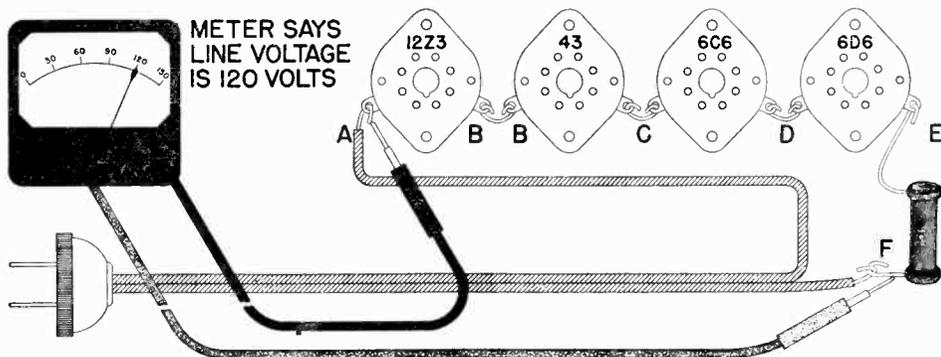


FIG. 9. This is how you would measure the power line voltage in a four-tube universal a.c.-d.c. receiver.

actly the same as the source voltage. An example of this is an ordinary electric lamp bulb. It is connected directly across the 115-volt voltage source, so the voltage drop across the lamp is likewise 115 volts.

**Experimental Proof.** Imagine that you are going to prove for yourself the basic rule *that all of the voltage drops across loads and resistors in a circuit do add up exactly to the source voltage.* You decide to use a universal a.c.-d.c. receiver having the tube arrangement shown in Fig. 9.

First of all, you measure the line voltage by connecting your voltmeter exactly as indicated on the diagram. Suppose you find it to be 120 volts. Now, if the basic rule is to be proved, you must show that the voltage drops across the tube filaments and across the resistor add up to this value.

You measure these individual volt-

MEASURE BETWEEN	VOLTAGE IS	COMMENTS
A and B-----	12 VOLTS	Filament voltage of Type 12Z3 tube
B and C-----	25 VOLTS	Filament voltage of Type 43 tube
C and D-----	6 VOLTS	Filament voltage of Type 6CG tube
D and E-----	6 VOLTS	Filament voltage of Type 6D6 tube
E and F-----	<u>71 VOLTS</u>	Voltage drop across resistor
<b>TOTAL</b> -----	<b>120 VOLTS</b>	This is equal to the Line Voltage

FIG. 10. Here are the results you would obtain if you made measurements on the receiver represented in Fig. 9 to prove that the sum of the voltage drops equals the line voltage.

ages one at a time and jot down their values in the manner shown in Fig. 10, then add up the five measured values. Yes, they do add to 120 volts, so you have proved conclusively that the resistor voltage drops in this series circuit add up to the source voltage.

**Terminals.** Whenever two termi-

nals are connected together by a wire, as are the two terminals marked B in Fig. 9, a connection can be made to either terminal for voltage-measuring purposes, because there is no noticeable voltage drop in the short length of wire which joins these terminals.

**Adding Voltage Drops.** The voltage across two or more parts is simply the sum of the individual voltage drops. Thus, if you measure between points A and C in Fig. 9, you will be measuring the sum of the voltage drops across the filaments of the types 12Z3 and 43 tubes. The data in Fig. 10 indicates that these voltages are 12 volts and 25 volts respectively, and consequently you would measure 37 volts between A and C.

The voltage drops across the resistors in a series circuit must add up to a value exactly equal to the source voltage. This is an important fact for you to remember.

**Polarity of Voltage Drops.** It is occasionally necessary to measure the voltage drop across a load with a voltmeter. Whenever we work with d.c. voltages, the + terminal of the voltmeter must go to the + terminal of the voltage being measured, and hence we may sometimes have to figure out the polarity of the voltage drop across a resistor.

When you know the direction in which electrons are flowing through a resistor or load, the polarity of the voltage drop across the part can be determined by the following simple rule:

*When electrons are flowing through a resistor, the polarity of the resistor terminals WITH RESPECT TO EACH OTHER is such that electrons ENTER the — terminal and LEAVE the + terminal.*

# What Makes Meters Work

**The Basic Meter.** The d.c. milliammeter is the heart of practically all meters used in ordinary radio work. Once you have a general idea of how this basic meter is constructed and how it works, you can very quickly learn how the d.c. milliammeter movement is used in d.c. voltmeters, ohmmeters, a.c. milliammeters, and a.c. voltmeters.

**Looking at a Meter.** From the outside, about all you can see of any meter are the terminals at the back, the glass window at the front through which can be seen the pointer which moves over the meter *scale*, and possibly also a part of the meter coil. Inside the case is where the differences between meters occur.

**It Works Like This.** The basic idea underlying a d.c. milliammeter can be expressed very simply. Suppose we have a permanent magnet of the modified horseshoe shape shown in Fig. 11A, with a much smaller permanent magnet pivoted between the poles of the large magnet.

Remembering the law of magnetism which says that like poles repel, you can see that when the small magnet is in the position shown in Fig. 11A, there will be a repelling action which

rotates the small magnet in the direction indicated by the curved arrow.

## D. C. Milliammeter

In a practical d.c. milliammeter, we have exactly the same situation except that our imaginary small pivoted magnet is replaced by a pivoted meter coil or electromagnet like that shown in Fig. 11B, through which we send the current to be measured.

The meter coil is pivoted on tiny jewel bearings so it will turn easily. A pointer attached to the top of the coil sweeps over the meter scale. Delicate spiral springs hold the coil in a position which places the pointer at zero on the scale when no current is flowing.

**How a Meter Coil Acts.** When current is sent through the meter coil, the coil becomes an electromagnet. For one direction of electron flow through the coil, this electromagnet might have the polarity indicated in Fig. 11B, which is the same as the polarity of the small pivoted magnet in Fig. 11A. The coil will react in the same way as the pivoted magnet; that is, the coil will rotate *clockwise*, and make the pointer move to the right.

The greater the current sent through

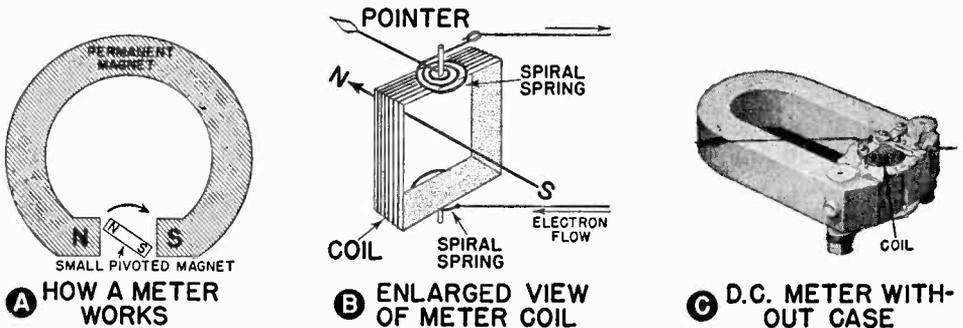


FIG. 11. How a d.c. milliammeter works, and what the inside of an actual meter looks like.

a meter coil, the stronger is the magnetic effect and the farther the coil rotates against the retarding action of the spiral springs. The meter coil always moves to a position at which the electromagnetic force which turns the coil is equal to the retarding force of the spring.

The divisions on the meter scale are spaced and numbered so the pointer will indicate the amount of current flowing through the meter coil.

**We Take a Meter Apart.** Figure 11C shows what you would see if you removed the case of a typical d.c. milliammeter. It should be pointed out, however, that radio men rarely if ever remove the case from a meter. Meters are even more delicate instruments than watches, and are easily damaged once the case is removed. Meters should be sent to the factory or to a meter repair expert when damaged.

**Uses for D.C. Milliammeters.** D.C. milliammeters are used by radio men to measure small values of direct current. Usually, milliammeters have coils which are designed to give a maximum (full-scale) reading for a current of 1 milliampere. Quite often, however, you will encounter milliammeters which are even more sensitive than this, and give full-scale readings on only a fraction of a milliampere.

**Warning!** Any current appreciably larger than the full-scale value of a milliammeter may damage the instrument. Excessively large currents will overheat the coil enough to melt the wire, and radio men then say that the meter is *burned out*.

Remember that whenever you use a milliammeter, make sure the current to be measured is within the safe current range of the meter.

**Second Warning!** Another equally important rule when using milliammeters is that a milliammeter must

always be placed *in series* with the circuit. In other words, you must actually cut into a circuit in order to connect a milliammeter properly. NEVER connect a milliammeter across a voltage source or across any part having a voltage drop, because this is almost certain to burn out the meter.

**Higher Meter Ranges.** Milliammeters and ammeters of all sizes employ essentially the same basic meter

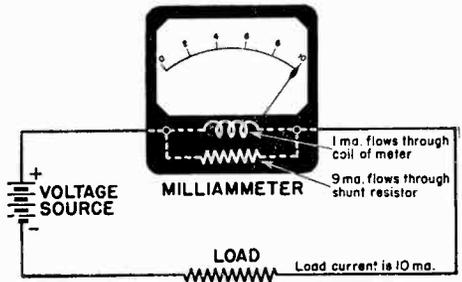


FIG. 12. This milliammeter, which reads currents up to 10 ma., is nothing more than a basic 0 to 1 ma. d.c. milliammeter with a shunt resistor.

construction illustrated in Fig. 11. Higher ranges are secured simply by adding to this basic meter a resistor which provides a short-cut or shunt path for some of the current being measured. The resistors used for this purpose are known as *shunts*, and are usually built right into the meter.

Figure 12 illustrates how a shunt is connected to a basic meter. By using various sizes of shunts with a basic 0 to 1 ma. d.c. milliammeter, you get milliammeters which can read from 0 to any higher maximum value such as 5 ma., 30 ma., 100 ma., or 500 ma. By using shunts still larger in dimensions, so that they have lower resistance and take a still greater portion of the current, you get ammeters which can read from 0 to any desired maximum current value in amperes.

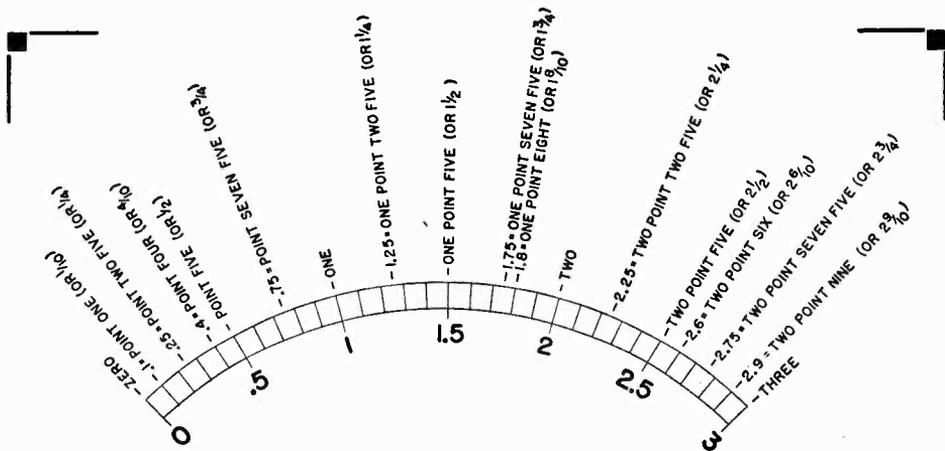


FIG. 13. The notes above this typical meter scale tell you what a radio man would read at various pointer positions. Figure out for yourself the value of each unmarked division line on the scale. Thus, the first line to the right of 1 is 1.1, the second is 1.2; the third (just past 1.25) is 1.3; the fourth (one line to the left of 1.5) is 1.4, etc.

**Polarity for Meters.** Radio men usually try to get the correct polarity the first time when connecting a d.c. milliammeter, because this saves quite a bit of time.

The correct polarity of connections for a milliammeter is such that the + terminal of the milliammeter goes to the + terminal of the voltage source, directly or through radio parts. The - terminal of the milliammeter then goes to the - terminal of the voltage source directly or through radio parts.

Just remember: *Plus to plus, and minus to minus.* This rule applies to d.c. ammeters, d.c. milliammeters and d.c. voltmeters.

If there is any doubt as to the correctness of a milliammeter connection, the radio man turns on the power for just a brief instant after making the connection and watches the meter pointer. If it starts to move backward (to the left of zero), he knows he must reverse the meter connections.

### Reading a Meter Scale

A careful study of the enlarged meter scale in Fig. 13 will give you a good start toward reading meters.

Once you understand what various pointer positions on this scale mean, you can apply your knowledge to practically any other meter scale which you may encounter in radio work.

**Marked Divisions.** When the pointer stops on a marked division such as .5, 1, 1.5, 2, 2.5 or 3, you simply read the printed number for that division line.

**Unmarked Divisions.** When the pointer stops on a small unmarked division, note the values of the marked divisions on either side, then figure out the value of the mark underneath the pointer just as you would figure out the value of one of the marks between 2 inches and 3 inches on a one-foot ruler.

**In Between Divisions.** If the pointer stops halfway between two unmarked divisions, determine the value of each of these unmarked scale divisions, then read a value halfway between them.

**Estimating is O.K.** It is entirely adequate in radio work to estimate meter readings roughly, when the pointer does not fall directly on a division line.

Study the diagram in Fig. 13 until you are sure you could give the value corresponding to each division line on this scale. The notes above the diagram will help you to determine what these values are and how they would be read by a radio man.

### D.C. Voltmeter

A d.c. voltmeter is nothing more than an ordinary d.c. milliammeter with a resistor placed in series with the meter coil.

This resistor uses up or drops a definite portion of the voltage being measured, and thus limits the current which can flow through the milliammeter.

**Multiplier Resistor.** The higher the resistance in ohms of a voltmeter resistor, the higher is the **voltage range** of the meter. The resistor in a voltmeter is usually called a *multiplier resistor*, for in a sense this resistor *multiplies* the useful range of the meter.

By using a printed scale made especially for use with the resistor in the meter, the pointer will correctly indicate the d.c. voltage which is being measured.

**Uses for Voltmeters.** Radio men use d.c. voltmeters to measure the d.c. voltages which exist between various tube electrodes and the chassis of a receiver, for these voltages are an important clue to trouble. In addition, d.c. voltmeters are used to check battery voltages, measure d.c. voltage drops across resistors and other radio parts, and for all other voltage measurements in direct current circuits.

**Polarity Is No Problem.** Although d.c. voltmeters must be connected with correct polarity before an accurate reading can be made, radio men ordinarily do not bother to figure out what the polarity is in case it is not marked.

They connect one voltmeter lead to the circuit, then tap the other voltmeter lead momentarily on the other circuit terminal while watching the meter pointer. If the meter pointer starts to move backward (to the left of zero on the scale), they disconnect the meter and reverse its connections.

**One Warning.** The important thing to remember when using a d.c. voltmeter is that the voltmeter range must be *higher* than the voltage being measured.

Whenever in doubt as to what a voltage may be, start with the highest-range voltmeter you have, and get a rough idea of what the voltage is before selecting a lower-range meter which gives a more accurate reading.

D.C. meters are built ruggedly



Courtesy Western Electric Co.

Meters tell this radio operator what's going on in each circuit, and thus help him to locate the defective tube or part speedily when trouble occurs. This transmitter is at the Springfield, Illinois, headquarters of the Illinois State Police radio system. It operates on a Federally-assigned frequency of 1,610,000 cycles (1610 kilocycles), which is only slightly higher than that of a broadcast station.

enough to stand considerable voltage overload. However, a voltage which is much too high for a meter may slam the pointer to the extreme right and bend it, or may even cause the meter coil to burn out.

### Ohmmeter

An ohmmeter is nothing more than a d.c. milliammeter with a small battery and a resistor mounted inside the meter and connected in series with the coil of the meter, essentially as shown in Fig. 14.

**Ohmmeter Uses.** An ohmmeter is used by radio men to measure the resistance values of resistors or other radio parts.

**Scale is Backward.** The commonest type of ohmmeter has a scale which reads from right to left, which is *backward* when compared to the other meters we have studied. In other words, zero ohms on the scale is at the extreme *right*, and the highest resistance value in ohms is at the left. A typical ohmmeter scale is shown on the meter in Fig. 14.

### Ohmmeter Zero Adjustment.

The resistor inside an ohmmeter is variable in value, and is adjusted by means of a knob or screwdriver adjustment so as to bring the pointer to 0 on the scale when the test probes are held together to give zero resistance. Since the ohmmeter battery voltage drops with use, the resistor must be adjusted from time to time. This is known as an *ohmmeter zero adjustment*, and takes only a few seconds to do.

**How an Ohmmeter Works.** When a resistor is placed between the test probes of the ohmmeter, this resistor reduces the amount of current which flows through the meter, and consequently the meter pointer takes a position somewhere on the scale. (Remember that the meter itself is still

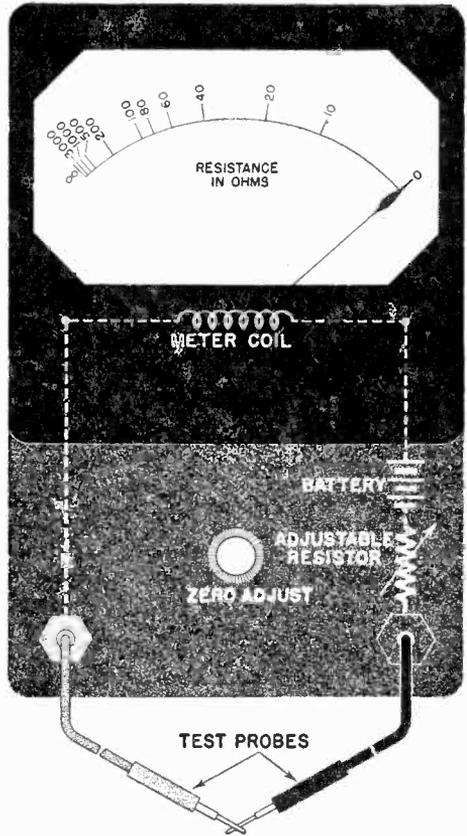


FIG. 14. Put a resistor and battery of the correct values in series with the coil of a d.c. milliammeter, and you have an ohmmeter for use in measuring resistance values. Polarity does not ordinarily matter when using an ohmmeter.

our basic d.c. milliammeter, with its pointer dropping to the extreme left when current is zero.)

The higher the value in ohms of the resistor being measured, the greater will be the total resistance in the ohmmeter circuit, the lower will be the current flowing through the circuit, and the closer to the left end of the scale will be the meter pointer. This is why resistance values on the scale of an ohmmeter must be zero at the extreme right and must increase gradually toward the left.

**What Infinity Means.** When the

test probes are apart, there is an *open circuit*. This is the same as having an *extremely high resistance* between the ohmmeter terminals. Under this condition, no current flows through the meter, and the pointer drops to the extreme left. We say that the resistance is then *infinitely* high, which means that it is the highest value it is possible to think of—many million million ohms. Sometimes we use the symbol  $\infty$ , called *infinity*, at the extreme left on the meter scale to designate an infinite resistance.

**Warning.** An ohmmeter must never be used in a circuit in which a voltage exists. Apparatus should always be disconnected from its power source before the ohmmeter is used. The ohmmeter has its own voltage source, and sends its own current through the part to be measured. Any additional voltage in the circuit would send extra current through the ohmmeter, possibly ruining the meter coil.

### A.C. Milliammeter

The a.c. milliammeter used most often in radio work is nothing more than an ordinary d.c. milliammeter with one additional part called a *rectifier*. This rectifier is a tiny unit mounted inside the meter case, and has the job of changing the alternating current being measured into a direct current. The d.c. milliammeter responds to this direct current, but its scale is made to read in terms of the alternating current being measured.

**Shunts.** The ranges of a.c. milliammeters are increased by using shunts, just as with d.c. milliammeters. These shunts are usually built right into the instrument.

**What the Meter Reads.** You will recall that an alternating current is one which varies continually from a maximum value flowing in one direc-

tion to a maximum value flowing in the opposite direction. What is the a.c. milliammeter going to read in a case like this?

Here is the answer—the meter reads the a.c. current value which will produce the same amount of heat as the corresponding direct current value. Thus, an alternating current of one ampere will produce *the same amount of heat* as a direct current of one ampere.

### A.C. Voltmeter

If you take an a.c. milliammeter and place a resistor in series with everything else inside the instrument, you have an *a.c. voltmeter*. The value of the resistor determines the range of the voltmeter.

**Uses.** A.C. voltmeters are used by radio men for checking the a.c. power line voltage, the filament voltages of tubes, and the various voltages across the secondary windings of the power transformer in a receiver or other piece of radio equipment. A.C. voltmeters are sometimes also used for measuring audio signal voltages.

**What the Meter Reads.** You will recall that an a.c. voltage is one which varies continually from a maximum value acting in one direction to a maximum value acting in the opposite direction. What is the a.c. meter going to read in this case?

The answer here is the same as for current—the meter reads the a.c. voltage value which will do just as much useful work as the corresponding d.c. voltage value. We call this a.c. voltmeter reading the *effective value* of the a.c. voltage. It so happens that the effective value of an a.c. voltage is equal to about seven-tenths of the maximum a.c. voltage acting in either direction.

**Effective Value of Voltage.** All you need to remember is that a.c. volt-

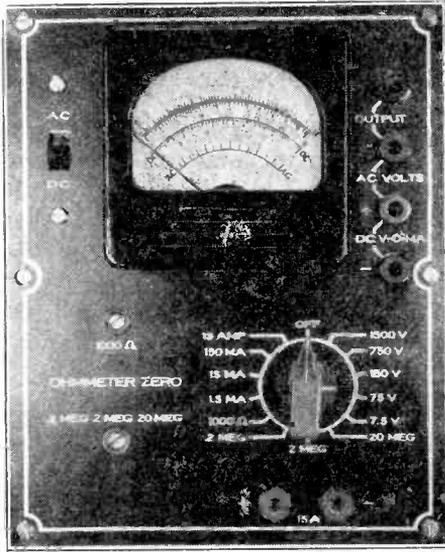


FIG. 15. Multimeter section of a typical professional all-purpose tester used by Radioicians.

meters measure the *effective* value of an a.c. voltage. The a.c. voltages you will deal with are effective values unless otherwise specified. Thus, if you have an ordinary a.c. power line coming to your home, its 115-volt value is an effective value, and the voltage actually reaches about 165 volts during the peaks of each cycle.

### Multimeters

Since every one of the meters ordinarily used in radio work consist of a d.c. milliammeter used alone or with resistors, rectifiers, shunts or batteries, it is only logical to make a single d.c. milliammeter serve for all purposes. This can be done by means of a switching arrangement which automatically connects the correct parts to the meter for a particular measurement. The resulting many-purpose instrument is known under various names, such as a *multimeter*, an *all-purpose tester*, a *multi-tester*, or a *volt-ohm-milliammeter*.

**All-Purpose Tester.** A typical ex-

ample of an all-purpose tester used by radio men is shown in Fig. 15. This one instrument is equal to twenty different separate meters, including five d.c. voltmeters with different ranges, five different a.c. voltmeters, three different d.c. milliammeters, one d.c. ammeter, four different ohmmeters and two different audio output voltage meters. This tester is thus capable of making practically all of the measurements required by radio men.

To prepare this all-purpose tester for use as a particular type of meter,

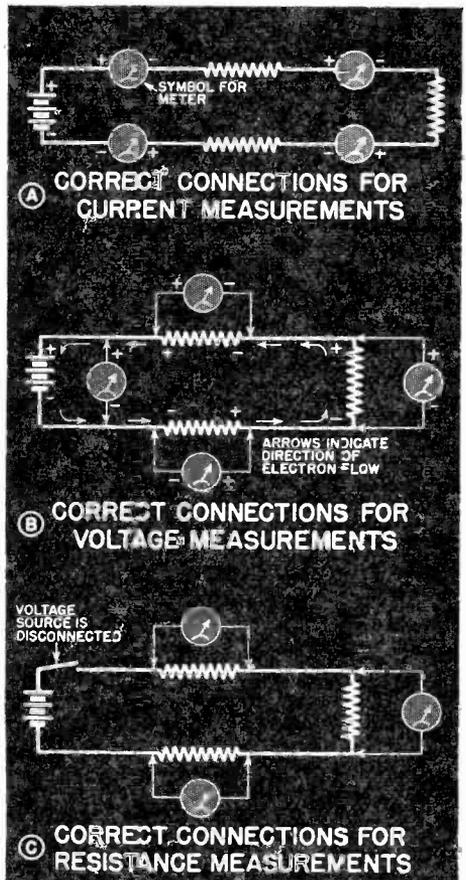


FIG. 16. Reference chart showing correct meter connections for current, voltage and resistance measurements and correct polarity for d.c. measurements.

**all you need do is plug the test leads** into the correct jacks on the panel, then set the two switches to the correct positions for the type of meter and range desired. Each jack and switch is clearly marked, so this can be done without even referring to an instruction book.

The meter in this tester has four scales. One serves for all d.c. readings, another for all a.c. readings, and the remaining two are used for the ohmmeter ranges. The two ohmmeter zero adjustments are of the screwdriver type, and are located at the lower left of the meter.

When the switch at the left of the meter is in its a.c. position, a rectifier is automatically inserted in the meter circuit for a.c. measurements. When this switch is in the d.c. position, the rectifier is disconnected.

Although for convenience and clarity we usually show individual meters in lessons, remember that a radio man

would normally use a multimeter or all-purpose tester for a measurement.

### Summary of Meters

The three important rules you should keep in mind whenever making meter measurements are:

1. For current measurements, always break into a circuit, so that the meter is in series with the circuit. This rule is illustrated in Fig. 16A. The current measurement can be made at *any one* of the four meter positions shown in this diagram.

2. For voltage measurements, connect to the points across which is the voltage to be measured. This rule is illustrated in Fig. 16B.

3. For ohmmeter measurements, connect *across* the resistance to be measured, but be sure to disconnect all batteries or other voltage sources, so no voltage exists in the external circuit. This rule is illustrated in Fig. 16C.

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## Voltage Sources for Radio Circuits

Since every radio circuit must have a *voltage source* of some kind, a general knowledge of the different kinds of voltage sources is highly desirable.

Many of the voltage sources used in radio are quite simple and familiar to most people. You may already know quite a bit about dry batteries and storage batteries, but even if you don't, you will undoubtedly find them easy to study.

The sections on d.c. and a.c. generators will particularly help you with radio work, because they give you a clearer understanding of what a.c. and d.c. voltages are like.

### Batteries

**Uses.** Dry batteries are used in battery-operated farm radios, in port-

able radio receivers, in the portable two-way radiophone sets of the Army and the U. S. Forest Service, in radio test instruments and in many other types of radio equipment. Storage batteries are the main voltage sources for radio equipment in airplanes, boats, automobiles, and other large mobile units. Thus, batteries are still used enough in the field of radio to make a general knowledge of their construction and connections desirable.

**Two Types.** Two distinct types of units or *cells* are used in batteries: 1. *Primary cells*, which are used in dry batteries, and cannot be recharged when exhausted; 2. *Secondary cells*, which are used in storage batteries, and can be recharged.



FIG. 17. Construction of a standard dry cell.

Both primary and secondary cells produce a d.c. voltage by chemical action, and supply direct current. Scientists would say that both types of cells convert chemical energy into electrical energy.

### Dry Cells

The *dry cells* generally used in portable radio receivers and in flashlights are perhaps the best known examples of primary cells. The chemical materials in a dry cell are gradually used up when current is drawn from the cell. When the supply of the chemical material is exhausted, the useful life of the cell is ended, and the cell must be replaced.

**Construction.** The construction of an ordinary dry cell is shown in Fig. 17. Flashlight cells are made in much the same way, but are smaller. In a flashlight cell, the negative connection is made *directly* to the exposed zinc case at the bottom of the cell,

instead of to a top terminal attached to this case.

**Voltage Produced.** All ordinary dry cells deliver the same voltage of slightly over  $1\frac{1}{2}$  volts (1.5 volts) when new. This voltage drops during use, and the cell must be replaced when its voltage becomes too low for its intended purpose.

**Current Furnished.** The larger the physical dimensions of a dry cell, the greater the amount of current it can deliver, or the longer it can deliver some definite value of current.

**Dry Cells in Series.** When a voltage higher than  $1\frac{1}{2}$  volts is required, dry cells are connected together in series until the desired voltage is obtained. In this series connection, the  $-$  terminal of one cell goes to the  $+$  terminal of the next cell. You would connect cells together in series whenever you needed a voltage equal to the sum of the individual cell voltages.

For example, when a voltage of 45

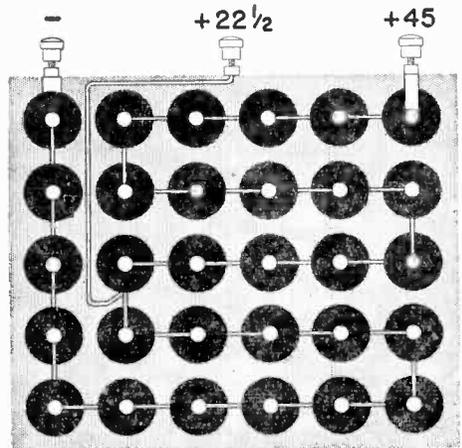


FIG. 18. How 30 dry cells are connected together in series to make a 45-volt B battery.

volts is desired, thirty individual dry cells are connected together in series in the manner shown in Fig. 18. This arrangement gives us a 45-volt B bat-

tery, of which a cut-away view is shown in Fig. 19.

**Dry Cells in Parallel.** When you require more current than it is desirable to draw from a single cell, you can connect additional cells *in parallel*. Each cell then furnishes its share of current to the load, so you increase the current-furnishing capacity without affecting the voltage. Examples of parallel connections are shown in Fig. 20.

**Remember This.** When you want



FIG. 19. Part of this 45-volt radio B battery has been cut away to show some of the individual dry cells.

to get the greatest possible voltage from a group of dry cells, connect them together *in series*. When you want to get the greatest possible current or the longest life from a group of dry cells, without any more voltage than is provided by a single cell, connect them together *in parallel*.

**Series-Parallel Connections.** When both higher voltage and higher current-delivering ability are desired, a series-parallel connection can be used. This involves connecting one group of cells together in series until it has the desired voltage, then connecting



FIG. 20. Parallel connections of dry cells. Remember that when cells are connected together in parallel, the voltage is that of one cell, but the current-furnishing ability is increased.

this group of cells in parallel with two or more similar groups of cells until the desired current-handling ability is obtained. Examples are shown in Fig. 21.

### Storage Batteries

The *storage battery* used in automobiles is the best-known example of a battery using rechargeable secondary cells.

**Construction.** A storage cell has two groups of plates, one group attached to the positive terminal and the other group attached to the negative terminal. These positive and negative groups of plates are made of lead (the metal, pronounced *LED*), and fit together alternately in the manner shown in Fig. 22A. Each plate has holes filled with the active chemical material of the cell, which is in paste form.

Between the plates are sheets of insulating material called separators, made either from porous wood or perforated rubber (Fig. 22B). Separators prevent the plates from touching each other and short-circuiting the cell.

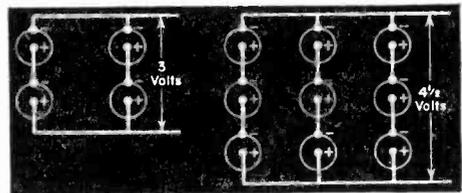


FIG. 21. Examples of series-parallel connections of dry cells.

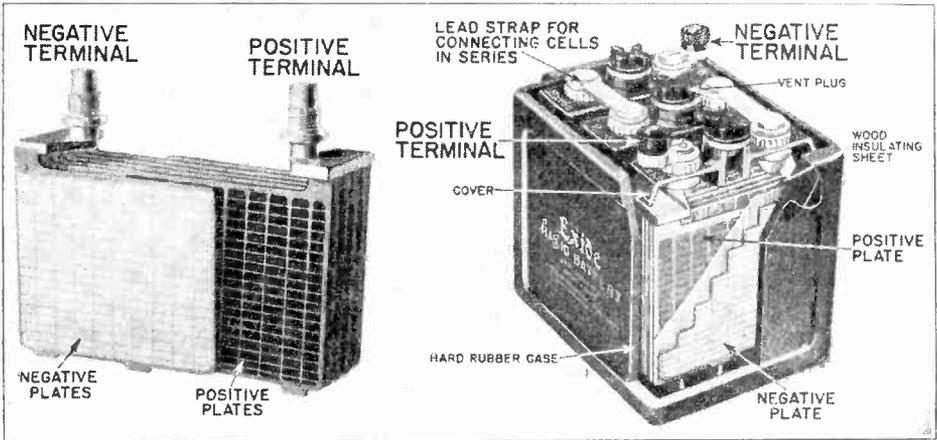


FIG. 22A. How the two groups of plates in a storage cell fit together. Insulating sheets prevent adjacent plate from touching

FIG. 22B. Cut-away view showing the construction of a typical 6-volt storage battery, which contains three storage cells connected in series.

**Hydrometer Test.** The plates of a storage cell are suspended in a container filled with a solution of sulphuric acid and pure water. The amount of sulphuric acid in the solution is an indication of how much the cell is charged (how much energy it contains in chemical form). The device shown in Fig. 23, called a *hydrometer* (pronounced *hie-DRUM-eh-tur*), is used to test the charge of a storage battery.

**Recharging.** The chemical materials change gradually to inactive forms when electrons flow out of the negative terminal of a storage battery during use. A storage battery is recharged by properly connecting it to a d.c. voltage source having a higher voltage than the storage battery. Charging is completed when the resulting reversed current has changed all of the chemical materials back to their original form.

**Voltage Produced.** The voltage of a storage cell is about 2 volts, and changes very little with use. For automotive purposes, three storage cells are connected in series, so that

their voltages add together to give the required 6 volts.

Storage batteries used for radio equipment in boats and airplanes often have six cells in series, to give 12 volts. The new tiny spill-proof storage batteries designed for portable radio receivers have only one cell, and hence deliver only 2 volts.

### Generators

The majority of radio receivers in use today operate from wall outlets



FIG. 23. A hydrometer like this is used to test the condition of each cell in a storage battery. A fully charged battery will give a reading of about 1.280, and a discharged battery will give a reading of about 1.120.

which furnish either a.c. or d.c. voltage. Behind each wall outlet is a complicated network of power lines which eventually leads to a central power station. Here steam turbines or water wheels turn huge generators which can supply thousands of amperes of current. Most of these power plants produce a.c. voltage, but there are still a few, in the older sections

of force extending from the N pole to the S pole.

The coil of wire is mounted between the poles of the magnet in such a way that it can be rotated. Connections are made to the two ends of the coil by means of *slip rings* and *brushes*, as illustrated in Fig. 24. The copper slip rings are mounted on the shaft but insulated from it. The

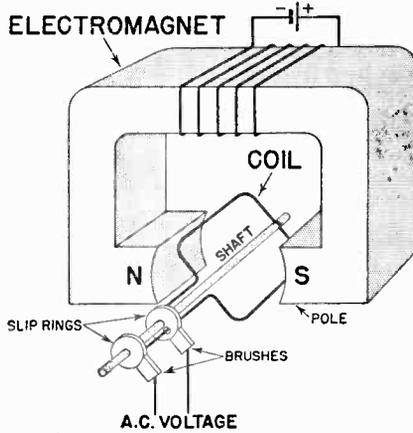


FIG. 24. Simple a.c. generator. Yes, it works!

of large cities, which generate d.c. voltages for home and business use.

A general idea of how these a.c. and d.c. voltages are produced is well worth while, because it will help you to understand how these voltages affect radio circuits.

### A.C. Generator

In its simplest form, an alternating current generator has only the two essential parts pictured in Fig. 24, an *electromagnet* and a rotating *coil*.

**Construction.** The electromagnet in this elementary generator is a horseshoe-shaped iron piece which is magnetized by a coil of wire connected to a d.c. voltage source. The electromagnet produces magnetic lines

of force extending from the N pole to the S pole. The brushes are blocks of carbon which are mounted in fixed holders. The brushes slide easily against the slip rings as the shaft rotates, and provide a complete path for electrons from the coil to any loads which are connected to the generator terminals.

**Operation.** When the coil is rotated, the number of magnetic lines of force which pass through the coil changes continually. This causes a voltage to be generated or induced in the coil.

**Polarity.** During one half of each complete revolution or turn of the coil, this voltage acts in one direction. During the other half of a revolution, the voltage reverses its *polarity* and acts in the other direction.

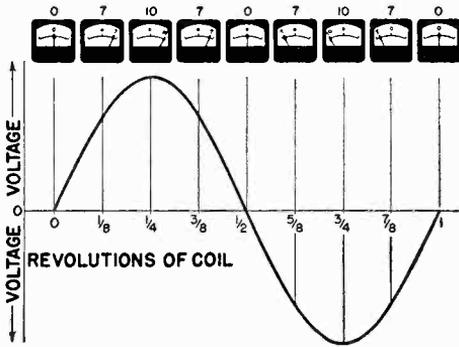


FIG. 25. One cycle of the voltage produced by a simple a.c. generator can be represented by the curve shown here. If the generator were being turned slowly enough so a zero-center d.c. meter could follow its variations, the meter readings at nine different coil positions during a complete revolution of the generator coil would be as shown here.

If we connect between the brushes of our simple a.c. generator a special type of d.c. meter which has zero at the center of its scale, and take photographs of the meter at each  $1/8$  revolution while turning the coil slowly, the meter photographs might appear as shown across the top of Fig. 25. These photographs give a good idea of how the voltage is varying both in value and polarity, so let's study them a bit.

During the first half-revolution of the coil (from 0 to  $1/2$ ), the meter pointer moves from 0 to the extreme right, and then back to 0. During the second half-revolution, the generated voltage has reversed its polarity, and the meter pointer moves from 0 to the extreme left and back to zero again.

#### Graph for an A.C. Generator.

Now, if we changed these meter readings into the type of simple graph used by radio men, we would secure a curve like that in Fig. 25. Distances up or down from the horizontal line represent voltage values at different instants of time (at different parts of a complete revolution). The curve is *above* the horizontal line for one po-

larity, and goes below the line when the voltage reverses its polarity.

**One Cycle.** One complete revolution of our simple generator coil produces *one cycle* of a.c. voltage. For each additional revolution, the voltage changes would repeat themselves in the same way as shown for the first revolution in Fig. 25. Consequently, the curve represents *one cycle* of our a.c. voltage.

**60-Cycle Power.** If we turn our simple generator fast enough so it makes 60 complete revolutions in one second, there will be 60 of these voltage cycles per second, and we will have 60-cycle electric power like that used in most sections of this country.

When a generator is rotated fast enough to produce 60 voltage cycles each second, no ordinary meter can follow the instantaneous voltage varia-

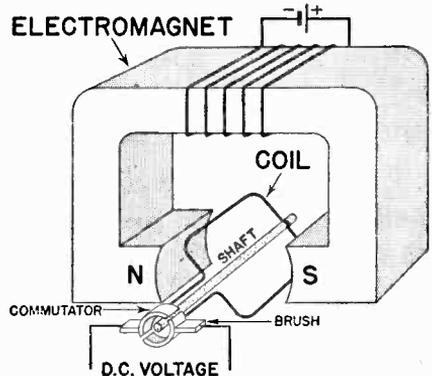


FIG. 26. Simple d.c. generator. It'll work, too! tions in each cycle. A cathode ray oscilloscope can follow fast changes, however, and for one cycle of a 60-cycle voltage it would show on its screen exactly the same curved pattern shown in Fig. 25.

#### D.C. Generator

A direct current generator is nothing more than an a.c. generator with a *commutator* in place of slip rings, as

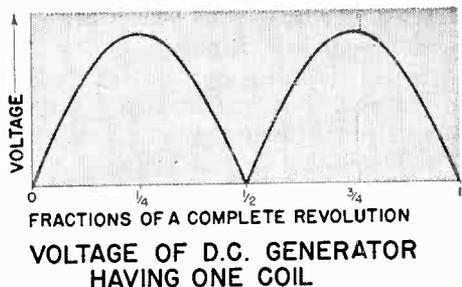


FIG. 27. Example of a diagram which might be used by radio men to show how the voltage of a simple d.c. generator varies during one complete revolution of the coil.

shown in Fig. 26. A commutator is a device which automatically reverses the connections between the generator and its load at the exact instant during each revolution when the generated voltage begins reversing its polarity.

**Polarity.** The automatic reversal of connections by the commutator makes the terminals of a d.c. generator have the same polarity at all times. As a result, a diagram showing the volt-

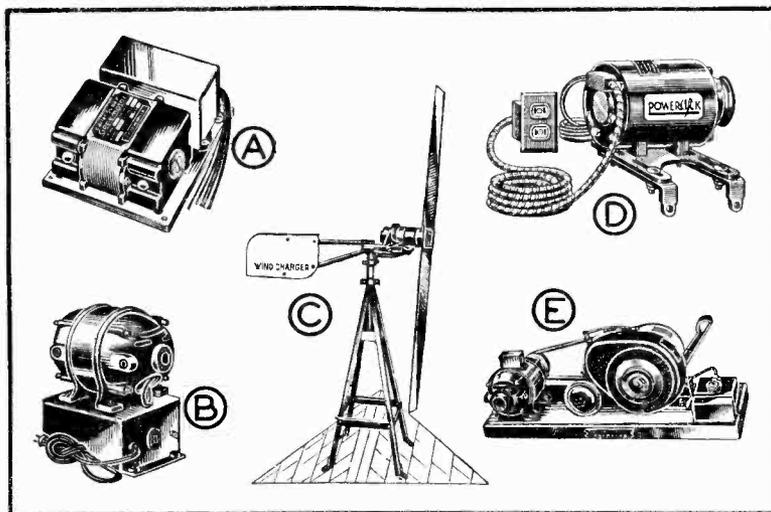
age between the brushes of a d.c. generator during one complete revolution would appear as in Fig. 27.

**Actual Construction.** In an actual d.c. generator such as would be used in a power plant and in certain types of radio equipment, there are many coils mounted on the rotating shaft of the generator, instead of just one coil. When the voltage in one coil is dropping, the voltage in another coil is rising, and the voltages add together to give an almost constant d.c. voltage.

**Filter.** Even a small variation in d.c. voltage is objectionable in certain types of radio equipment, but this variation can be removed with a filter made up of coils and condensers.

**Field Current.** In commercial a.c. and d.c. generators, the electromagnet coil is known as the *field coil* because it provides the magnetic field.

In a d.c. generator, some of the direct current produced by the generator



Examples of generator units used with radio equipment. ♦A—Dynamotor, a combination motor and generator operating from a 6-volt storage battery and generating the high d.c. voltages required by police radio equipment or public address amplifiers. ♦B—Dynamotor which changes 32 or 115 volts d.c. to 115 volts a.c. ♦C—Wind-driven generator used for charging storage batteries of farm radios. ♦D—115-volt a.c. generator which can be driven by the fan belt of an automobile or truck. ♦E—Combination 115-volt a.c. and 6-volt d.c. generator driven by a gasoline engine, used with high-power portable radio equipment.

is used for the field coil, thereby eliminating the need for a battery.

A commercial a.c. generator always has a small d.c. generator mounted on its shaft or built right into the a.c. generator housing, to produce the di-

which connects to the power line, and three secondary windings providing the three required voltages, as shown in Fig. 28. You thus see that a transformer can be made either to step up or to step down the line voltage.

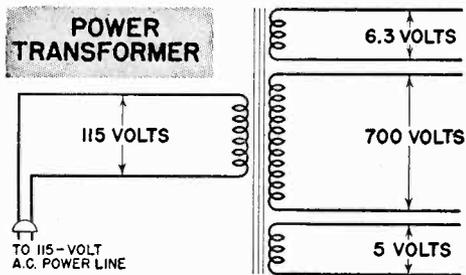


FIG. 28. Diagram of a power transformer similar to those used in a.c. radio receivers.

rect current required for the field coils of the a.c. generator.

### Transformers

In a.c. circuits, the problems of securing a desired a.c. voltage value is greatly simplified because we can change an a.c. voltage to any desired higher or lower value by means of a transformer.

Here is a practical example: A radio receiver which is to operate from a 115-volt a.c. power line requires 5 volts for the filament of the rectifier tube, 6.3 volts for the filaments of the other tubes, and 700 volts for the power pack which is to furnish d.c. voltages to the various tube electrodes.

The power transformer for this receiver will have one primary winding

### Looking Ahead

The general knowledge of radio broadcasting and receiving which you gained from the first two lessons of your Course has now been reinforced by the highly practical information in this lesson.

You have learned a great deal about voltage, current and resistance in radio circuits. You have a working knowledge of voltage source, load and meter connections. You are familiar with the different kinds of voltage sources used in radio work.

You are ready, therefore, at this early point in your N.R.I. Course to take up a typical modern superheterodyne radio receiver. In the next lesson, you will find out what this receiver is like, how it is operated, what each part is, why and how the parts might fail, and what radio men would do to repair the receiver when various parts fail.

Not only will this next lesson prove clearly the importance of the material you have already studied, but it will carry you right ahead to still more interesting and more practical work. You will actually be learning some of the different ways of repairing receivers and radio parts.

# Lesson Questions

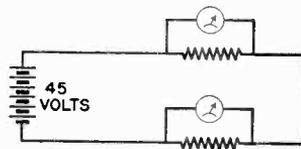
Be sure to number your Answer Sheet 3FR-2.

Place your Student Number on *every* Answer Sheet.

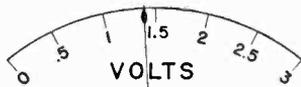
*Send in your answers for this lesson immediately after you finish them, as instructed in the Study Schedule. Do this, and get the greatest possible benefit from our speedy personal grading service.*

1. What type of voltage reverses its polarity regularly?
2. Name the *unit* of current which is equal to one thousandth of an ampere.
3. If you want to reduce the amount of direct current which is flowing in a radio circuit, what radio part should you insert in the circuit?
4. If you *increase the source voltage* in a complete radio circuit, what will happen to the current?
5. If you *decrease the resistance* in a complete circuit, what will happen to the current?
6. In what way would you connect together the filaments of radio tubes if all the filaments required the same current, and the available voltage was equal to the SUM of all the filament voltage ratings?
7. When can the filaments of radio tubes be connected *in parallel*?

8. Will the voltage drops across the resistors in this series circuit add up to a value *LESS THAN, EQUAL TO, or MORE THAN* the 45-volt source voltage?



9. What reading is indicated by the pointer on this voltmeter scale? (Study Fig. 13 carefully before you answer this question.)



10. Tell how you would connect four  $1\frac{1}{2}$ -volt dry cells together to get a total voltage of 6 volts?

## GETTING YOUR "SECOND WIND"

After an opening burst of speed, a champion long-distance runner drops down to a steady natural pace. This brief period of relaxation releases that reserve of power called "second wind." He is then able to overtake and pass the now nearly exhausted leaders in the race.

No matter what you are doing, you also can do it better once you get your *second wind*. Instead of fighting that sleepy feeling which sometimes comes when you are studying, stop and relax for a few minutes so as to release your own supply of reserve power. Here are some ways to relax, all worth trying.

Get up and exercise for a few minutes. Get a drink of cold water. Step outside for a few *deep* breaths of fresh air. Take a brisk walk up and down the road or once around the block.

If you "shake yourself awake" in one of these ways, you'll find it isn't at all hard to get that second wind which enables you to study longer and makes the going easier. Then you'll get in some real worthwhile studying—then you'll get things done!

J. E. SMITH.

**GETTING ACQUAINTED WITH  
RECEIVER SERVICING**

4FR-2

**NATIONAL RADIO INSTITUTE**

ESTABLISHED 1914

**WASHINGTON, D. C.**



# STUDY SCHEDULE NO. 4

To master this fourth part of your N.R.I. Course most speedily and thoroughly, divide your study into the steps given below. First read the specified pages at your usual reading speed. Next, reread them slowly one or more times, endeavoring always to *understand* as well as remember. Finish with one quick reading to tie the ideas together in your mind, then write on scrap paper your answers to the questions specified for that step. After finishing the lesson, copy all answers neatly on an Answer Sheet.

- 1. **We Study a Modern Radio Receiver**.....Pages 1-5  
Read this section at least once, to find out *why* a knowledge of radio receiver servicing problems is important to *YOU*, to get a *preview* of how receivers are taken up in this lesson, to study a *radio map* which will guide you through a modern superheterodyne receiver, to become acquainted with the knobs and dials on the front of a receiver, and to learn how the chassis is removed from the cabinet of a receiver. Answer Lesson Question 1.
- 2. **Radio Receiver Tubes**.....Pages 6-9  
Since a tube is one of the most important parts in modern receivers, you study first the five tubes used in the average superheterodyne receiver which serves as our example for this lesson. Tubes will mean a lot more after you finish this interesting section. Answer Lesson Questions 2 and 3.
- 3. **Above-Chassis Parts**.....Pages 10-16  
You take up one by one the parts which are mounted on top of the chassis of our superheterodyne receiver—the loudspeaker, the output transformer, the gang tuning condenser, the trimmer condensers and the i. f. transformers. You find out how each one is constructed, what it does, how it might fail, and how it is repaired or replaced. Answer Lesson Questions 4, 5 and 6.
- 4. **Under-Chassis Parts**.....Pages 17-25  
You turn the chassis upside-down, and study in turn, the soldered joints, resistors, condensers, r. f. coils, power transformer, power cord, on-off switch and volume control, tone control, dial lamp and tube sockets. Answer Lesson Questions 7, 8 and 9.
- 5. **Radio Receiver Servicing Techniques**.....Pages 26-28  
Here's your first real introduction to actual servicing techniques. You learn how a radio man uses various techniques to find first the defective section or stage, then the defective circuit, and finally the defective part. Answer Lesson Question 10.
- 6. **Mail Your Answers for Lesson 4FR-2 to N. R. I. for Grading.**
- 7. **How To Remove and Overhaul the Chassis**.....RSM Booklet No. 4  
This is another of the series of Booklets that are to give you practical, how-to-do-it training in fixing radios. Read this Booklet now, then keep it handy for future reference. The "shop training" you get from these RSM Booklets is going to prove valuable to you throughout your service career, so you will want to read and review these Booklets many times. NOTE: Some of the Study Schedules in future Lessons may refer to "Job Sheets" instead of RSM Booklets. Read the correspondingly numbered RSM Booklet in such cases. These Booklets are an improved and enlarged version of the older Job Sheets, and are replacing them.
- 8. **Start Studying the Next Lesson, on Resistors.**

# GETTING ACQUAINTED WITH RECEIVER SERVICING

## We Study a Modern Radio Receiver

**I**F RADIO servicing is to be your chosen work, you will unquestionably deal with radio receivers most of the time. And no matter what other branch of radio you intend to get into, you will still work with receivers part of the time.

*For example*, in practically every broadcast station there is at least one radio receiver. This receiver is tuned to the station frequency, and enables the radio operator on duty to hear exactly what the thousands of radio listeners are hearing. A radio operator at a huge transmitter would certainly feel embarrassed if he had to call in a

radio serviceman to fix his own monitor receiver when trouble developed.

The two-way radio communication stations used on boats, airplanes and other mobile equipment provide more examples. Failure of just one tiny part in the radio receiver can put one of these stations out of action just as effectively as failure of the largest part in the station's transmitter.

The fundamental principles *which make a radio receiver work* also apply to transmitters, test instruments, and all other radio apparatus. Yes, radio receiver parts and circuits work exactly like the corresponding parts and

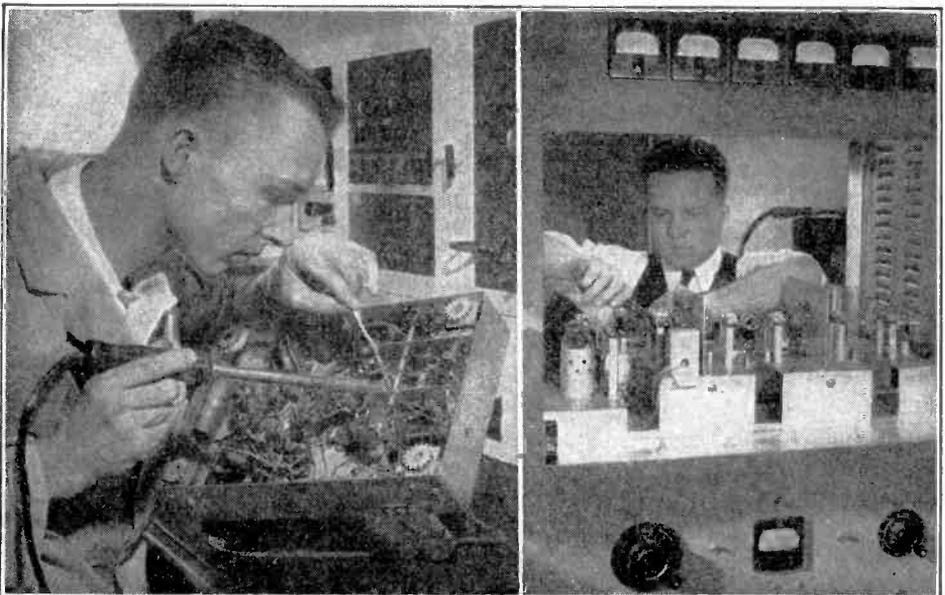


Photo by F. C. Wilkinson

Courtesy General Electric Co.

Radio men use receiver servicing techniques for all types of radio equipment. Thus, the method which the man at the left is using to repair a modern all-wave superheterodyne receiver will apply equally as well to the television transmitter being serviced by the man at the right.

circuits you find in all other types of radio equipment, wherever you may be—in the radio room of an ocean liner, at the radio controls in a transport air liner, at the sending key of a forest fire lookout station, in the control booth of the public address system at a huge football stadium, or in any of the broadcast and short-wave radio stations throughout the world.

Once you learn how to repair radio receivers by using modern professional servicing techniques, you will have the foundation training for operating and repairing practically any other piece of radio apparatus.

### Preview of This Lesson

In this lesson, you imagine that you have in front of you the table model receiver pictured in Fig. 1. It is an average modern receiver, using the latest *superheterodyne* circuit (pronounced *SOO-pur-IET-w-o-DINE*, and often abbreviated as *superhet* or *super*). With the aid of special illustrations, you learn about the parts in this receiver and their general purposes, just as if the set were right in front of you in your own home.

Starting with the front of the receiver, you learn what each control knob does and how it is used. The meanings of the printed numbers and scales on the receiver tuning dial are clearly explained. You find out what to do when a customer loses one or more of the front-panel knobs or controls.

You remove the “works” (the *chassis*, pronounced *CHASS-iss*) from the receiver cabinet, then examine in turn each of the parts on top of the chassis, including the radio tubes, the loud-speaker, the output transformer, the gang tuning condenser, and some new parts which you take up for the first time here. In the same way, you study



FIG. 1. This modern superheterodyne receiver is used as an example throughout this lesson.

the many smaller parts which are located underneath the chassis.

Throughout this lesson, the practical viewpoint is stressed. You learn a great deal about how a part is made, what it does, how it can fail or give trouble, and how it is repaired or replaced.

### Radio Map of a Superhet

Your study of this lesson will be much easier if you first get a general idea of what a superheterodyne circuit is. You are already familiar with the other common type of receiver circuit—the t.r.f. circuit, so it will be particularly interesting to find out why the superheterodyne can “run circles” around a t.r.f. receiver of the same size.

Figure 2 is the “radio map” which gives you the facts about the circuit arrangement of the superheterodyne receiver used as the example for this lesson. As you can see, the first stage in our receiver is the *MIXER-FIRST DETECTOR*. It receives two signals:

1. *The desired modulated r.f. carrier signal*, which is produced in the receiving antenna by radio waves.

2. The r.f. signal, which is self-generated in the R.F. OSCILLATOR stage in such a way that its frequency is exactly 456 kc. higher than the carrier frequency of the desired incoming signal.

Now comes the action which makes the superhet so different from a t.r.f. receiver. The mixer-first detector combines its two incoming signals to give an entirely new signal called the i.f. signal.

The new modulated i.f. carrier sig-

After the carrier frequency of the desired incoming signal has been changed to 456 kc. by the mixer-first detector, the resulting i.f. signal is fed into the I.F. AMPLIFIER. There it receives a tremendous amount of amplification, because an i.f. amplifier stage which works at *only one* carrier frequency is many times more effective than an amplifier stage which must work at many different frequencies. (In a t.r.f. receiver, you know, the r.f. amplifier stages must handle any car-

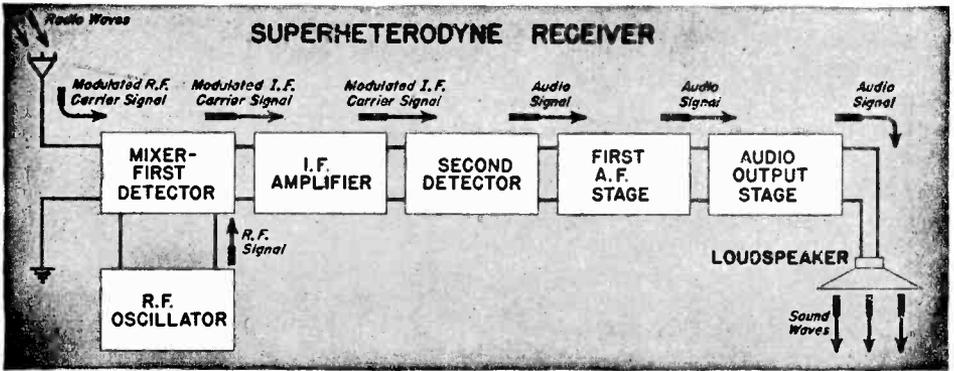


FIG. 2. Arrangement of stages in the superheterodyne receiver used as an example in this lesson.

nal in this receiver (and in many other superheterodyne receivers) *always* has a carrier frequency of exactly 456 kc. (This value is the difference between the frequencies of the r.f. signal and the incoming carrier signal.)

Strictly speaking, we should call this new signal the *modulated intermediate frequency carrier signal*, because it still has the original audio signal modulation. Since you are going to work with radio men, however, you will undoubtedly want to talk their own language, and say "eye of signal" just as they do.

The frequency of the i.f. carrier signal is *in between* the r.f. carrier frequency and audio frequencies. This is why the new signal is called an *intermediate* frequency signal.

rier frequency which the listener tunes in.)

The strengthened i.f. signal enters the *SECOND DETECTOR*. This stage separates the audio signal from the i.f. carrier signal, and also gets rid of the i.f. carrier since it is no longer needed. The second detector thus gives us the audio signal we have been working for all this time.

The audio amplifier and loudspeaker, following the second detector of this super, are exactly like the audio amplifier and loudspeaker of a t.r.f. receiver. Thus, the audio signal gets one boost in strength from the *FIRST A.F. STAGE*, and gets another boost from the *AUDIO OUTPUT STAGE* before being fed to the *LOUDSPEAKER*. Here the audio signal

is changed into *SOUND WAVES* which are the desired radio program.

**You See How Radio Men Work.** After becoming thoroughly acquainted with the various parts in this receiver, you will see how radio servicemen go about locating and repairing trouble in the receiver.

Once the trouble is located, the defective part is easily replaced or repaired. It is in finding the cause of the trouble that a complete knowledge of the purposes and actions of radio parts counts.

### Front-Panel Parts

**Control Knobs.** The general arrangement of the controls on this receiver is typical of that employed in most modern receivers, and is shown in Fig. 3.

To turn on this set you turn the *volume control* knob clockwise until a click is heard. This turns on the power pack, by completing the circuit for the power cord which goes to a wall outlet.

Further clockwise rotation of this knob increases the volume of loudness of the program coming from the loud-speaker.

When you tune in a station with the *tuning control* knob, you are doing two things: 1. Rotating the gang tuning condenser to the correct setting for reception of the station; 2. Moving a pointer to the position on the tuning dial at which the frequency of the station is indicated.

The *tone control* knob allows you to change the tone of the sound coming from this receiver, by reducing the strength of higher audio frequencies. The lower audio frequencies will then predominate, and give a mellow bass tone which emphasizes drums, large bass horns and men's voices.

On some receivers, it is also possible to set the tone control to reduce the strength only of the *lower* audio fre-

quencies. The high frequencies of violins, cornets, bells and women's voices will then be emphasized. Tone controls are provided because people's tastes differ regarding tone just as much as tastes differ regarding food, clothes, or almost anything else.

**Tuning Dial.** It is common practice among receiver manufacturers to mark the tuning dial in kilocycles, with the last zero in the frequency value omitted. The dial shown in Fig. 3 is an example of this practice. If you add a zero to each number printed on this dial, the numbers represent *frequency in kilocycles*.

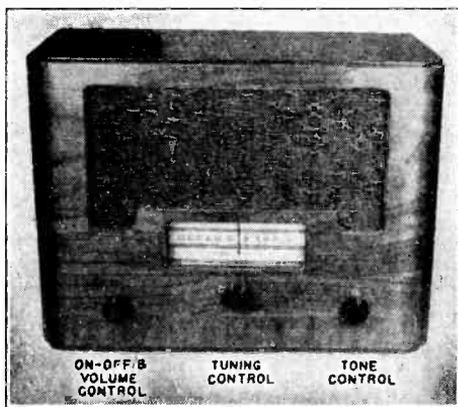


FIG. 3. Front view of the modern superheterodyne receiver used as an example throughout this lesson. If you add a zero to each number on the tuning dial, the numbers will represent the frequencies in kilocycles to which the receiver can be tuned.

Radio station frequencies are usually expressed in kilocycles for convenience, so the term *kilocycle* (abbreviated *kc.*) is one which you should remember. (One kilocycle is equal to 1000 cycles.)

The tuning dial shown in Fig. 3 extends from 550 kilocycles to 2000 kilocycles, hence our superhet receiver covers the entire *broadcast band* and certain police bands just above the broadcast band.

**Replacing Control Knobs.** When a Radiotrician encounters a receiver

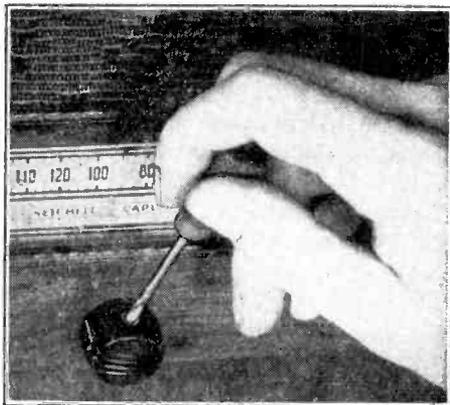


FIG. 4. Loosening the set screw of a receiver control knob with a small screwdriver. Some knobs may have two set screws.

having a broken or missing control knob, he will first try to obtain a new knob exactly like those already on the set.

If an exact replacement knob is not available, he will secure an entire set of new knobs, having a design and color which will harmonize with the finish of the receiver cabinet.

### Removing the Chassis

In order to get a good look at the radio parts in our receiver, we will have to remove the *chassis* from the cabinet. The chassis includes the metal *chassis base* and all parts mounted on this base.

**Removing Control Knobs.** We first loosen the tiny set screw in each control knob, using a small screwdriver as illustrated in Fig. 4, then remove the knobs.

Sometimes knobs are held in position by stiff flat springs inside the knobs, instead of screws. These knobs are removed by pulling firmly. If it is difficult to get a good grip on a small knob, slip a handkerchief between the knob and the cabinet,

and pull on the ends of the handkerchief.

**Removing Chassis Screws.** Next, we use a medium-size screwdriver to loosen the two wood screws which fasten the chassis base to the bottom of the receiver cabinet, as illustrated in Fig. 5.

In some receivers, the chassis is fastened with bolts which pass through the bottom of the cabinet. These are removed either with a screwdriver or socket wrench, depending upon the type of bolt head employed, after turning the receiver on its side so

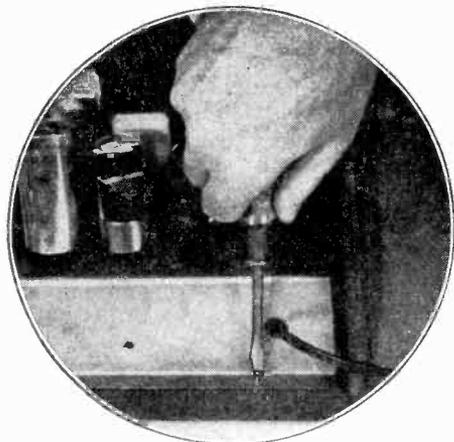


FIG. 5. Using a medium-size screwdriver to remove the chassis hold-down screws at the back of our superheterodyne receiver.

that the bottom of the cabinet is accessible.

The chassis-removing procedure just described applies to practically any radio receiver, so let's repeat it here as a practical radio servicing rule: *To remove the chassis from the cabinet of a radio receiver, first take off the control knobs, then remove the chassis hold-down screws or bolts.* If the chassis will not come out yet, look for additional hold-down screws around the dial and loudspeaker.

# Radio Receiver Tubes

Now we can slide the chassis out from the back of its cabinet. When this is done, all of the parts on top of the chassis base are clearly visible, as shown in Fig. 6. Study this view for a few minutes, to become familiar with the positions of the tubes and the labeled large parts.

Before you can understand how parts work together in radio circuits, and before you try to locate defective parts, you should have a general idea of how these radio parts look, how they are built, and how they act on signals and circuit currents when operating properly. Therefore, we concentrate upon radio parts in this lesson, starting with tubes.

The five tubes in our superheterodyne radio receiver are all on top of

the chassis base, and are easily recognized even though one of them is almost completely enclosed in a metal shield.

**Tube Shields.** First of all, we'll take off that metal shield and see what the hidden tube looks like. We remove the clip from the top cap of the tube, then pull the shield straight up with one hand as shown in Fig. 7.

Yes, there's just an ordinary radio tube inside this shield, a type 6K7 tube serving in the i.f. amplifier stage. It's an important tube, for it receives extremely weak desired signals and boosts them a tremendous amount. But undesired signals would get into this tube circuit, so we must use the metal *shield* to protect the tube elec-

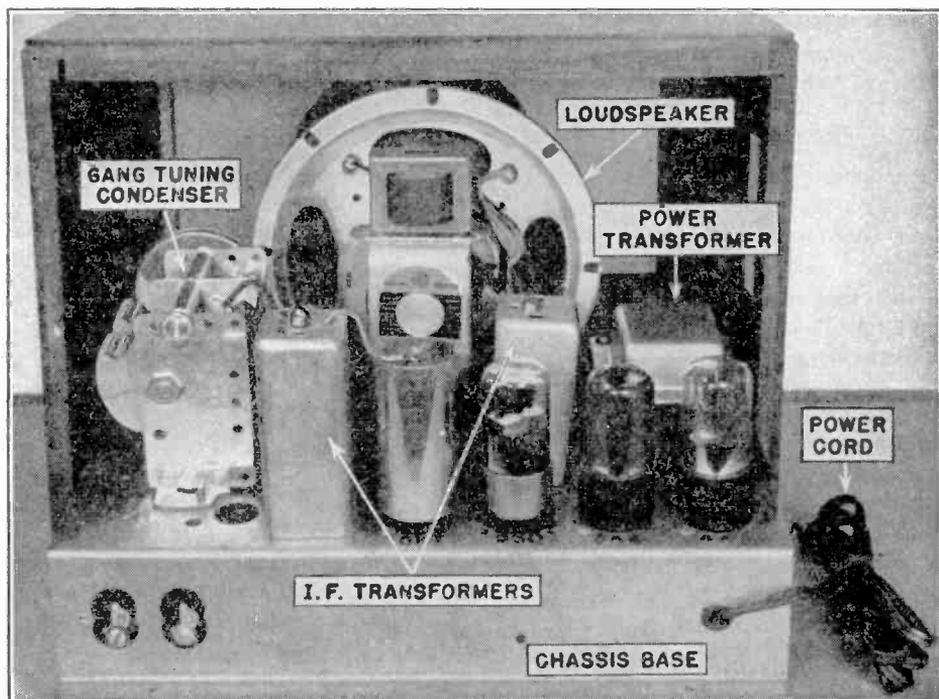


FIG. 6. Rear view of our superheterodyne receiver, with the chassis almost out of the cabinet.

trodes from stray fields (stray radiated signals) of other parts. Without a shield, the set would probably squeal at each station as you tuned across the dial.

**Tube Bases.** The tubes in this receiver all have *octal* bases, which means that each one can have as many as eight prongs. Some of the tubes require all eight prongs for connecting purposes, either because the tubes have extra grids or because two or more sets of electrodes are combined in one tube.

When a tube requires less than eight connections to its socket, some of the prongs are omitted during manufacture without changing the positions of the others. Thus, the 6X5

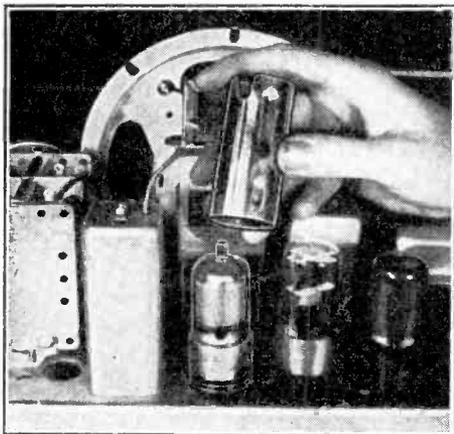


FIG. 7. Removing the shield from a tube.

rectifier tube taken from our set and shown in Fig. 8 has only six prongs.

**Tube Numbers.** Each of the tubes in our receiver has an identifying number of its own, printed right on the glass envelope or on the base of the tube. Furthermore, each tube is intended for a particular socket on the chassis base of this receiver. Since an octal-base tube will fit in any octal-base socket, the sockets must be identified in some way so as to make sure

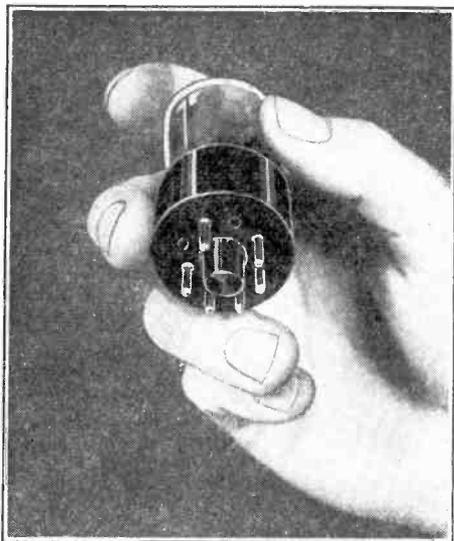


FIG. 8. This is what you see when you remove a 6X5 tube from a receiver and look at its base.

that the correct tube will be placed in each socket.

**Socket Markings.** The number of the tube intended for each socket has been clearly printed on the chassis base of our receiver, in the manner shown in Fig. 9. This illustration shows how the set would appear from the rear if only the tubes were above the chassis. The functions of the tubes are indicated by notes on the illustration.

Sometimes tube sockets themselves are marked. More often, however, there are no markings anywhere near the sockets. Instead, there is a diagram attached to the chassis or the inside of the cabinet, showing which tube goes in each socket.

In a few sets, there will be no markings at all. You can either mark all the sockets yourself before taking out the tubes, or take out only one tube at a time for testing.

**Removing Tubes.** To remove a tube for testing or replacement, pull it straight up while wiggling it slightly

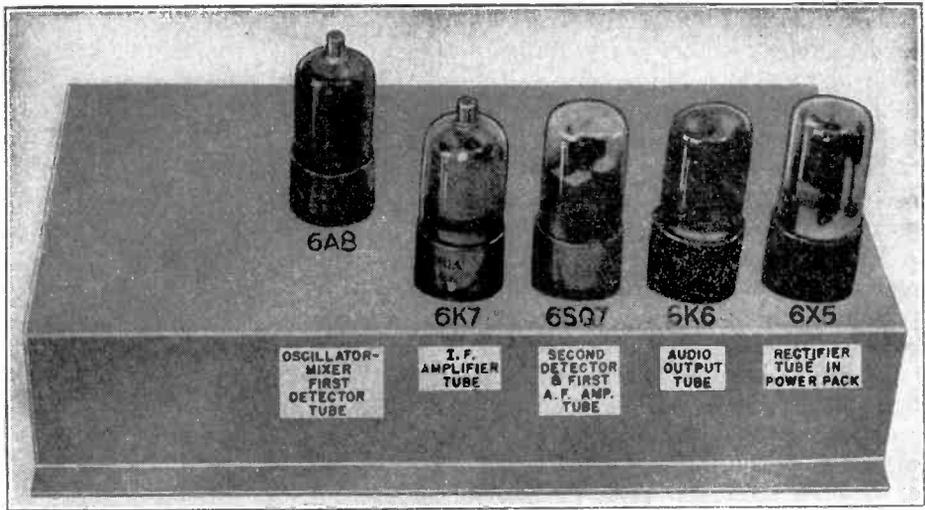


FIG. 9. This is how the back of our superheterodyne receiver chassis would look if only the tubes were visible.

from side to side. Tube sockets are purposely made to grip tube prongs tightly, so as to provide good electrical contact and prevent tubes from bouncing out during shipment of a receiver. Thus, it may take quite a bit of force to pull out a tube.

Here is a practical tip which may prove mighty useful when you are trying to remove a tube from a crowded corner of a radio chassis. While pulling and wiggling the tube with your right hand, insert a screwdriver under the tube base with your left hand and twist the screwdriver so as to pry up the tube. Of course, the receiver must be turned off when this is done, for otherwise the screwdriver might short some power supply circuit and cause damage.

**Tubes Get Hot.** All radio tubes heat up when in operation, but some types of tubes get much hotter than others.

The tubes used in battery and portable receivers stay the coolest. All-metal tubes, on the other hand, sometimes get so hot in normal use that you

can actually burn your fingers on them.

The tubes in our superheterodyne receiver all use glass envelopes, which in some cases get almost as hot as metal envelopes. Allow a minute or so for the tubes to cool before attempting to remove them, or use a handkerchief or piece of cloth to protect your hand when removing hot tubes.

This precaution applies particularly to the audio output tube and the rectifier tube in a receiver, for these two tubes usually get the hottest. The reason is that the audio output tube has a higher plate current than any other tube except the rectifier tube. As a matter of fact, the rectifier tube handles all the plate and grid currents drawn by all the other tubes in the receiver.

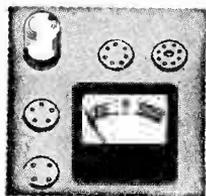
**Testing Tubes.** When a receiver comes into a radio service shop for repair, tubes are usually the first things which the serviceman checks. He removes the tubes from the chassis and tests them, one at a time, in a special tube-testing instrument like that shown in Fig. 10.

For each tube, he first adjusts the tester controls according to a tube chart furnished with the tester, then plugs the tube into a socket on the tester. If the meter in the tester indicates BAD, the tube is defective.

Although tube testers have a number of different sockets and controls, they are really quite simple to use once the operating instructions are carefully studied. There is no need for you to secure a tube tester yet, however—in fact, you will not need one until you are doing a considerable volume of radio service work.

**How Tubes Fail.** How do radio tubes go bad? Three common reasons are a burned-out filament, low emission, and shorted electrodes.

*Burned-Out Filament.* The filament in a radio tube is normally operated at red heat or even brighter. The thinnest parts of the filament wire get the hottest. The intense heat eventually causes the filament wire to melt and break open at a thin spot, causing complete failure of the tube.



When the meter pointer in the tube tester does not move at all from its position at the lowest end of the BAD region, the serviceman knows

the tube has a *burned-out filament*.

*Low Emission.* Another common cause of radio tube failure is aging of the chemical coating on the **cathode**. This chemical coating gives off electrons when heated by the filament. As the coating wears off during use, fewer and fewer electrons are given off, and the total electron flow from cathode to plate through the tube is correspondingly reduced.

Eventually, the number of electrons given off or emitted is so much reduced that the tube can no longer do its full job in the receiver. Radio men then say that the tube has *low emission*. A tube tester would indicate that the tube was BAD.

When a serviceman encounters a



FIG. 10. Tubes are tested by radio men with tube testers like this.

tube which has low emission, he discards it and puts in a new tube.

*Shorted Electrodes.* Another type of radio tube failure is *shorted electrodes* inside a tube. Changes in temperature may cause an electrode to sag out of position enough so that it touches a nearby electrode inside the tube. We then say that the tube has shorted electrodes. Thus, the filament may touch the surrounding cathode and cause a *short*.

Practically all tube testers have a separate red light on the panel of the instrument. This red light glows whenever any two of the electrodes are shorted together. When this light glows, the radio man knows that he will have to replace the tube anyway, so he ordinarily does not bother to figure out which electrodes are touching.

**More About Tubes.** You will learn a lot more about radio tubes and their electrodes later in the course in a lesson devoted entirely to radio tubes. Furthermore, in one of the N.R.I. Practical RMS Booklets, you get complete step-by-step instructions for testing tubes with typical modern tube testers.

# Above-Chassis Parts

Although other superheterodyne radio receivers may look quite different from the average receiver chosen for our example in this lesson, all superheterodyne receivers use the same fundamental principles of operation and similar groups of ordinary radio parts. Corresponding parts in superheterodyne receivers may look different, but their important purposes and actions will be quite similar.

## The Loudspeaker

Let us consider the loudspeaker next, for it is the largest radio part in our receiver, and has the actual job of changing the final strong audio signal into the sound waves of a radio program.

**Electrodynamic Loudspeaker.** As you previously learned, a modern dynamic loudspeaker has only three important parts: A diaphragm or cone, a *voice coil*, and a *magnet*.

In the loudspeaker used in our superheterodyne receiver, an electromagnet provides the magnetic field, and consequently the loudspeaker is known as an *electrodynammic loudspeaker*. The coil of the electromagnet is called the *field coil*.

The direct current required by the field coil is obtained from the power pack of the receiver. An electrodynamic loudspeaker will therefore have two extra wires, going from the field coil to the proper power pack terminals underneath the chassis. The field coil and its leads can be seen in Fig. 11, which shows the loudspeaker all by itself on the chassis of our receiver.

Through its connection to the power pack, the field coil also does an important filtering job, about which you

will learn more later when you study power packs.

**How It Works.** When an audio signal current is sent through the *voice coil* of our electrodynamic loudspeaker, this voice coil produces a varying magnetic field which reacts with the constant magnetic field of the field coil.

The two coils thus alternately attract and repel each other. The field coil electromagnet cannot move because it is attached to the frame, so

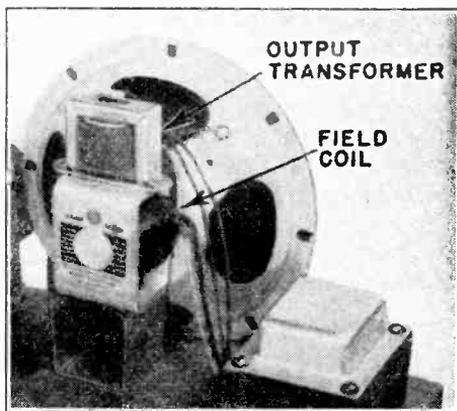


FIG. 11. Here's a close-up view of the loudspeaker in our superhet receiver.

only the voice coil moves back and forth. Since the voice coil is attached to the cone, the cone likewise moves the surrounding air back and forth, producing the sound waves of radio programs.

**Damaged Cone.** When the stiff paper cone of a loudspeaker is accidentally torn, or when it becomes warped or cracked due to aging, radio programs no longer will sound natural, and will sometimes be accompanied by buzzing or scratchy sounds.

When a Radiotrician encounters a damaged cone, he either replaces the

entire cone-voice coil assembly with an exact duplicate replacement assembly obtained from the set manufacturer or a radio parts distributor, or installs an entirely new loudspeaker. Curiously, with small loudspeakers it is often cheaper and more satisfactory to put in a new loudspeaker than to replace or repair a damaged cone.



**Off-Center Voice Coil.** One common loudspeaker trouble is an off-center voice coil. This coil moves in and out in a very small air gap between the pole pieces. Slight warping of the cone or slight shifting of the various parts of the loudspeaker throw the coil off center in the air gap, and the coil then rubs against the pole pieces.

When the voice coil is off center, radio programs will sound distorted on the loud bass notes which are present when a man talks or sings, but will probably be perfectly clear for women's voices, which have few low notes. You may hear a rubbing sound when you push the cone in and out with your fingers after the set is turned off.

When the voice coil is properly centered, you should be able to move it in or out without having the coil rub against the pole pieces between which it moves.

Practically all of the older loudspeakers and even a great many modern loudspeakers have provisions for recentering the voice coil. These loudspeakers will either have one screw inside the voice coil or several screws arranged around the outside of the voice coil. When these screws are loosened, the voice coil can be moved a small distance sidewise in any direction for recentering purposes.

With the screws loose, small narrow strips of celluloid or cardboard are inserted between the inside of the voice coil and the central iron core of the loudspeaker to provide automatic centering. The screws are then tightened, after which the centering strips can be removed.

There is a growing trend toward permanent fastening of the voice coil to the loudspeaker frame. The loudspeaker used in our superheterodyne receiver is an example of this, and its voice coil cannot economically be re-centered by the radio serviceman. If for any reason the voice coil gets off center, the loudspeaker can be returned to a loudspeaker cone manufacturer or a radio supply firm for installation of a new cone-voice coil unit, or the entire loudspeaker can be replaced.

**Open Field Coil.** A broken wire (usually due to corrosion) is about the only trouble encountered in the field coil of an average electrodynamic loudspeaker. The trouble usually occurs at one end of the coil, at the point where the flexible insulated field coil lead is connected to the enamel-covered coil wire. If the trouble is at the outer end of the coil, it can often be repaired after removing the outer layer of insulating cloth from the coil.

When the field coil opens up and the break cannot be repaired, it is sometimes possible to remove the defective field coil and install a new one. In many of the smaller modern loudspeakers, however, the core of the field coil is permanently anchored in position during manufacture, making it impossible to remove the field coil. The entire loudspeaker must then be replaced.

**Replacing a Loudspeaker.** After a receiver has been in use for three or four years, the paper used in its cone

becomes quite brittle, and this appreciably affects the tone quality of the loudspeaker. The installation of a new cone or a new loudspeaker will usually provide a definite improvement in tone quality, with a corresponding increase in customer satisfaction at your work.

When replacing a loudspeaker with an exact duplicate replacement unit obtained from the receiver manufacturer or a radio supply house, all you need to be careful about is the connections. Make a rough sketch showing the terminals to which the voice coil and field coil leads of the old loudspeaker were connected, and use this as your guide for connecting the leads of the new loudspeaker.

### The Output Transformer

The small radio part mounted on top of the loudspeaker in Fig. 11 is the *output transformer*. Its job, as you know, is to transfer audio signals from the audio output tube of the receiver to the loudspeaker in such a way that the loudspeaker gets the low-voltage, high-current signal it requires for most efficient operation.

**Construction.** The output transformer has two coils of wire, wound one over the other on an iron core. The primary coil contains many turns of fine wire. The secondary has fewer turns but uses heavier wire. The primary coil is connected to the output of the receiver, while the secondary coil is connected to the voice coil of the loudspeaker.

**Open Coil.** A break in either of the coils of wire in the output transformer will cause failure of a radio receiver. When such a break occurs, the serviceman says that the output transformer is *open*, and replaces the entire transformer. If the old transformer is held in place with rivets, he either drills out the rivets or cuts off

their spread-out ends with side-cutting pliers. There are usually only four connections to unsolder, so this is a fairly simple service job. Either rivets or nuts and bolts can be used for mounting the new transformer.

Once the trouble has been isolated to the audio output stage by professional servicing techniques, the output transformer can be checked by two simple ohmmeter measurements, one across each winding.

### The Gang Tuning Condenser

In Fig. 12 is a clear illustration of another highly important large part in our superheterodyne receiver, the *gang tuning condenser*. It has two sections, which work with the two r.f. coils located underneath the chassis. A tuning condenser is never used by itself; it is always used with a coil, and *tunes* the coil to a particular frequency. You will find this interesting situation many, many times in radio—different parts acting together to produce special results which the parts could not give separately.

One gang tuning condenser section is connected to an r.f. coil in the input circuit of the mixer-first detector input, and tunes the r.f. coil to the frequency of the desired incoming signal. The other gang tuning condenser section is connected to the r.f. oscillator coil in the oscillator circuit, and tunes this coil to a frequency exactly 456 kc. higher than the desired incoming carrier frequency.

**Rotor and Stator.** Each section of our gang tuning condenser in Fig. 12 has two important parts:

1. A *rotor*, which in this particular unit has eleven aluminum plates. Both rotors are mounted on the same condenser shaft.

2. A *stator*, which consists here of ten aluminum plates mounted on but *insulated* from the condenser frame.

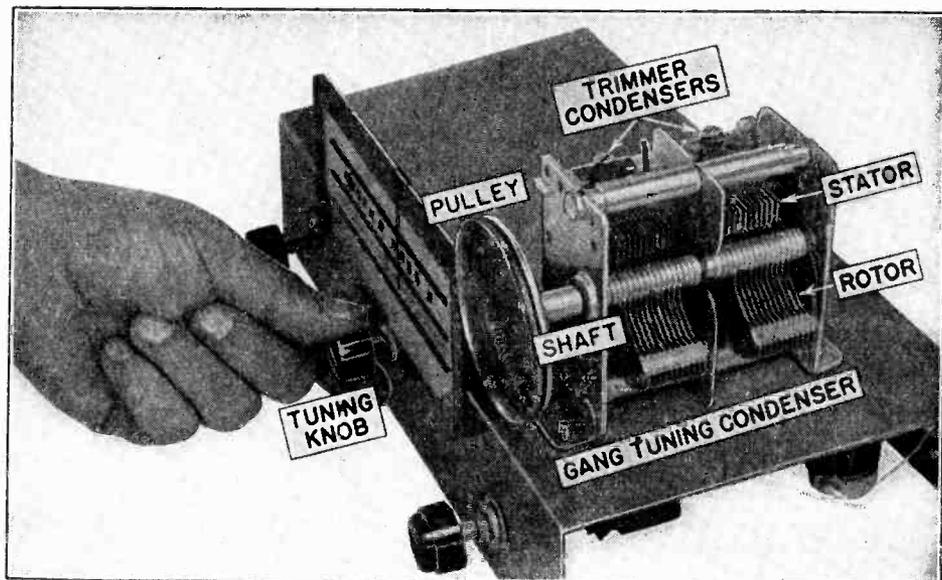


FIG. 12. View of the gang tuning condenser in our superheterodyne receiver, with important parts identified.

The rotor of each tuning condenser section *rotates*, but the stator of each section is stationary, staying in the same position all the time.

**How the Capacity Varies.** When the shaft of the gang tuning condenser is turned in one direction, the rotor plates go farther in between the stator plates, without ever touching the stator plates. This increases the electrical size or *electrical capacity* of each tuning condenser section.

Turning the shaft in the opposite direction brings the rotor plates out from the stator plates, reducing the electrical capacity of each section.

It is this change in the electrical capacity of each gang tuning condenser section which tunes the receiver to different stations. In later lessons you will learn exactly how this is done.

**Dial Cord.** The tuning control knob in our superheterodyne receiver is quite some distance from the gang tuning condenser shaft, as you have probably noticed already in Fig. 12.

Now, what makes the tuning condenser shaft rotate when we turn the tuning control knob? Also, what makes the dial pointer move to the correct dial number for the station whose program is coming from the loudspeaker?

Figure 13 shows the answer—simply a length of braided dial cord and a few pulleys. The cord runs around a small pulley on the tuning control knob shaft, around a large pulley on the gang tuning condenser shaft, and around two idler pulleys. A spring inside the large pulley keeps the cord tight.

The pointer is attached to the dial cord at the position shown. When you turn the tuning control knob in a clockwise direction, the sliding pointer moves to the right along the tuning dial, and the gang tuning condenser shaft turns in a clockwise direction.

The average radio set owner never sees this cord-and-pulley system, because it is at the front of the chassis, and cannot be seen when the chassis is in its cabinet. For this reason, a

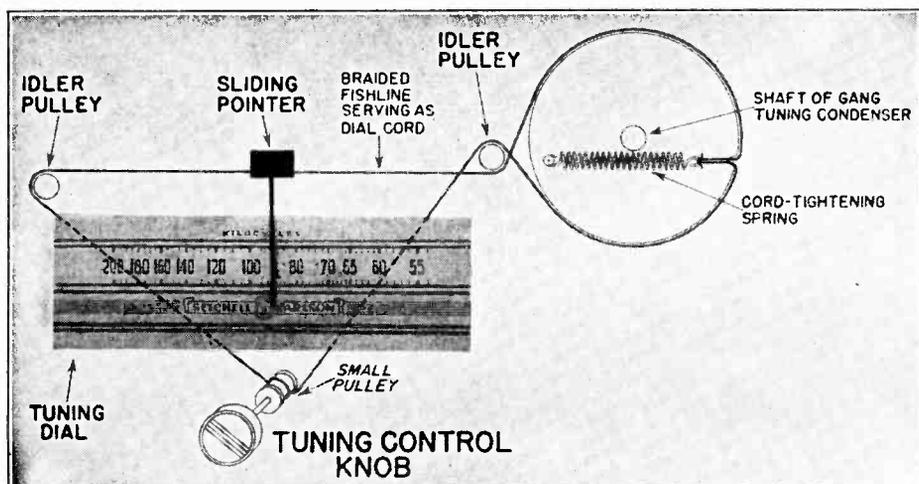


FIG. 13. This diagram shows how a length of fishline is used to make the pointer slide over the tuning dial and make the gang tuning condenser rotate when the tuning knob is turned.

radio serviceman will usually be called even for a mechanical trouble such as a broken dial cord.

**Restringing a Dial Cord.** The diagram in Fig. 13 also gives you the necessary information for restringing the dial cord of our superheterodyne receiver. After removing the old dial cord, tie one end of the new cord to the spring, bring the cord entirely around the system exactly as shown in the diagram, then tie the other end of the cord to the spring and cut off surplus cord.

Now tune in a local station whose frequency you know, then push the pointer over the cord at the frequency number of this station. There are spring clips behind the pointer for gripping the cord. If restringing is properly done, the pointer will now move to the correct dial number for whatever station you tune in.

**Bent Rotor Plates.** Defects in a gang tuning condenser are generally easy to locate and repair. For instance, one of the rotor plates may become bent so that it touches an adjacent stator plate. This shorts one

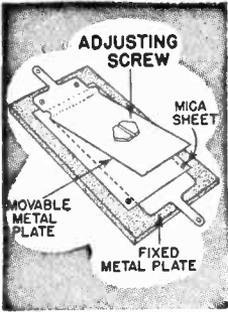
of the tuning circuits, and the receiver no longer plays. In other cases, the short is only momentary, and causes noise in the loudspeaker as the receiver is tuned.

Sometimes you can hear the condenser plates scraping against each other if you listen closely while turning the tuning control knob. The remedy simply involves straightening the bent plate with a putty knife or a thin-bladed screwdriver.

### Trimmer Condensers

On top of each section of the gang tuning condenser is a small adjustable device known as a *trimmer condenser*. The purpose of these two trimmer condensers is to take care of any small differences which might exist between the two sections of the gang tuning condenser.

When the two tuning condenser sections are correctly matched by adjusting the "trimmers," both sections will automatically be tuned correctly for reception of a desired station when the tuning dial pointer is at the station frequency.



**Construction.** Each trimmer condenser consists of one fixed and one movable metal plate, separated by a sheet of *mica*, an insulating material. Tightening the adjusting screw brings the

movable plate closer to the fixed plate, and thus increases the electrical size or capacity of the trimmer. Loosening the screw reduces the capacity of the trimmer.

The construction of a trimmer condenser is so simple that this unit rarely becomes defective. Correct adjustment is the chief problem of the radio man when working with the trimmer condensers in a receiver.

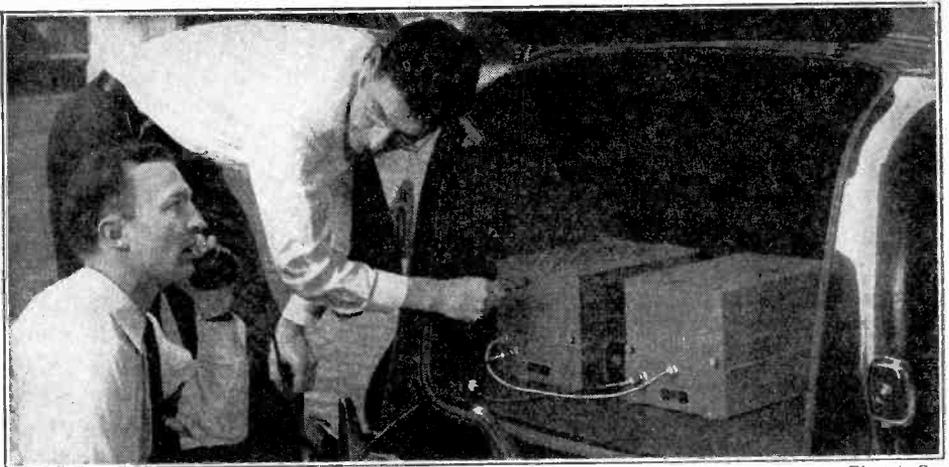
**Adjustment.** Although the actual adjustment of a trimmer condenser with a screwdriver or wrench is an extremely simple job, considerable knowledge of radio servicing techniques and radio fundamentals is nec-

essary in order to know when and how much to turn the screwdriver or wrench during the alignment procedure. This is another highly practical technique which you are going to learn in the N.R.I. Course.

### I. F. Transformers

Also on top of the chassis base of our superheterodyne receiver are two parts which look like square metal cans. Through the two holes in the top of each can, we can see adjusting screws if we look carefully. These screws are fairly reliable identifying clues which tell that the parts are *i.f. transformers*. The square cans are simply *shields* which protect the inner coils from the effects of stray electric and magnetic fields. These shields also keep the fields of the coils from affecting other parts.

**Construction.** Figure 14 shows what you would see if you removed a few nuts, then lifted up the square shield of one of the *i.f. transformers*. Inside this shield are two small coils mounted on a wood rod. Above the coils are two trimmer condensers, one



*Courtesy Western Electric Co.*

The radio man at the right is adjusting a trimmer condenser in the transmitter unit of a two-way mobile police radio set, while the other man listens to the effect of the adjustment with a headset plugged temporarily into a test jack on the transmitter

connected across each coil. These are the trimmer condensers which you adjust with a screwdriver through the holes in the top of the shield during alignment of the receiver.

**What I.F. Transformers Do.** When the gang tuning condenser and its coils are adjusted or tuned to a desired incoming carrier signal, these parts cannot always keep out undesired signals having different frequencies. I.F. transformers are much more selective, however.

The only signal which can get through a properly adjusted i.f. transformer without serious reduction in strength is the one whose carrier frequency is the i.f. value of the receiver (456 kc. in this case).

The oscillator-mixer-first detector automatically changes the carrier frequency of the desired incoming signal to the i.f. value, but changes undesired signals to other frequencies which cannot get through. One i.f. transformer transfers the desired signal from the mixer-first detector to the i.f. stage, and the other i.f. transformer transfers the desired signal from the i.f. stage to the second detector.

**Open Coils.** The coils in an i.f. transformer are made from small-size insulated copper wire. This wire sometimes breaks due to vibration or jarring, to strains set up by changes in temperature, to corrosion at soldered joints, or to swelling of the coil form in damp weather. Any break in the coil wire interrupts the signal circuit, thereby causing complete failure of the receiver. If the break is at a joint, it can sometimes be repaired by soldering. If not, a new i.f. transformer must be installed.

Sometimes a connection may be weakened by corrosion without being completely open. This defect can



FIG. 14. This is what you see inside the i. f. transformer when you lift the metal shield can.

cause loud crackling noises in the loudspeaker as the connection opens and closes at irregular intervals.

These are just a few of the many receiver troubles which cannot be seen by examining the parts. A systematic and efficient trouble-locating procedure such as you learn in the N. R. I. Course is absolutely necessary for locating defects like open i.f. transformers.

### Power Transformer

We now have only one device left to consider above the chassis base of our superheterodyne receiver. This is the *power transformer*, a large and heavy unit located at the right of the loudspeaker when looking at the back of the chassis. Since all of the leads of this radio part are located underneath the chassis, however, let us postpone its study until we turn the chassis base upside down.

# Under-Chassis Parts

Turning the chassis of our superheterodyne receiver upside down, as in Fig. 15, we find that most of the actual wiring and practically all of the small radio parts are underneath. As a result, the bottom of the radio chassis may look somewhat complicated to you now.

Don't let this worry you, however. A radio man never has to figure out all at once what every single part under the chassis is doing. When he works

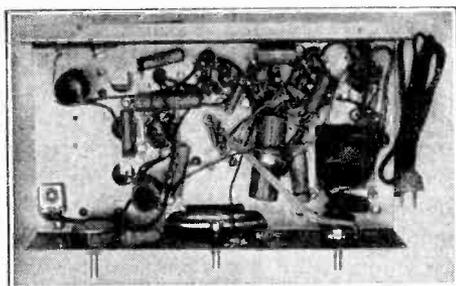


FIG. 15. When you turn the chassis of our superheterodyne receiver upside down for a casual inspection, this is what you see. A radio man doesn't care how complicated a set looks underneath its chassis, however, because he looks only for certain parts or circuits and disregards all the rest of the things underneath the chassis.

underneath the chassis, he is looking for one particular circuit or part, and disregards all the other parts.

In this section, you will see a number of under-chassis views, each with only certain groups of parts showing. These views really represent what the trained eyes of a radio man might see as he works on the chassis.

## Soldered Joints

Poorly soldered joints between groups of wires or between wires and terminals are a possible cause of receiver failure, but bad joints are just about the easiest defects to repair.

There are over 75 soldered connections which can go bad in our superheterodyne receiver. A loose joint can cause annoying crashing sounds to be heard with the radio program from the loudspeaker when the receiver is jarred, or can momentarily affect receiver performance in other ways.

**Servicing Technique.** Here is how you could go about locating a bad joint. Take the chassis out of the receiver cabinet and set the chassis upside down in such a way that you can see all of the joints under the chassis. Now take a stick of wood and push against each of the joints or leads in turn under the chassis while the receiver is in operation and tuned to a station.

If you hear the crashing sound from the loudspeaker as you move one of the joints or its leads, you know that the joint is loose and should be resoldered. A joint can be loose and defective even though it looks good, because it takes only a very small separation between wires to cause trouble.

After you learn a bit more about radio fundamentals and servicing techniques, you will become able to tell which section or stage has a defective joint, and will then have to check only the joints in that section.

## Resistors

A careful count reveals that there are nine small carbon resistors mounted underneath the chassis of our superheterodyne receiver. The actual-size view in Fig. 16 of one of these resistors is typical of all the rest, for carbon resistors look much alike. The construction of this resistor is shown in Fig. 17.

Each resistor is in a current-carrying circuit. The current flowing through

a resistor produces a voltage drop across it. This voltage drop controls voltage and current in various ways both in the d.c. supply circuits for tubes and in the circuits which transfer signals from tube to tube.

**Resistor Color Code.** Carbon resistors are by far the commonest of all resistors used in radio equipment. Each resistor is marked with rings of color which tell the electrical size in

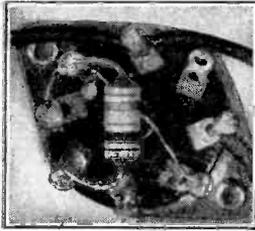


FIG. 16. Actual-size view of a carbon resistor under the chassis.

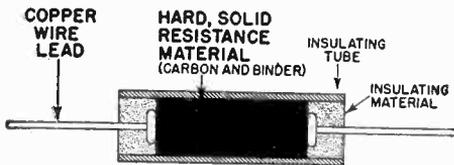


FIG. 17. Inside story of a carbon resistor. The two wire leads (pronounced LEEDS) make contact with the ends of the resistance material.

ohms just as accurately as would printed values. Elsewhere in the course, you will be shown how to read the color rings on resistors.

**Resistor Troubles.** A resistor can fail because of a break in one of its leads, usually inside the resistor or at the point where the wire lead enters the body of the resistor. A crosswise crack in the carbon resistance element can also cause failure of a resistor. In both cases, the radio man says that the resistor is *open*, for the break or crack blocks electron flow through the resistor.

**Locating a Bad Resistor.** A defective resistor can always be located by making tests with servicing instruments, chiefly with an ohmmeter.

**Burned-Out Resistors.** Resistors also become defective due to overheating, which occurs when an excessively large current flows through the resistor.

Overheating may cause the resistor to open up or change its electrical value, but radio men will simply say that the resistor has *burned out*. Such a resistor can usually be spotted at a glance because its colors and markings are blackened by the heat. A blackened resistor is not necessarily defective, however; it may have been overloaded at some time, but not enough to cause failure.

### Condensers

Four different types of condensers are to be found underneath the chassis of our superheterodyne receiver—*paper* condensers, *electrolytic* condensers, *mica* condensers, and *trimmer* condensers. The illustration in Fig. 18 shows all of these condensers, with other parts painted out for clearness.

In basic electrical action, all four types of condensers are alike, in that they all control voltage and current in signal and supply circuits. The chief differences in condensers occur in the method of construction and in the electrical sizes.

**Paper Condensers.** The condenser used more than any other type in radio equipment is the *paper condenser*. This name comes from the fact that the material used for insulation inside the condenser is waxed *paper*. The construction is shown in Fig. 19.

**Loose Leads.** A poor connection between a foil strip and one of the wire leads is a common cause of trouble in a paper condenser. A defect of this nature can sometimes be located by wiggling the body of each condenser under the chassis while the receiver is turned on. The defective con-

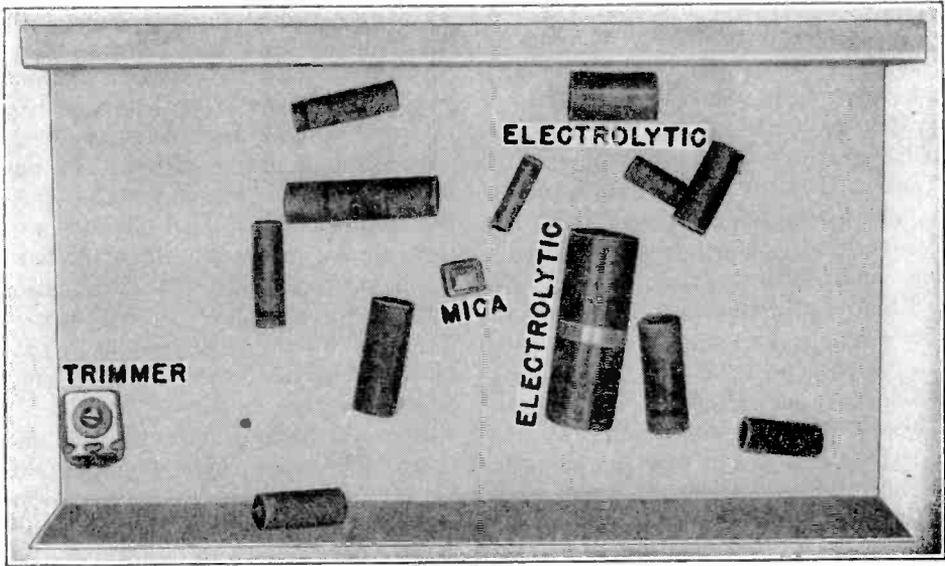


FIG. 18. If only the condensers underneath the chassis of our superhet receiver were visible, this is what you would see. The condensers not marked are paper condensers.

denser will usually cause a change in loudspeaker volume when this is done.

*Shorted or Leaky Condensers.* If the waxed paper insulation between the two foil strips of a paper condenser fails completely (becomes punctured, so that the metal strips touch each other through a small hole in the paper), the condenser is said to be *shorted*.

Just as the high voltage between the points of an auto spark plug can burn a hole through a sheet of paper, so can high voltage burn a hole through the insulation of a paper condenser. This paper insulation weakens with age, so that eventually the normal circuit voltage is sufficient to short the condenser.

If the insulation becomes *partly* defective, so that a few electrons can flow from one strip to the other through the paper, the condenser is said to be *leaky*. Simple measurements with an *ohmmeter* will locate shorted or leaky paper condensers.

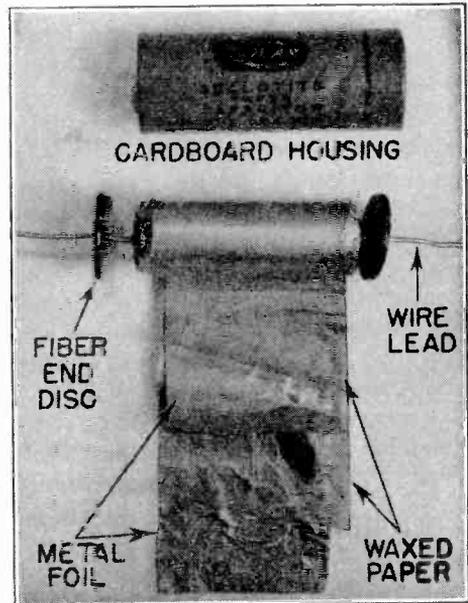


FIG. 19. This photo shows you what's inside a paper condenser. Two long strips of metal foil separated by strips of waxed paper are rolled together, with the two foil strips projecting from opposite ends of the roll and making contact with the two wire leads. The cardboard housing and fiber end discs keep out moisture.

*Capacity.* Since one entire lesson of your N.R.I. Course is devoted to condensers of various types, the manner in which condensers are rated according to electrical size and working voltage need not be taken up in this lesson. Now we just want to make a general acquaintance with the different kinds of condensers, so let's pass on to the next type.

**Electrolytic Condensers.** Two of the condensers in our receiver are of a special type known as *electrolytic condensers* or simply *electrolytics*. These condensers look much the same as paper condensers, and are made in a similar manner except that there is no paper between the rolled-up aluminum sheets. Instead, the sheets are separated by a moist chemical paste which produces an *insulating film* on the aluminum sheets through *electrolytic action*. (A scientist would explain that electrolytic action occurs when an electric current is sent through chemical materials.)

*Leaky Electrolytics.* Electrolytic condensers are common causes of radio receiver trouble, for the pasty chemical material spoils with age, reducing the effectiveness of the insulating film. Radio men say that the condenser has then become *leaky*, because the insulating film allows electrons to *leak through* the condenser when they should be held back.

*Connections.* The large electrolytic condenser in this receiver is really two electrolytic condensers combined in one housing. Three insulated leads, colored red, black and blue respectively, come out of one end of the cardboard housing.

The leads are colored for two reasons, to distinguish between the two sections of the condenser and to indicate the correct polarity of connections. An electrolytic condenser will

work as a condenser only when connected in a certain way. Correct connections are designated by calling one lead *positive* and the other *negative*; the positive lead must go to the positive terminal of the circuit. In our large electrolytic condenser, the black wire is the negative lead for both sections of the condenser. The red wire is the positive lead for one condenser section, and the blue wire is the positive lead for the other section. This information is marked on the condenser housing.

It is common practice to combine two or more electrolytic condensers in one housing in this way. A careful radio man always makes a rough sketch of connections before removing an old condenser, to reduce chances for errors when connecting the new unit.

**Mica Condensers.** There is only one *mica condenser* in our receiver. It is a tiny unit encased in molded brown bakelite, and located just about in the exact center underneath the chassis. (See Fig. 18.) The mica condenser gets its name from the fact that sheet *mica* is used as the insulation between the metal plates. The construction is shown in Fig. 20.

*Possible Troubles.* Mica condensers seldom fail in use, but are more costly than paper condensers. Occasionally a wire lead may break at the point where it enters the bakelite body of a mica condenser. The break may not always be visible, but can be located by wiggling each of the mica condensers in turn while the receiver is in operation. When a defective condenser is moved in this manner, it affects the loudness of the sound coming from the loudspeaker.

The mica insulation can fail due to moisture or to excessively high voltages. Adjacent metal plates then

touch each other, causing a *shorted* mica condenser. A simple ohmmeter test will reveal this trouble once the defective condenser is located.

**Trimmer Condenser.** As you can see in Fig. 18, there is only one trimmer condenser mounted underneath the chassis. This unit is constructed very much like the trimmer condensers already studied.

Just as with the other trimmer condensers in this receiver, the chief problem is *knowing how* to make the simple screwdriver adjustment which is required during alignment of the receiver.

This particular trimmer condenser is technically known as the *oscillator paddler*, because it is connected in series with the tuned circuit of the oscillator. You will learn later that this trimmer is adjusted to make the tuning dial readings more accurate at the low-frequency end of the tuning dial.

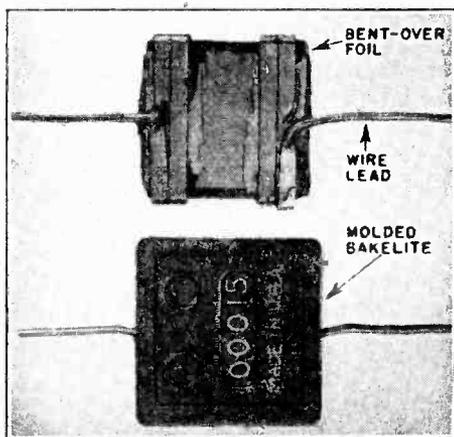
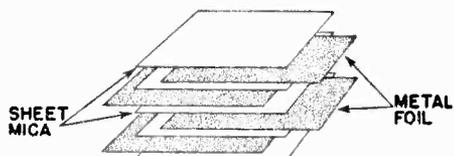


FIG. 20. How a mica condenser is made.

## R. F. Coils

There are two *r.f. coils* (radio frequency coils) under the chassis. Each r.f. coil has the basic construction shown in Fig. 21, in which there are two windings placed side by side on a waxed cardboard or bakelite coil form. These windings may either be in a single layer as shown in Fig. 21, in many layers wound one over the other in a criss-cross manner as they are on one of the coils in this receiver, or even with one coil wound right over the other.

One of the r.f. coils is known as the *antenna coil*, because it has a direct

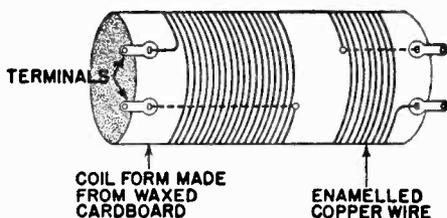


FIG. 21. Construction of an r.f. coil.

connection to the antenna terminal of the receiver. Its job is to transfer the incoming modulated r.f. carrier signal from the antenna circuit to the grid circuit of the mixer-first detector tube.

The other r.f. coil is known as the *oscillator coil*, because it is a part of the tuned circuit of the r.f. oscillator.

These r.f. coils work hand in hand with the two sections of the gang tuning condenser to let only the one desired carrier signal get through the receiver.

R.F. coils have much the same troubles as i.f. transformer coils. Thus, they may have broken wires or poor connections at terminals. Sometimes adjacent turns of wire may touch each other and cause shorted turns. When a radio man finds that an r.f. coil is defective, he invariably replaces the coil with a new one, be-

cause it is difficult to make satisfactory repairs with the very fine wire used in winding these coils.

### Power Transformer

When you see a large, heavy unit like that in Fig. 22 on top of the chassis of a radio receiver, you can be sure it is the *power transformer*.

All connecting wires for the power transformer in our receiver are underneath the chassis. The wires have various colors for purposes of identification.

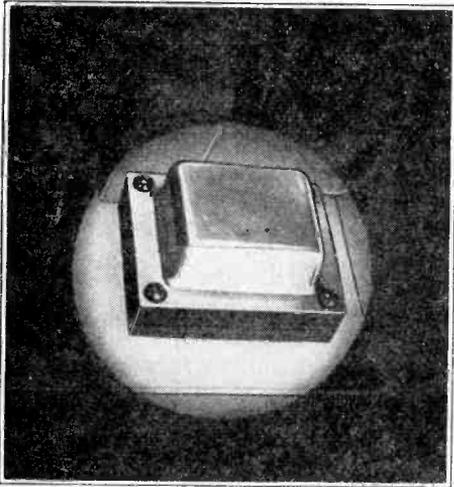


FIG. 22. The power transformer in a radio set is a heavy, bulky unit. It is covered with a metal cap for two reasons, to protect prying fingers from the high voltages which exist in the coils of wire inside, and to prevent the magnetic effects inside from escaping and interfering with other parts

**Construction.** A power transformer consists of a number of separate coils of insulated wire, wound one over the other on a core made from thin strips of soft steel stacked together.

**Purpose.** The power transformer in a radio set transforms the a.c. power line voltage to the various higher and lower a.c. voltage values which are required by the tubes in the receiver. The more tubes there

are, the larger must be the power transformer if it is to supply these tubes with the required electrical power without getting too hot.

**Possible Troubles.** There are a number of defects in receiver circuits which can place an *excessive load* on the power transformer. Overloading causes a power transformer to get hot, and this excessive heat damages the insulation between the windings of the power transformer.

When the insulation on the wires or between the coils of a power transformer loses its insulating qualities because of heat, adjacent turns touch each other and cause a short circuit. This allows more current to flow, and produces still more heat. The condition becomes worse and worse. Eventually the trouble becomes so bad that the wire in the transformer melts and opens up at some point, causing complete failure of the receiver.

A radio serviceman often relies upon his nose to tell him when a power transformer is bad. Failure of this part is usually due to the overheating just described. The resulting smell of burned insulation is easily recognized, hence a radio man suspects the power transformer first of all whenever he encounters this strong, unpleasant odor of burned insulation.

An ordinary break in one of the wires inside a power transformer is rather rare, and yet may happen occasionally. Either voltmeter or ohmmeter measurements at the power transformer would reveal this trouble. Poor connections at the power transformer terminals would have a similar effect.

Defective power transformers are replaced rather than repaired. The replacement offers no difficulty if

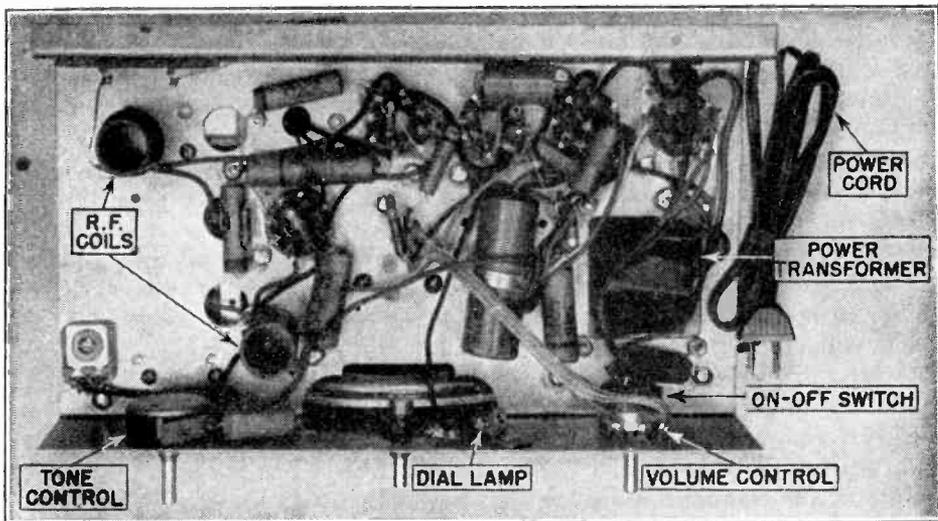


FIG. 23. Additional under-chassis parts of our superhet receiver are identified in this view.

you make a careful diagram of the original connections before disconnecting the damaged transformer.

#### Other Under-Chassis Parts

As you can see in Fig. 23, there are still quite a few parts underneath the chassis of our receiver which have not yet been taken up. Most of these are highly important parts which often require attention by a radio serviceman, so a brief study of each will be well worth while.

**Power Cord.** One trouble which occasionally baffles even the most expert radio man because of its very simplicity and obviousness is a *defective power cord*. This cord shows clearly in Fig. 23, and serves to connect the receiver to an a.c. power line. The cord has a standard wall outlet plug at one end, and the two leads at the other end connect to two points underneath the chassis of the receiver.

The most likely places for trouble to occur in a power cord are at the plug, at the receiver connection, and at the point where the cord passes through a hole in the chassis. Ex-

amine all these places carefully for poor connections or damaged insulation.

The rubber used in a power line cord ordinarily lasts only about five years. Therefore, whenever you encounter an old receiver having brittle or cracked rubber insulation on the power line cord, it is best to install a new cord. The replacement job is quite simple, as it involves only the resoldering of two connections.

Always examine the power cord plug carefully whenever you work on a receiver, to make sure that there are no loose strands of wire which might eventually touch the opposite terminal of the plug and cause a short which would blow the fuses in the house. Tighten the screws in the plug if they are loose.

**On-Off Switch and Volume Control.** As you already know, a single control knob serves for both of these parts. The on-off switch is mounted at the back of the volume control. The switch is connected in series with one of the power cord wires, so that it can open the power supply circuit.

The on-off switch rarely gives trouble. About the only thing you need watch for is a poor connection at either of the two switch terminals.

The volume control is definitely a trouble-maker, however, in any radio receiver. In this set, it is in the circuit of the second detector, and governs the strength of the audio signal which is fed into the first a.f. stage by the second detector.

The volume control in this receiver is a type of variable resistor which consists of a strip of carbon resistance material, and a rotating contact arm which sweeps over the carbon material. The contact arm can be set to make contact at any desired point along the resistance material.

If the carbon material in a volume control wears away due to repeated use, or if the contact arm becomes loose, we have the equivalent of a poor connection. The radio man says the volume control has become *noisy*, because a noise is heard from the loudspeaker every time the control is adjusted.

The only remedy for a noisy volume control is replacement with a new volume control. It is usually customary to order a new switch at the same time, since the two are furnished as a single unit.

If you make a rough picture diagram of the connections to a defective volume control before removing it, no trouble should be experienced in making connections to the new unit. Even experienced servicemen follow this procedure, to reduce chances for mistakes.

**Shielded Wire.** One of the leads going to the volume control in our receiver is enclosed in a tube of braided wire which serves as a shield. This shield is necessary to prevent that

particular wire from being affected by stray electric and magnetic fields around the chassis. Without this shielded wire, the set might have an annoying squeal or hum.

**Tone Control.** The tone control circuit in this receiver consists of a variable resistor connected in series with a condenser in the audio output stage. The condenser side-tracks the higher audio frequencies, and the variable resistor determines *how much* the higher frequencies will be side-tracked or suppressed. This variable resistor thus controls the *tone* of the program coming from the loudspeaker.

The tone control resistor is less likely to cause trouble than the volume control, but the two parts are similar and do have the same types of trouble.

**Dial Lamp.** Mounted behind a hole in the front of the chassis is a small socket containing a *dial lamp*, also known as a *pilot lamp*. This lamp illuminates the tuning dial when the receiver is turned on, and thus also serves to indicate whether or not the receiver is turned on. The dial lamp in our receiver requires a voltage of 6.3 volts, the same as the filaments of the tubes in the set. The lamp is therefore connected in parallel with the tube filaments.

There is only one wire connection to the dial lamp and to each tube filament, because the circuit to the other terminal is in each case completed through the metal chassis. This arrangement is common practice in radio sets, and is illustrated in Fig. 24.

A dial lamp has about the same life as an ordinary electric light bulb in a home, and hence requires replacement occasionally. The easiest way to insure getting the correct new lamp

is to take the old bulb to the radio supply store and ask for a similar new bulb.

When ordering a dial lamp by mail from a catalog, select one having the same type of base, the same voltage rating, and the same color of glass bead inside the lamp as that in the original lamp. The color of this glass bead is a clue to the current required by the lamp.

In some receivers, the color of the glass bead is unimportant, but in others the wrong pilot lamp can defi-

ous terminals of the old socket before you remove it.

When sockets are riveted to the chassis, they can be removed by drilling out the rivets. The new socket can be mounted with bolts and nuts or with new rivets.

**Conclusion.** Remember that failure of a radio receiver is due to some simple breakdown in one or more of its parts. The breakdown is purely mechanical, such as a broken wire, defective terminal or parts touching each other. Usually the breakdown

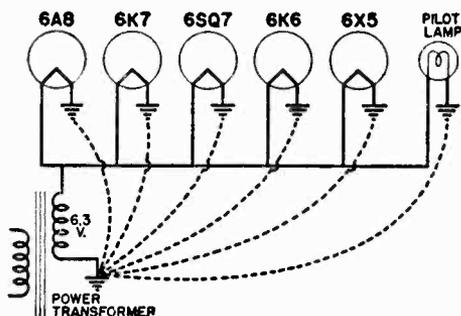


FIG. 24. Filament circuit, with dotted lines indicating the connections which are completed through the chassis of the receiver. All tube filaments and the pilot lamp require 6.3 volts, and the transformer secondary provides this voltage.

nately cause tubes in the receiver to burn out, as you will learn later.

**Tube Sockets.** The five tube sockets on the chassis of this receiver are of modern design, and rarely if ever give trouble. However, on older sets you will occasionally encounter tube socket contacts which do not grip the tube prongs tightly enough. If it is impractical to tighten the socket clips sufficiently, a new socket must be installed. The installation of a new tube socket should offer no difficulty to you even right now, if you make a careful sketch showing how connections were made to the vari-

ous terminals of the old socket before you remove it.

can be seen if you go to the trouble of taking the defective unit completely apart. Of course, it is impracticable to take apart every unit in a receiver when hunting for trouble. You won't have to do this, because the knowledge of how radio parts work in circuits and the radio servicing techniques you learn in the N.R.I. Course will enable you to locate defective parts speedily with simple test instruments. In some cases, you will even be able to spot the guilty part by noting the manner in which an ailing receiver is operating.

# Radio Receiver Servicing Techniques

If you have twelve identical radio receivers all with the same trouble, and give these receivers to twelve different radio servicemen for repair, the chances are pretty good that each man will use *a different method* for locating the trouble!

This is an entirely normal condition, because radio servicing techniques depend a great deal upon the amount of training and experience a man has. All twelve men will undoubtedly find the trouble, but some will find it much faster than others, because their methods are more efficient.

To illustrate some of the methods which could be used to find trouble in a receiver, let us imagine that a defective receiver has come into your radio shop for repair.

**Check of Performance.** All servicing methods start with a thorough check of performance, to see just how the radio receiver is misbehaving. Oftentimes this check gives valuable clues to the source of trouble.

To make a performance check at a service bench, you plug the receiver power cord into a wall outlet, make antenna and ground connections if the set does not have a built-in aerial, then turn on the receiver and attempt to tune in stations. You note how the volume control, the tone control and any other controls affect performance. Your training and experience have taught you to pay particular attention to unusual squeals, noises or hum, because each tells its own story of trouble.

**Testing Tubes.** Since bad tubes are common causes of trouble in receivers and since tubes are easily tested, your next step is a test of each

tube in the receiver. Tubes should be tested at some point in the servicing procedure anyway, so this might as well be done first of all since it is so easy to do.

A beginner might prefer to check the tubes later if he didn't have a tube tester, for he would usually have to take the tubes to some reliable store for testing.

An expert serviceman might be able to figure out where the trouble is just by listening to the receiver, and clear up the trouble before making a routine check of the tubes. In the great majority of cases, however, you will find that tube testing comes first on the serviceman's list of trouble-hunting techniques.

Your aim in servicing is to narrow down the trouble first to a section of the receiver, then to a stage, and finally to just one circuit and to a particular part in that circuit.

The average superheterodyne receiver can be divided into six sections for servicing purposes. The *preselector* section includes all circuits between the antenna and the mixer-first detector. The *frequency converter* section includes the local r.f. oscillator stage and the mixer-first detector stage. The *i.f. amplifier* includes all i.f. stages and i.f. transformers. The *second detector* is just one stage. The *audio* section covers everything following the second detector, including the loudspeaker. The *power pack* section includes the rectifier tube and its associated parts.

Sometimes an experienced, properly trained serviceman can tell the section containing the trouble just by listening to how the receiver is misbehaving. If the faulty section has more

than one vacuum tube stage, he then makes simple tests which isolate the defective stage.

Other servicemen prefer to start right in with tests which locate the defective stage. Let us select just two of these tests now, and see how you would carry them out.

**Circuit Disturbance Test.** For a dead receiver, the simple circuit disturbance test is one of the fastest and most effective of these tests for locating the defective stage. It is carried out by introducing an electrical disturbance in each tube circuit in turn while the receiver is turned on.

You start at the audio output stage (next to the loudspeaker) and work backward through the stages when carrying out the circuit disturbance test. The disturbance which you introduce causes a click or thump in the loudspeaker when all the stages between the point of disturbance and the loudspeaker are good, but no click is heard when you arrive at the defective (dead) stage.

The electrical disturbance for this test can be inserted in a stage by removing a tube momentarily, by touching the top cap of the tube if there is one, by removing and replacing the top cap, or by shorting the grid of the tube momentarily to the cathode of the tube.

Since you must go from tube to tube in a definite order if a circuit disturbance test is to be effective, one requirement for making this test is the ability to identify the various tubes on top of the chassis. An entire lesson later in your Fundamental Course is devoted to this highly practical problem of finding your way around in a receiver chassis.

**Stage-Isolating Procedure.** For some receiver troubles, such as low volume, you introduce a test signal

in one stage after another, working toward the antenna. Again the loudspeaker tells when you have come to the defective section or stage, because the introduced signal will increase in volume for good stages but not when you move through the defective stage. For each other type of receiver complaint, there are one or more professional techniques for tracing the trouble.

**Locating the Defective Part.** Having located the defective stage in our ailing receiver, you could now follow the practice of some servicemen and begin testing the parts in the defective stage, one after the other. This eventually locates the defective part, but it can take a lot of valuable time.

As a professional radio serviceman, however, you would make additional general tests which gradually narrow the trouble down to a particular circuit in the stage. Then, by testing only a few radio parts at the most, you would locate the defective part in the quickest possible time.

It is in locating the defective stage and circuit that we find such a great variety of servicing techniques. There is no one technique which can be considered the best for all jobs; each has its own advantages, and will work best in a particular job. If you choose radio servicing for your advanced study, you will learn these servicing techniques and will also learn when to use them most effectively.

**Short Cuts.** After years of experience in radio servicing, a man learns to associate certain ailments and symptoms of a receiver with definite defects. He can then take short cuts through the general servicing procedure which bring him more rapidly to the defective part. One important

requirement for taking any short cuts, however, is a thorough knowledge of radio fundamentals. Unless these are thoroughly mastered, radio servicing becomes an unprofitable guess-and-dry proposition.

**Guess-Work Won't Pay.** If you consider how many different tubes, resistors, coils, condensers, transformers, connections and other parts there are in a radio receiver, the folly of expecting to service radio receivers simply by testing one part after another without thinking becomes clearly evident.

**You Can Start Soon.** Once you learn the fundamental facts about radio parts and circuits as presented in the early lessons, and learn how to perform some of the simpler tests and servicing techniques as outlined in your practical N.R.I. RMS Booklets, you will be in a position to start the N.R.I. practical training plan for securing radio receiver servicing experience right in your own home. After completing this training plan according to our instructions and mastering the essential radio information given in Fundamental Lessons, you will be able to do service work nearly as well as the average radio technician.

By the time you have finished your N.R.I. Course and followed all our experience-getting suggestions, you will be way ahead of the average radio technician. You will be using speedy, modern servicing techniques which allow you to fix any radio set whatsoever, and which enable you to handle

much more work than poorly-trained competitors can do in a similar period of time. Yes, N.R.I. graduates definitely get *more business* and make *more money* than less well-trained men.

After a defective part is located and replaced, the wise serviceman checks performance to make sure that there are no other defective parts. In addition, he readjusts or rather aligns the receiver for best performance, and cleans both the chassis and cabinet so as to restore the receiver as nearly as possible to its new condition. This final check-up and clean-up can prevent future trouble, and is definitely appreciated by customers. These extra services help to justify the fair charge which servicemen should make for their knowledge and experience.

### Looking Ahead

Having now completed the groundwork for your study of radio, you are ready to add to your knowledge of how radio parts work and what they do.

In the next lesson you will study different types of resistors, and learn how they are used in practical radio circuits to control voltage and current. The two following lessons will cover coils and condensers in the same thorough manner.

This next group of three lessons will thus complete your knowledge of three important parts used in radio circuits.

# Lesson Questions

**Be sure to number your Answer Sheet 4FR-2.**

**Place your Student Number on every Answer Sheet.**

***Send in your answers for this lesson immediately after you finish them, as instructed in the Study Schedule. Do this, and get the greatest possible benefit from our speedy personal grading service.***

1. If there is no set screw on the control knob of a receiver, how would you remove the knob?
2. Why is a shield placed around one of the tubes in our superheterodyne receiver?
3. Name three ways in which radio tubes commonly fail.
4. If a radio receiver sounds distorted, and you can hear a rubbing sound when you push the loudspeaker cone in and out with your fingers, what is the trouble?
5. You are tuning a radio receiver towards one end of the dial. The set suddenly goes dead as you tune, and all you can hear is the noise caused by the gang tuning condenser plates scraping against each other. How would you correct this trouble?
6. State briefly the purpose of the small trimmer condensers which are mounted on top of the gang tuning condenser.
7. What type of meter is ordinarily used to locate shorted or leaky paper condensers?
8. What happens to a power transformer when you overload it by drawing too much current from it?
9. How many connections must be unsoldered in order to disconnect the power cord of an a.c. superheterodyne receiver like that studied in this lesson?
10. At what stage do you start, when using the simple circuit-disturbance test in a dead receiver?

## THE VALUE OF COURTESY

A recent survey showed that people complain more about discourteous clerks than about any other fault a business could have. In fact, many people pay extra at higher-priced stores just to get the courtesy and respect they feel entitled to.

Regardless of whether you work for someone else or have a radio business of your own, plain ordinary courtesy can bring many extra dollars to you.

Genuine courtesy comes from the heart. When you say "*thank you*," be sure you sincerely mean it.

Courtesy becomes a habit if practiced long enough. Be courteous to everyone—to members of your family, to those who don't buy from you, even to the very lowest persons who serve you—then you can be sure you'll be courteous when it really counts.

Give your courtesy with a smile. There is an old Chinese proverb which says, "*A man who doesn't smile shouldn't keep a shop.*" And you're keeping a "shop" even if you are selling only your ability and knowledge to an employer. A friendly smile is itself courtesy of the highest type, bringing you unexpected returns in actual money as well as friendship.

J. E. SMITH

**RADIO RESISTORS  
AND  
HOW THEY ARE USED**

5FR-4

**NATIONAL RADIO INSTITUTE**

ESTABLISHED 1914

**WASHINGTON, D. C.**



# STUDY SCHEDULE NO. 5

Use this schedule to help you master the fifth part of your N.R.I. Course. Unless other instructions are given, study one step at a time, as follows: Read the specified pages first at your usual reading speed, then reread slowly one or more times until you understand the entire section. Finish with one quick reading to tie the ideas together in your mind, then figure out your answers to the Lesson Questions specified for that step.

- 1. Why Resistors Are Important in Radio. . . . . Page 1 of this lesson  
One careful reading of this introductory section is enough to show you the importance of learning about resistors.
- 2. Quickly review pages 3 to 13 of Lesson 3FR-2.  
These pages contain three sections: What Radio Men Measure; How Voltage and Resistance Affect Current in Simple Circuits; Load Connections for Radio Circuits. It is highly essential for you to review this material, so do it now. One quick reading should be enough.
- 3. Ohm's Law Tells "How Much" for Resistors. . . . Pages 2 to 6 of this lesson  
Ohm's Law, the radio man's most important rule for figuring out what *has* happened, *is* happening or *will* happen in a circuit containing resistance, is presented here in a complete, easy-to-understand manner. After you master this section, answer Lesson Questions 1, 2, 3, and 4.
- 4. Using Resistors to Reduce Voltage. . . . . Pages 7 to 9  
In this short section, you learn what to do when the source voltage is too high for the load, and master Kirchhoff's famous Voltage Law. Answer Lesson Question 5.
- 5. Resistors in Parallel. . . . . Pages 10 to 17  
Another short section, giving useful information on resistors in parallel and showing how Kirchhoff's Current Law applies to radio circuits. Answer Lesson Questions 6 and 7.
- 6. Wattage Ratings of Resistors. . . . . Pages 18 to 21  
Resistors give off heat when in use. If a resistor cannot withstand this heat, it burns out. The wattage rating is your guide as to whether a particular resistor will stand up in a particular circuit without becoming overheated. That's why this section is so important for practical radio work. Answer Lesson Questions 8 and 9.
- 7. How Resistors Are Made. . . . . Pages 21 to 28  
The characteristics, advantages and disadvantages of various types of resistor construction are taken up. Answer Lesson Question 10, then read "Looking Ahead" on page 28.
- 8. Mail Your Answers for Lesson 5FR-4 to N.R.I. for Grading.
- 9. How To Restraining Dial Cords and Set Push Buttons. . . RSM Booklet No. 5  
This is another of the series of Booklets that are to give you practical, how-to-do-it training in fixing radios. Read this Booklet now, then keep it handy for future reference. The "shop training" you get from these RSM Booklets is going to prove valuable to you throughout your service career, so you will want to read and review these Booklets many times. NOTE: Some of the Study Schedules in future Lessons may refer to "Job Sheets" instead of RSM Booklets. Read the correspondingly numbered RSM Booklet in such cases. These Booklets are an improved and enlarged version of the older Job Sheets, and are replacing them.
- 10. Start Studying the Next Lesson on Coils.

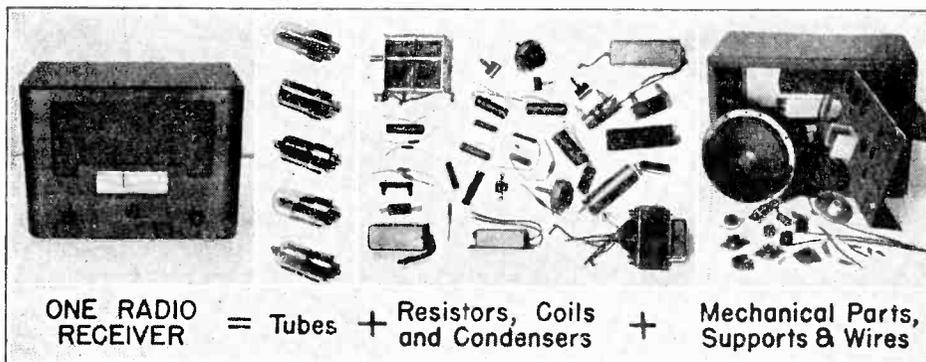
# RADIO RESISTORS AND HOW THEY ARE USED

## Why Resistors Are Important in Radio

**I**N addition to tubes, there are only three kinds of radio devices *which control voltages and currents in radio circuits*, and thereby make radio possible. These are *resistors, coils and condensers*. Terminals, wires, moving contacts or other mechanical parts merely provide connections or supports for these three basic radio parts.

You'll find resistors in power packs, reducing the high d.c. output voltage to the various lower voltage values required by the tubes in transmitters or receivers.

You'll often find resistors in tone control circuits, working with condensers to give you a choice of tone—mellow, brilliant or deep bass.



Thus, volume controls are combinations of *resistors* and mechanical parts. Audio, power and r.f. transformers are combinations of *coils*. I.F. transformers are combinations of *coils* and *condensers*. Loudspeakers are combinations of *coils* and mechanical parts.

**Jobs for Resistors.** You'll find resistors in practically every radio tube circuit. Sometimes they work alone or work with coils and condensers to keep radio signals in their correct paths. Sometimes they furnish the correct d.c. operating voltages for the tube electrodes. Sometimes they serve as loads for radio tube circuits.

In dozens and dozens of other places in radio equipment, you'll find resistors working silently and effectively to help make possible the magic of modern radio. Certainly, resistors deserve every bit of the attention we shall give them.

**A Trio of Lessons.** This lesson gives you practical information about resistors and the laws which govern their performance in radio circuits. The next two lessons give you the important basic facts about coils and condensers. Thus, in just three lessons you master the fundamental facts about three important radio parts.

# Ohm's Law Tells "How Much"

## Radio's Simplest Resistor Circuit.

In Fig. 1 is the simplest complete circuit we can have in radio apparatus. The source provides exactly the correct voltage value for the resistance load, so the source and load are connected together directly by wires.

This simple circuit has only three values—the source voltage, the current, and the load resistance. In a previous lesson, you learned that these values are related to each other in a logical way. For instance, the current can be increased either by raising the source voltage or lowering the load resistance. Likewise, the current can be reduced by lowering the voltage or raising the resistance.

In most radio jobs, it is sufficient to know only in this general way how the current is going to change when either the voltage or resistance are changed. There will be times, however, when you want to know *exactly how much* the new current value will

be.



The basic radio law which gives you this definite information is known to all radio men as Ohm's Law, because it was first discovered and proved to be true by the scientist

George Simon Ohm, in 1826.

Ohm's Law expresses the exact relationship between voltage, current and resistance for any circuit containing only resistors. You'll find later that you can also make this law work for other radio parts, too.

**Ohm's Law for Current.** To find the current, *divide the voltage value by the resistance value.*

The voltage value should be in *volts*

(as it usually is), and the resistance value should be *ohms*. The result will then be the current in *amperes*. For example, if the voltage is 50 volts and the resistance is 25 ohms, the current will be 2 amperes ( $50 \div 25 = 2$ ).

**Formula for Ohm's Law.** Radio men usually prefer to work with a simplified version of Ohm's Law, in

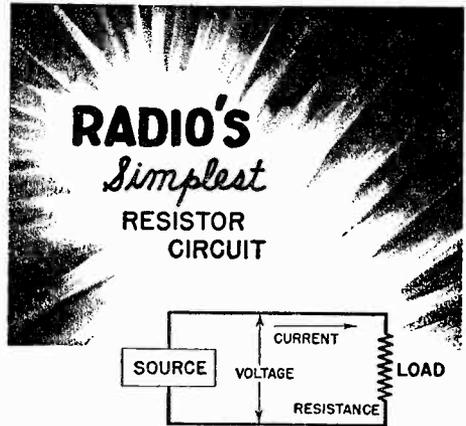


FIG. 1. Simplest basic circuit used in radio. The source voltage is exactly correct for the load, so these two parts are here connected together directly by wires. When the source voltage is too high for the load, however, the connection is made with a combination of wires and radio parts, chiefly resistors. The connecting system is sometimes known as a **TRANSMISSION SYSTEM**, because it serves to "transmit" power from the source to the load.

which letters and arithmetic signs replace words. These letters are an important part of the language of radio, so let us get acquainted with them now.

**R** is always used to represent **RESISTANCE**.

**I** is always used to represent **CURRENT**. (This may not seem logical, but is done to avoid confusion because the letter C is used for condensers. If you like, you can think of I as standing for *intensity of current*.)

**E** or **V** is used to represent **VOLTAGE**. (**E** comes from *electromotive force*, which is another name for a source voltage.)

By using these letters, we can condense Ohm's Law into a simple formula which is much easier to use.

Instead of saying that current (**I**) is equal to voltage (**E**) divided by resistance (**R**), we can simply say:  $I = E \div R$ .

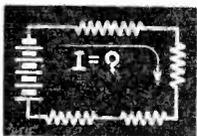
You will occasionally find Ohm's Law expressed in an even simpler manner. One letter is placed *under* the other to indicate division. The short line between the letters means that the *value above the line is to be divided by the value below the line*. This dividing line can either be slanting or horizontal. When we express Ohm's Law in this simplest possible form, here is what we get:

$$I = \frac{E}{R}$$

$$I = E/R$$

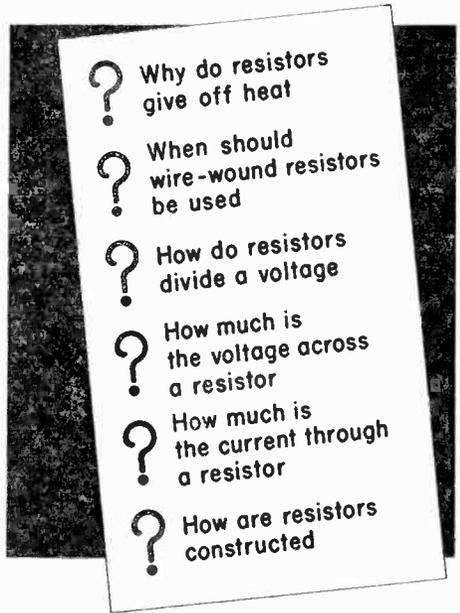
Both formulas are pronounced *I equals E over R*, and both mean the same thing—that the amount of current is equal to the amount of voltage divided by the amount of resistance.

Ohm's Law can be used to find the exact current value flowing through a complete d.c. circuit or through any portion of the circuit. Of course, if you are figuring the current in a *complete circuit*, all your values should be total values for the entire circuit. Likewise, when figuring the current in *part of a circuit* or in a single resistor, all your values must be for just that part of the circuit. A few simple examples will illustrate these entirely logical rules for applying Ohm's Law.



the battery voltage: let's say it is 90

Suppose we want to find the current in this series circuit. The voltage which acts on the *complete* circuit is

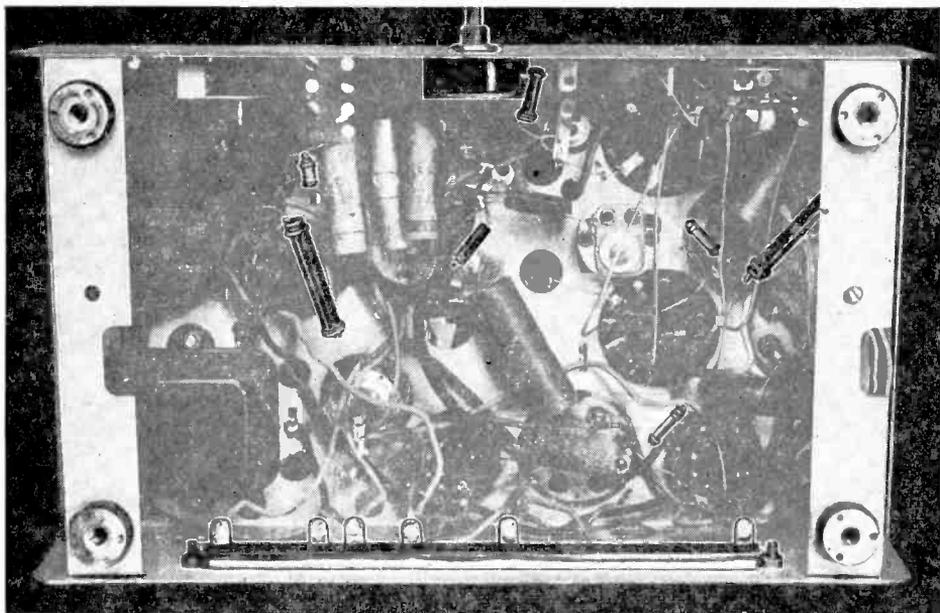


Here are just a few of the resistor questions which are answered in this lesson.

volts. The total circuit resistance is the sum of the individual resistor values since all four resistors are in series; let's say these values are known and add up to 30 ohms. The current value is equal to the voltage value divided by the resistance value, so we divide 90 by 30. This gives 3 amperes as the circuit current.

Now suppose we need to find the current through the single cathode resistor in this tube circuit, and haven't time to unsolder the cathode connection and insert an ammeter. We measure the voltage across the resistor with a d.c. voltmeter—let's say we find it's 12 volts. *Next*, we read the resistance value marked on the resistor—and let's say this is 300 ohms. Remembering  $I = E \div R$ , we divide 12 by 300, and get .04 ampere as the current through the resistor. Notice that we didn't have to pay any attention to the rest of the





No matter whether it's the voltage, current or resistance value you want to know for any one of the resistors underneath the chassis of this receiver, Ohm's Law can tell you how much the value will be.

circuit; we wanted to know the resistor current, so we used only the voltage and resistance values of the resistor itself.

One thing you must realize when using Ohm's Law is that you are dealing with *numbers*. You are not dividing volts by ohms. You can't do that any more than you can divide dollars by people. However, you can divide a *number* of dollars by a *number* of people, to get something entirely different—the *number* of dollars each person gets. That's what we do in radio—we divide the *number* of volts by the *number* of ohms, and get something entirely different—the *number* of amperes flowing. Keep this in mind, and you will never be asking, "How many volts are there in an ohm?" This question would be just like asking how many people there are in a dollar!

#### **Finding Voltage or Resistance.**

So far, we have used Ohm's Law only

to find the current when we know the voltage and resistance values. But sometimes we know what the current is, and want to find one of the other values instead. This can still be done with Ohm's Law—in fact, whenever *any two* of the three values (voltage, current and resistance) are known, Ohm's Law will help you find the third value.

It is perfectly possible to use our already-familiar current formula to find voltage or resistance, even though this might be a bit inconvenient. For example, if I was 2 amperes and E was 10 volts, but R was unknown, we could start with  $I = E/R$ . Putting our two known values in place of the letters, we would get  $2 = 10/R$ . This tells us that 10 divided by the value of R gives 2. There is only one value of R which will work here—5. Thus, we find that the resistance is 5 ohms.

In much the same way, the current formula could be used to find the

value of  $E$ . It works out all right, but in many cases it is easier and quicker to use Ohm's Law in two other forms, one for finding voltage and the other for finding resistance. While you are reading about these, keep in mind that the three versions of Ohm's Law are just convenient different ways of expressing *the same basic principle*.

**Ohm's Law for RESISTANCE.** To find the resistance, *divide the voltage value by the current value*. Thus, if the voltage is 10 volts and the current is 2 amperes, the resistance will be 5 ohms ( $10 \div 2 = 5$ ).

This version of Ohm's Law can likewise be expressed in simpler forms, as  $R = E \div I$  or  $R = E/I$ .



The filament circuit in a portable radio receiver makes an interesting example for this resistance-finding formula. The voltage is 1.5 volts, and let's say the rated filament current is .1 ampere for the tube shown here. First we write:  $R = E \div I$ . Next, we substitute the two known values:  $R =$

$1.5 \div .1$ , and divide, thus getting 15 ohms as the resistance of the filament when hot.

**Ohm's Law for VOLTAGE.** To find the voltage, *multiply the current value by the resistance value*. Thus, if the current is 2 amperes and the resistance is 5 ohms, the voltage will be 10 volts ( $2 \times 5 = 10$ ).

This version of Ohm's Law can be written as  $E = I \times R$ . Sometimes the multiplication sign is omitted, giving  $E = IR$ . (In formulas, letters are often placed side by side like this to indicate multiplication.)

To illustrate a practical example of this voltage-finding formula, we can take the plate circuit of almost any

vacuum tube and draw this one complete circuit by itself as shown in Fig. 2. We want to find out how much d.c. voltage to expect across plate load resistor  $R$ . All we do know is that this is a 50,000-ohm resistor with a

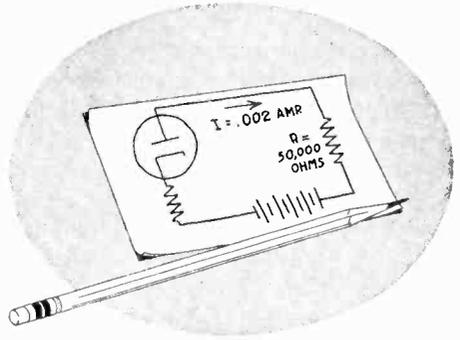


FIG. 2. Drawing the plate circuit of a tube by itself in this way simplifies any Ohm's Law figuring you might want to do for the resistors in the circuit.

current of .002 ampere flowing through it. These two values are all we need to use the formula  $E = I \times R$ . We get  $E = .002 \times 50,000$ . Multiplying this out gives 100 volts as the voltage drop which would normally exist across  $R$ .\*

**What About A.C. Circuits?** Ohm's Law holds true for all simple d.c. circuits and for all a.c. circuits which contain only resistors, provided that voltage values are in *volts*, current values are in *amperes*, and resistance values are in *ohms*.

\* As a special feature of the NRI Consultation Service, a Data Sheet on resistors will be sent to you with your graded answers for this Lesson. This Data Sheet contains a number of additional examples of applying Ohm's Law to practical resistor circuits, as well as a reference listing of all the formulas taken up in this Lesson.

You are not *required* to study this supplementary material; it is provided only as additional information for those who want to learn more about this subject. Remember, you won't have to do any involved calculating when you do radio servicing—these examples are given to help you understand what happens when parts of different resistances are used.

## Useful Data for Ohm's Law

In radio, current values are often in milliamperes, and resistance values are occasionally in megohms. Once in a while, you will even encounter millivolts and microvolts instead of volts. Before you can use these values with Ohm's Law, you must change them to amperes, ohms and volts.

**Changing Milliamperes to Amperes.** You will recall from a previous lesson that a milliampere is a thousandth of an ampere. Therefore, a value in milliamperes *must be divided by 1000* to change it to amperes. This is easily done *by moving the decimal point THREE places to the LEFT*. Here are a few examples:

.3 milliampere	=	.0003 ampere
1 milliampere	=	.001 ampere
25 milliamperes	=	.025 ampere
600 milliamperes	=	.6 ampere
1000 milliamperes	=	1 ampere

**Changing Megohms to Ohms.** In radio work, we often deal with resistance values running up into millions of ohms. To avoid having to write large numbers a larger unit of resistance called the *megohm* is used. One megohm is equal to one million ohms.

To change a resistance value in megohms into ohms, all you need to do is multiply by 1,000,000. The easiest way to do this is *by moving the decimal point SIX places to the RIGHT*. Here are some examples:

.1 megohm	=	100,000 ohms
.75 megohm	=	750,000 ohms
1 megohm	=	1,000,000 ohms
2.5 megohms	=	2,500,000 ohms
10 megohms	=	10,000,000 ohms

**Changing to Volts.** A millivolt is one thousandth of a volt, so a value in millivolts must be divided by 1000 to change it to volts. This can be done by moving the decimal point *THREE places to the LEFT*.

A microvolt is a millionth of a volt.

A value in microvolts **must be divided** by 1,000,000 to change it to volts. This can be done by moving the decimal point *SIX places to the LEFT*.

**Symbols for Ohms.** Resistance values appear so many times in radio diagrams that a single Greek letter is, for convenience, used instead of the word *ohm*. The two ways of writing this letter, or rather symbol, are shown in Fig. 3. You will find that

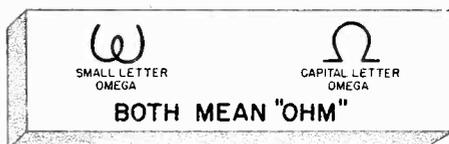


FIG. 3. Symbols used in radio for ohm. (The letter is omega, pronounced oh-MEE-gah. It is the last letter of the Greek alphabet.)

some diagrams use one way, some the other. On your own diagrams, you can choose whichever symbol is easier for you to make, because both are perfectly correct.

In rare cases, you may find both symbols appearing on the same diagram. The small letter  $\omega$  then represents *ohms*, and the capital letter  $\Omega$  represents megohms. Remember—this applies only when both symbols appear on the same diagram.

**Abbreviations.** On circuit diagrams the abbreviation *MEG.* is often used for megohms. Thus, a 10-megohm (ten million ohm) resistor would be marked 10 MEG. on a diagram.

You will sometimes find either the letter *M* or the letter *K* used to represent *thousands of ohms*. Whenever you see an *M* or *K*, just read it as thousand ohms. Thus, 250M and 250K would both be read as two hundred and fifty thousand ohms.

Figure 4 shows some of the most commonly used ways of specifying a typical resistance value on diagrams.

# Using Resistors to Reduce Voltage

**A Common Radio Problem.** When the source voltage in a d.c. circuit is *higher* than is required by the load, the source cannot be connected directly to the load with wires, because the excessively high source voltage would send too much current through the load and possibly damage it.

"Why not change the d.c. source voltage, or get a different load which can handle this higher voltage?", you ask. Either is a perfect solution, but unfortunately they are not always possible. The high source voltage may be needed for other circuits, and the particular load may be essential for satisfactory operation of the radio set. In many cases, the only practical solution is to place between the source and load some part which will take part of the source voltage, thus lowering the load voltage to the required value.

**A Resistor Will Do It.** If we place a resistor between source and load in the manner shown in Fig. 5, the current flowing through the resistor will

produce a voltage drop across it. This voltage drop cannot act on the load, so less voltage is available for the load. By choosing the proper resistance value for this series resistor, we can reduce the load voltage to any desired value.

A series resistor is often called a *voltage-dropping resistor*, because it uses up, wastes or "drops" the undesired portion of the source voltage.

A series voltage-dropping resistor can be anywhere in the circuit—next to the source, next to the load, on the



FIG. 5. Basic radio circuit in which the source voltage is too high for the load.

other side of the source, on the other side of the load, or anywhere in between source and load—*because the same current flows through all parts of a series circuit.*

There are two ways to look at this series resistor. First, we can think of it as increasing the total circuit resistance. (When resistors are in series, their ohmic values add together.) Increasing the circuit resistance reduces the current to the correct value for the load. When a load gets its correct current, it is also getting its correct voltage.

Here is the second way to think of this series resistor. Whenever current flows through a resistor, the voltage drop produced across this resistor is equal to the current value multiplied by the resistance value ( $E = I \times R$ ). The load voltage is then equal to the source voltage minus the voltage drop across the resistor.

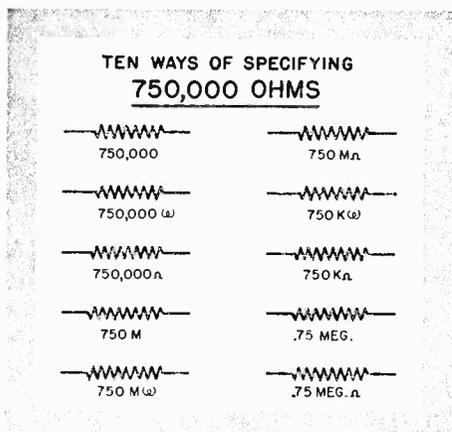


FIG. 4. You may find resistor values specified in any of these ways on circuit diagrams used in radio work. Don't try to memorize these ways now, because you can easily figure them out when necessary, and need refer to this diagram only when in doubt.

The important thing to remember is that a series resistor of the correct value will reduce the load voltage in a simple series circuit to its correct value.

**Can Voltage Vanish?** In the circuit of Fig. 5, we have three different voltages: 1. The source voltage; 2. The voltage drop across the series resistor; 3. The load voltage, which is also a voltage drop.

The relationship between these voltages is highly important because it tells you how much voltage to expect in actual radio circuits. Before we take up these voltage relationships, however, study the circuit in Fig. 5 a bit and see if you can answer the following questions from reasoning and what you already know:

- Can the load voltage ever be higher than the d.c. source voltage?

- Can the sum of the two voltage drops ever be HIGHER than the source voltage?

- Can the sum of the two voltage drops ever be LOWER than the source voltage?

- Can any of the source voltage just vanish?

**Kirchhoff's Voltage Law.** If you answered "no" to all four questions you were entirely correct, and already have a clear understanding of the highly important voltage law now to be taken up. This fundamental law is known as *Kirchhoff's Voltage Law*, in honor of the scientist who first proved that it always holds true.

**KIRCHHOFF'S VOLTAGE LAW**

**In a complete circuit, the sum of the voltage drops is always equal to the source voltage.**

**Examples.** Like Ohm's Law, Kirchhoff's Law has a great number of practical applications in radio circuits. Examples involving real radio problems always help in remembering important ideas, so let us consider now a few of the ways in which this law is used by radio men. These examples will demonstrate even more clearly that the law is "just what you would naturally expect."

**Filament Circuit.** In the simple filament circuit shown in Fig. 6, the storage battery sends current through the tube filament and series resistor *R*. If you measured each voltage in the circuit with voltmeters connected

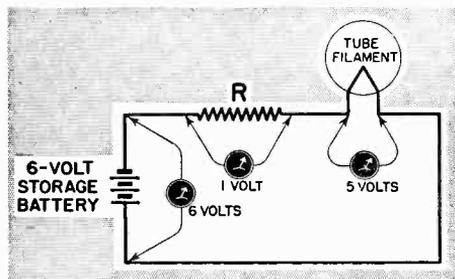


FIG. 6. Example of a circuit in which the source has too high a voltage for the load. Series resistor *R* cuts the load voltage down to the required value of 5 volts.

as indicated in the diagram, you would find that adding the filament voltage drop of 5 volts to the resistor voltage drop of 1 volt gives the source voltage value of 6 volts, just as predicted by Kirchhoff's Voltage Law.

Nearly every universal a.c.-d.c. radio receiver has a series resistor in its filament circuit, to use up that portion of the power line voltage which is not needed by the filaments in series. Later in this lesson, you will get acquainted with the line cord resistors and resistance tubes which are used for this purpose in a.c.-d.c. receivers.

**Plate Circuit.** Kirchhoff's Voltage Law holds true for any complete circuit no matter how many other cir-

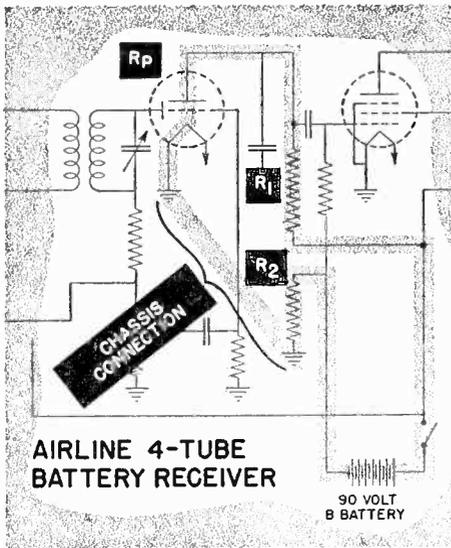


FIG. 7. This portion of an actual receiver circuit diagram is given here only to illustrate how voltages can be checked in one complete circuit regardless of nearby circuits. Note: No dots at cross-overs mean no connections.

cuits or radio parts are connected to various points in the circuit. For example, if a radio man were to pick up the service manual of an Airline Model 93BR-462A portable receiver and trace along the plate supply circuit of one tube in the manner shown by the shading in Fig. 7, he would ignore all the other circuits, and apply the voltage law just to that one circuit.

Thus, if he took a d.c. voltmeter and measured in turn the voltage across resistor  $R_1$ , resistor  $R_2$  and plate-cathode resistance  $R_p$  while the receiver was in operation, then added these three voltage drops together, the result would be equal to 90 volts, the voltage of the B battery which is the source in this circuit.

The plate-cathode path through the tube in Fig. 7 is marked  $R_p$  and considered as a resistance because it acts exactly like a resistor in this circuit. In other words, it obeys Ohm's Law. To find the value of this resistance,

you would divide the plate-cathode voltage by the plate current ( $R = E \div I$ ). You'll learn more about this when you study the lesson on tubes.

Note that numbers and letters are used after the letter  $R$  to distinguish between the three resistors in this plate circuit. This is always done whenever more than one resistor is shown on a diagram. You can pronounce these notations as  $R$  sub one and  $R$  sub pee, or simply as  $R$  one and  $R_p$ .

**Variable Series Resistors.** In some circuits, we use a *variable resistor* or *rheostat* in series with the load, and adjust its resistance value until the load is getting its correct voltage and current.

The construction of one type of variable resistor is shown in Fig. 8. The position of the contact arm can be changed by rotating the shaft of the unit. Since the movable arm is connected to one terminal, the position of the arm determines how much resistance there is between the terminals.

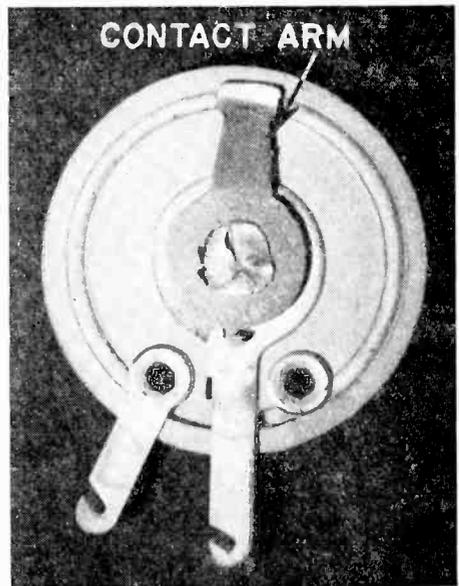


FIG. 8. Adjustable resistor or rheostat.

# Resistors in Parallel

When two resistors are connected in parallel to a voltage source, as they are in Fig. 9A, both resistors are acted on by the same voltage. This means that each of the parallel resistors gets the same current as if it were connected all by itself to the battery (Figs. 9B and 9C).

The current through each parallel resistor will be determined by Ohm's Law:  $I = E \div R$ . Thus, the current  $I_1$  through resistor  $R_1$  in Fig. 10 will depend upon the source voltage and the ohmic value of  $R_1$ . Likewise, the current  $I_2$  through  $R_2$  will depend upon the source voltage and the ohmic value of  $R_2$ .

Since the source in Fig. 10 must

supply current to two resistors in parallel, it is logical to expect that the source current  $I$  will be the *sum* of currents  $I_1$  and  $I_2$  flowing through the parallel resistors. This is quite correct, and we express it as a fundamental radio rule known as Kirchhoff's Current Law. Here it is:

## KIRCHHOFF'S CURRENT LAW

The currents flowing toward a terminal in a radio circuit must equal the currents flowing away from that terminal.

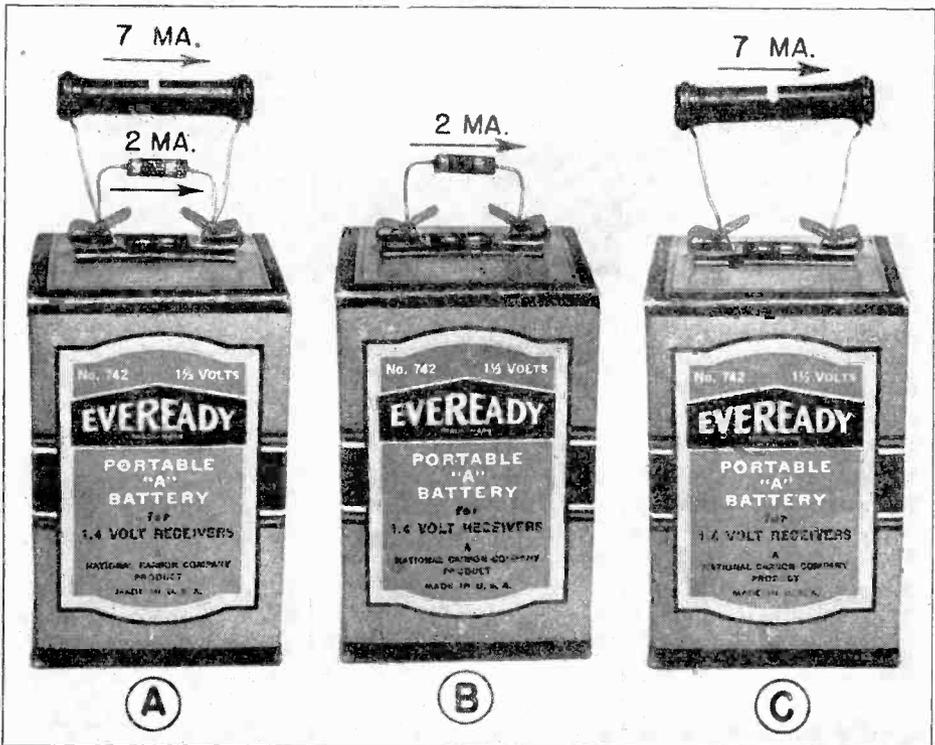


FIG. 9. These photos illustrate the basic fact that when resistors are connected in parallel to a voltage source, each draws the same current it would get if connected by itself to the source.

In the circuit of Fig. 10, this means that current  $I$  flowing toward terminal  $a$  must be equal to the sum of currents  $I_1$  and  $I_2$  flowing away from terminal  $a$ . Likewise,  $I_1 + I_2$  flowing toward  $b$  must equal  $I$  flowing away from  $b$ .

**Combined Resistance.** Since the sum of two values is larger than either one alone, total current  $I$  in the circuit of Fig. 10 is larger than either of the resistor currents. There is only one thing which can make it possible for the source to deliver this larger current—the combined resistance of  $R_1$  and  $R_2$  must be lower than that of either resistor alone. (Remember that lowering the total resistance in a circuit is one way to make the current

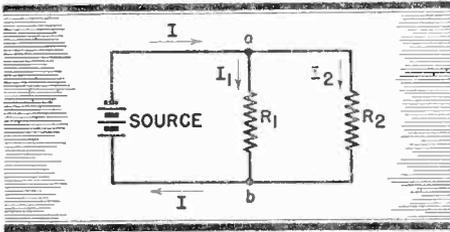


FIG. 10. Two resistors in parallel.

increase.) Our conclusion is, then, that the combined resistance of two resistors in parallel is *less* than that of either resistor alone.

**Different Resistors in Parallel.** When two parallel resistors have different values, the combined resistance will always be *less than that of the smaller resistor*. This is true because the smaller resistor gets the most current. Any resistor placed in parallel with it will make the total circuit current go up, and this means that *the combined resistance of two parallel resistors is less than that of the smaller resistor*. See Fig. 11 for two illustrations of this fact.

**Identical Resistors in Parallel.** When two parallel resistors are alike in ohmic value, the same current must

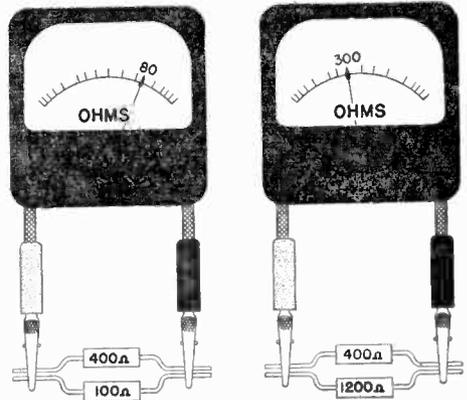


FIG. 11. When two different resistors are in parallel, the combined resistance is **LESS** than that of the smaller resistor. Note: Zero is at the extreme right on the scale.

flow through each. The total current is then twice the current through one resistor. For any given voltage value, the current can be doubled only by cutting the resistance value in half. The combined or equivalent resistance of two equal resistors in parallel is *thus exactly half* that of one resistor.

Likewise, the combined resistance of three equal resistors in parallel is exactly one-third that of one resistor. Actual hook-ups of two and three equal resistors in parallel are shown in Fig. 12.

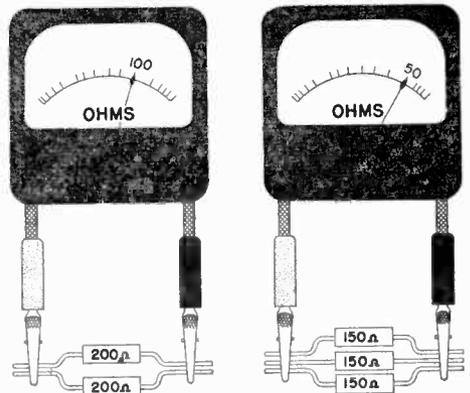


FIG. 12. When two equal resistors are in parallel, the combined resistance is exactly half that of one resistor. When three equal resistors are in parallel, the combined resistance is one-third that of one resistor.

The general rule is: To find the combined resistance of *equal* resistors in parallel, divide the resistance of one resistor by the number of resistors. *Example:* Four 40,000-ohm resistors in parallel have a combined resistance of  $40,000 \div 4$ , which is 10,000 ohms.

**Parallel Resistance Formula.** The foregoing general information is usually sufficient for a radio man when making resistance measurements in a receiver. There is, however, a fairly simple way of figuring out exactly the combined resistance of two resistors in parallel. Just multiply the two resistance values together, then divide by the sum of the resistance values, and you have the combined resistance. Of course, all values must be in ohms.

As a formula, this method of figuring parallel resistance values appears even simpler. Let  $R_1$  and  $R_2$  represent the values in ohms of the two resistors. The combined resistance  $R$  of the two resistors in parallel then is:

$$R = \frac{R_1 \times R_2}{R_1 + R_2}$$

In this kind of formula, the horizontal line means that everything above the line is to be divided by everything below the line, after the indicated multiplication and addition have been carried out. A few examples will illustrate this.

*Example 1.* Find the combined resistance of two 10-ohm resistors in parallel.

**SOLUTION:** Write the parallel resistance formula, then repeat it with the two known values in place of  $R_1$  and  $R_2$ , as follows:

$$R = \frac{R_1 \times R_2}{R_1 + R_2} \quad R = \frac{10 \times 10}{10 + 10}$$

Now multiply the two top numbers, add together the two bottom numbers, then divide the results as follows, to get 5 ohms as the combined

resistance (exactly half the value of one resistor):

$$R = \frac{100}{20} \quad R = 5 \text{ ohms}$$

*Example 2.* Find the combined resistance of 3 ohms and 6 ohms in parallel.

$$R = \frac{3 \times 6}{3 + 6} \quad R = \frac{18}{9} \quad R = 2$$

The combined resistance is thus 2 ohms. As we expected, this is less than the value of the smaller resistor.

**Combining Large and Small Resistors.** When two unequal resistors are in parallel and one is more than about 25 times larger than the other, the combined resistance will be very close to the smaller resistance value. In practical work, a radio man wouldn't even bother to figure out the value; he'd say the combined resistance was about the same as the value of the smaller resistor.

*Example 1.* Find the combined resistance of 1 ohm and 25 ohms.

$$R = \frac{25 \times 1}{25 + 1} \quad R = \frac{25}{26}$$

$R = .96$  ohm, which for practical purposes is 1 ohm.

*Example 2.* Find the combined resistance of 300 ohms and 10,000 ohms.

$$R = \frac{300 \times 10,000}{10,300} \quad R = \frac{3,000,000}{10,300}$$

$R = 291.2$  ohms, which for practical purposes is 300 ohms.

**Three or More In Parallel.** When more than two different resistors are in parallel, disregard all resistors which are more than 25 times larger than the smallest. If three resistors are left, combine two of them by means of the formula, then combine the result with the third resistor. By taking resistance values two at a time in this way, you can figure out the combined resistance of any number of resistors in parallel. The **combined**

resistance will always be less than that of the smallest resistor in the group.

*Example: Find the combined resistance of 2 ohms, 5 ohms, 10 ohms, 250 ohms and 1 megohm in parallel.*

First, we cross out 250 ohms and 1 megohm, since they are more than 25 times larger than 2 ohms. Now we combine 2 ohms with 5 ohms by means of our formula, and get 1.43 ohms. Next, we combine 1.43 ohms with 10 ohms, and get 1.26 ohms as the combined resistance of these five resistors in parallel. Note that this is less than the value of the smallest resistor in the group.

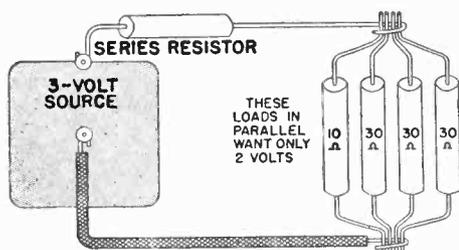


FIG. 13. A series resistor is often used to reduce the voltage for loads in parallel.

**Parallel Loads With Series Resistor.** In radio, we quite often have a number of loads in parallel, all requiring the same voltage, and a source which has too high a voltage for the loads. A series resistor will reduce the load voltage here, the same as it did for a single load resistor. All you have to do is figure out the combined resistance of all the loads in parallel, then use this value as if it were for a single load resistor.

An arrangement of parallel loads with a series resistor is shown in Fig. 13. The parallel loads require 2 volts, but the source furnishes 3 volts. As a review of the formulas you've already learned, let's figure out what the series resistor value should be.

First, we find the combined resistance of the load. The three 30-ohm resistors have a combined resistance of 10 ohms, and combining this with the 10-ohm resistor gives 5 ohms as the combined resistance of the group. (We chose easy values purposely here, but you could do this for any values by using the parallel resistance formula.)

Next, we find the current which this combined load resistance will draw through the series resistor. We have a 5-ohm load and it requires 2 volts;  $I = E \div R$ , so  $I$  is  $2 \div 5$ , which is .4 ampere.

We want to find  $R$  for the series resistor, so we bring out another Ohm's Law formula:  $R = E \div I$ . We've already found that  $I$  is .4 ampere. Kirchhoff's Voltage Law tells us that  $E$  (the voltage drop across the series resistor) plus 2 volts (the load voltage) must be equal to 3 volts, so  $E$  must be 1 volt ( $3 - 2 = 1$ ). Substituting these values, we find that  $R$  is  $1 \div .4$ , which is 2.5 ohms, the value of the series resistor.

*Example.* The filament circuit arrangement in Fig. 14 is an excellent example of how a series resistor reduces the voltage for parallel loads in a receiver. The filaments of the five tubes are connected in parallel between terminals  $a$  and  $b$ , with the connections between ground symbols be-

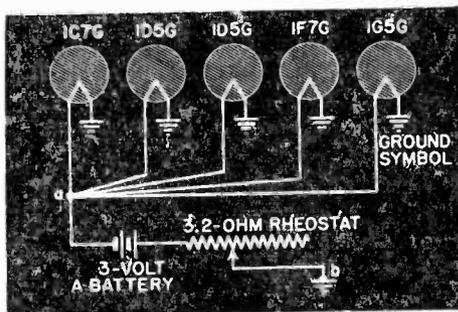


FIG. 14. Filament circuit arrangement of a 5-tube battery-operated radio set.

ing completed by the metal chassis of the receiver.

Each tube filament requires 2 volts, so the voltage between *a* and *b* must be 2 volts at all times. The battery delivers 3 volts when new, but this voltage drops gradually as the battery runs down. This is why a rheostat (variable resistor) is used as a series resistor for reducing the load voltage to 2 volts.

During normal use of this receiver, the owner adjusts the rheostat about once a week by means of a knob at the back of the receiver, in the position shown in Fig. 15. He *reduces* the ohmic value of the rheostat, so

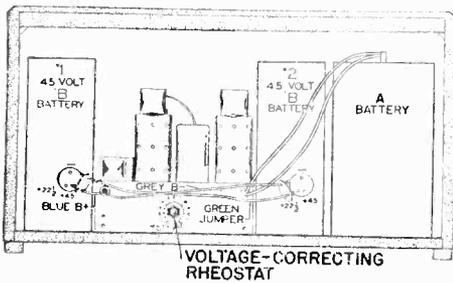


FIG. 15. Rear view of Belmont Model 524 radio receiver, showing the location of the control knob for the series rheostat which adjusts the filament voltage. The set owner is instructed to turn the rheostat one marked division further each week, which gives approximate correction of filament voltage during normal use.

that the voltage drop across it becomes less and less as the battery runs down. When the battery is down to 2 volts, the rheostat is down to zero resistance, and further drops in voltage can no longer be corrected. The receiver eventually stops working, and a new 'A' battery must then be installed.

**Problem: Load Voltage Varies Too Much.** When the load is a resistor, a series resistor does a perfect job of reducing the load voltage to the correct value. When the load is a vacuum tube which acts like a re-

sistance, however, we find that the load voltage just won't stay where it should.

A typical situation of this kind is presented in Fig. 16. The vacuum tube load in this particular example requires 40 volts at all times, and the series resistor is chosen to provide this

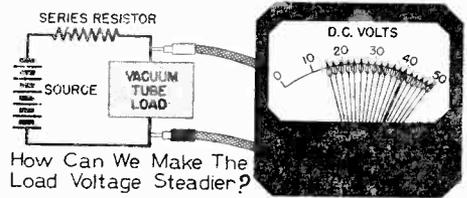
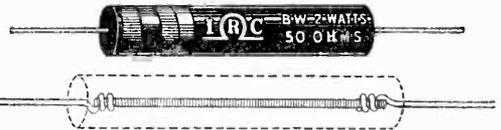


FIG. 16. Here's an interesting practical radio problem: The load is a vacuum tube whose resistance varies so much that a voltmeter across the load may read any value between 15 volts and 49 volts when the voltage is supposed to be 40 volts at all times. This lesson shows you how just one extra resistor of the proper value, properly connected into the circuit, will "save the day."

voltage when the tube resistance is at its normal value of 4000 ohms. During normal operation, the tube resistance can drop all the way down to 1000 ohms or go up to 6000 ohms, and this explains why the d.c. voltmeter in Fig. 16 is showing dozens of different readings ranging from 15 volts to 49 volts.



This low-wattage, low-value wire-wound resistor appears exactly like a carbon or metalized resistor. The resistance wire is wound on a core of insulating fiber, wire terminal leads are crimped over each end, and an insulating material similar to bakelite is molded over the entire assembly.

Each change in load resistance changes the total circuit resistance and the circuit current flowing through the series resistor, thus changing the amount of voltage which the series

resistor leaves for the load. What's to be done about it?

Clearly, our problem is to make the load voltage independent of the load resistance. We can do this by placing across the load a resistor having considerably lower resistance than the load, as shown in Fig. 17. (This resistor is usually called a *shunt resistor*, because *shunt* and *parallel* have the same meaning in radio.)

After the shunt resistor is connected, we must change the series resistor value so the load still gets 40 volts initially. Now the voltmeter just barely flickers around 40 volts when the tube resistance varies as it did before. For all practical radio pur-

current and all the circuit voltages likewise stay about the same. Thus, the shunt resistor does a fine job of steadying the load voltage.

A shunt resistor is sometimes called a *bleeder* resistor, because it draws or "bleeds" from the source a certain amount of current which flows through the series resistor but never gets a chance to do useful work in the load.

**Voltage Regulation.** When the load voltage varies excessively with load resistance, radio men say that the circuit has *POOR voltage regulation*. When the load voltage stays essentially the same despite variations in load resistance, the circuit has

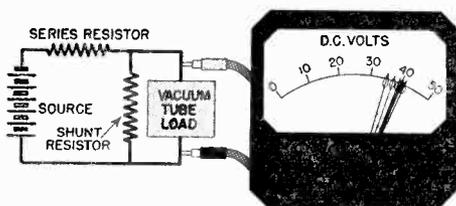


FIG. 17. With a suitable shunt resistor value in our circuit, the load voltage can vary only over the limited range of 35 volts to 41 volts, which is hardly enough variation to worry a practical radio man.

poses, we have a perfect solution to our problem. Let's see now why this single extra resistor has such an effect on the behavior of the circuit.

It is the combined resistance of shunt and load which determines how much voltage drop there is across the load. If we use a shunt resistor having much lower resistance than the load, the combined resistance will be just a little smaller than the shunt resistance. Furthermore, variations in load resistance will have little effect on the combined resistance because even the lowest load resistance value is still many times larger than the shunt value.

With the combined resistance staying essentially constant, the circuit

*GOOD voltage regulation.* A shunt or bleeder resistor across a load thus *improves* the voltage regulation of the circuit.

### Review Example

The circuit in Fig. 18, representing the plate supply circuits of three radio tubes in a simple public address amplifier, is an excellent illustration of how a serviceman can use Kirchhoff's Laws to speed up trouble-shooting in radio circuits.

All trouble-shooting is based upon a knowledge of *what should exist* when everything is working properly. Thus, a Radiotrician working on this circuit knows from Kirchhoff's Voltage Law that the d.c. voltage drops in

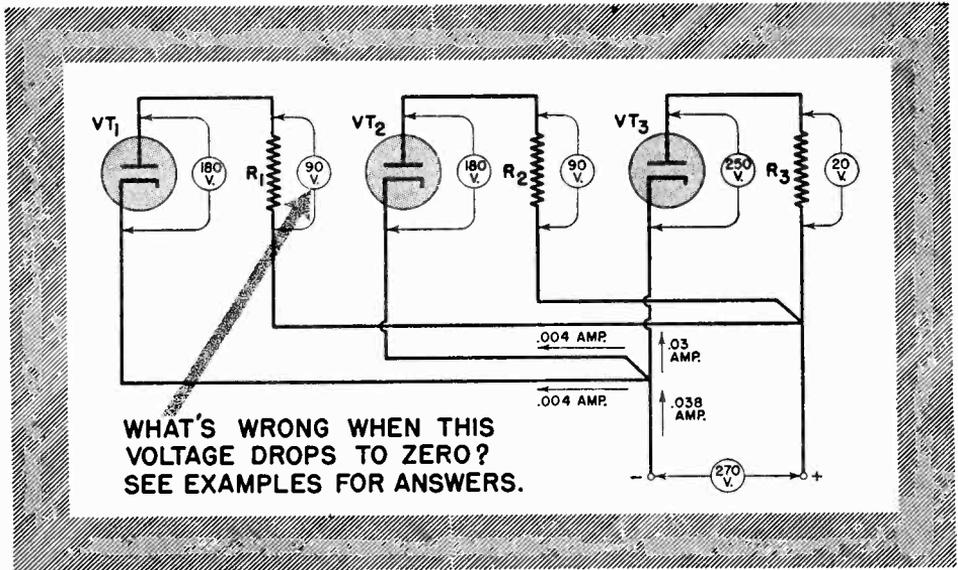


FIG. 18. Simplified plate supply circuits of a public address amplifier, showing voltage and current values a Radiotrician would measure if these circuits were free of trouble.

each complete plate circuit should add up to the source voltage of 270 volts. Voltage values which he might measure during normal operation are indicated in the diagram in Fig. 18, so let's check them.

In tube circuit  $VT_1$ , we add the tube drop of 180 volts to the 90 volts across  $R_1$ , and get just what the law says we should—270 volts. Tube circuit  $VT_2$  gives exactly the same result. In tube circuit  $VT_3$ , we add the tube drop of 250 volts to the 20 volts across  $R_3$ , and again get exactly 270 volts.

Normal circuit current measurements are also indicated in Fig. 18, and can be checked quickly against Kirchhoff's Current Law. Thus, adding .004 amp., .004 amp., and .03 amp. gives .038 amp., which is the source current.

Here are two examples of how a Radiotrician would use his knowledge of radio laws to help locate trouble in this circuit.

*Example 1.* Instead of the voltage and current values shown in Fig. 18,

suppose there is no voltage across resistor  $R_1$ , and tube  $VT_1$  has the full source voltage of 270 volts. In addition, the plate current for tube  $VT_1$  is much higher than the expected value of .004 ampere. What is wrong?

No voltage across  $R_1$  can mean that  $R_1$  is shorted and current is flowing around it through the shorting path, or can mean that no current is flowing through the entire circuit which includes  $R_1$ . We know that plate current is flowing through the circuit, so  $R_1$  must be shorted. Full source voltage across tube  $VT_1$  confirms the fact that there is no series resistance in the circuit, so the Radiotrician investigates  $R_1$  to see if it is defective or if its leads are touching each other.

*Example 2.* Measurements indicate the same voltage situation as for Example 1, but this time there is no plate current.

Absence of current means that either  $R_1$  or the tube is open. (When a tube does not pass current, it acts just like an open resistance.) Full

voltage across the tube proves that continuous paths exist from the tube terminals to the voltage source, so  $R$  cannot be open. This means that the tube is guilty this time. Whenever a break occurs in a complete circuit such as at the tube here, full source voltage appears *across the break*, just as if your meter were connected directly to the source terminals.

### Controlling Load Voltage with Potentiometers

In radio, it is often necessary to have a circuit arrangement which makes it possible to change the voltage of a load by turning a knob. You have already seen one way of doing this, by means of a rheostat in series with the load. A rheostat can reduce the load voltage only a certain amount, however—never all the way down to zero.

When complete control of load voltage from the maximum value to zero is required, we use a special type of variable resistor known as a potentiometer (pronounced *poe-TEN-shi-om-eh-ter*). The basic circuit is shown in Fig. 19.

As you can see, a potentiometer is simply a rheostat having three terminals instead of two. The entire resistance element of the potentiometer is connected across the source. The load is connected between the movable contact ( $P$ ) and one of the end terminals of the resistance element, and hence the load can be connected across any desired portion of the resistance element by changing the position of the movable contact.

When  $P$  is set at  $X$ , the load gets the full source voltage (lower left in Fig. 19).

When  $P$  is moved downwards, such as to the position shown at the top in Fig. 19, the upper potentiometer

section ( $R_1$ ) is a series resistance, and the lower potentiometer section ( $R_2$ ) is a shunt resistance connected directly across the load. The voltage drop across  $R_1$  now cuts down the amount of voltage available to the load. Note that both the load current and the current drawn by  $R_2$  flow through series resistance section  $R_1$ , and produce this voltage drop.

Moving  $P$  farther downward reduces the load voltage still more. When  $P$  is moved all the way down to  $Y$ , as at the lower right in Fig. 19, the load voltage is zero.

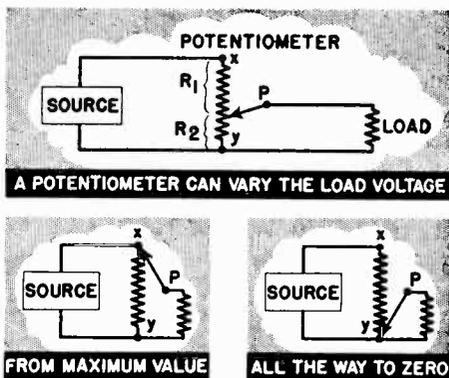


FIG. 19. Basic circuit using a potentiometer.

A potentiometer is often thought of as a voltage divider, because it divides the source voltage in the desired proportion between two parts of a circuit (in this case, it divides the voltage between  $R_1$  and the load- $R_2$  parallel combination).

When the total resistance of the potentiometer is much lower than the load resistance, the current drawn from the source depends chiefly on the resistance value of the potentiometer. The source current then stays essentially the same regardless of the position of the movable contact; this is a highly desirable condition in some radio circuits.

# Wattage Ratings of Resistors

If you glance through any catalog or listing of resistors, you will see that in addition to the ohmic value, each resistor has a *wattage rating*. This rating tells a radio man whether or not a resistor can "take it" in a particular circuit without getting too hot or burning out.

The rate at which a resistor produces heat is its *power in watts*. This value must for safety be lower than the wattage rating, because the wattage rating represents the maximum rate at which the resistor can safely produce heat. Since practical radio men work with resistors just about every day, the subjects of power, wattage ratings and watts are an important part of our study of resistors.

**What is POWER?** Let us suppose that we have two quart pans full of water, which we want to boil. If we set one pan on a small electric hot plate and the other on the large super-speed burner of an electric range, it may take the small burner three times as long to boil the water. Both burners accomplish the same result, however, for both have to supply the same total amount of heat to the quart of water.

The large burner accomplishes its work at three times the *rate* of the small burner. *The rate of doing work*—in this case, of producing heat—is called *power*.

This definition of power holds true for radio, too, because a resistor produces heat at a definite rate when current flows through it. Before we consider power problems in resistors, however, let us first get acquainted with the practical *units* used by radio men for specifying amounts of power.

**Units of Power.** In radio and electricity, power is measured in *watts* or in *kilowatts*. One kilowatt is equal to

1000 watts, hence one watt is one-thousandth of a kilowatt. Small radio sets draw about 40 watts, which is the same as the power drawn by a 40-watt soldering iron or a 40-watt electric lamp bulb. Large sets draw as much as 150 watts.

In radio, we occasionally encounter a smaller unit of power called the *milliwatt*. This is equal to one-thousandth of a watt, and is used chiefly in designating the strength of audio signals.

**Figuring Power.** To find the amount of power which a voltage



This wire-wound resistor looks much like a carbon or metallized resistor. It is made with various ohmic values, ranging from 5 ohms to 70,000 ohms, and in two different wattage ratings—5 watts or 10 watts. A red dot on the resistor changes to brown when the resistor is overloaded (when the wattage rating is exceeded). The dot changes back to red when the overload is corrected.

source is delivering to a circuit, all you need to do is *multiply the amount of voltage by the amount of current*. The voltage should be in volts and the current in amperes; the result will then be the electrical power in watts which is being delivered to the circuit by the voltage source. Thus, if a 6-volt battery is furnishing 2 amperes, the power being delivered is 12 watts ( $6 \times 2 = 12$ ).

To find the amount of power which is producing heat in a resistance, simply *multiply the current* flowing through the resistance *by the voltage drop* across the resistance. Voltage must be in *volts* and current in *amperes*. The result will then be the power in *watts* which is producing heat. Thus, if a current of 4 amperes

produces a voltage drop of 5 volts across a resistor, the power in the resistor is 20 watts ( $5 \times 4 = 20$ ).

**Power Formula.** In radio, we let the capital letter  $P$  stand for power. The simple formula which expresses what has just been said about figuring power is:

$$P = E \times I$$

Remember that  $P$  stands for the power *in watts*,  $E$  stands for the voltage *in volts*, and  $I$  stands for the current *in amperes*.

This power formula applies to all direct current circuits, and to all a.c. circuits containing only resistance.

Radio men use the power formula chiefly to figure how much power a resistor is handling. A given resistor can handle only a certain amount of power without getting too hot and destroying itself, because all of the power supplied to a resistor is used to produce heat. This is the reason why resistors have *wattage ratings* in addition to their resistance values.

**Wattage Ratings of Resistors.** The wattage rating of a resistor is the *maximum* amount of power in watts which the resistor can handle without becoming excessively hot. Naturally, a resistor can safely handle any power value which is *less* than its wattage rating.

The wattage rating depends essentially on the physical size of a resistor (length, thickness and surface area), and is entirely independent of the ohmic value. In plain language, the wattage rating depends upon how fast the resistor can get rid of the heat produced by its resistance element.

The larger the physical size of a resistor, the more heat it can withstand and the higher will be its wattage rating. A 10-watt resistor will thus be larger in dimensions than a 1-watt resistor.

As a matter of fact, in a radio store

you could get a resistor with a suitable wattage rating by holding up the old resistor and saying, "I want a 500-ohm resistor which is this big or bigger." When ordering a resistor from a mail order radio supply firm, however, you would have to specify the wattage rating as well as the ohmic value.

Most of the resistors which you will encounter in radio circuits have low wattage ratings, such as  $\frac{1}{4}$ ,  $\frac{1}{2}$ , 1 or 2 watts. A  $\frac{1}{4}$ -watt resistor is less than

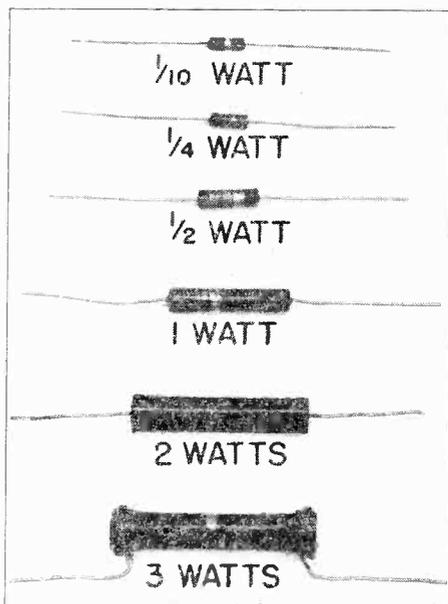


FIG. 20. Examples of carbon resistors having different wattage ratings.

an inch long, and is scarcely any thicker than a piece of insulated hook-up wire. Figure 20 shows how the physical dimensions of resistors vary with the wattage rating.

**Margin of Safety.** It is good practice to provide a margin of safety by using a resistor having a wattage rating which is about twice the actual power it must handle. This prevents resistor failure even though the power should for some reason get considerably higher than the normal value.

Another reason for providing a margin of safety is that wattage ratings are for resistors mounted in free air. When resistors are mounted in enclosed places, such as under a shallow chassis where there is little circulation of air, the heat-dissipating ability is greatly reduced.

*Examples.* If a resistor is to handle 4 watts, choose a 10-watt resistor for safety. If it is to handle .4 or .6 watt, use a 1-watt resistor. If it is to handle 1 watt, use a 2-watt resistor. As a general rule, try to use a wattage rating which is about twice the actual power being handled.

**Rheostat Wattage Ratings.** Wattage ratings of variable resistance units, such as rheostats and potentiometers, apply to the *entire resistance element*. When the position of the movable contact is such that only a part of the resistance element is in the circuit, the wattage rating is *proportionately reduced*.

Thus, if the movable contact of a rheostat is at the mid position, the unit can safely handle only half of the wattage rating. If the movable contact is set so only about one-eighth of the total resistance is in the circuit, the unit can safely handle only one-eighth of the wattage rating.

Potentiometers and rheostats invariably burn out near a terminal, because only a small portion of the resistance element is then in the circuit, and the useful wattage rating is correspondingly small.

A burned-out potentiometer is shown in Fig. 21. When the contact arm was set near the right-hand terminal, the wire which forms the resistance element became so hot that it melted and the surrounding fiber of the unit became badly charred.

If you encounter a receiver in which a certain resistor, rheostat or potentiometer becomes too hot and perhaps

even burns out frequently, yet no other parts in the receiver are defective and causing excessive current to flow, you should replace the overheated unit with another having the same ohmic value but a *higher wattage* rating.

**Wattage Ratings of Parallel Resistors.** When two or more resistors which have *the same ohmic values* and the same wattage ratings are connected in parallel, the wattage rating of the combination will be the *sum* of the ratings of the individual resistors.

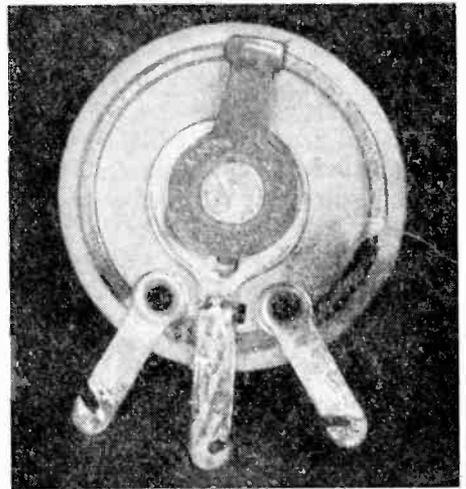


FIG. 21. Burned-out wire-wound potentiometer.

Thus, if two 100-ohm, 1-watt resistors are connected in parallel, the combined wattage rating is  $1 + 1$ , or 2 watts. (The combined resistance is determined in the usual way by the parallel resistance formula, and is 50 ohms in this case.) If four 40,000-ohm, 3-watt resistors are connected in parallel, the combined wattage rating is  $3 + 3 + 3 + 3$ , which is 12 watts.

When resistors having different wattage ratings and different ohmic values are connected in parallel or in series, the power must be figured for each resistor separately, to make sure that no resistor will be overloaded.

**Other Power Formulas.** Since the *resistance* value affects the amount of current which flows through a resistor, resistance also affects the amount of power being handled by a resistor. If we know the resistance value and either the voltage or current value, we can figure the power. Here are the formulas:

When the current and resistance are known, use this formula:  $P = I \times I \times R$ . This is usually written as  $P = I^2 \times R$  or simply as  $P = I^2R$  (*pronounced P equals I squared R*). The small numeral 2 at the upper right of a letter means that the value is to be multiplied by itself. Thus, if the current through a resistor is .5 ampere

and the resistance is 40 ohms, the power will be  $.5 \times .5 \times 40$ , which is  $.25 \times 40$ , or 10 watts.

When the voltage and resistance are known, use this formula:  $P = E \times E \div R$ , which is usually written as  $P = E^2/R$  (*pronounced P equals E squared over R*). If the voltage drop across a resistor is 20 volts and the resistance is 80 ohms, the power will be  $20 \times 20 \div 80$ , which is  $400 \div 80$ , or 5 watts.

*Important:* You are not expected to memorize any of the formulas in this lesson. You can always refer to this lesson when you have need for a particular formula. Eventually, after using a formula a few times, you'll find that you know it without ever having tried to remember it.

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## How Resistors Are Made

**Many Kinds of Construction.** Resistors used in radio equipment not only have different ohmic values and different wattage ratings, but they also differ greatly in actual construction. Thus, fixed resistors can have wire-wound, carbon or metallized construction.

Some types of construction give greater accuracy than others; some are cheaper than others when mass production methods are used; some provide the high wattage ratings required for resistors which must withstand a lot of heat; some construction methods give extremely low or extremely high ohmic values.

We will first consider the materials from which resistors are made, then study the three types of construction methods used for resistors, and finally get acquainted with typical resistors used in radio.

### What Resistors Are Made Of

If we took a piece of ordinary No. 18 lamp cord or bell wire long enough

to reach all the way across the country from New York to San Francisco, and measured the total *resistance* of this wire, the result would be less than 100,000 ohms! To get 1 megohm of resistance, a value commonly used in radio circuits, we would have to run this copper wire all the way around the world and still add on about 5000 miles of wire.

You say, "Use smaller wire, because it has much higher resistance per foot of length. Some copper wire is made as fine as human hair."

All right, suppose we did use copper wire the size of a human hair. It would still take 190 miles of the wire to get a megohm of resistance, and that's a lot of wire to put into a radio part.

All this leads us to the conclusion that the resistance of copper is too low for use in resistors. This low-resistance characteristic of copper makes it ideal, however, for the hook-up wire used in connecting radio parts together, and for constructing radio parts

which *should* have low resistance, like coils and transformers.

What we need for practical radio *resistors* are materials having much higher resistance than copper. Two such materials are *nichrome* (used in wire-wound resistor construction) and *carbon* (used in carbon resistor construction). In addition, there is a *metallized* resistor construction which is often used in place of carbon but has slightly different characteristics.

**Resistivity.** The easiest way to compare the resistances of different materials is by considering pieces of wire which are all the same length and thickness, each made from a different material. The resistance in ohms of a specified length of wire is technically known as the *resistivity* of the material.

If, for convenience, we choose a length and thickness which will make the copper sample have a resistance of exactly 1 ohm, then similar samples of other common resistance materials will have approximately the ohmic values given in Table I. These values vary considerably in some cases, depending upon the purity of the metal and the manner in which it is made into wire.

**Length.** Increasing the *length* of a piece of resistance material without changing its thickness is just like placing resistors in series to get more resistance. Increasing the length *INCREASES* the amount of resistance.

Technically speaking, the resistance of a conductor is *directly proportional* to its length. In other words, 2 feet of wire will have two times the resistance of a 1-foot length; 7 feet will have seven times the resistance of a 1-foot length; one inch of wire will have one-twelfth the resistance of a 1-foot length.

**Thickness.** Increasing the thickness (cross-sectional area) of a piece

of resistance material is just like placing resistors in parallel. As you will recall, resistors in parallel have less resistance than either resistor alone. Increasing the thickness thus *DECREASES* the resistance. Actually, doubling the cross-sectional area cuts the resistance exactly in half.

Reducing the thickness of a piece of resistance material *INCREASES* the amount of resistance, because this is

Silver .....	.9 ohm
Copper .....	1 ohm
Aluminum .....	1.7 ohms
Iron, soft .....	6 ohms
Manganin .....	25 ohms
Constantan .....	29 ohms
Nichrome .....	60 ohms
Carbon .....	2000 ohms

Table I. Resistances of wire samples which all have the same length and thickness but are made from different materials. The last five materials are used in resistors and are taken up later in this lesson. Note that carbon has 2000 times as much resistance as copper.

like slicing a resistor lengthwise and using only part of it. Cutting the cross-sectional area in half doubles the resistance.

## Resistor Construction

**Wire-wound Construction.** Resistors made from metal wire are known as *wire-wound resistors*. The metal wire used most often in resistor construction is an alloy called *nichrome*, made chiefly from nickel and chromium. As indicated in Table I, nichrome has about 60 times the resistance of copper. This means that the *resistivity* of nichrome is about 60 times that of copper.

The nichrome wire is wound on a flat strip of fiber, on a rod of porcelain, or on some other stiff insulating material. The turns may be spaced out so they do not touch each other, or the wire may have enamel insulation.

When higher-wattage wire-wound resistors are required, they are wound on large cement or baked clay forms. The wire used is made to have the largest possible cross-sectional area, so there is a large surface area to conduct the heat to the form and to the surrounding air.

If a resistor of low ohmic value is required, a few turns of fine wire wound on the form would give the desired resistance, but this short wire would not get rid of its heat quickly enough to give the desired wattage rating.

Thicker and longer wire having the same resistance value is better, because it will get rid of heat faster. Increasing length and thickness both increase the surface area which radiates heat, so the resistor can then radiate more heat safely. This makes possible a higher wattage rating.

As you know, heat is produced whenever current flows through a resistance. The larger the current, the more heat we get. Nichrome wire can get red hot without breaking or melting; in fact the material used for the heating elements of electric stoves, toasters, electric heaters and soldering irons is usually nichrome.

But nichrome is by no means the complete solution to our resistor-making problem. We would still need a little over three miles of the smallest size nichrome wire to get a million ohms of resistance, and the cost of that much wire would be too high for practical purposes.

Nichrome wire is ordinarily used only for fairly low resistance values (below about 50,000 ohms for both fixed resistors and rheostats or potentiometers). Even then, wire-wound resistors made from nichrome (or similar metal alloys sold under various other trade names) are used chiefly when the resistor must carry considerable current, when the ohmic value of

the resistor must be highly accurate, or when the ohmic value must stay the same despite changes in temperature.

**Carbon Construction.** This is the construction method which makes possible the tiny high-value fixed resistors, rheostats and potentiometers found in modern radio sets.

By itself, carbon has a resistance ranging from 500 to 2500 times the resistance of copper. By mixing carbon particles with certain cements or clay products, however, we can increase the resistance to more than a million times that of a corresponding amount of copper. Cements or binders serve the added purposes of giving needed "backbone" or strength to carbon, and keeping the carbon particles in position so the resistance stays fairly constant. Carbon is really a good all-around resistance material for general use.

By varying the proportion of carbon to binder material and forming the mixture into rod-shaped pieces, fixed carbon resistors having the same physical size (length and thickness) can be made in any desired value from about 50 ohms to about 15 megohms.

A carbon resistor scarcely *half an inch* long can have a resistance of a million ohms. Compare this with the *miles* of metal wire which would be needed to get such a high resistance.

The ends of each carbon resistance element are copper-plated so terminal leads can be soldered to them. The completed unit is then painted with various colors of enamel so as to specify its ohmic value according to the standard resistor color code (explained elsewhere in the Course).

Sometimes carbon resistors are placed in tubes of insulating material or coated with an insulating material. More often, the insulating material is molded over the carbon resistance element. Color code markings are then applied over the insulation. Insula-

tion prevents short-circuits (accidental short-cut paths for current) when resistors touch each other or other parts in a crowded chassis.

The chief reason for making carbon resistors in larger physical sizes is to secure higher wattage ratings. A larger resistor has more surface area to radiate the heat which is generated in the resistor, and can therefore handle more electrical power without becoming too hot. Three watts is about the highest rating in which carbon resistors are ordinarily made, however, for carbon-cement mixtures cannot withstand much heat.

Carbon resistors are the lowest-priced of all resistors, and for this reason are widely used in ordinary radio equipment.

Carbon rheostats and potentiometers are commercially available in values from about 500 ohms to about 10 megohms. Thus, values below 500 ohms are ordinarily obtainable only as wire-wound units, and high ohmic values are available only as carbon units.

**Metallized Construction.** In this process of making resistance elements for resistors, a liquid solution of certain metals is produced by a secret chemical process. The backbone of the resistor, usually a glass or porcelain rod or tube, is coated with this liquid and heated. This evaporates the liquid, leaving on the glass a hard, thin metallic coating which serves as the resistance element. By varying the strength of the metallic solution and by varying the amount of solution sprayed on the backbone, any desired resistance value between about 250 ohms and 20,000,000 ohms can be obtained.

**How Temperature Affects Resistance.** Practically all metals *increase* in resistance a small amount when heated. For example, a piece of *nichrome* wire having a resistance of 1

ohm when in ice water would have a resistance of about 1.04 ohms when placed in boiling water (this is a temperature rise of 180° Fahrenheit). A piece of *copper* wire having a resistance of 1 ohm at freezing temperature would increase much more, to about 1.39 ohms, when placed in boiling water.

Constantan and Manganin are special metal alloys which are little affected by temperature. Thus, a 1-ohm piece of Manganin would increase only to 1.001 ohms when transferred from freezing water to boiling water. These and other similar alloys are used chiefly for making precision resistors used in meters and in laboratory equipment requiring highly accurate and stable resistance values.

Carbon actually drops in resistance when temperature is increased, and this drop can be quite large when carbon resistors heat up during use.

When a material having a resistance of one ohm is heated or cooled one degree, the resulting *CHANGE IN OHMIC VALUE* is known as the *temperature coefficient of resistance*.

Metals are said to have a *positive* temperature coefficient of resistance, because they *increase* in ohmic value when temperature is increased. Carbon, on the other hand, has a *negative* temperature coefficient of resistance, because it *decreases* in ohmic value when heated.

Pilot lamps and tube filaments are perhaps the most striking examples of how the resistance of a metal increases with temperature. In these parts, the change in temperature from a cold filament to a red or white-hot filament may be several thousand degrees.

The hot resistance of a tungsten filament may be as much as ten or fifteen times its cold resistance. Thus, a particular pilot lamp might have a resistance of 4 ohms if measured with an ohmmeter when cold, and a re-

sistance of 40 ohms when hot. (The hot resistance cannot be measured with an ohmmeter, but can be figured out with Ohm's Law if the pilot lamp voltage and current are known or are measured.)

Tube filaments do not increase quite as much when hot, because these filaments do not get as hot as those in pilot lamps. Just remember the general fact—that a tube or lamp filament has a higher resistance when hot than when cold. This will explain seemingly erroneous results which might be obtained when using lamps as resistances in experimental circuits. It also explains why tube filaments often burn out right after a receiver is turned on. The cold tube filaments have a low resistance, hence an excessively large initial current flows when the set is first turned on.

Lamps with carbon filaments are now quite rare, but it is nevertheless worth pointing out that these lamps have a *much lower* resistance when hot than when cold.

Iron or nickel are resistance materials which increase in resistance even more rapidly than nichrome when heated. This characteristic is utilized to advantage in the special resistance tubes which serve as filament voltage-dropping resistors in some universal a.c.-d.c. receivers. These resistance tubes are so designed that when normal current flows through the resistance wire and produces normal heat, all tube filaments in the receiver get correct voltages.

If the line voltage goes above normal, however, the increased current through the resistance tube makes its resistance increase, so the resistance tube takes most of the increase in line voltage. As a result, the tubes still receive normal voltage, and are thus protected from burn-outs.

In the same way, when the line voltage drops lower than normal, the resistance of this resistance tube drops, the voltage drop across the tube becomes lower, and the tube filaments still get essentially the same voltage even though the line voltage has dropped.

**Tolerance of Resistors.** In resistors, just as in other manufactured parts, there is no such thing as a perfectly accurate unit. Accuracy costs money, because it means additional testing and more careful adjustment of resistor values. A 100-ohm, 1 watt resistor which is accurate to within 20% may cost only four cents, whereas the same resistor with an accuracy of 1% may cost fifty cents, and with .1% accuracy may cost \$2.

The per cent variation from the specified ohmic value is called the *tolerance* of a resistor. A tolerance of 20%, sometimes expressed as  $\pm 20\%$  (*plus or minus twenty per cent*), means that any value in the range from 20% under to 20% over the specified resistor value is tolerated (passed) during manufacture.

Thus, a 100-ohm resistor with  $\pm 20\%$  tolerance may actually be anywhere between 80 ohms and 120 ohms. ( $20\%$  is  $.2$ ;  $.2 \times 100 = 20$ ;  $100 - 20 = 80$ ;  $100 + 20 = 120$ ). A 100-ohm resistor with 5% tolerance may be anywhere between 95 and 105 ohms.

A 500,000-ohm resistor with 20% tolerance can be anywhere between 400,000 and 600,000 ohms.

A resistor tolerance or accuracy of 10% or even 20% is entirely satisfactory for general use in radio circuits, because resistance values are not critical in most circuits. Only in measuring instruments and in certain special radio circuits is it necessary to use more accurate resistors.

Normally, carbon resistors are made only to an accuracy of 20%. For special applications, however, carbon resistors are available with a tolerance of 10% or 5%, depending upon circuit requirements.

Metallized resistors have a normal accuracy of 10%, but here again, more accurate units are available at increased cost. Even 2% accuracy is obtainable, because these resistors are more stable (less affected by temperature changes) than carbon resistors.

Wire-wound resistors can be made as accurate as .1%, but 5% is the normal tolerance for general-purpose wire-wound resistors. Greater permanent accuracy and greater power-handling ability are the two main features of wire-wound resistors.

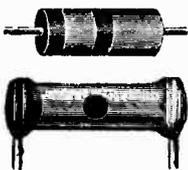
The tolerance encountered in actual resistors makes it unnecessary to figure resistance values with extreme accuracy. This is why practical radio men think nothing of using a 1000-ohm resistor when their figuring indicates that 900 ohms is needed.

### Types of Resistors Used in Radio Work

**FIXED RESISTOR.** A radio part having an essentially constant ohmic value, used to introduce a definite resistance value permanently into a circuit. Fixed resistors have two terminals, leads or lugs, one at each end of the resistance element.

Three types of fixed resistor construction are found in radio: 1. Carbon Resistors; 2. Metallized Resistors; 3. Wire-wound Resistors.

◆ **1. Carbon Resistor.** A fixed resistor having as its resistance element a mixture of carbon and a clay-like material which has been formed into a rod and baked until hard. Carbon resistors



are made in various wattage ratings from  $\frac{1}{10}$  watt up to 3 watts. They are the lowest-priced of all resistors. Standard tolerance is 20%, but resistors with 10%, 5% or 2% tolerance are available at higher cost. Carbon resistors always decrease in ohmic value when heated. Both insulated and non-insulated types are in common use. The wire leads come out either from the ends or sides.

◆ **2. Metallized Resistor.** A fixed resistor having wire leads attached to the ends of a metallized resistance element, with the entire unit encased in cement, bakelite or other insulating material. They have essentially the same appearance, wattage ratings and ohmic values as carbon resistors, and can ordinarily be used interchangeably in radio circuits.



◆ **3. Wire-Wound Resistor.** A fixed resistor made of nichrome wire wound on a strip or rod of insulating material. The two ends of the resistance wire are soldered, welded, riveted or clamped to terminal lugs or leads.

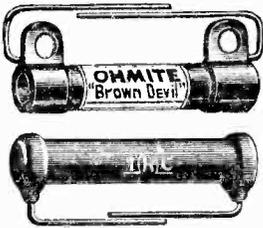
Flat wire-wound resistors with exposed wire usually have ohmic values



ranging from 1 ohm to 1000 ohms, and a wattage rating of only  $\frac{1}{2}$  watt. These resistors are therefore suitable only for circuits where both resistance and power-handling requirements are low.

Flat wire-wound resistors wrapped in heavy fiber paper, then encased in a piece of sheet steel, are sold under the trade name "Candohm." The metal shield protects the resistance wire from damage and helps to get rid of heat.

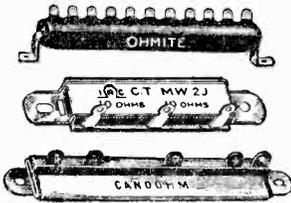




In wire-wound resistors which must handle a great deal of power, the resistance wire is wound on a porcelain

tube and is covered with a baked-on layer of vitreous (glass-like) porcelain enamel or a special heat-resisting cement.

**TAPPED RESISTOR.** A fixed wire-wound resistor having one or more extra permanent terminals or taps along the length of the resistance element, and used chiefly as a voltage divider in the power pack of a radio set. Some tapped resistors have the



SYMBOL FOR TAPPED RESISTOR

resistance wire completely exposed, but more often the wire is protected with a vitreous enamel coating, a combination bakelite and steel housing, or a fiber and steel wrapping. Carbon and metallized resistors are not made with taps, but two or more of these resistors can be used in series to get taps.

**ADJUSTABLE RESISTOR.** A fixed wire-wound resistor having one or more semi-movable extra terminals or taps which can be clamped in position at desired points along the length of the resistor. A semi-adjustable rheostat is obtained by using one sliding terminal and one end terminal. A potentiometer is obtained by using one

sliding terminal and both end terminals. A screwdriver or wrench is required to change the resistance value. Adjustable resistors are used chiefly as voltage dividers in power packs.



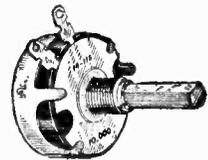
SYMBOL FOR ADJUSTABLE RESISTOR OR POTENTIOMETER

**RHEOSTAT.** A resistor which can be changed in ohmic value by rotating a knob. In wire-wound rheostats, bare nichrome resistance wire is wound around a strip of flexible insulating material. The strip is then bent into a ring and mounted so a contact arm can be rotated over the edges of the turns of wire. One terminal of the rheostat goes to one end of the resistance wire, and the other terminal goes to the movable contact.

When the contact arm is at the fixed terminal, the resistance between the two terminals is zero. When the contact arm is moved *away from* the fixed terminal, the resistance between the terminals increases. When the contact



WIRE-WOUND



CARBON



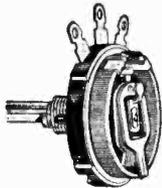
SYMBOLS FOR A RHEOSTAT

arm is farthest away from the fixed terminal, the resistance is a maximum.

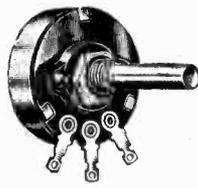
In one type of carbon rheostat, a shallow groove in a round disc of bakelite is filled with a mixture of carbon and cement. In another type, a washer-shaped disc of paper or the

inside of a paper ring is sprayed with a solution containing carbon. This circle of resistance material is broken at one point to form the resistance element over which the contact slides.

**POTENTIOMETER.** A rheostat having terminals at both ends of the resistance element, making three terminals in all. Construction is exactly the same as for rheostats; in fact, a potentiometer can be used as a rheostat by leaving one end terminal unconnected. Carbon units have a maxi-



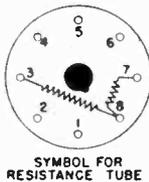
WIRE-WOUND



CARBON

mum wattage rating of one watt or less, hence are used chiefly in low-current signal circuits. Wire-wound units ordinarily have 5-watt ratings in radio work, but larger units are available.

**RESISTANCE TUBE.** A length of iron, nickel or nichrome wire mounted in either a glass or metal radio tube housing, and used in series with the filaments of tubes in universal a.c.-d.c. receivers for voltage-dropping purposes. A resistance tube is plugged into the standard tube socket provided for it on top of the receiver chassis. In this open-air position, the heat produced by the tube is dissipated much faster than by an equivalent under-chassis resistor.



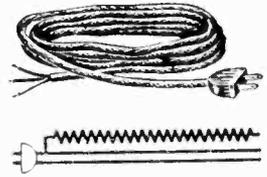
SYMBOL FOR RESISTANCE TUBE

Resistance tubes using iron or nickel wire are sometimes known as ballast tubes, because they offset variations in

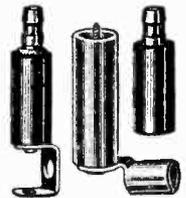
line voltage by changing in resistance when the line voltage changes.

Resistance tubes usually have one or more taps, so pilot lamps can be connected across part of the resistance. The schematic symbol for a resistance tube shows the internal connections between the resistance wire and the base prongs or socket terminals.

**LINE CORD RESISTOR.** A spirally wound resistance wire covered with asbestos, running alongside the two regular stranded copper wires of the power cord in a universal a.c.-d.c. receiver. The cord gets warm when in use, since it must get rid of the same amount of heat as a resistance tube.



**SUPPRESSORS.** Fixed carbon or wire-wound resistors used in series with spark plug and distributor cables of automobile ignition systems to suppress interference which might affect auto radio reception. They have special terminals to permit easy mounting on spark plugs and distributor.



### Looking Ahead

In the next lesson, you will learn about coils in much the same way that you have just learned about resistors. You will find that coils act like resistors in some ways, but coils act entirely different from resistors in alternating current circuits. The manner in which a coil can change the amount of its opposition instantly when the frequency of the current changes is one of the truly fascinating actions in radio.

# Lesson Questions

Be sure to number your Answer Sheet 5FR-4.

Place your Student Number on *every* Answer Sheet.

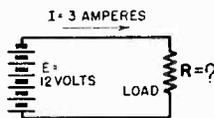
Send in your answers for this lesson immediately after you finish them, as instructed in the Study Schedule. Do this, and get the greatest possible benefit from our speedy personal grading service.

1. What does each of the following represent:

- (a) E
- (b) I
- (c) R
- (d)  $\omega$
- (e) MEG.

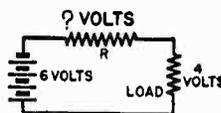
2. Give the formula you should use to find the amount of current flowing in a circuit, if you know the source voltage and the total resistance of the circuit.

3. In the circuit at the right, the load voltage E is 12 volts, and the load current I is 3 amperes. The formula to use is  $R = E \div I$ . Is the load resistance value: 4 ohms; 36 ohms; or  $\frac{1}{4}$  ohm?



4. How much voltage would be measured across a 5 ohm resistor if the current flowing through it is 4 amperes?  $E = IR$

5. In the circuit at the right, the source voltage is 6 volts, and the load voltage is 4 volts. How much is the voltage drop across series resistor R?



6. Currents of 3 amperes, 7 amperes, and 2 amperes are flowing toward a terminal. How many amperes of current are flowing away from that terminal?

7. If a 5 ohm, 10 ohm, and another 10 ohm resistor are connected in parallel will the total resistance be: 25 ohms; 20 ohms; or less than 5 ohms?

8. Write one of the formulas for finding the power dissipated in a resistor.

9. About how much higher than the actual power should the wattage rating of a resistor be to give a margin of safety?

10. When the filament of a vacuum tube is heated does the filament resistance: increase; decrease; or remain the same?

## A PLAN FOR YOUR FUTURE

In a radio interview a few minutes after a championship heavyweight boxing match, one of the fighters stated his plans for the future in just fifteen words:

*"I'm going to get myself in shape, fight my own fights, and listen to nobody!"*

You can use these dynamite-packed words as your plan for the future, too. Here's the way:

**"GET MYSELF IN SHAPE."** You're doing this right now, because the N.R.I. Course gets you in shape for a career in radio. But remember that it takes the *complete* N.R.I. Course, with all its associated practical work, to get you *completely in shape*.

**"FIGHT MY OWN FIGHTS."** In real life, the only person who can bring you success is **YOU** yourself. Expecting somebody else to do your work and fight for your success is just wishful thinking.

**"LISTEN TO NOBODY."** Even friends and relatives will at times ridicule your studies—they can't help it, because seeing you get ahead makes them feel uncomfortable about their own laziness. So, remember human nature, and don't give anyone a chance to discourage you.

J. E. SMITH.

**HOW TUNED CIRCUITS  
FUNCTION**

**COUPLING RADIO CIRCUITS**

9FR-3

**NATIONAL RADIO INSTITUTE**

ESTABLISHED 1914

**WASHINGTON, D. C.**



# Study Schedule No. 9

For each study step, read the assigned pages first at your usual speed. Reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind, then answer the Lesson Questions for that step. Study each other step in this same way.

1. Introducing Resonant Circuits . . . . . Pages 1-2

This short section introduces the L-C circuit, and tells how it is different from the filter circuits you have studied so far.

2. Series Resonant Circuits . . . . . Pages 3-9

By comparison with familiar circuits, you are shown how a resonant circuit of the series type functions. Here is a remarkable circuit, capable of stepping up voltage, due to the cancellation effects produced by phase. Study and reread this section. Be sure you understand how the series resonant circuit works before going on, as you will find this information valuable when you study r.f. circuits. Answer Lesson Questions 1, 2 and 3.

3. Parallel Resonant Circuits . . . . . Pages 10-14

The simple act of putting the coil and condenser in parallel instead of in series gives a completely different circuit action. Yet, both circuits will accept or reject signals if properly used. Answer Lesson Questions 4 and 5.

4. Practical Resonant Circuit Facts . . . . . Pages 14-19

The resistance within a resonant circuit has much to do with its ability to select, as you learn from the section on the Q factor. There are also sections giving practical ways of distinguishing series and parallel circuits, and on tuning these circuits. Answer Lesson Questions 6 and 7.

5. How Resonant Circuits are Used . . . . . Pages 19-23

More practical data—showing typical methods of putting resonance circuits to use selecting or rejecting signals. Answer Lesson Question 8.

6. Coupling Radio Circuits . . . . . Pages 24-28

There are four ways of connecting a voltage source to a resonant circuit. These methods of coupling and an introduction to inductive coupling are given here, again from the practical "how it works" angle. Answer Lesson Questions 9 and 10.

7. Mail Your Answers for this Lesson to N.R.I. for grading.

8. Start Studying the Next Lesson.

# HOW TUNED CIRCUITS FUNCTION

## COUPLING RADIO CIRCUITS

### Introducing Resonant Circuits

**S**EPARATING different frequencies which have become mixed is an important task in radio. An antenna picks up signals from many different stations and we want to pick out just one of these. Within the radio, we have cases where we prevent hum, correct the tone quality and do other desirable things by making the proper separation. As you learned in a previous lesson, coils and condensers can be used

for this purpose. If we want to pass on to the load only frequencies *below* a certain value, we can use a coil *between* the source and load, or we can use a condenser in parallel with the load. To make the cut-off more sharp, we may use both.

Similarly, if we wish to pass on to the load only frequencies *above* a certain value, we merely reverse the connections. This time, we use a con-



*Courtesy General Electric Co.*

Resonant circuits are used to select the desired station from among all others which are broadcasting. When you tune a radio, you are varying either the L or the C of these circuits to make their reactances balance at the desired frequency.

denser between the source and load; we use a coil in parallel with the load; or use both.

These circuits are known as filter circuits — called *high-pass* filters if they allow only frequencies above a certain cut-off frequency to pass, and called *low-pass* filters if they allow only frequencies below the cut-off frequency to pass.

► Notice that these circuits use only a *single* coil or a *single* condenser, or else they use one in series and the other in parallel with the load. There is *another combination* of a coil and condenser in which *both* are used together. This combination acts in a different manner altogether, forming the basis for new groups of filters.

► As you will soon see, there are many instances where we do not want *all* the frequencies above a certain value, or *all* the frequencies below a certain value. Instead, we may want to pass a certain narrow *band* of frequencies and exclude *all others above or below* this particular frequency range. Or, we may want to *reject* a narrow band of frequencies, yet pass all others *above and below* this narrow band. We can get these actions by: 1, using a coil-and-condenser combination between the load and the source; 2, using the proper combination in parallel with the load; or 3, using the proper combination *as* the load. This combination is known as an “L-C” circuit (L is the symbol for inductance; C is the symbol for capacitance).

► This particular action occurs because of the manner in which the reactances of these parts vary with the frequency. As you know, an increase in frequency causes a higher inductive reactance, while the same increase in frequency causes a lower

capacitive reactance. In other words, if we increase the frequency applied to a coil-condenser combination, the reactance of the coil will increase while the reactance of the condenser will decrease. *Hence, with any coil-and-condenser combination, there is some frequency at which the two reactances are exactly equal*, as will be shown in this lesson. A very special action occurs at this frequency.

By choosing different coil and condenser values, we can “move” this action to a different frequency. This makes it possible for us to “tune” a radio receiver (by using variable coils or condensers) so that we can select the desired signals and exclude signals from other stations operating on other frequencies. *The L-C circuit used for frequency selection is called a tuned circuit or a resonant circuit.*

► Another important use for this characteristic of a resonant circuit is in the generation of radio waves. The fact that this particular combination of parts can be made to accept some one frequency and exclude others makes it possible to use this combination with vacuum tubes in such a way that radio frequency waves of a desired frequency are generated. Thus, the L-C circuit can be used in transmitters and in oscillators to *produce* any desired frequency, while other L-C circuits can be used to accept or reject a particular frequency from among many others.

► Before we can see how L-C circuits are used for these purposes, we must learn how they may be connected together and how they act at resonance. Therefore, in this lesson we will show first how the two main types of resonant circuits function, then discuss some of the ways in which they are used in radio.

# Series Resonant Circuits

There are two distinct types of resonant circuits, each having its own peculiar behavior. In a *series resonant circuit*, the source of voltage, the coil, and the condenser are all *in series*. In a *parallel resonant circuit*, the source of voltage, the coil, and the condenser are all *in parallel*. We will consider series resonant circuits first.

► The easiest way for you to understand the important actions that take place in a series resonant circuit is to analyze several circuits in turn. We'll start with a circuit composed of an a.c. generator and a variable resistance; then we'll add an inductance; next, substitute a capacitance for the inductance; and finally, combine the inductance, capacitance, and resistance into a resonant circuit. At each step, we'll see what current flows in the circuit and what voltages exist. This step-by-step procedure will show you exactly what effect each circuit element has on a resonant circuit.

For our explanation, we'll assume that the a.c. generator in the circuit produces 110 volts with a frequency of 500 cycles. This frequency is chosen purely for convenience, because it allows us to form a resonant circuit with inductance and capacity values which are easy to handle in calculations. Actually, any frequency—even r.f.—could be used and would show us the same fundamental facts, if the parts values were properly chosen.

**The Resistance Circuit.** Fig. 1A shows a circuit with which you're already familiar. It is made up of a variable resistor—value set at 120 ohms—in series with the a.c. source. An ammeter is put in the circuit, and a voltmeter is placed across the resistor to show the current and voltage that exist in the circuit. As you

already know, all the source voltage will be developed across the resistance, so the voltmeter reads 110 volts. And Ohm's Law for this circuit tells us the current flowing will equal the voltage divided by the resistance, so our ammeter reads .92 ampere (actually .917 as it is  $110 \div 120$ , but the meter reading will be so close to .92 that we would accept this as our reading. Other similar *practical* values are used in this lesson.) You learned from your previous lessons that this current is in phase with the voltage.

**The Resistance-Inductance Circuit.** Now, suppose we add a 100-millihenry (mh.) coil in series with the resistance; this will give us the circuit shown in Fig. 1B. The ammeter now shows a current flow of about .33 ampere. The reason for the decrease in current is that we have added the reactance of the coil to the circuit.

The voltage developed across the coil is shown by a meter to be 103 volts, while that across the resistance

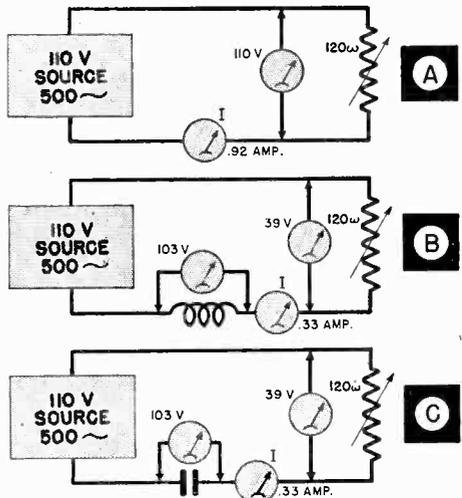


FIG. 1. When either a coil or a condenser is used in series with a resistor, the circuit current is reduced, as we would expect.

is 39 volts—a total of 142 volts, which is considerably more than the 110 volts generated by the source. As you learned in your lesson on coils, *phase* causes this effect. The voltage across the coil is out of phase with the circuit current (the voltage *leads* the current by  $90^\circ$ ), while the voltage across the resistor is in phase with the current.

**The Capacitance-Resistance Circuit.** Now, let's replace the 100-mh. coil with a 1-mfd. condenser as in Fig. 1C. At 500 cycles, the reactance of a condenser of this size is approximately the same ohmic value as that of the 100-mh. coil. (If you wish, you can check this fact by figuring the reactances of the coil and of the condenser by the methods given in earlier lessons.) We see at once that this is true when we measure the current and voltages in the circuit. The voltage across the condenser is 103 volts, and that across the resistor is 39—a total of 142 volts, exactly what we found in the resistance-inductance circuit. And the current is again .33 ampere.

As you know from your study of condensers, the voltage across the condenser is *out of phase* with the circuit current (the voltage *lags* the current by  $90^\circ$ ) while the resistor voltage is in phase with the current.

## THE RESONANT CIRCUIT

So far, the actions of the components of the three circuits we've analyzed have been familiar. If we combine the three parts, then, we will have a circuit with a condenser, a coil, and a resistor all in series with an a.c. source, and you might expect the current to decrease further.

But—something unusual happens! The voltages and the circuit current increase tremendously when compared with the values of Figs. 1B and 1C. As Fig. 2 shows, a current of .91

ampere flows, and the voltage across the resistor rises to 109 volts. These values are almost what they were in the simple resistance circuit shown in Fig. 1A. Evidently, as far as the resistor voltage and the circuit current are concerned, the coil and the condenser might as well not be in the circuit.

► See what happens to the condenser and the coil voltages! Both increase to 285 volts—well over *twice* the source voltage. The circuit current has increased to almost three times what it was in the circuits shown in Figs. 1B and 1C. As we would expect, the voltage drop caused by this current flow through the coil and condenser reactances has also increased almost three times—from 103 volts to 285 volts. Just what makes this increase possible?

There is a very simple reason for these increases. The condenser voltage is  $90^\circ$  *behind* the circuit current, and the coil voltage is  $90^\circ$  *ahead* of the circuit current. Therefore, the two voltages are  $180^\circ$  out of phase with each other, which means their effects are exactly opposite. And, since the voltages are equal in size and opposite in effect, they cancel one another completely so far as the *circuit* is concerned. A voltmeter placed between points X and Y in Fig. 2 would show no voltage. (That is,

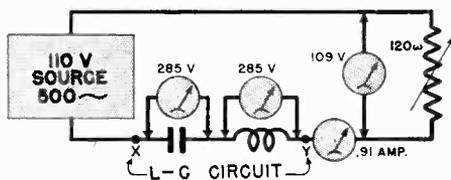


FIG. 2. When both the coil and the condenser are connected in series and are resonant to the applied frequency, their reactances effectively disappear. The circuit current is now determined by the source voltage and the resistance. The voltage drops across both the coil and the condenser are large, and their values are the product of the circuit current multiplied by the respective reactances.

it would if the coil were a perfect inductance, without resistance. But any practical coil has a certain amount of resistance, which will cause a slight extra voltage drop across the coil that the condenser voltage will not cancel. This is the reason why the full source voltage does not appear across the resistor in Fig. 2—one volt has been dropped in the coil resistance.)

This canceling effect of the voltages across the coil and the condenser is always present to some extent in a series L-C circuit, whether or not resonance exists. That is, even when the coil and the condenser reactances are not equal, some cancellation will occur. If the condenser voltage is smaller than the coil voltage, it will cancel a part of the coil voltage, and so cause a slight increase in circuit current over that which would flow if the coil were used alone. The same condition holds if the coil voltage is smaller than the condenser voltage. But the canceling effect becomes extremely noticeable only when the coil and the condenser reactances are equal for the particular frequency fed into the circuit—that is, when the circuit is at resonance. Then we get the enormous increase in circuit current we found in the circuit of Fig. 2, where practically all that limits the current is the resistor.

**Reducing R.** Fig. 3 shows what happens if we reduce the size of the resistor in our resonant circuit. Here, the electrical sizes of the circuit components are the same as in Fig. 2, except that the ohmic value of the resistor is reduced from 120 to 50 ohms. The circuit current increases to 2.1 amperes—and the condenser and the coil voltages (the result of this current flow through their impedances) jump to 670 volts! This shows that the value of the resistance in a resonant circuit is very important in

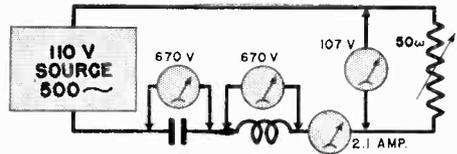


FIG. 3. The voltages measured here prove that the circuit current in a resonant circuit depends only on the circuit resistance.

determining the amount of current flowing. Notice, however, that less voltage appears across the resistor in the circuit of Fig. 3 than was across it in Fig. 2—107 volts for the former as against 109 volts for the latter. This difference is caused by the increased voltage drop across the resistance of the coil when the current is increased.

We cannot, therefore, produce an almost infinite current or an almost infinite voltage across the coil or across the condenser by reducing resistor  $R$  to zero. The resistance of the coil will limit the current flow, even if there is no other resistance in the circuit. However, if we used a coil with very low resistance and carefully removed as much resistance as possible from the remainder of the circuit, we could get a very high current in our resonant circuit (and so a very high voltage across the coil and across the condenser). Of course, in practice, the source voltage usually will range from a small fraction of a volt to just a few volts, so even at best the condenser or the coil voltage is small. However, in such a circuit, it is sometimes necessary to limit the source voltage so the current will not rise to such values that the condenser voltage ratings are exceeded.

**Resonant Voltage Step-Up.** One of the most useful effects in a series resonant circuit is the fact that a high voltage—much higher than the source voltage—can exist across the coil and across the condenser at the resonant frequency of the combination. This

is called *resonant voltage step-up*. By connecting across either the coil or the condenser (but not across both), we can use the voltage across that part to operate another circuit. The resonant circuit in effect has “amplified” the signal to which it is tuned, but will not so treat other frequencies. Hence, the circuit “selects” particular frequencies by increasing these more than others. You will meet and use this effect constantly in your later work in radio.

**Conditions in a Series Resonant Circuit.** Let’s repeat the substance of what you’ve just learned, so it will be firmly fixed in your mind. In a series resonant circuit *at resonance* we have:

1. Maximum circuit current, determined by the resistance in the circuit and the source voltage.

2. Maximum and equal voltages across the coil and across the condenser, determined in size by the circuit current and by their reactances.

3. The L-C part of the circuit (the section between points X and Y in Fig. 2) *acts like a resistor of low ohmic value*; this resistance is almost entirely concentrated in the coil.

► Remember—this condition of series resonance exists for the circuits we’ve been considering *only at one particular frequency*. We deliberately chose the values of inductance and capacity used so that their reactances would be equal at 500 cycles, the frequency of the source. *Their reactances would not be equal at any other frequency*, so, with the particular condenser and coil we have used, *our circuit cannot be resonant at any other frequency*. In other words, our circuit is *tuned to resonance at 500 cycles*.

### VARYING L AND C

Let’s see what would happen if we were to use other values of capaci-

tance or inductance in our circuit.

**Varying C.** Suppose we were to vary the value of  $C$  (by inserting condensers of different sizes) while leaving the *source frequency*, the *coil*, and the *resistor* just as they are in Fig. 2. We are particularly interested in the circuit current  $I$ , since it determines what the voltage across  $L$  or  $C$  will be.

Fig. 4 shows a plot of the circuit current  $I$  for various values of  $C$ . With condensers smaller than 1 mfd., circuit current  $I$  drops rapidly; the

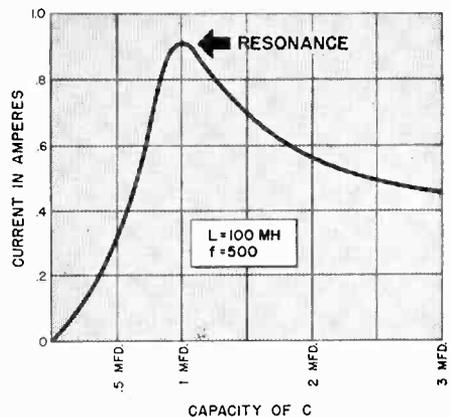


FIG. 4. This graph shows how the current varies in a series resonant circuit when the value of capacity  $C$  is varied from 0 to 3 mfd. Note that at 500 cycles and with a 100 mh. coil, resonance and maximum current occur when the capacity of  $C$  is 1 mfd., making its capacitive reactance equal to the inductive reactance of  $L$ .

smaller the value of  $C$ , the lower the current. The reason is quite simple; reducing the electrical value of  $C$  increases its reactance above that of  $L$ , and that portion of this capacitive reactance which is not cancelled by  $L$  acts to limit current flow.

With condensers larger than 1 mfd., the ammeter reading likewise drops; increasing the electrical value of the condenser makes the reactance of  $C$  lower than that of  $L$ , with the result that a portion of the inductive reactance is left over to limit current flow.

► Incidentally, when we tune our series resonant circuit away from resonance, the circuit will act the same as *that reactance which is the larger or is predominant*. Lowering the value of  $C$  here makes the capacitive reactance the larger, and the series resonant circuit therefore acts like a condenser, while increasing  $C$  makes its reactance less and the circuit acts like an inductance.

**Varying  $L$ .** If the inductive reactance (instead of capacitive reactance) were varied by inserting various values of  $L$  in the series resonant circuit, similar results would be obtained, as it is the amount of reactance left over (not balanced out) which governs the current. This is

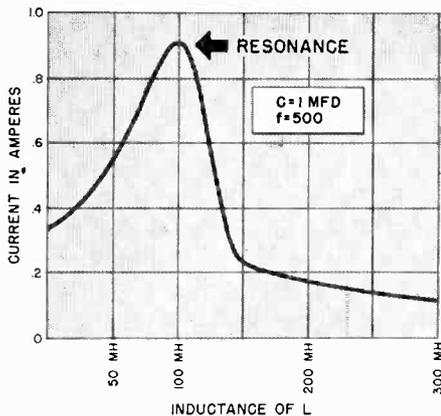


FIG. 5. Varying  $L$  instead of  $C$  produces this kind of graph. Resonance again is obtained when the coil and condenser reactances are equal.

shown by Fig. 5—a graph of the circuit current for various values of  $L$ .

√ **Varying Both  $L$  and  $C$ .** With a 500-cycle source, we know that resonance can be obtained with a 1-mfd. condenser and 100-mh. coil. Could resonance also be obtained at this same frequency with other values of  $L$  and  $C$ ? Suppose we try using a 50-mh. coil and vary  $C$ . If we do, we will find that we get resonance (maxi-

mum current) with a 2-mfd. condenser. Now notice—multiplying together the values of  $L$  and  $C$  in our original circuit (Fig. 2) gives a figure of 100 ( $100 \times 1$ ). We get exactly the same figure for a circuit with a 50-mh. coil and a 2-mfd. condenser ( $L \times C = 50 \times 2 = 100$ ). As a matter of fact, we would find, if we tried other values of  $L$  and varied  $C$  to bring the circuit to resonance at 500 cycles, that multiplying together the values of  $L$  in millihenrys and  $C$  in microfarads at resonance always will give the number 100 for a 500-cycle source (this figure is sufficiently accurate for practical purposes). The larger the value of  $L$ , the smaller will be the value of  $C$  required for resonance.

► If we changed the source frequency to, say, 1000 cycles, then changed either  $L$  or  $C$  until resonance was reached, we would find that the product of  $L$  (in millihenrys) multiplied by  $C$  (in microfarads) would be 25. At some other frequency, we would find the product of  $L$  times  $C$  would be equal to some other number; in other words, there is one number corresponding to each resonant frequency. Tables are available giving these numbers for different frequencies, so an engineer who is assembling a resonant circuit need only look up this number for the frequency in question, then choose values of  $L$  and  $C$  which, when multiplied together, will give this number. (Although this number is equal to  $L$  times  $C$ , engineers for convenience often omit the word “times” and simply call it the  $L-C$  value.)

This relationship between  $L$  and  $C$  at any resonant frequency makes it easy for engineers to predict beforehand what values of  $L$ ,  $C$ , and frequency will give resonance in a particular circuit. When any two of these values are known, they can find the

third either by means of a chart, a table, or a mathematical formula.\*

► We just said that the product of  $L$  and  $C$  for a circuit resonant at 500 cycles is 100, while  $L$  times  $C$  for a circuit resonant at 1000 cycles is 25. This is an example of a general rule which you will find useful to remember—the lower the  $L$ - $C$  value of a series resonant circuit, the higher the frequency to which the circuit is resonant. In other words, decreasing the value of either  $L$  or  $C$  increases the

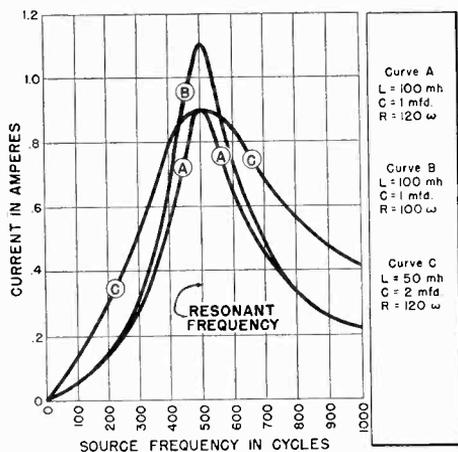


FIG. 6. Here is what happens when you vary the frequency of the source in a series resonant circuit. These curves prove that for any given set of values for  $L$  and  $C$ , there is only one frequency at which resonance and maximum current will be obtained. Notice that it is possible to obtain resonance with a different value of  $L$  and  $C$ , providing the  $L$ - $C$  product is the same. Also, reducing the circuit resistance (curve B) produces a sharper curve with a higher peak.

resonant frequency of a series resonant circuit, while increasing the value of  $L$  or  $C$  decreases the resonant frequency.

\* For those who are interested in mathematics, this formula is:

$$f = 159,000 \div \sqrt{LC}$$

In this formula,  $f$  is the resonant frequency in cycles,  $L$  is the inductance of the coil in microhenrys (1 millihenry = 1000 microhenrys) and  $C$  is the condenser capacity in microfarads. This formula applies to both series and parallel resonant circuits.

For example, using a 100-mh. coil and a condenser of 2 mfd. does not give resonance at 500 cycles, but does give resonance at the frequency corresponding to an  $L$ - $C$  value of 200. As we expect, the new resonant frequency is lower than 500 cycles—it is near 355 cycles. Hence, when we change either the coil or the condenser, we tune away from one frequency but establish resonance at another. Thus, an  $L$ - $C$  circuit is always resonant to some frequency, although it may not be the desired frequency.

**Resonance Curves.** The most important characteristic of a resonant circuit is how much better it accepts the desired resonant frequency than the undesired frequencies on either side. In radio receivers, for example, we want the resonant circuits to tune in a desired carrier frequency signal while rejecting the adjacent undesired carriers. To find how well a resonant circuit can do this, we refer to a curve obtained by plotting current against source frequency. (Such a curve is commonly called a *resonance curve*, because it shows how circuit conditions change as we pass through resonance.) Three such curves for series resonant circuits are shown in Fig. 6; let us see how much they can tell.

► Curve A in Fig. 6 is for the series resonant circuit in Fig. 2; it was obtained by keeping the source voltage constant while varying the frequency and measuring the circuit current. You can see that there is quite a drop in circuit current as you go above and below the resonant frequency of 500 cycles. The sharper the curve at resonance, the greater will be this drop in current or, as radio men say, the better will be the selectivity of the circuit. (Since such a curve shows how selective a circuit is, it is often called a *selectivity curve*.) You will learn later in the course, however,

that a certain amount of broadness in the curve at resonance is also desirable in order to keep distortion at a minimum. Let us see how we can get a more peaked curve, and a broader curve.

► You will recall that the rheostat ( $R$ ) was set at 120 ohms in Fig. 2; if we change the setting to 100 ohms, the circuit resistance at resonance will be lower and we will secure a more peaked resonance curve. This is evident in Fig. 6 from the fact that curve  $B$  for 100 ohms has a higher and sharper peak than curve  $A$  for 120 ohms. Lowering the resistance in a series resonant circuit thus improves selectivity, as it allows a higher current at resonance. This means the resonant step-up is greater, but the voltages at other frequencies are not stepped up, so the difference between the voltage at resonance and the off-resonance values is greater.

► Suppose we use a 50-mh. coil and a 2-mfd. condenser, resetting  $R$  to 120 ohms. We still get resonance at 500 cycles, as the  $L$ - $C$  value of 100 has not been changed, but now curve  $C$  in Fig. 6 represents conditions in the circuit. Note that this curve is considerably broader at resonance; this indicates that a wider range of frequencies around the resonant value will be passed almost equally well. If we experimented further in this same way, we would find that the lower the value of  $L$ , while circuit resistance is kept constant, the broader will be the resonance curve obtained. (Incidentally, resonance or selectivity curves

like those in Fig. 6 are also known as *frequency response curves* or simply as *response curves*.)

### Resonance Current and Voltage.

We have pointed out several times that a circuit will be resonant at 500 cycles regardless of what values of  $L$  and  $C$  we use, provided only that the product of  $L$  times  $C$  equals 100. The circuit current will be about the same at resonance whether we use a large coil and a small condenser or a smaller coil and a larger condenser. This is because the reactances of  $L$  and  $C$  cancel at resonance, no matter what their values are, and only the resistance in the circuit limits the current flow.

The voltages across the coil or condenser at resonance, however, will depend upon the values of  $L$  and  $C$ . If we obtain resonance by using a 50-mh. coil, for instance, the voltage across it will be only about half what it would be across a 100-mh. coil. You can readily see the reason for this—the voltage across the coil is equal to the current times the coil reactance. A 50-mh. coil has only half the reactance of a 100-mh. coil and, since the current remains constant, reducing the coil reactance one-half reduces the voltage across it by one-half. For this reason, in series resonant circuits a high  $L$ -to- $C$  ratio is used when the greatest resonance voltage step-up is wanted. (A large inductance and a small capacity are used.) Don't confuse this ratio with the  $L$ - $C$  value; the ratio is  $L \div C$ , while the  $L$ - $C$  value is  $L \times C$ .

# Parallel Resonant Circuits

When the source for a resonant circuit is connected in parallel with both the coil and the condenser, as in Fig. 7, we have a *parallel resonant circuit*. The values of  $L$ ,  $C$ , and  $R$  used in this circuit are the same as those in the series resonant circuit in Fig. 2, so you might expect that circuit current and voltages would be about the same for Fig. 7 as they are for Fig. 2. But actual measurements would show that they are completely different.

For one thing, the circuit current in Fig. 7 (measured on an a.c. ammeter in series with the generator) is very low. (As you recall, the circuit current in a series resonant circuit is high.) And, practically all the generator voltage appears across the resonant circuit, between terminals  $X$  and  $Y$ , with almost no voltage across resistance  $R$ . This is completely different from the series resonant circuit.

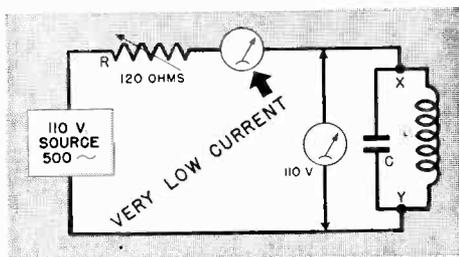


FIG. 7. At resonance, a parallel resonant circuit reduces the line current to a very low value.

In the series circuit, nearly all the generator voltage appears across resistance  $R$ , and almost none across the resonant circuit. (Of course, there is always a high voltage across each of the *parts* in the series circuit, but these voltages cancel one another. Across the *whole*  $L$ - $C$  circuit, between terminals  $X$  and  $Y$  in Fig. 2, there is almost no voltage.)

► Suppose we set up a circuit like that in Fig. 8, with a.c. ammeters inserted in the parallel resonant circuit to measure currents  $I_1$  and  $I_2$  through  $C$  and  $L$ , respectively. We would discover another surprising fact—there is plenty of current inside the parallel circuit itself!  $I_1$  and  $I_2$  are both high, and equal to one another. Yet, outside the parallel circuit, there is practically no current.

► Clearly, this resonant circuit for some reason acts like a very high resistance *across its terminals* ( $X$  and  $Y$  in Fig. 8), but at the same time offers very little opposition to current flow *in the resonant circuit itself* (the circuit made up of  $L$ ,  $C$ , and the two ammeters). The reason for this peculiar combination of actions is really quite simple. Here, our reference factor is the source voltage which is across both the coil and the condenser. You know that the current flow in a coil is  $90^\circ$  behind the coil voltage, while the condenser current is  $90^\circ$  ahead of the condenser voltage. Since the source voltage is now across both the coil and the condenser, it becomes the coil voltage and the condenser voltage. As one current is ahead  $90^\circ$  and the other is behind  $90^\circ$ , this makes the *currents*  $180^\circ$  out of phase. (Compare this with the series resonant circuit, where the *current* is the reference value and the *voltages* are  $180^\circ$  out of phase.)

The fact that the currents  $I_1$  and  $I_2$  are  $180^\circ$  out of phase means that the condenser discharges during the half cycle in which the coil stores up energy; on the next half cycle, the coil releases energy while the condenser stores it. When the impedances of the coil and the condenser are equal (that is, at resonance) the energy stored by

the condenser equals the energy released by the coil, and vice versa. Then the condenser absorbs all the energy released by the coil, and in turn releases this energy to be absorbed by the coil. Thus the coil and condenser pass current back and forth

as the *resonant resistance* of the circuit.

**Resonant Current Step-Up.** You have just seen that the current through either the coil or the condenser in a parallel resonant circuit is higher than the line current at resonance. This is *resonant current step-up*—the ability of a parallel resonant circuit to circulate a current many times greater than the current fed into it. This particular characteristic is very useful in radio circuits, as you will soon see.

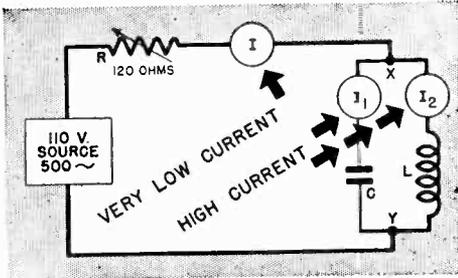


FIG. 8. The current within a parallel resonant circuit is very high, compared to the line current, due to resonance current step-up.

inside the resonant circuit, causing the high readings on the meters in the circuit. The source has to supply only enough power to make up for the small power losses in the small resistances within the coil and the condenser, so the current outside the resonant circuit is small.

► Whenever a coil and a condenser are in parallel in an a.c. circuit (regardless of whether or not their reactances are equal), one will be feeding current back into the line while the other is drawing current. Therefore, the line current at any instant (taking phase into account) will be the difference between the two currents. When the reactances are equal, as at resonance, the line current is a minimum, while both the coil and condenser draw a current determined by their reactances and the voltage across the resonant circuit. At resonance, then, a parallel resonant circuit behaves as a *resistor of high ohmic value*, reducing line current almost to zero. The resistance of a parallel resonant circuit at resonance is known

### VARYING R, L, C, AND F

**Varying the Coil Resistance.** Since all coils have a certain amount of resistance acting in series with their inductance, let us see how increases in this resistance affect a parallel resonant circuit. Instead of analyzing circuits using coils with the same inductance and various values of resistance, let us assume a circuit like that in Fig. 9, where we have a rheostat  $R_1$  in series with our coil. If we were to start with  $R_1$  at zero resistance, then increase it gradually, we would find that the coil current goes down only a little, the condenser current remains the same, but the line current goes up. This means that when the resistance in series with the coil in a parallel resonant circuit is increased, the *resonant resistance of the L-C cir-*

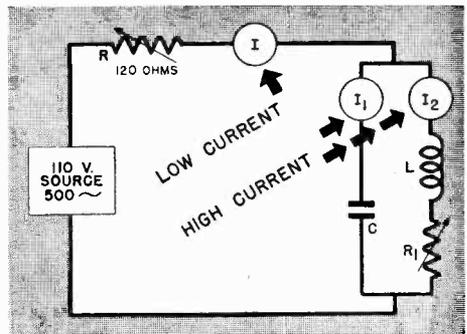


FIG. 9. Resistance within the resonant circuit ( $R_1$ ) reduces the current step-up.

*cuit* (between points  $x$  and  $y$  in Fig. 8) is *decreased*. This allows more line current to flow, and makes the line current value more nearly equal to the condenser current. We can conclude, then, that inserting resistance in a parallel resonant circuit *decreases* both the resonant current step-up ratio (the ratio of coil current to line current) and the resonant resistance.

**Varying  $C$ .** With the parallel resonant circuit in Fig. 9 tuned to resonance and both  $R$  and  $R_1$  at their zero resistance settings, let us see what effect varying  $C$  would have on the line current while the source frequency is kept constant at 500 cycles. Fig. 10, in which the line current is plotted for various values of capacity, shows the result we would get.

Starting with a very small value of capacity for  $C$ , we would find that condenser current is practically zero, as it offers a very high reactance. The line current is then determined by the coil reactance ( $R$  is set at zero). Hence, the coil and line currents are both about .35 ampere.

As the capacity of  $C$  is gradually increased, its reactance decreases, so we would find that the condenser current goes up, line current goes down, and coil current remains unchanged. Coil and condenser currents become more nearly equal as the capacity of  $C$  approaches its resonant value. Finally, the condition of resonance is reached, where line current is at its minimum value and the coil and condenser currents are equal.

Increasing the capacity of  $C$  above the resonant value causes line current to go up again, while condenser current continues to increase. With a very high capacity for  $C$  the reactance of  $C$  becomes so low that coil current becomes negligible in comparison with the extremely high condenser current, and meters  $I$  and  $I_1$  of Fig.

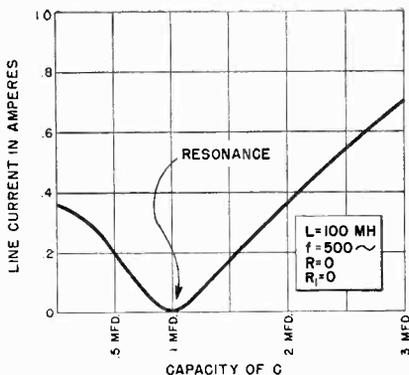


FIG. 10. This graph tells how the line current changes when the capacity of the condenser ( $C$ ) in a parallel resonant circuit is varied.

9 read almost the same values.

► Lowering the capacity of  $C$  below the resonant value (the value giving resonance at the source frequency) makes the *reactance* of the condenser greater than the reactance of the coil. A parallel resonant circuit acts the same as the part drawing the most current (having the *least* reactance), so in this case the parallel circuit *acts like a coil*. On the other hand, raising the capacity of  $C$  above the resonant value makes the reactance of the condenser less than that of the coil, so the parallel circuit *acts like a condenser*.

► Actually, when we reduce or increase the value of  $C$  from the resonant value, we make the parallel circuit resonant to some other frequency. The frequency to which the circuit will be resonant with the changed value of  $C$  is, of course, the frequency at which the reactance of the coil will equal the reactance of the new value of  $C$ . From what you already know about how the reactances of coils and condensers vary with frequency, it will be easy to see that *decreasing* the value of  $C$  *raises* the frequency to which the circuit is resonant, while *increasing* the value of  $C$  *lowers* the resonant frequency.

**Varying L.** If we were to vary inductive reactance (instead of capacitive reactance) by using different sizes of coils for  $L$  while keeping the source frequency constant, we would find that low-inductance values of  $L$  had the same effect as high-capacity values of  $C$ , and high values of  $L$  gave the same results as low values of  $C$ .

**Varying Frequency.** Now suppose we see what would happen in the circuit shown in Fig. 9 if we were to set both  $R$  and  $R_1$  to zero resistance and vary the frequency of the source without changing its voltage or the coil and condenser values. Suppose we start with a very low frequency (less than one cycle per second). This is very nearly the equivalent of direct current, and you know that  $C$  will act as an open circuit (very high reactance), while  $L$  will act as a short (very low reactance). This means that a very high current will flow through the coil and ammeters  $I$  and  $I_2$ .

Increasing the frequency gradually from zero to 500 cycles makes the reactances of  $L$  and  $C$  more and more nearly equal, with the parallel resonant circuit acting in the same manner as a coil (because below 500 cycles

the reactance of  $L$  is lower than that of  $C$ ). Line current drops to almost zero at 500 cycles, and condenser current increases gradually, being equal to the coil current at 500 cycles.

► When the source frequency is 500 cycles, we have the resonance condition already described.

► As the source frequency is increased above 500 cycles, coil reactance goes up and condenser reactance goes down, making our parallel resonant circuit behave as a condenser, with line current increasing again to supply the extra current drawn by the condenser. At a very high frequency, the reactance of  $L$  will be so high that it will act as an open circuit, but the reactance of  $C$  will be practically zero, and line current will be just as high as it was for very low frequencies.

**Varying Both L and C.** If we were to insert various values of  $L$  in Fig. 9 and use the correct values of  $C$  to give resonance at 500 cycles, we would find in each case that the same number is obtained by multiplying together the values of  $L$  and  $C$ . In other words, the product  $L-C$  is a constant for a given frequency, just as it was for series resonant circuits. Furthermore, it is the *same* number as the  $L-C$  value for series resonance. Also, the formula for determining the resonant frequency when  $L$  and  $C$  are known (already given you in a previous footnote) applies to parallel resonant circuits as well as series resonant circuits.

**Resonance Curves.** We can draw the same kind of resonance curves for our parallel resonant circuit that we draw for the series circuits by plotting frequency against line current, as in Fig. 11—the entire 110 volts is applied, as  $R$  is reduced to zero. When  $L$  is 100 mh. and  $C$  is 1 mfd., curve  $A$  is secured, while when  $L$  is 10 mh. and  $C$  is 10 mfd., curve  $B$  is the result. As you can see from these curves,

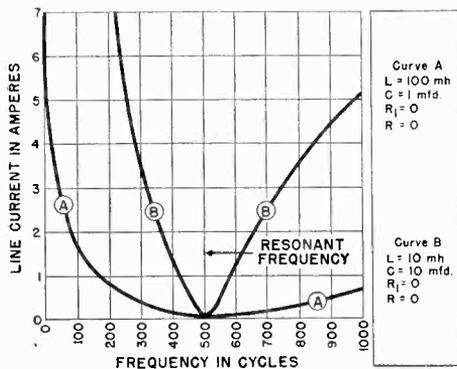


FIG. 11. The effects of variations in frequency on the line current for a parallel resonant circuit are given on this graph for two different  $L-C$  ratios.

lower values of  $L$  in parallel resonant circuits make line current increase more rapidly off resonance. In other words, a parallel resonant circuit

tunes more sharply if  $L$  is small, so a low  $L$ -to- $C$  ratio is desirable here. A small inductance and a large condenser are used for greatest selectivity.

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## Practical Resonant Circuit Facts

Tuned circuits are found most commonly in the r.f. stages of radio apparatus, where they are used to separate signals coming from different stations. This separation is possible when the stations operate on different frequencies, as we will show you later in this lesson by means of typical circuits.

However, there are several points we should clear up before going on to these practical circuits. We should get an idea of the amount of resonance step-up obtainable; learn how to recognize series and parallel circuits; and learn how these circuits may be adjusted or tuned to resonate with different frequencies. We will now take these subjects up in order.

### Q FACTOR

You have learned that the resonant voltage step-up or resonant current step-up of any resonant circuit increases as the resistance in the circuit is decreased. Since a high step-up ratio usually is wanted, engineers seldom insert resistors intentionally in resonant circuits. However, any resonant circuit has a certain amount of loss caused by the effective a.c. resistances of the coil, the condenser, and the wiring in the circuit. Usually we can assume this loss is concentrated in the coil. And we can see what effect such a loss has on a circuit by using a quantity called the "Q factor" of the coil.

The Q factor is really a measure of

the merit of the coil. It can be computed for any coil from the formula  $Q = X_L \div R$ , where  $R$  is the total a.c. resistance of the coil, and  $X_L$  is the inductive reactance of the coil. (Test instruments are available which will measure the Q of a coil.) As practically all the loss in a resonant circuit is concentrated in the coil, we usually assume that the coil Q is also the Q of a resonant circuit using that coil. This factor applies to both series and parallel resonant circuits.

► Of what use is the Q factor to us in our study of resonant circuits? For one thing, it allows us to compare two coils of the same inductance to see which has the more desirable characteristics for use in a tuned circuit. If two coils have the same inductance, but the Q factor of one is higher than the other, we know at once that the coil with the higher Q will give sharper tuning (better selectivity) in a resonant circuit. (This follows from what you have already learned. You know that a circuit tunes more sharply if the resistance in it is small, and from the formula just given, it is obvious that if two coils have the same inductance, the coil with the higher Q factor will have the smaller resistance.)

Also, a knowledge of the Q factor of the coil used in a resonant circuit will let us find out several things about the circuit itself. In a series resonant circuit, the applied voltage multiplied by the Q gives the voltage

available across either the coil or the condenser. In a parallel resonant circuit, the line current multiplied by  $Q$  gives the current flowing in the tuned circuit; also, multiplying  $Q$  by the coil reactance will give us the resonant resistance of the parallel resonant circuit.

► Notice—we used the a.c. resistance of the coil in our formula for  $Q$  factor. This is not the same amount as the d.c. resistance of the coil, which we could measure with an ohmmeter. The a.c. resistance of a coil is always much higher than the d.c. resistance, particularly when the coil is used in r.f. circuits. This is because r.f. current does not travel through the entire cross-section of the wire in a coil, but instead tends to travel close to the wire surface. The tendency becomes more marked as the frequency of the current is increased. This is known as the “skin effect.”

The skin effect means that the actual current-carrying area of the wire in a coil is considerably reduced as the frequency of the current increases, and therefore the resistance of the coil increases. To this resistance must be added dielectric losses in the coil form, shield losses, and, in a coil which has an iron core, losses in the core. All these factors make up the a.c. resistance of the coil as they represent losses. This resistance has all the characteristics of resistance, and must not be confused with reactance, which the coil also has. Engineers frequently draw a resistance symbol in series with the coil symbol to indicate that the coil resistance is in series with the inductance.

Since both the a.c. resistance and the inductive reactance increase as the frequency of the circuit current increases, it might seem as if these increases would cancel one another, so that the  $Q$  factor of the coil (the ratio

of the reactance to the a.c. resistance) would remain constant for all frequencies. In the interest of greater operating efficiency, many coil designers try to achieve this effect. Usually, however, the a.c. resistance of a coil increases much more rapidly than does the coil reactance as the frequency of the circuit current is raised. Therefore, it is generally true that the  $Q$  factor of a coil becomes smaller when higher frequency currents flow through it. In recognition of this fact, the  $Q$  factor of a coil always is specified as having been measured at some particular frequency.

### COMPARING RESONANT CIRCUITS

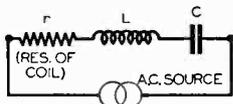
A comparison of series and parallel resonant circuits is given elsewhere in this lesson, in a table. Here are a few additional facts:

► When a resonant circuit is designed for a particular frequency, inductance and capacity values are chosen which will give the  $L \times C$  value necessary at that frequency. However, we often want a resonant circuit which can be tuned over a band of frequencies at will. For example, the r.f. circuits of your radio tune over the broadcast band, covering frequencies from 550 kc. to 1500 or 1600 kc. In these cases, the *maximum* inductance or capacity value of the variable part is found, then the circuit is designed to tune to the *lowest* frequency in the band. Then, as the variable capacity or variable inductance value is decreased, the circuit tunes to higher frequencies and thus is made to cover the desired band. In this case, notice that there are a number of frequencies applied simultaneously, and the resonant circuit is adjusted to “tune in” or accept the one in which we are interested.

► When a choice of several  $L$  and  $C$  values is possible and *maximum* selec-

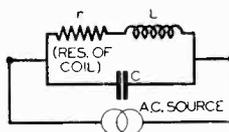
# Comparison of Series and Parallel Resonant Circuits

## SERIES RESONANT CIRCUITS



1. The coil, the condenser and the A.C. voltage source are all in series.
2. Resonance occurs when the reactance of  $L$  is equal to the reactance of  $C$ .
3. At resonance, source current is a *maximum* (very high).
4. At resonance, a series resonant circuit acts like a *resistor of low ohmic value*.
5. At resonance, the voltages across  $L$  and  $C$  are equal in magnitude but 180 degrees out of phase with each other.
6. At resonance, the same current flows through the entire circuit.
7. At resonance, the voltage across either  $L$  or  $C$  may be greater than that of the source, giving resonant voltage step-up.
8. At resonance, increasing the value of coil resistance  $r$  lowers the circuit current, thereby *lowering* the resonant voltage step-up.
9. Off resonance, the circuit acts like that part which has the *higher* reactance.
  - a. Increasing  $C$  above its at-resonance value makes the circuit act like a coil.
  - b. Reducing  $C$  below its at-resonance value makes the circuit act like a condenser.
  - c. Increasing  $L$  above its at-resonance value makes the circuit act like a coil.
  - d. Reducing  $L$  below its at-resonance value makes the circuit act like a condenser.
  - e. Applying a *higher* frequency than the resonant one makes the circuit act like a coil.
  - f. Applying a *lower* frequency than the resonant one makes the circuit act like a condenser.
10. The product  $LC$  is constant for any given resonant frequency.
11. Increasing  $L$  or increasing  $C$  lowers the resonant frequency.
12. Decreasing  $L$  or decreasing  $C$  raises the resonant frequency.
13. The Q factor of the circuit is essentially equal to the coil reactance divided by the A.C. resistance of the coil.

## PARALLEL RESONANT CIRCUITS



1. The coil, the condenser and the A.C. voltage source are all in parallel.
2. Resonance occurs when the reactance of  $L$  is equal to the reactance of  $C$ .
3. At resonance, source current is a *minimum* (very low).
4. At resonance, a parallel resonant circuit acts like a *resistor of high ohmic value*.
5. At resonance, the voltages across  $L$ ,  $C$  and the source are all the same in magnitude and phase.
6. At resonance, the currents through  $L$  and  $C$  are essentially equal in magnitude but are 180 degrees out of phase.
7. At resonance, the current through either  $L$  or  $C$  is greater than the source current, giving resonant current step-up.
8. At resonance, increasing the value of coil resistance  $r$  increases line current, thereby *lowering* the resonant current step-up.
9. Off resonance, the circuit acts like that part which has the *lower* reactance.
  - a. Increasing  $C$  above its at-resonance value makes the circuit act like a condenser.
  - b. Reducing  $C$  below its at-resonance value makes the circuit act like a coil.
  - c. Increasing  $L$  above its at-resonance value makes the circuit act like a condenser.
  - d. Reducing  $L$  below its at-resonance value makes the circuit act like a coil.
  - e. Applying a *higher* frequency than the resonant one makes the circuit act like a condenser.
  - f. Applying a *lower* frequency than the resonant one makes the circuit act like a coil.
10. The product  $LC$  is constant for any given resonant frequency.
11. Increasing  $L$  or increasing  $C$  lowers the resonant frequency.
12. Decreasing  $L$  or decreasing  $C$  raises the resonant frequency.
13. The Q factor of the circuit is essentially equal to the coil reactance divided by the A.C. resistance of the coil.

tivity is desired:

A. For series resonance, use a large inductance and small capacity, giving a high  $L \div C$  ratio.

B. For parallel resonance, use a small inductance and large capacity, giving a low  $L \div C$  ratio.

► After the  $L$  and  $C$  values have been chosen, the  $Q$  factor then determines the actual voltage or current step-up. The higher the  $Q$  (the lower the a.c. resistance), the greater the step-up.

**Distinguishing Between Series and Parallel Resonant Circuits.** One question often asked even by experienced radio men is, "How can I tell whether a resonant circuit is of the series or parallel type when I see it in a circuit diagram or in radio apparatus?" It's not really very difficult. Here's a simple procedure that will give you the right answer every time:

1. Locate the coil and the condenser forming the resonant circuit.

2. Locate the source of a.c. voltage for this resonant circuit. Typical sources will be receiving antennas, plate circuits of vacuum tubes, and voltages induced from other circuits.

3. If the source for the resonant circuit is *in series* with the coil and the condenser, you have a series resonant circuit.

4. If the source for the resonant circuit is connected *directly in parallel* with both the coil and the condenser, you have a parallel resonant circuit.

## VARIABLE INDUCTANCES

You know that a circuit can be tuned either by varying either the capacity of the condenser or the inductance of the coil, but how do we actually vary these things? You know the answer as far as variable condensers are concerned, since they have been fully covered in another lesson. Variable inductances were mentioned

in a previous lesson, but now let's spend a little time learning more about how variable inductances work, because inductance tuning is becoming increasingly more important in modern radio receivers. (It has been used in transmitters for years.)

You will recall that anything which increases or decreases the magnetic flux linkage per ampere of current flowing through a coil will increase or decrease the inductance of the coil. For example, increasing the number of turns on the coil increases its inductance, and increasing the amount of flux flowing through the coil like-

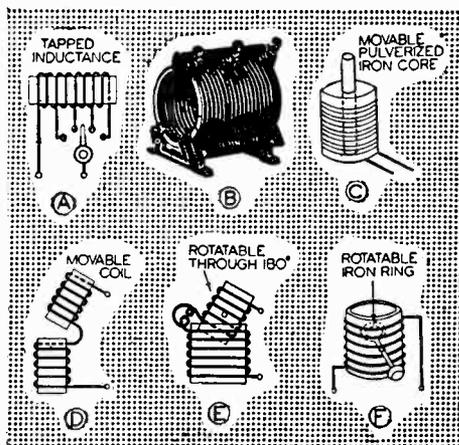


FIG. 12. Methods of varying the inductance of a coil.

wise increases its inductance. One way of building a variable inductance, then, is to use a coil which has the number of turns required to give the maximum desired inductance for a circuit, and provide some means of using fewer turns when we want to reduce the inductance in the circuit. One scheme which has been widely used for this purpose is shown in Fig. 12A. Here, various turns on the coil are connected by means of wire leads (taps) to metal contacts over which a contact arm moves, thus permitting

the operator to select as many or as few turns as he requires.

Earlier radio receivers used this tapped inductance extensively, while a variation of it is still being used in transmitters. Fig. 12B shows a transmitting coil with movable clamp type contacts which can be placed on any desired turns. Only those turns which carry current are effective in producing flux linkages.

► You can increase flux through a coil and thereby increase its inductance by placing inside the coil a material which offers less opposition than air to the flow of magnetic lines of force.

For low frequencies, iron and steel in either solid or laminated (thin sheets in layers) form make satisfactory coil cores, but for radio frequencies, solid or laminated cores act as a multitude of tiny short-circuited secondary coils which serve to reduce the useful magnetic flux. Radio manufacturers get around this difficulty by pulverizing the steel core material and mixing this steel powder with liquid bakelite (a good insulator). The resulting mixture can be molded into a core of the desired shape for use with radio frequency coils. The metal particles are insulated from each other by the bakelite and so do not produce the short-circuiting effect just mentioned. However, enough iron is in the mixture to reduce the reluctance of the magnetic field and thus to increase the inductance. If the core is designed so that it can be moved in and out of the coil, as indicated in Fig. 12C, we have another form of variable inductance. This style of inductance variation is used most commonly today in radio receivers.

► Two coils are often connected together in series. If the coils are some distance apart, so that there is no

coupling between them, their combined inductance is the sum of the separate inductances. As the coils are moved closer to each other, so that the flux of one links the turns of the other, the combined inductance will increase if the two fluxes are in the same direction and will decrease if the two fluxes are in opposite directions. A variable inductance of this type is illustrated in Fig. 12D.

A more convenient way of using the combined inductance of two coils to produce a variable inductance is shown in Fig. 12E. Here, one of the coils is made smaller and mounted on a shaft inside the other. It is possible, by rotating the inner coil, to vary the combined inductance of these coils over a wide range. When the coils are arranged so that their currents flow in the same direction, the fluxes will be in the same direction and the combined inductance will be high. When the inner coil is turned halfway around ( $180^\circ$ ) so that the currents flow in opposite directions, the fluxes will oppose each other and the inductance will be greatly reduced. A variable inductance of this type is known as a *variometer*.

► Another method of constructing a variable inductance is illustrated in Fig. 12F. A rotatable metal ring is mounted in the center of the core; when this ring is at right angles to the axis of the coil, it acts as a short-circuited secondary winding having one turn. The induced voltage causes a high current in this secondary, which in turn produces a flux which partially balances out the coil flux, reducing the inductance of the coil. Rotating the ring lessens the amount of flux which can induce a voltage in the ring. This reduces the "bucking" effect of the flux produced by the ring and increases the coil inductance. A metal disc will work equally as well as a

ring in this variable inductance, for a disc really consists of a great many rings, one inside the other. A very important fact to remember is that the presence of any conducting material near a coil, such as a metal shield surrounding a coil, will reduce the inductance.

► Of course, a fixed condenser is used in the tuned circuit when the inductance is variable, while a fixed induct-

ance is used with a variable condenser for any one tuning range. Where several ranges are to be covered, as in a receiver designed to tune over short-wave bands as well as the broadcast band, tapped coils (or separate coils, exchanged by a switch) usually are used for the extra bands. You will learn more about this when you study r.f. stages and all-wave receivers.

## How Resonant Circuits Are Used

Now, since you have learned how resonant circuits function, let us put both the series and the parallel circuits to use in practical radio applications.

### WAVE TRAPS

#### Using a Parallel Resonant Circuit.

When a radio receiver is located so near a powerful short-wave or broadcast band transmitter that signals from this station interfere with those of stations broadcasting on adjacent frequencies, a wave trap similar to that shown in Fig. 13A is quite often used. The wave trap weakens the signal of the undesired local station before this signal gets into the amplifying section of the receiver. A parallel resonant circuit made up of coil  $L$  and condenser  $C$  is inserted in the antenna lead-in wire and tuned to resonance at the frequency of the undesired signal.

Fig. 13B shows schematically how this wave trap works. The antenna and ground system can be considered a voltage source, while the radio itself acts as a load for this source. Our wave trap is inserted in the circuit between the source and the load. Now, the source voltage will always divide between the wave trap and the load

in proportion to their impedances. When the source voltage (that is, the input signal) has the frequency to which the wave trap is tuned, the wave trap has a very high impedance, while the load (the radio) has its usual moderately low impedance. Therefore, nearly all of this particular signal voltage is developed across the wave trap, and very little is fed to the radio. At other frequencies, to which the wave trap is not tuned, the trap will have much lower impedance than the radio. Then most of the signal voltage will be developed across the radio, and very little will be wasted in the trap. Thus, our wave trap acts as a *signal rejector*; it keeps one narrow band of frequencies out of the radio, but lets all frequencies above and below this band go on to the set. This circuit is actually a filter. Since it eliminates a narrow band of frequencies and lets those above and below pass, it is called a *band-elimination* filter.

A coil and a condenser connected to form a parallel resonant circuit which can be used as a wave trap are shown in Fig. 13C. The trimmer type condenser need be adjusted only once, at the time when the wave trap is installed.

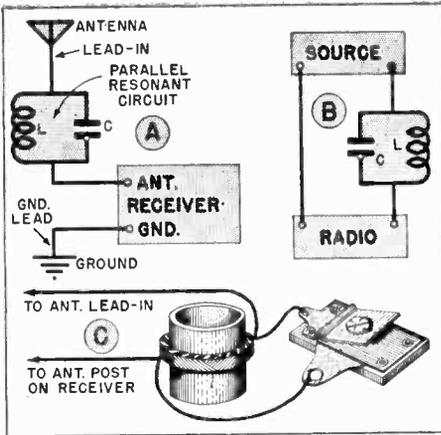


FIG. 13. Using a parallel resonant circuit as a wave trap.

### Using a Series Resonant Circuit.

A series resonant circuit, when connected to the antenna and ground terminals of a receiver in the manner shown in Fig. 14A, also serves as a wave trap. We can analyze the action of this trap with the schematic diagram in Fig. 14B. Again, the antenna-ground system is represented as a voltage source and the receiver is shown as a load. The series resonant circuit is across this load. Notice—we have included an impedance  $Z$  in the voltage source. This is the impedance of the antenna-ground system. (This impedance was present also in the circuit we just discussed, but was not mentioned because there it was unimportant in the analysis of that circuit.)

Since the load and the series resonant trap are in parallel, there always will be the same voltage drop across each. The source voltage will divide between the impedance  $Z$  of the source and the combined impedances of the trap and the receiver. At the resonant frequency of the trap, its impedance will be very low—and, of course, the combined impedances of the trap and the receiver will be even lower. Thus, when the input signal is of the frequency to which the trap is tuned, practically all the voltage will be dropped across impedance  $Z$ , and very little will be dropped across the trap and the receiver. At other frequencies, the trap-receiver combination will have much higher impedance than  $Z$ , and nearly all the input signal voltage will be dropped across them. Again, we have a *signal rejector* which keeps one narrow band of frequencies out of the radio receiver.

The actual connections of a coil and condenser used as a wave trap of the series resonant type are shown in Fig. 14C. A completely assembled wave trap in a neat metal housing is illustrated in Fig. 14D. This is of the series resonant type, with connections like that in Fig. 14A; hence its two terminals are connected directly to the antenna and ground terminals of the receiver. The condenser can be adjusted by inserting a screwdriver through a hole in the top of the housing, and turning the adjusting screw.

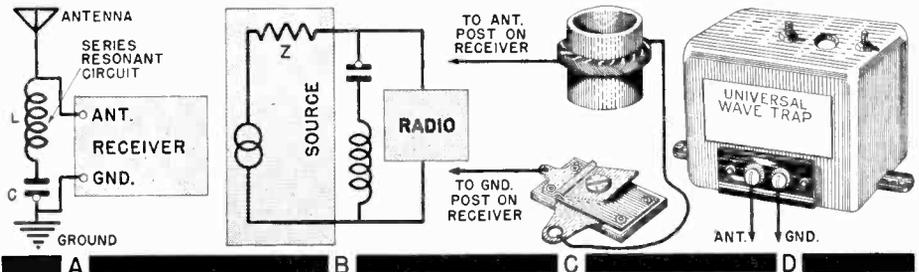


FIG. 14. How a series resonant wave trap should be connected.

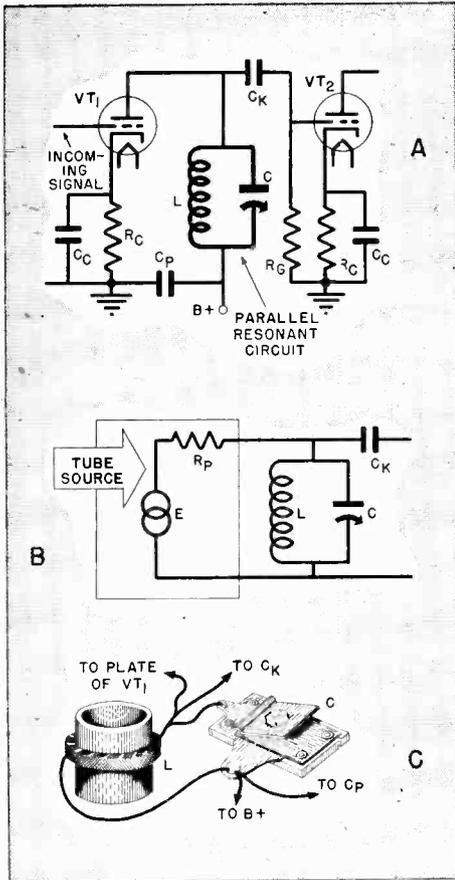


FIG. 15. A parallel resonant circuit used as a plate load in a vacuum tube circuit. A variable condenser may be used instead of the trimmer shown at C.

► Be sure to remember two very important facts about the two types of wave traps. A *parallel* resonant trap is connected *in series* with the receiver. A *series* resonant trap, on the other hand, is connected *in parallel* with the receiver. (A parallel resonant trap is often called a *series trap*, and a series resonant trap is often called a *parallel trap*, because of the way they are connected into the circuit. These terms, however, refer only to the method of connection, not to the kind of resonant circuit used.)

**Wave Trap Data.** Wave traps of the parallel resonant type generally

give best results on receivers which have a low input impedance, while those of the series resonant type will work best with receivers having a high input impedance. When the impedance of the receiver input is unknown, the practical radio man usually tries both types, selecting the one which gives the greater interference reduction.

Adjusting a wave trap is a simple matter. Connect it to the receiver in the proper manner, tune the receiver to the offending station, and then adjust the condenser in the wave trap until the offending signal is as weak as possible. If more than one station interferes, use one wave trap for each interfering station. Many all-wave superheterodyne receivers have built-in wave traps, which can be adjusted by the serviceman to tune out offending low-frequency code stations.

A wave trap is not a cure-all; there are many types of interference which it cannot reduce. You will learn more about the types of interference in later lessons.

## TUNING CIRCUITS

**Parallel Resonant Circuits in R.F. Amplifiers.** Tuned or resonant circuits are widely used for feeding radio frequency signals from one tube to another, as in tuned r.f. amplifiers. The tuned-impedance or high-gain amplifier circuit shown in Fig. 15A is an example. We are interested primarily in the action of the  $L$ - $C$  circuit drawn in heavy lines, since the other parts of this circuit will be taken up in detail later in the Course.

For our purpose, we can represent the input side of this circuit (tube  $VT_1$  and the  $L$ - $C$  circuit) by the schematic diagram in Fig. 15B. Here we consider tube  $VT_1$  to be an a.c. voltage source with a resistance  $R_p$ . You can see at once that any voltage produced by the tube source will

divide between the resistance  $R_P$  and the parallel resonant circuit.

If we tune the parallel circuit to resonance at the frequency of the incoming signal, it will have a very high resistance at this frequency. Consequently, most of the signal voltage of tube  $VT_1$  will be developed across the parallel circuit and will be passed on to the grid of tube  $VT_2$  through  $C_K$ . The parallel resonant circuit will have a low impedance for input signals of other frequencies, however, so they will be largely dropped in  $R_P$  and little energy will be passed on.

A parallel resonant circuit connected this way passes on signal voltages at the frequency to which it is tuned, and rejects voltages at all other frequencies. This means that the circuit possesses selectivity characteristics which make it very valuable for separating r.f. signals, so we can "pick out" and accept those from a desired station.

The picture diagram in Fig. 15C shows you the actual connections for a parallel resonant circuit, just as you might find them in the chassis of an r.f. amplifier. Notice how much simpler it is to draw the schematic diagram (Fig. 15A) than the picture diagram. Except for keeping certain leads (such as grid and plate leads) as far apart as possible, the manner in which parts are physically connected together is generally not important as long as the proper terminals are connected together *electrically*.

**Series Resonant Circuits in R.F. Amplifiers.** Another typical r.f. amplifier which uses a resonant circuit is shown in Fig. 16A. There is no difficulty here in recognizing the resonant circuit made up of condenser  $C$  and coil  $L$ , but the question is: Where is the source of voltage for this circuit? We must locate this source before we can tell whether the circuit

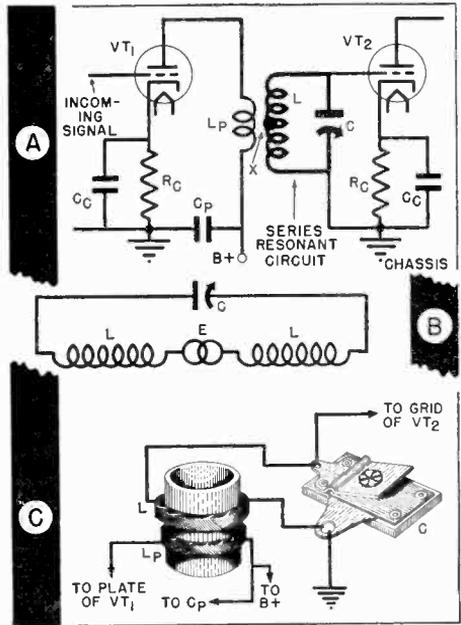


FIG. 16. A series resonant circuit used in a vacuum tube circuit. The connections at C are typical, although the coil and condenser may be of different shape.

is of the series or the parallel type.

The answer to this question lies in the transformer made up of coils  $L_P$  and  $L$ . The r.f. signal current supplied by the plate of tube  $VT_1$  flows through coil  $L_P$  and induces an r.f. voltage of corresponding wave form in coil  $L$ . The field about  $L_P$  links with  $L$ , inducing a voltage WITHIN  $L$ , just the same as if a voltage source were placed in series with  $L$ . In fact, we can represent the resonant circuit by the diagram in Fig. 16B, where the voltage induced within  $L$  is shown as the generator  $E$ .

By tracing from the generator in Fig. 16B, you can see that to make a complete circuit we must go through part of coil  $L$ , through condenser  $C$ , and through the other part of coil  $L$  back to the generator. Therefore, this is obviously a *series* circuit—as is any resonant circuit in which the voltage is induced in the coil.

The identification of a circuit of this kind puzzles many radio men. Remember this one fact, and you'll have no difficulty in identifying the circuit:

**Whenever a voltage is induced in the coil of a tuned circuit, the circuit is series resonant.**

When  $L$  and  $C$  in this series resonant circuit are tuned to the frequency of the induced signal voltage, they act as a resistance of very low ohmic value. A large r.f. current, therefore, flows through the series resonant circuit, developing a correspondingly large voltage across condenser  $C$ , and this voltage acts on the grid and cathode of tube  $VT_2$ . At all other frequencies, the  $L$ - $C$  circuit acts as a high reactance. The r.f. currents at these other frequencies are therefore low, so the voltages developed across  $C$  are low, thus the desired signals are favored. A series resonant circuit thus gives selectivity besides giving resonant amplification of the desired signal. Chassis connections as they might be for a series resonant circuit in an r.f. amplifier are shown in Fig. 16C.

**Resonant Circuits in a Superheterodyne Receiver.** You have just seen how series and parallel resonant circuits each contribute a certain amount of selectivity. In superheterodyne receivers, series and parallel resonant circuits are often connected together, in the manner shown in Fig. 17A, to secure even greater selectivity. Again the resonant circuits are shown in heavy lines.

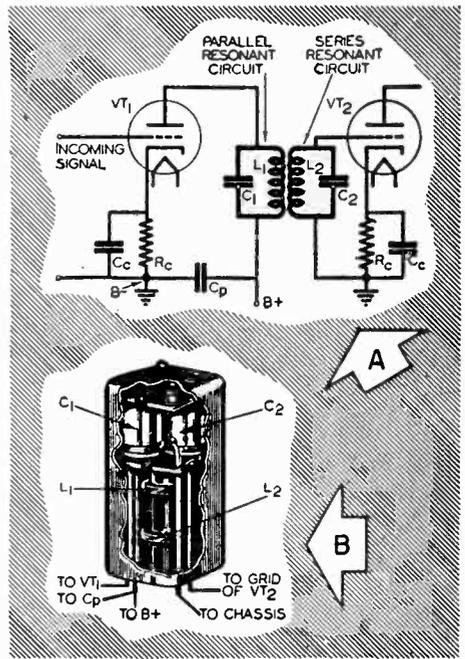


FIG. 17. In the i.f. section of a superheterodyne, both series and parallel resonant circuits are used.

First consider only circuit  $L_1$ - $C_1$ ; clearly, this is a parallel resonant circuit as is that shown in Fig. 15A, because tube  $VT_1$  is its signal source. At resonance, then, a large current flows through coil  $L_1$  and condenser  $C_1$ .

Now let us bring the  $L_2$ - $C_2$  circuit into our picture. The r.f. current flowing through coil  $L_1$  induces in coil  $L_2$  an r.f. voltage which acts in series with  $L_2$  and  $C_2$ , just as in Fig. 16A. Thus  $L_2$  and  $C_2$  form a series resonant circuit. The actual chassis connections for this double resonant circuit might be as shown in Fig. 17B.

# Coupling Radio Circuits

Now that we are well acquainted with resonant circuits and with a few of their applications, we are ready to take up the different ways in which a resonant circuit can be coupled (connected) to a source of voltage.

## ANTENNA COUPLING METHODS

The antenna circuit of a radio receiver is one of the simplest and best-known applications for coupling systems, so we will use it as our practical example in connection with this study of coupling methods. Suppose we have an antenna and a ground,

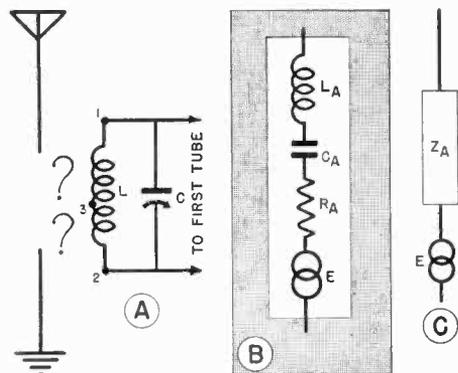


FIG. 18. How can we couple the resonant circuit to the antenna-ground?

and want to connect or couple them to the resonant or tuned circuit which feeds the first r.f. stage of a radio receiver, as shown in Fig. 18A.

The antenna-ground circuit will be a source of voltage, as it picks up energy from the traveling radio waves. However, like all other sources, the antenna-ground circuit has an impedance. It has a certain amount of resistance and also has some inductive and some capacitive reactance. Hence, the antenna-ground circuit can be represented as in Fig. 18B. As you can see, it is a series resonant circuit

itself, but in practice antennas are so dimensioned that they are resonant only in the short-wave bands. Thus, at the lower or broadcast frequencies, the combination is far off resonance, and acts like a condenser in series with a resistance.

We can lump together all the opposition as an impedance, as shown in Fig. 18C. We don't need to know the exact amount of either the impedance of  $Z_A$  or the voltage of  $E$ , but we do want to know how we can couple from this circuit into the resonant circuit.

There are four simple ways to couple an antenna to a single resonant circuit:

1. Direct coupling.
2. Capacitive coupling.
3. Direct inductive coupling.
4. Inductive coupling.

Let us consider them one by one.

**1. Direct coupling**, as shown in Fig. 19A, is the first and simplest solution to this problem. The entire antenna voltage is applied across our  $L$ - $C$  circuit, making it a *parallel resonant circuit*. The resonant circuit then becomes the load, and forms a voltage divider with the antenna impedance  $Z_A$ , as shown in Fig. 19B.

As you have learned, this tuned circuit will act as a resistance of high ohmic value to a signal having the resonant frequency to which  $L$  and  $C$

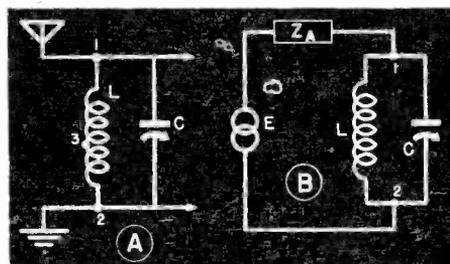


FIG. 19. Direct coupling.

are tuned. At this desired frequency, then, the resistance of the input circuit is much higher than the antenna impedance, and the greatest part of the signal voltage will exist across the resonant circuit and be passed on to the grid of the first tube. At all other frequencies, however, this resonant

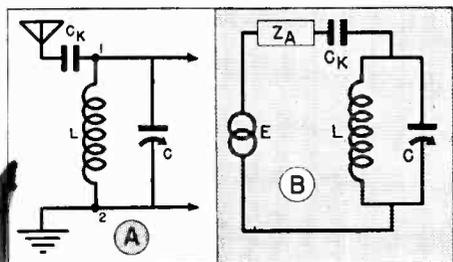


FIG. 20. Capacitive coupling.

circuit will act as a low impedance, and a far smaller voltage will be developed across it.

**2. Capacitive coupling**, as illustrated in Fig. 20A, is the second method. As you can see, it is simply direct coupling with a low-capacity condenser  $C_K$  inserted in the antenna lead-in.

Using the condenser gives an improvement over the direct coupling circuit shown in Fig. 19. In direct coupling, the selectivity is more sharp the higher  $Z_A$  becomes. The voltage division for off-resonant signals is not effective until the resonant circuit impedance has dropped to some value near (or below) the  $Z_A$  value. Therefore, when  $Z_A$  is small, the parallel resonant impedance must drop a great amount before off-resonant signals are eliminated. This means signals near the resonant frequency are not cut out, so the tuning is broad.

Inserting the condenser  $C_K$  as shown in Fig. 20A places its reactance in series with the antenna impedance as shown in Fig. 20B. Hence, it is added to the series impedance, increasing the

effectiveness of the circuit. Now the parallel impedance does not have to drop so far before its value compares with that of the series impedance, so the circuit rejects frequencies closer to the resonant value.

**3. Direct inductive coupling** of the form shown in Fig. 21A applies the antenna voltage to only a part of the resonant circuit. By having the tap 3 well up on the coil (near 1), the circuit still acts as a parallel resonant circuit, but only a part of its impedance will be between 2 and 3, and hence in the antenna-ground circuit. This can result in a better voltage-dividing action, and transformer step-up can be used to increase the voltage slightly. The position of point 3 can be chosen to secure maximum gain with fair selectivity or low gain with good selectivity, as desired.

**4. Inductive coupling**, shown in Fig. 22A, is perhaps the most widely used of all methods for coupling the antenna system to a resonant input circuit. (Inductive coupling is also known as transformer coupling or sometimes as indirect inductive coupling.) Antenna current flowing through

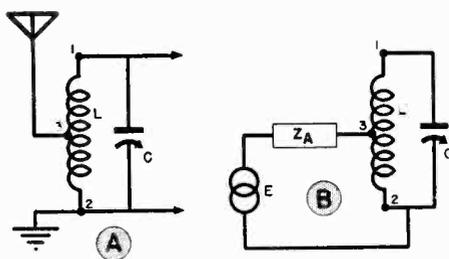


FIG. 21. Direct-inductive coupling.

coil  $L_1$  sets up a varying magnetic flux  $M$  which links coil  $L$  and induces in it a voltage which acts as if it were in series with  $L$ . Inductive coupling can be designed to give very high gain, to give high selectivity, or to give a satisfactory compromise between the two, as desired, by varying the num-

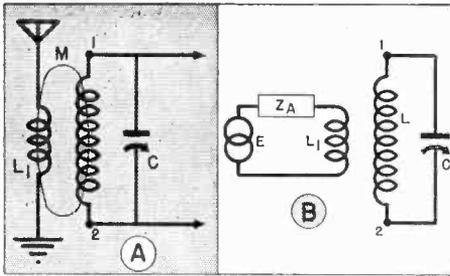


FIG. 22. Inductive coupling.

ber of turns in coil  $L_1$  and its position with respect to coil  $L_2$ .

► The coupling methods shown in Figs. 19 to 22 for antenna circuits are essentially the same as those you will encounter in a host of other radio applications, the most common of which is the coupling together of the plate circuit of one tube and the grid circuit of the following tube. We will study these circuits in detail in later lessons, when we take up complete radio stages.

### INDUCTIVE COUPLING FACTS

Any inductively coupled circuit can be simplified to the three essential parts shown in Fig. 23: a source, a coupling transformer, and a load. The source may or may not have appreciable impedance, and may be delivering either an a.f. or an r.f. signal voltage. The transformer may be of the air-core type, if its two windings (coils) are separated only by air or some other insulating material, or it may be of the iron-core type. The

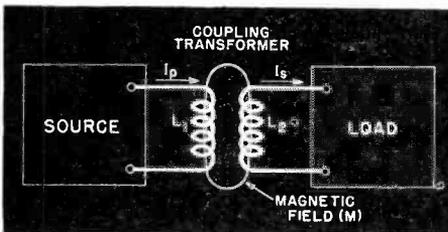


FIG. 23. Any inductively coupled circuit can be represented by this simplified circuit.

load likewise may have impedance. Either the source, the load, or both, may be resonant circuits. In an inductively coupled circuit of this sort, the source sends through the transformer primary (coil  $L_1$ ) a current  $I_P$  which induces in coil  $L_2$  a voltage that acts directly on the load, making current  $I_S$  flow through the load.

The current flow through the load means that the load is taking power from the coupling transformer, and further, this power must be given to the coupling transformer by the source. Hence, the source "feels" the load through the coupling, and the current flow  $I_P$  is automatically ad-

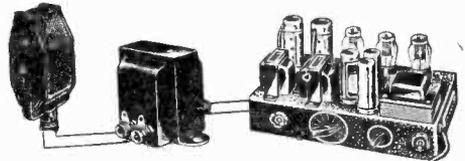


FIG. 24. An audio frequency example of inductive coupling.

justed so that the required power is delivered, provided the coupler is properly designed. You will go into this in more detail in later lessons.

**Practical A.F. Application.** A very common and practical low-frequency application of inductive coupling is that shown in Fig. 24, where a microphone serves as a source of a.f. signals, an audio amplifier serves as a load, and an iron-core coupling transformer serves to transfer to the source the effects of the load. This coupling transformer can be designed to deliver maximum voltage or maximum power to the load.

**Practical R.F. Application.** A radio frequency application of inductive coupling is the little "pill-box" you see connected between the two halves of some short-wave antennas in an arrangement similar to that shown in Fig. 25A. A close-up view of this box

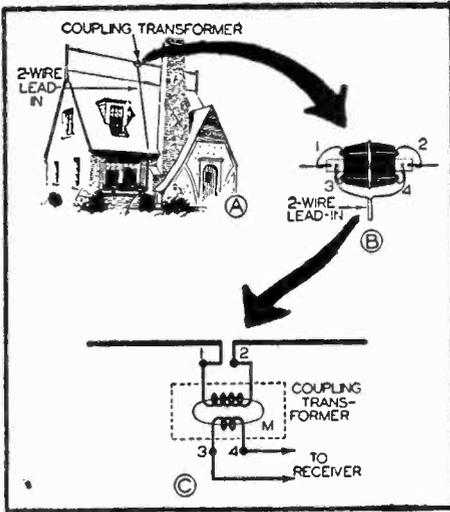


Fig. 25. An r.f. application of inductive coupling.

appears in Fig. 25B. Generally, all you will find inside it are two coils forming an air-core transformer, which is connected between the two halves of the antenna and the two-wire lead-in (transmission line) in the manner shown in Fig. 25C.

► We might get a little in advance of our subject here to explain why this transformer is used. You know that an antenna has an impedance at its connecting terminals; likewise, the input of a radio receiver has impedance. As later lessons will show, maximum *power* can be transferred from one circuit to another only when their impedances are equal—or, as radio men say, *matched*. The simplest way to match impedances is to couple the circuits with a transformer which has a certain calculated number of turns in its primary and secondary windings. Thus, the transformer in Fig. 25B is used as an impedance-matching transformer so that maximum signal power will be transferred from the antenna to the set.

**Types of Inductive Coupling.** The action of a coupling transformer, as

you already know, depends upon the fact that current flowing through the primary winding produces a varying magnetic flux. *When all of the magnetic flux produced by the primary winding links with (passes through) the entire secondary winding,* we have what is known as *unity coupling*. While unity coupling is just about impossible to secure in a practical transformer, a very close approach to it can be made; this is called *tight coupling*. On the other hand, when only a part of the magnetic flux links with the secondary winding, the coupling is said to be *loose* or *weak*, and the load will have less effect upon the source. If none of the magnetic flux links with the secondary winding, there is *zero coupling*. Let us consider how tight or weak coupling can be secured in actual coupling transformers.

**Tight Coupling.** When the two wires which form the primary and secondary windings of a coupling transformer are wound side by side on the coil form, as indicated in Fig. 26A, we have tight coupling; now practically all of the flux produced by the primary winding must link with or

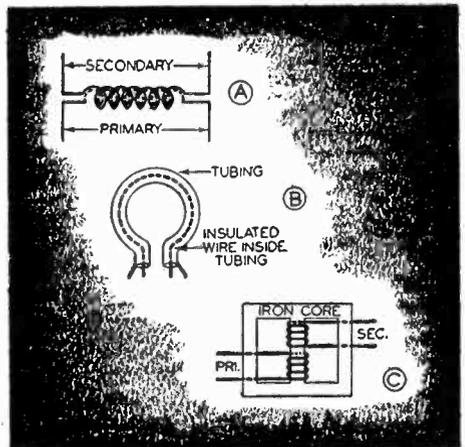


FIG. 26. Unity coupling can be secured as shown here.

pass through the entire secondary winding. In ultra-high-frequency apparatus, the same result often is secured by using one or more turns of copper tubing as the primary coil, and threading an insulated wire through this tubing to serve as the secondary coil. This is illustrated in Fig. 26B. When an iron core is used, as in Fig. 26C, tight coupling is secured without the necessity for having the turns of the two windings close together. In fact, the primary and secondary may be two separate coils placed one inside or alongside the other, as long as nearly all of the magnetic flux passes

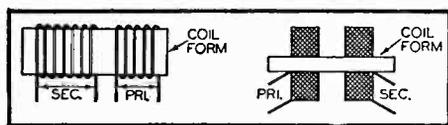


FIG. 27. Loose coupling.

through both coils. Iron cores are used in coupling transformers designed for audio amplifier circuits.

**Loose Coupling.** Where loose coupling is desired, as in some r.f. transformers, the two coils generally are arranged in one of the two ways shown in Fig. 27, so that there is considerable space between the coils. Only a small fraction of the magnetic flux thus links with the secondary coil.

► When two coils are loosely coupled to each other, we really have a condition where a small part of the primary coil is *tightly coupled* with a small part of the secondary coil, and the remaining parts of the primary and secondary coils are not coupled to anything. These uncoupled parts of

the primary and secondary coils can be considered as separate, extra coils in their respective circuits. Thus we can, if we wish, think of a loosely coupled circuit as being a tightly coupled circuit with an extra coil in its primary circuit and an extra coil in its secondary circuit. It is often very convenient to analyze a loosely coupled circuit this way. These extra inductances are known as the *primary leakage inductance* and the *secondary leakage inductance*, since they exist because some magnetic flux *leaks off* without linking the other coil.

As you learn more about radio apparatus, you will see how important is this left-over or leakage inductance in the actual operation of radio circuits which employ loose coupling. Also, you will discover that moving the original coils closer together increases the coupling, reduces the values of these leakage inductances, and affects the operation of the circuit in many other ways.

**Looking Forward.** Now that we have a general understanding of resistors, coils, condensers, vacuum tubes, and tuned circuits, we can proceed to connect all these into actual vacuum tube circuits and study their operation in connection with tubes. This we will do in our next lesson, in which you will be introduced to a number of typical circuits actually being used in radio receivers and other radio apparatus. You will probably be surprised, after you have completed your study of this next lesson, to see how much you have learned from the first ten lessons in your Course.

THE N. R. I. COURSE PREPARES YOU TO BECOME A  
**RADIOTRICIAN & TELETRICIAN**  
(REGISTERED U.S. PATENT OFFICE) (REGISTERED U.S. PATENT OFFICE)

# Lesson Questions

Be sure to number your Answer Sheet 9FR-3.

Place your Student Number on *every* Answer Sheet.

*Send in your set of answers for this lesson immediately after you finish them, as instructed in the Study Schedule. This will give you the greatest possible benefit from our speedy personal grading service.*

1. Does *reducing* the resistance of a series resonant circuit: 1, *increase*; or 2, *decrease* the selectivity of the circuit?
2. When a series resonant circuit is tuned to resonance, is the circuit current flow: 1, *at its highest value*; or 2, *at its lowest value*?
3. Suppose the capacity of C in a series resonant circuit is increased. Will the circuit now tune to: 1, the *same* frequency; 2, a *higher* frequency; or 3, a *lower* frequency?
4. At resonance, does a parallel resonant circuit act like: 1, *a coil*; 2, *a condenser*; 3, *a high resistance*; or 4, *a low resistance*?
5. When not tuned to resonance, does a parallel resonant circuit act like the part: 1, having the *higher* reactance; or 2, the part having the *lower* reactance?
6. To obtain the highest possible selectivity, would you use a coil having: 1, a high Q; 2, a low Q; or 3, a medium Q?
7. What will happen to the inductance of an air-core coil when a powdered-iron core is placed inside the coil?
8. When a voltage is *induced* in the coil of a resonant circuit, is the circuit: 1, *series* resonant; or 2, *parallel* resonant?
9. Name the four methods of coupling an antenna to a resonant circuit.
10. What condition is necessary for *unity* coupling?

## DETERMINATION

Did you ever watch a steamroller — an old-fashioned one with clanging gears, escaping steam and great clouds of black smoke belching from the stack? Wasn't it a thrill to see that great machine level out obstacles in its path, leaving a perfectly smooth roadway? The word "determination" always brings to my mind that picture of a steamroller — a machine which goes places once the "full steam ahead" lever is thrown.

So, too, are *you* determined to go places, to achieve success and happiness. One by one you are completing your lessons, studying hard and making sure you understand everything; step by step you are approaching that greatest of all goals—SUCCESS.

Of course, the way is long and not always easy. But whenever the going gets a little tougher than usual or you feel a bit discouraged, just bring out that old determination, and back it up with every single ounce of ambition you have.

Be a human steamroller, always confident of your own power, always moving ahead, always succeeding. Keep that determination of yours alive every single minute. I'm with you, ready to help you over the bumps at any time — *I'm determined to make you win.*

J. E. SMITH

**HOW TUBES WORK IN  
TYPICAL RADIO STAGES**

10FR-4

**NATIONAL RADIO INSTITUTE**

ESTABLISHED 1914

**WASHINGTON, D. C.**



# STUDY SCHEDULE NO. 10

For each study step, read the assigned pages first at your usual speed. Reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind, then answer the Lesson Questions for that step. Study each other step in this same way.

1. How Radio Systems are Divided into Stages . . . . . Pages 1-5

Now that you have learned the characteristics of radio parts, you are ready to combine them with tubes into radio stages. This section gives an introduction to the names and purposes of radio stages as they are found throughout a radio system. Use this "map" as you go along, to understand just how each kind of stage fits into the over-all picture. Answer Lesson Question 1.

2. More Facts About Tubes . . . . . Pages 5-13

Radio stages work because it is possible to arrange parts and voltages so that tubes will do certain things. The more you can learn about tubes, the easier you can understand stages. Answer Lesson Questions 2 and 3.

*Review!*  3. Basic Amplifying Stage . . . . . Pages 13-18

All tube amplifying stages can be reduced to a simple, basic stage—an input device or signal source, a load to which the signal is delivered, and a power supply. Here are the details of how this basic circuit works; how changes in voltage or load value affect the operation; how signals are kept in their correct paths by by-pass condensers; how a tube can produce its own bias. This is a VERY IMPORTANT section. Read it over and over, until you feel you know everything in it. Then review it from time to time. Answer Lesson Questions 4, 5 and 6.

4. Typical Amplifying Circuits . . . . . Pages 18-22

Now that you understand the basic stage, you are introduced to some of its variations. The major difference is in the type of coupling—that is, in the different ways of getting signals into and out of the stage.

5. Tubes as Signal Generators and as Signal Mixers . . Pages 23-28

Here is a brief introduction to radio stages which do other things in addition to amplifying signals. You are *not* expected to fully understand oscillators, modulated stages and frequency converters from this discussion, but you should learn what these stages do to the radio signal. Answer Lesson Question 7.

6. The Diode in Radio Systems . . . . . Pages 28-33

Power supplies and demodulator stages commonly use the two-element (diode) tube. This introduction to these stages shows how it is used. Answer Lesson Questions 8, 9 and 10.

7. A Complete Superheterodyne Receiver Circuit . . Pages 33-36

A schematic diagram is used to show how radio signals progress through a superheterodyne receiver. This review of the lesson shows the manner in which all kinds of stages are brought together in a receiver—how each stage performs a necessary operation so that the reproducer can deliver the intelligence we want.

8. Mail your Answers for this Lesson to N.R.I. for Grading.

9. Start Studying the Next Lesson.

# HOW TUBES WORK IN TYPICAL RADIO STAGES

## How Radio Systems are Divided into Stages

**T**HIS lesson is a milestone in your Course in Radio Fundamentals. Up to this time you have been learning the characteristics of *radio parts*. You will now begin to learn how these parts are combined into groups in which tubes are used—groups that are called *radio stages*.

**In this lesson, we are going to do two things: 1, give you a preview of the radio stages you will study in detail in the next few lessons; and 2, give you a basic understanding of the requirements underlying ALL stages.** Once you understand *stages*, it will be easy for you to combine them into *sections* and thus form a complete receiver or transmitter.

► To help you see the relationship between stages, we will first describe a complete radio system. Pay particular attention to the *names* and *functions* of the stages—and refer to this “radio map” from time to time so that you can keep these stages properly in mind. (Later in this lesson, we will give a description of a complete superheterodyne receiver in somewhat more detail.)

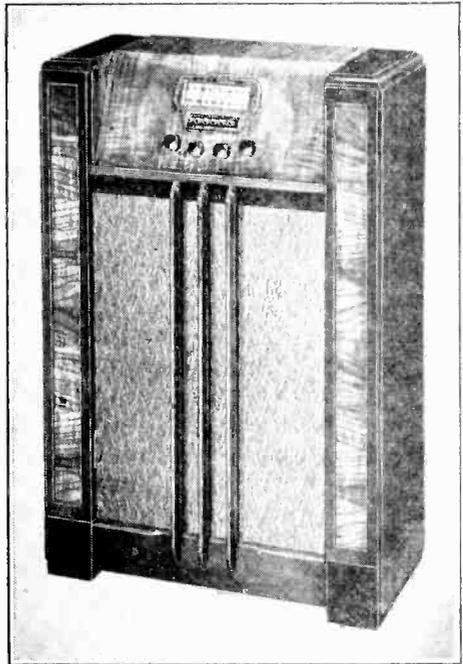
Naturally, we will not try to give *every detail* of the operation of radio stages in this one lesson. A preview of this kind is intended to give you enough basic facts to help you understand the full details to be found in later lessons.

### RADIO SYSTEMS

All radio systems have one object—to convey intelligence from one point to another. The intelligence can be

anything—a sound or sight (television) program; commercial (voice or code) messages going from point to point; a guide-path for aircraft or ships; or messages intended for police or fire departments.

The original intelligence cannot be carried to distant points without some form of conversion or without the use of a carrier. Even the loudest sounds cannot be heard more than a few



Courtesy Galvin Mfg. Corp.

This Motorola is typical of the console type radio receivers you are likely to meet if you become a serviceman. What is inside the cabinet? A superheterodyne receiver using the basic stages you are studying in this lesson.

miles and close-up details of a scene are visible for only a few feet. We can record sounds or take photographs, but these are not instantaneous methods of transmission. It is also true that by converting sounds or scenes into electrical impulses it is possible to carry them over wire lines. However, at best, wire circuits have limits, so radio steps in and provides a means of sending intelligence instantly to any desired receiving point.

Radio transmission is made practical by the fact that high-frequency signals will travel through space, and it is possible to make these signals "carry" the intelligence. At the receiving point, a *copy* of the original intelligence signal can be obtained from the "carrier" signal and used to operate a reproducer system. Fig. 1 shows just such a system in block-diagram form. Let's review the basic actions in each stage.

## THE TRANSMITTER

**The Carrier Signal.** At the transmitter, the "carrier" signal is generated by a vacuum tube stage known as an *oscillator* (indicated at *A* in Fig. 1). The carrier is called an r.f. (radio frequency) signal since it is in the r.f. range. At each station, the frequency of this signal is set at the value assigned to that particular station. An American broadcast station using the amplitude modulation (a.m.) system will be assigned a frequency within the 550 kc.-to-1600 kc. broadcast band, while an f.m. (frequency modulation) broadcast station will be assigned a carrier frequency somewhere within the 88- to 106-mc. range.

Amplified copies of this r.f. signal are obtained from a series of vacuum tube amplifying stages (in section *B* of Fig. 1) which deliver the amount of power needed to make the proper com-

bination with the intelligence signal.

**The Intelligence Signal.** The intelligence (sound waves or visible scenes) is converted into a corresponding electrical signal by the pickup device at *D*. Amplified copies of this electrical signal are obtained from the amplifiers at *E*.

**Modulation.** At *C*, the carrier and the intelligence signals are combined. This process of combining is called *modulation*, because the intelligence signal *modulates* or varies the carrier signal. At present, two systems of modulation are in use: *amplitude modulation* (a.m.) and *frequency modulation* (f.m.). In amplitude modulation, the *amplitude* of the carrier signal is varied in proportion to the amplitude of the intelligence signal; in frequency modulation, the *frequency* of the carrier signal is varied in proportion to the amplitude of the intelligence signal. In either system, the rate at which variations in the carrier signal occur corresponds to the frequency of the intelligence signal.

► The modulated carrier signal may be, and frequently is, amplified before being applied to the transmitting antenna or radiator *G*. In later lessons, we'll go into the subject of antennas and find why they radiate—here, all we'll say is that the signal is radiated into space in the form of a varying electromagnetic field which is known as a radio wave.

## THE RECEIVER

When the radio wave encounters receiving antenna *H*, it induces a voltage in the antenna. This voltage causes a varying current flow through the lead-in; this, in turn, induces a voltage in the r.f. preselector *I*. All radio waves striking the antenna—whether they come from the desired station or from other stations—induce voltages in the

r.f. preselector, so we use "tuned" or resonant circuits (coil-condenser combinations) in the preselector section to start the process of selecting the desired signal from all the others received. (Here you see the reason for assigning different frequencies to transmitters; only by frequency separation can we select the signals of the desired stations from among all others broadcasting at the same time.)

The preselector usually must tune over a wide range of frequencies, since, of course, we generally like to tune to different stations. Now, it is difficult to obtain the same amount of amplification and selectivity at different frequencies over wide ranges. It is far easier to obtain high gain and good selectivity from an amplifier operating at a *fixed* frequency, especially if this *fixed* radio frequency is in the lower r.f. range. These facts have led to the creation of the *superheterodyne* receiver, in which incoming signals, regardless of frequency, are converted into a fixed lower frequency for amplification.

**The Superheterodyne.** *Briefly the fundamental principle of the superheterodyne circuit is: The incoming signal is combined with a "local" r.f. signal so that a lower frequency r.f. signal is produced.*

Now, let's see what stages are used in a superheterodyne receiver to perform the action we have just described. In Fig. 1, preselector *I* is used to give an initial separation of the signals present in the receiving antenna *H*. The preselector is not required to give much amplification nor much selectivity; these are obtained after the frequency mixing has occurred. The mixer-first detector stage *J* is fed the modulated r.f. signal from the preselector and is also fed another (unmodulated) r.f. signal from the *local r.f. oscillator stage K*. (This is called a "local"

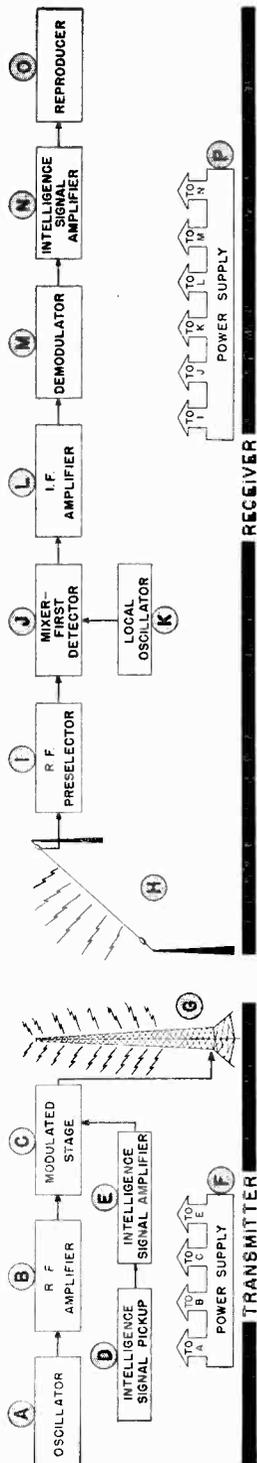


FIG. 1. A complete radio system. After the sound or picture signal is combined with the r.f. carrier at the transmitter, the combined signal is radiated through space by the antenna system G. At the receiving location, the radio wave is picked up by receiving antenna H. The superheterodyne receiver selects the desired signal, amplifies and demodulates it, then amplifies the intelligence signal further so as to operate the reproducer. This block diagram applies to both a.m. and f.m. sound systems as well as to television. The major differences are in the r.f. and intelligence frequencies handled, and in the types of modulation and demodulation stages. Hence, this diagram can be applied to ALL radio systems.

oscillator to distinguish it from the oscillator in the transmitter.)

The result of the frequency combination (called "frequency conversion") is a new carrier which has a frequency known as the *intermediate frequency* (abbreviated i.f.) and which is modulated by the intelligence signal. The i.f. is usually equal to the difference between the carrier frequency and the local r.f.\*

► The preselector and the local oscillator are tuned simultaneously by a single control which rotates their tuning condensers or varies their coil inductances in step. Thus, as we tune to a particular station, the local oscillator is automatically adjusted to the frequency necessary to produce the desired combination.

► The tuned circuits at the input of the i.f. amplifier ( $L$  in Fig. 1) tend to select the desired i.f. carrier (carrying the intelligence signal) from among all others offered by the mixer-first detector. Each stage in the i.f. amplifier makes an amplified copy of the desired signal, and the resonant circuits in each stage serve to further select the desired signal from among others.

When the modulated i.f. carrier signal has been amplified to the limits of the i.f. amplifier, it is fed into a *demodulator* stage. (This stage is another *detector*, and is called the *second detector* to distinguish it from the *mixer-first* detector.)

**Demodulation.** The demodulator  $M$  serves a very important purpose; it is here that the intelligence signal is

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\*Here is an example: If the local oscillator of a receiver generates a 1460-kc. signal when the r.f. section of the receiver is tuned to a 1000-kc. station, the i.f. will be 460 kc. ( $1460 - 1000 = 460$  kc.). This is called the i.f. value because it is normally *intermediate* in value between the r.f. carrier frequency and the frequencies of the intelligence signal.

separated from the carrier. The carrier has been necessary to convey the intelligence signal through space. The intelligence signal must remain with the carrier until the desired carrier can be selected from among all others. Then, when the carrier has served its purpose, it is cast aside and a reasonably accurate duplicate of the original electrical intelligence signal is obtained.

► The intelligence signal is similar to that existing at  $D$  in the transmitter, and usually must be amplified further before it can operate the reproducer. The intelligence signal amplifier  $N$  is called an *audio* amplifier if the signal corresponds to sound, or is called a *video amplifier* if the signal corresponds to a television picture scene.

**Reproduction.** After the intelligence signal has been sufficiently amplified, it must pass through a *reproducer*  $O$  which will convert the electrical signal into the original form of the intelligence. For audio signals (voice or music), a *loudspeaker* is used. This device produces sound waves in accordance with the variations in electrical current. For television, an image reconstructor tube "paints" the television scene on a screen at one end of the tube. Code signals may operate a sounder or an automatic typewriter. Facsimile signals may operate a device which chemically reproduces a picture on special paper. And so it goes—whatever the original signal, a suitable device restores the original sound, scene, or impulse into a recognizable or usable form.

► This completes the journey of our intelligence signal. It has travelled through many radio stages at both the transmitter and the receiver. *Most of the stages made amplified copies, which in turn were amplified by suc-*

ceeding stages. Some stages did nothing but amplify, others performed specialized duties, while still others did both. This has been just a bare outline of the process of radio transmission and reception—we have not tried to show the *number* of stages in each section nor have we shown all that may be done in some of the more elaborate systems. Instead, we have given you a brief over-all picture to make it easier for you to see the relationships between the *stages* you are about to study.

**Importance of Tubes.** One important fact stands out about these stages: whatever their function, *vac-*

*uum tubes are used in all of them,* except possibly the intelligence pickup *D* and the reproducer *O*. (In a television system, special tubes are used at *D* and *O* too.)

Because of this widespread use of tubes, you must have a thorough understanding of fundamental tube operation before you study stages. We will now expand some of the information presented in your earlier lesson on tubes, thereby giving you a more complete picture of tube action under different operating conditions. Then we will be ready to study the operation of each *basic* radio stage.

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## More Facts About Tubes

There are many kinds of tubes—some designed for specific purposes and some for general purposes. Basically, they all operate over rather similar characteristic curves; the particular actions obtained depend upon the *operating* point chosen and upon the devices used as “loads.” Although tubes with two, three, and four grids are better than triode (single grid) tubes for certain purposes, we will consider only diode and triode tubes in this lesson. This will let you concentrate on the important features of the stages described without becoming confused by extra grids. (You will take up multi-grid tubes as you study each stage in detail in later lessons.)

### HOW THE GRID CONTROLS THE PLATE CURRENT

From your earlier studies you know that, in a triode tube with a fixed plate voltage, the grid voltage determines the amount of plate current which will

flow at any particular instant. When no signal voltage is applied to the grid, the bias voltage (a d.c. voltage placed between the grid and the cathode) determines the amount of plate current flow; the plate current which flows under these conditions is often called the “operating” current. (This value is the initial or starting plate current which is then varied by the signal.)

In most receiver circuits the grid is not allowed to become positive with respect to the cathode, because this would cause an undesirable flow of grid current. (In transmitter r.f. power amplifier circuits and in oscillator circuits, the grid is allowed to go positive, as we shall see later.) Hence, we usually have a *negative* grid bias. The amount of bias depends on the point at which we desire to operate on the characteristic curve.

A typical characteristic curve is shown in Fig. 2. For a tube which has this characteristic, a negative bias of  $-3$  volts places the operating point at

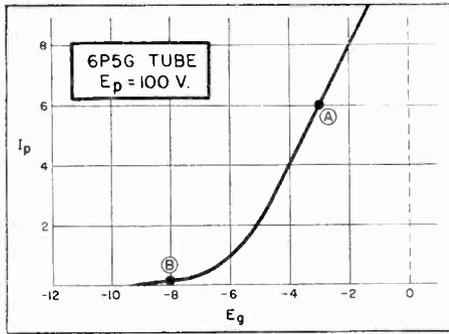


FIG. 2. The characteristic curve for a 6P5G tube. Although characteristic curves are similar in shape, the values apply to only one tube type and to one particular supply voltage value.

A (on the straight portion of the curve). A bias of  $-8$  volts causes the operating point to move to B, while a bias of more than  $-9$  volts will "cut off" the plate current. If a different plate voltage is used, then the curve will "move" so that other bias values will be required for these conditions. For this reason, curves must be labeled also as to what plate voltage is being applied.

► Naturally, tubes differ in their plate currents and bias requirements, although the general shape of their characteristic curves may be very similar.

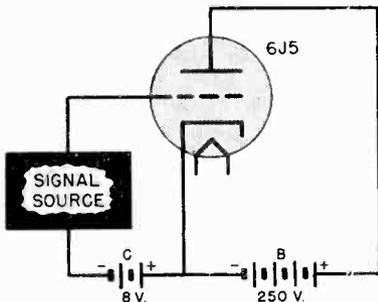


FIG. 3. Typical bias and plate voltage connections are shown here. However, the filament supply is NOT shown. Naturally it IS connected to the filament terminals and must be of the proper value to produce temperature saturation. Its omission on diagrams is purely to simplify the circuit so that you can concentrate on the actions being described.

Another tube may require a bias of  $-50$  volts to operate at point A, and may then pass a current of 30 ma. instead of the 6-ma. value shown in Fig. 2. Thus, the same curve may fit both tubes, except that the values of current and voltage differ. Therefore, characteristic curves must be labeled also for the particular tube to which they apply.

► Now, with our operating point determined by some suitable bias, let's see what happens when we apply a sine-wave signal to the circuit shown in Fig. 3.

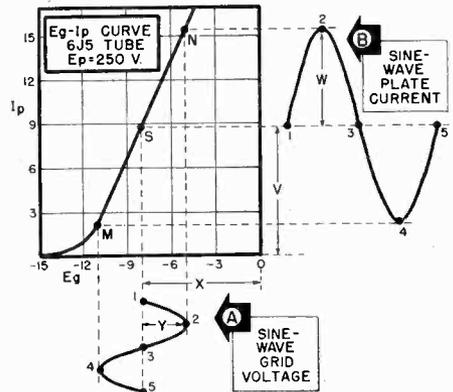


FIG. 4. The  $E_g-I_p$  curve for a 6J5 tube, with a plate voltage of 250 volts. It is assumed that the proper filament voltage is applied to the tube.

Fig. 4 is an  $E_g-I_p$  (grid voltage-plate current) curve for the triode tube used in Fig. 3. With the plate and bias voltages shown in Fig. 3 (which are typical for this tube), the operating point is at S, and the operating plate current value is nearly 9 ma. We know that the applied signal makes the grid voltage alternately more and less negative than the grid bias, so when the signal shown at A (in Fig. 4) is applied, the grid voltage will be made up of the bias value  $x$  plus or minus the value of the signal voltage  $y$  at that instant. Since the bias voltage is greater than the

largest value of the signal voltage, the grid remains negative at all times and there is no grid current flow. The grid is purely a *potential-operated device*—its *voltage* at any instant determines the plate current value.

Let's see how a change in the grid voltage affects the plate current. Suppose the grid voltage changes from point 1 to point 2 on curve A (making the grid less negative by 3 volts). Tracing vertically from each of these points up to the  $E_g-I_p$  curve, then horizontally over to curve B, we find that this grid voltage change causes a plate current change from point 1 to point 2 on curve B. The plate current changes from its operating value of 9 ma. (point 1) to a value of about 15.5 ma. (point 2)—an increase of about 6.5 ma.

Similarly, changes in grid voltage from point 2 to point 3, from point 3 to point 4, and from point 4 to point 5 on curve A cause plate current changes as shown by similarly-numbered points on curve B. Each grid voltage swing in a *less* negative direction causes an *increase* in plate current, while each swing in a *more* negative direction causes a *decrease* in plate current.

► Another important fact is that the plate current is a *pulsating d.c.*; that is, it is made up of a d.c. component  $v$  (equal to the no-signal operating current) and a sine-wave a.c. component  $w$  which causes the total plate current to vary above and below its no-signal operating value. (The plate current cannot reverse its *direction* of flow; hence, the a.c. variation can exist only as a *component* or part of the pulsating plate current. When we speak of an a.c. plate current, we mean *only* this varying component, not the total plate current.)

► The signal actually applied to an amplifier rarely has a sine-wave shape; it usually is more complex. However,

a complex signal is made up of many different sine-wave voltages in combination, so the tube response to a sine-wave signal corresponds to its response to that component of a complex signal. This makes it possible to use sine-wave signals when studying radio circuits. However, as Fig. 5 shows, if we know the exact wave shape of a complex signal, we can prove that the plate current has the same form as the grid voltage, provided that the  $E_g-I_p$  curve is linear (straight) over the operating region, as is the section  $M-N$  of Figs. 4 and 5.

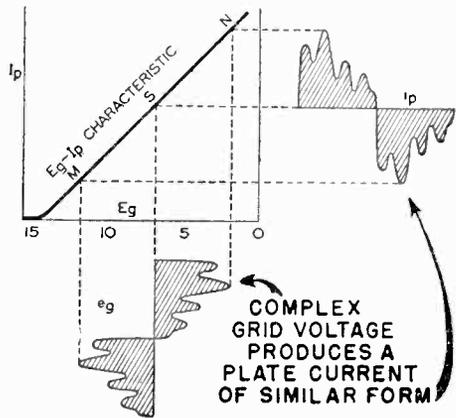


FIG. 5. The signal currents handled by most radio and television tubes have irregular wave forms much like that shown here for  $e_g$ . It can be proven by mathematics, however, that any wave, no matter how complicated, can be broken up into a number of simple sine waves. We can simplify our work by considering only sine-wave signals, for we know that radio circuits will behave the same for a particular sine wave as they do for this component of a complex wave.

## CLASSES OF AMPLIFIERS

Not all amplifying stages operate on the straight portion of their characteristic curves; sometimes an exact "carbon copy" of the grid voltage is not wanted. This means that amplifiers must be classified according to the part of the characteristic over which they operate.

► When an exact duplicate of the *en-*

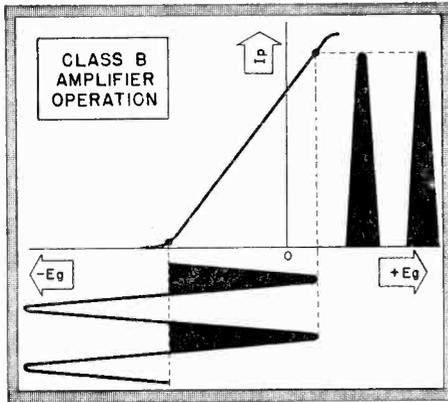


FIG. 6. In class B amplifiers, only half of the incoming signal is effective in producing plate current pulses.

entire input signal is wanted, the operating point is chosen near the center of the straight portion between the zero grid-voltage value and the lower bend or curved portion of the characteristic curve, and operation is confined to this straight portion. This is called class A amplification. Class A amplification is relatively distortionless (the output is practically proportional to the input at all times) provided: 1, that the signal voltage is not so large that some curved portion of the characteristic is encountered; and 2, that the grid never goes positive.

► For some purposes, we do not want the plate current variations to be exactly like the entire signal applied to the grid circuit. We may, for example, apply a bias sufficient to cut off plate current, as shown in Fig. 6. Now when we apply a signal, we will get half-wave pulses in the plate circuit—in other words, very little (if any) plate current will flow until the positive signal alternations overcome the bias. This is called class B operation and is very necessary for certain purposes.

► Going a step further, we may even apply a bias many times that required for plate current cut-off, as shown in

Fig. 7. Only the positive tips of the grid signal can cause plate current to flow. This is called class C operation.

► If we allow the grid signal to swing over into the positive region (Figs. 6 and 7), class B and class C amplifiers become far more efficient than class A (that is, they produce more output for the same plate supply power). However, the resulting distortion generally limits such operation to special stages where the distortion can be overcome or does not matter. We shall explain these uses in later lessons.

### FAMILIES OF CURVES

In practical radio work, tables of tube characteristics furnish practically all the needed information, so curves are rarely used. However, characteristic curves are extremely valuable to you now as a student in helping you to understand just how tubes work—so let's take a few minutes to see how these curves are plotted and what they reveal.

**Plotting Curves.** To make curves, the tube manufacturer uses a circuit similar to that shown in Fig. 8. This circuit is arranged so that any desired C voltage can be applied, the plate voltage is variable, and the plate current can be measured. Normal fila-

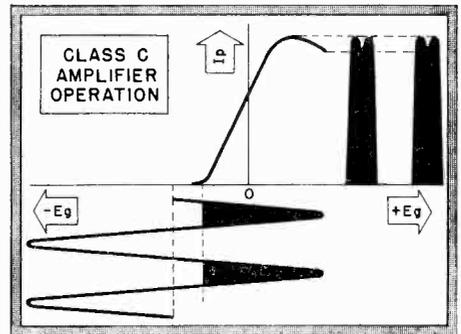


FIG. 7. In class C operation, even less than half of the incoming signal is used to produce plate current pulses.

ment voltage is applied.

To make  $E_g-I_p$  curves, the first step is to choose some value of plate voltage, by adjusting tap  $P_2$  on the B battery. Then the grid voltage is varied in steps from zero to a maximum negative value by moving the slider from terminal 1 toward terminal 2 on the potentiometer  $P_1$  in the grid circuit. The plate current is measured and plotted for each particular grid voltage value. The resulting curve is the  $E_g-I_p$  characteristic curve of the tube for that particular plate voltage.

This process is repeated for other values of plate voltage until a series or family of  $E_g-I_p$  characteristic curves is secured. They are similar to those shown in Fig. 9 (which are for a 6C5 tube).

► Although a family of curves can be made to tell almost everything about the behavior of a tube, radio men compress some of the most important information in characteristic curves into quickly usable forms. Ratings like horsepower, maximum speed, and gasoline consumption tell much about the performance of an automobile; tube performance is likewise expressed by the following ratings:

1. Amplification factor.
2. A.C. plate resistance.
3. Mutual conductance.

You've already met the first two of these in an earlier lesson. We'll review them briefly, then discuss the third rating (which is really a com-

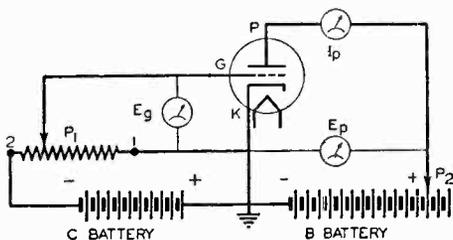


FIG. 8. A laboratory circuit used to determine the characteristics of tubes.

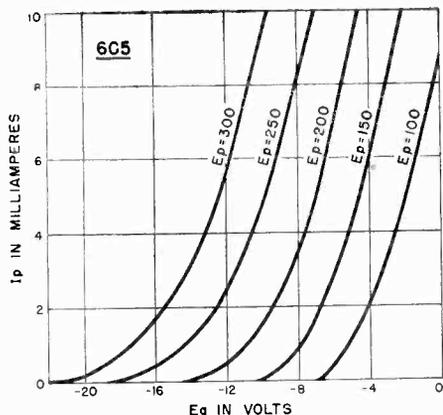


FIG. 9. A family of characteristic curves.

bination of the other two).

### AMPLIFICATION FACTOR

This factor is important because it tells us how good a tube is as a voltage amplifier.

Suppose we use Fig. 10, showing two curves taken from Fig. 9, to illustrate this. If the 6C5 tube is operating at point  $a$  on curve  $B$  (for a plate voltage of 200 volts), then a plate current increase of 4 ma. can be obtained by changing the grid voltage 2.5 volts (from  $-8.5$  to  $-6$  volts); this moves the operating point to  $c$ . Similarly, with the bias voltage fixed at  $-8.5$  volts, a 4-ma. change can be obtained by increasing the plate voltage 50 volts, from  $a$  on curve  $B$  to  $b$  on curve  $A$ . (Curve  $A$  is for 250 volts, which is 50 volts higher than curve  $B$ .) We can thus get a certain change in plate current either by leaving the plate voltage fixed and changing the bias, or by leaving the bias fixed and changing the plate voltage. However, notice that the plate voltage change is far larger than the grid voltage change needed to give the same plate current change.

If we now divide 50 (plate voltage change) by 2.5 (grid voltage change) we get 20. This means that the grid is *twenty times as effective as the plate*

(for this particular tube) in controlling the plate current. This value is called the *amplification factor*, and in plain language it means that, for this tube, a 1-volt grid change will affect the plate current just as much as will a 20-volt plate-voltage change. In other words, a 1-volt signal on the grid can act like a 20-volt variation in the plate circuit of this tube.

► The amplification factor *does not* tell directly how much the grid controls plate current, but it does express the relative effects of the grid and plate voltages on the plate current. The amplification factor can be defined technically as follows:

**Amplification factor is equal to the plate voltage change which will produce a certain plate current change, divided by the grid voltage change which will produce the same plate current change.**

Amplification factor, also called amplification ability, amplification number, or amplification constant, is generally designated by the Greek letter mu ( $\mu$ ), which is pronounced "mew." Amplification factor depends mostly upon the construction of the tube—particularly upon the position of the grid between the cathode and plate and upon the spacing between the meshes or turns of the grid wire. The closer the grid is to the cathode and the closer together are the grid wires, the larger will be the  $\mu$  of the tube, because the grid then will have a greater control over the electron movements.

► The values of  $\mu$  given in tube characteristics charts are only *averages*, as tubes of the same type may vary as much as 20% from the rated value. This deviation applies to other tube ratings as well, for radio tubes are delicate devices, made so compactly that errors of a few thousandths of an inch in the position of an electrode will greatly affect the tube ratings. These

differences in tube characteristics are permitted because close similarity is generally not necessary.

### A.C. PLATE RESISTANCE

The opposition which the plate-cathode path of a tube offers to the flow of a.c. is expressed in ohms, just as is any other resistance, and is commonly abbreviated as  $r_p$ .

The value of  $r_p$  for a tube is found by applying to the plate-cathode an

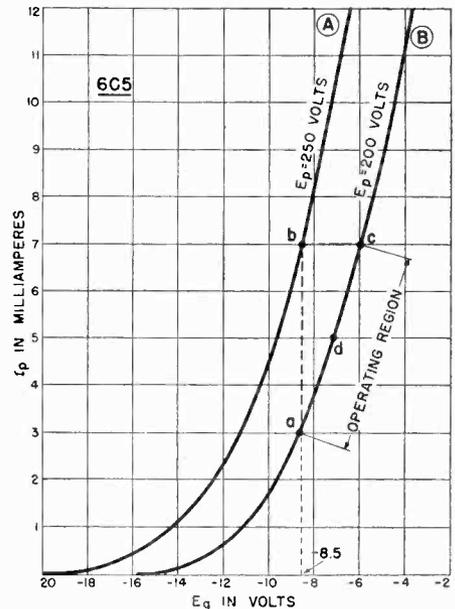


FIG. 10. Two of the curves taken from the family in Fig. 9.

a.c. voltage and measuring the resulting change in plate current, or it can be found from the characteristic curves. For example, referring to Fig. 10, a plate voltage change of 50 volts (*a* to *b*) produces a plate current change of 4 ma. when the tube is biased at  $-8.5$  volts. As we can consider the *change* to be equivalent to an a.c. variation, we need only use Ohm's Law to find the a.c. resistance. In this case, then,  $r_p$  is equal to  $50 \div .004$ , or 12,500 ohms. Notice—this value applies only

to the particular operating point chosen (—8.5 volts bias). Other operating voltages would result in a different a.c. plate resistance.

**D.C. Plate Resistance.** A radio tube is peculiar in that it offers a different plate-to-cathode resistance to a steady voltage than it does to a varying voltage. When only d.c. voltage is applied to the grid and the plate, a definite amount of d.c. plate current will flow. By dividing the d.c. plate voltage by the d.c. plate current (Ohm's Law), we get the d.c. plate resistance of the tube for that particular grid bias value. Thus, at point *a* in Fig. 10, the d.c. plate voltage is 200 volts and the d.c. current is 3 ma., so the d.c. resistance is  $200 \div .003$ , which is about 66,000 ohms. (This is for the chosen value of grid bias only—since changing the grid bias changes the plate current and therefore changes the d.c. plate resistance, the bias must always be specified when a value of d.c. plate resistance is given.) This is quite different from the a.c. resistance.

► Strictly speaking, the designations "a.c." or "d.c." should always be given when the name "plate resistance" is used. However, for amplifiers, the *a.c. behavior* of a tube is the most important, for it tells us what the tube will do when signals are applied; d.c. resistance is seldom of interest. *Whenever you see the plate resistance of a tube mentioned, you can assume the a.c. value is meant, unless it is definitely marked "d.c."*

## MUTUAL CONDUCTANCE

The third important tube rating, called mutual conductance or transconductance, is equal to the amplification factor divided by the a.c. plate resistance. This rating tells directly how much the grid voltage controls the plate current; it is defined as follows:

**Mutual conductance of a tube is equal to the a.c. plate current divided by the a.c. grid voltage, when there is no load in the plate circuit.**

Mutual conductance, abbreviated as  $g_m$  (or  $s_m$ ), is expressed in *mhos* (pronounced "mose"). If we divide the a.c. plate current (the *varying* portion of the plate current) in amperes by the a.c. grid voltage in volts, we obtain mhos. However, the mutual conductance of tubes is but a small fraction of a mho, so, to avoid an awkward decimal number, the *micromho*, equal to one-millionth of a mho, is used in radio work. (Incidentally, "mho" is "ohm" spelled backwards.)

Mutual conductance (sometimes known as transconductance) shows *how much* a.c. plate current will be produced by an a.c. signal in the grid circuit. If the grid voltage in volts and the mutual conductance in mhos are known for a tube circuit, such as an amplifier tube circuit whose load has negligible resistance, the a.c. plate current in amperes is obtained simply by multiplying the two values together.

► When a tube ages, its electron emission is lowered and its  $g_m$  is reduced. This is why  $g_m$  is often measured when testing radio tubes, because it indicates the worth of the tube.

► Values of mutual conductance are given on tube charts. They can also be obtained from the  $E_g-I_p$  curves. In Fig. 10, a grid voltage change of 2.5 volts (from *a* to *c*) produces a plate current change of .004 ampere. Since a change corresponds to an a.c. value, the mutual conductance of the 6C5 tube (for operating point *d* in the middle of the operating region of curve *B*) will be  $.004 \div 2.5$  or .0016 mhos. This is the same as 1600 micromhos.

► An example of a tube chart is given in Fig. 11. Note that  $r_p$ ,  $g_m$ , and  $\mu$  vary greatly among the tubes listed.

## EQUIVALENT TUBE CIRCUITS

The tube ratings we have discussed so far are those of the tube alone, with no load in the plate circuit. These ratings are useful, but they do not tell directly how much amplification can be obtained in a practical stage, where a load must be used. When we wish to find the *actual stage gain*, it is far simpler to consider an equivalent circuit (such as was shown in an earlier lesson). This circuit deals only with the a.c. voltages and the a.c. currents in the plate circuit of the tube.

### Equivalent A.C. Plate Voltage.

First, let us explain what is meant by *equivalent a.c. plate voltage*. You know

a.c. grid voltage in its *effects*.

We use the tube amplification factor to find this equivalent voltage. Since this factor shows how many times more effective is a change in grid voltage than is a change in plate voltage, the plate voltage change is equal to the grid voltage change multiplied by the  $\mu$  of the tube. Hence, we say that the a.c. grid voltage  $e_g$  multiplied by the  $\mu$  of the tube is the *equivalent a.c. plate voltage*  $\mu e_g$ . This equivalent a.c. plate voltage is used in calculations just as if it were an actual voltage; in fact, most engineers speak of it as the a.c. plate voltage.

**The Circuit Without a Load.** Now, let us draw the equivalent circuit of an

TYPE	CLASS	BASE	FILAMENT RATING		USE	PLATE VOLTS	NEGATIVE GRID VOLTS	SCREEN VOLTS	PLATE CURRENT MA.	SCREEN CURRENT MA.	$r_p$ PLATE RESISTANCE OHMS	$\mu_m$ MICROMHOS MUTUAL CONDUCTANCE	$\mu$ AMPLIFICATION FACTOR	OHMS LOAD FOR STATED POWER OUTPUT	UNDISTORTED POWER OUTPUT MILLI-WATTS
			VOLTS	AMPS											
6K5GT/G	TRIODE	5-U	6.3	0.30	AMPLIFIER	100 250	1.5 3.0	---	0.35 1.10	---	78,000 50,000	900 1400	70 70	---	---
6K6GT/G	PENTODE	7-S	6.3	0.40	POWER AMP.	100 250 315	7.0 18.0 21.0	100 250 250	9.0 32.0 25.5	1.6 5.5 4.0	104,000 68,000 75,000	1,500 2,300 2,100	---	12,000 7,600 9,000	350 3,400 4,500
6K7GT/G	PENTODE	7-R	6.3	0.30	AMPLIFIER	90 250	3.0 3.0	900 100	5.4 7.0	1.3 1.7	300,000* 800,000	1,275 1,450	400 1,160	---	---
6L5G	TRIODE	6-D	6.3	0.15	AMPLIFIER	100 250	3.0 9.0	---	4.0 8.0	---	10,000 9,000	1,500 1,900	15 17	---	---
6L6	BEAM AMP.	7-AC	6.3	0.90	POWER AMP.	250 350	14.0 18.0	250 250	72.0 54.0	9.0 2.9	22,500 33,000	6,000 9,200	---	2,500 4,200	6,800 10,800
6Q7	DUODIODE TRIODE	7-V	6.3	0.30	DET. AMP.	100 250	1.5 3.0	---	0.35 1.1	---	88,000 58,000	800 1,200	70 70	---	---

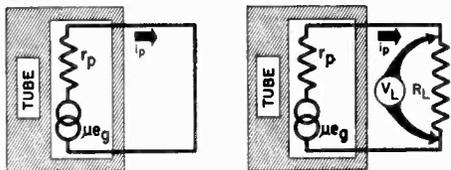
FIG. 11. A section of a typical tube chart. Reading from left to right we find that the columns give: 1, the type number on the tube, assigned according to the RMA code; 2, the class of tube, whether a triode, pentode, etc.; 3, the base connections are given in another section of the chart (not shown here) but the key letter and number in column 3 refer to similarly-keyed base views; 4 and 5, the filament voltage and current; 6, the use to which the tube can be put; 7 through 11, the voltages and currents of the various plate, screen grid and control grid electrodes; 12, 13 and 14, the plate grid resistance, mutual conductance and amplification factor; 15, the proper load value; 16, the power output. This chart only shows the characteristics of a few of the many hundreds of different tubes used in modern radio receivers. Notice how even these tubes vary widely in characteristics.

that an a.c. grid voltage, or signal, acting in one circuit of a tube (the grid circuit) produces an a.c. plate current in another circuit (the plate circuit). Thus, we are dealing with two different circuits connected only by the action of a tube. It is always simpler to study a single circuit at a time. For this reason, we transfer the *effects* of the a.c. grid voltage to the plate circuit; we forget the grid circuit and assume that there is only a plate circuit containing an a.c. plate voltage *equivalent* to the

amplifier having no plate load. Considering only the a.c. actions, we find that the only opposition to the flow of current in the plate circuit is the a.c. plate resistance  $r_p$ , and that the only a.c. voltage acting in the circuit is the equivalent a.c. plate voltage  $\mu e_g$ .

Therefore, we can consider the a.c. plate circuit to be as shown in Fig. 12A—an a.c. source and a resistance  $r_p$ . The a.c. plate current flows through  $r_p$  and all the equivalent a.c. voltage is dropped in this internal tube resistance.

### EQUIVALENT TUBE CIRCUITS



A: NO LOAD      B: WITH LOAD

FIG. 12. Equivalents of plate circuits with and without a load.

**Equivalent Tube Circuit With Load.** The circuit without a load is useless, as all the signal energy is lost in the tube plate resistance. Hence, in a practical amplifier, we have to use a load in the plate circuit so that some of the signal energy can be developed outside the tube where it can be put to use. This load  $R_L$  is in series with the resistance  $r_p$ , as shown in the equivalent circuit of Fig. 12B. The insertion of this load will not change the operation of the tube, provided the same d.c. operating voltages are still applied to the tube. (If the load is a resistor and we want to operate the tube at a definite plate-to-cathode d.c. voltage, then the supply must furnish this plate voltage

plus the drop which occurs because of the d.c. plate current flow through the load resistor. The values of  $\mu$  and  $r_p$  will then be those shown in the tube characteristics chart.)

With a load in the plate circuit, the equivalent a.c. plate voltage is divided between  $r_p$  and  $R_L$ .\* That portion of the a.c. plate voltage which is developed across  $R_L$  is the useful portion of the signal voltage and can be applied to other circuits.

\*For example, let's suppose  $e_g$  is 2 volts;  $\mu$  is 10;  $r_p$  is 10,000 ohms; and  $R_L$  is 50,000 ohms. The voltage division will be in the ratio of  $R_L$  to the total ( $R_L + r_p$ ) resistance, so the amplified signal voltage developed across  $R_L$  will be:

$$\begin{aligned} &= \mu e_g \times \frac{R_L}{R_L + r_p} \\ &= (10 \times 2) \times \frac{50,000}{50,000 + 10,000} \\ &= 20 \times \frac{50,000}{60,000} \\ &= 20 \times \frac{5}{6} = 16.66 \text{ volts} \end{aligned}$$

Thus, in this case, the original 2-volt signal has produced a voltage of about 17 volts across the load resistor. With a  $\mu$  of 10, the value of  $\mu e_g$  is 20, but we can never get all this as there is always some loss across  $r_p$ .

## The Basic Amplifying Stage

Any amplifier, whatever its purpose, can be reduced to the simple circuit shown in Fig. 13. As shown here, the basic requirements for a tube amplifier are: 1, a signal source; 2, the proper operating voltages; 3, a load to which the signal is delivered.

**Signal Source.** The signal source which feeds into the grid-cathode terminals of a tube may be any device capable of supplying the desired signal voltage. The basic operation of a tube stage will be the same regardless of the

source—whether the signal source is a photoelectric cell, a microphone, or a television camera tube, or whether the signal is obtained from a preceding stage.

**Amplifier Load.** The load is the device to which the tube delivers a copy of the voltage applied to the grid. As there must be a path for the d.c. supply voltage to the tube, the load (or a coupling part) will be a resistor, a coil, or a transformer.

**Supply Voltages.** We have already

discussed the importance of supply voltages in this and other lessons. The plate voltage and grid voltage are chosen to cause operation at the particular point on the characteristic curve desired for that particular amplifier. Under those conditions, the applied signal will cause the proper variation in the plate current to give the desired output voltage across the load.

### VOLTAGE AND POWER AMPLIFICATION

Amplifiers can have one of two purposes: 1, if an amplifier makes the signal *voltage* across its load as large as possible (much larger than the grid signal voltage), it is called a *voltage amplifier*; or 2, if an amplifier makes the signal power in the load as large as possible, it is called a *power amplifier*. In this latter case, the a.c. load current multiplied by the a.c. load voltage should be as large as possible.

Voltage amplifiers are needed to build up the strength of weak signals,

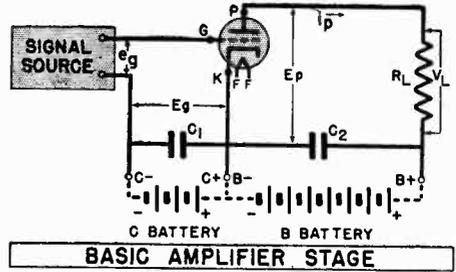


FIG. 13. By inserting the proper signal source, the proper load value, and the correct operating voltages, this basic circuit can be made to fit ANY AMPLIFIER USE!

while power amplifiers are needed to operate most reproducers and to build up the power at the transmitter to the desired level. The basic circuit shown in Fig. 13 will be the same in either case—we can get either maximum signal *voltage* or maximum signal *power* by the proper choice of a tube and by adjusting the value of the load.

To see just how the load can have this effect, let's go back to our equivalent circuit shown in Fig. 12B. To get the greatest *voltage* amplification, we

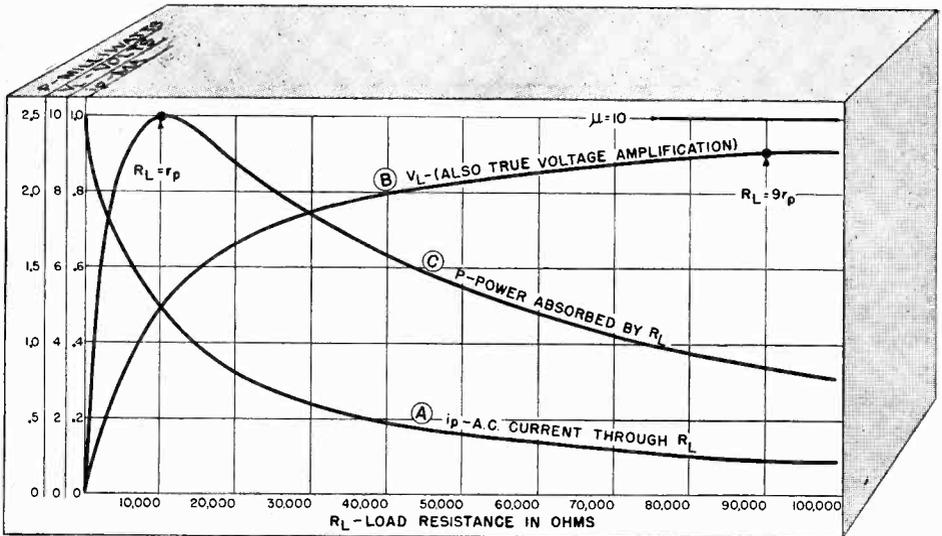


FIG. 14. These curves show the effect that various load values have on the plate current, load voltage, and power output. The higher the load value the greater the VOLTAGE gain. The load should equal the source impedance for greatest POWER output.

want  $V_L$  ( $i_p$  times  $R_L$ ) to be as large as possible, while for power amplification we want the power absorbed by the load ( $V_L$  times  $i_p$ ) to be as large as possible. To illustrate the effects of the load resistor, let's assume that  $r_p$  is 10,000 ohms, that  $\mu$  is 10, and the a.c. grid voltage is 1 volt.

**A.C. Plate Current.** From Ohm's Law, the a.c. plate voltage divided by the total a.c. resistance in the plate circuit gives the a.c. plate current  $i_p$ . (This a.c. plate current is only a part of the total plate current—it is just the variation caused by the signal. We are not interested in the d.c. current, so we ignore it.) By assuming different values of  $R_L$  and plotting the resulting plate current\* we would get the curve *A* of Fig. 14. As we expect, increasing  $R_L$  (that is, increasing the total resistance) causes  $i_p$  to be reduced.

**A.C. Output Voltage.** The load voltage is simply the product of  $i_p$  and  $R_L$ . By multiplying together these values for each size of load resistor, we will get curve *B* of Fig. 14. Although  $i_p$  is decreased by larger values of  $R_L$ , their product is increasingly larger—that is,  $R_L$  increases faster than  $i_p$  decreases.†

This voltage across  $R_L$  is the *output voltage*; the larger  $R_L$  is the larger the output voltage becomes. *The true stage gain is equal to the output divided by the input*, and since we chose an input

\*The a.c. plate voltage is  $\mu e_g$ , and the total resistance is  $R_L + r_p$ , so

$$i_p = \frac{\mu e_g}{R_L + r_p}$$

†We could arrive at the same result by remembering that the a.c. plate voltage  $\mu e_g$  divides between  $R_L$  and the plate resistance  $r_p$ ; as  $R_L$  is increased, a greater part of the a.c. plate voltage is developed across it. In fact, we can calculate the load voltage for any particular input voltage from the relation:

$$V_L = \mu e_g \times \frac{R_L}{R_L + r_p}$$

grid voltage of 1 volt, in this particular case the voltage  $V_L$  is equal to the true voltage amplification.

**Power Output.** The load power is the product of  $i_p$  and  $V_L$ —the a.c. plate current multiplied by the a.c. voltage across the load. By multiplying together the values obtained on curves *A* and *B* of Fig. 14, we get curve *C*, which represents the load power for various values of load resistance. ► There are a number of very important facts revealed by the three curves in Fig. 14. These are the points you should remember:

**1. Effect of Load on A.C. Plate Current:** The insertion of a load in the plate circuit of an amplifier tube *decreases* the a.c. plate current; the greater the ohmic value of the load, the less the current.‡ When the plate load resistance is exactly equal to the a.c. plate resistance, the a.c. plate current will be reduced to exactly half its no-load value (the value when  $R_L$  is not in the circuit).

**2. Effect of Load on Amplification:** Increasing the plate load resistance increases the stage gain, so the maximum gain is obtained when  $R_L$  is many times larger than  $r_p$ . The maximum possible amplification is  $\mu$ , the amplification factor of a tube, but this limit can never be reached in actual practice, since there is always some loss in voltage across  $r_p$ . When  $R_L$  is equal to  $r_p$ , only 50% or one-half the total possible amplification is obtained, and when  $R_L$  is nine times the value of  $r_p$ , about 90% of the maximum amplification is obtained. (Practical voltage amplifying stages using *triode* tubes usually have load values about 7 to 10 times the value of  $r_p$ .)

‡‡It is assumed that the supply voltage is adjusted so the d.c. plate-cathode voltage and the d.c. plate current do not change.

**3. When Maximum Power is Delivered to the Load.** The load in a triode tube amplifier gets the maximum power when the resistance of the load is *equal* to the a.c. plate resistance of the tube; the power then decreases gradually as  $R_L$  is further increased. Thus, maximum power output is obtained when the load resistance equals or matches the a.c. plate resistance. ► From this, you can see that the *value* of plate load resistance used in a stage is important. In your service work, you will be called upon many times to replace the part used as a load. Naturally, the replacement part should be approximately the same value as the original, although generally you can

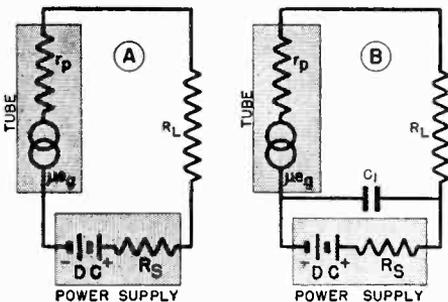


FIG. 15. How the effects of the power supply resistance are eliminated by a by-pass condenser.

use a somewhat higher value if an exact replacement part cannot be obtained.

► Incidentally, maximum *undistorted* power output is not obtained with the same load value as is maximum power, as we shall see later. With *triode* tubes, distortion is less when the load value is about twice the a.c. plate resistance. Tube charts give load values for maximum *undistorted* power output.

### BY-PASS CONDENSERS

Returning to the circuit of Fig. 13, you will notice that condensers  $C_1$  and  $C_2$  are connected across the batteries. Now, batteries have some internal re-

sistance and, as they grow older, this resistance increases. Also, all power supplies (used in place of batteries) have considerable impedance. If the signal current has to flow through this impedance or resistance, several undesirable actions will occur.

► The equivalent circuit containing this added resistance is shown in Fig. 15A. Obviously, the added  $R_s$  increases the total resistance, thus reducing  $i_p$  and reducing the output voltage ( $i_p$  times  $R_L$ ) across  $R_L$ .

We can minimize this effect by connecting a condenser  $C_1$  across the power supply, as shown in Fig. 15B. By choosing a capacity large enough to have a very small reactance to the signal current, we find that the condenser will act practically as a short circuit for signals around  $R_s$ , so that little signal energy is lost in  $R_s$ . We can then ignore  $R_s$  and can consider the a.c. circuit to be  $r_p$ - $R_L$ - $C_1$ , with the position of  $C_1$  so low that the circuit is, for all practical purposes, the same as was shown in Fig. 12B. (Of course, the condenser, if in good condition, does not pass appreciable d.c., so the d.c. operating voltages are still supplied to the tube through  $R_L$  in the usual manner.) Because  $C_1$  offers another path for a.c. around the undesired path, it is called a *by-pass* condenser.

► There is another reason for keeping the signals out of the power supply. Usually the power supply furnishes d.c. voltage to more than one stage. If we let signals get into the power supply, they may flow into the other stages powered by the supply. Such coupling is undesirable—it may cause squealing and howling, or it may reduce the gain; you will learn about this in later lessons.

► Finally, we don't want stray a.c. voltages to come from the supply, along with the desired d.c. The same

by-pass condenser helps here too. If we represent the unwanted a.c. in the supply by generator  $e$  in Fig. 16, you can see that there are two paths for its current; 1, through  $R_s$ - $R_L$ - $r_p$ ; and 2, through  $R_s$ - $C_1$ . As  $C_1$  has a low reactance, it forms a filter with  $R_s$ , such that most of the undesired a.c. is dropped in  $R_s$ . Thus, the condenser again acts practically like an a.c. short circuit, reducing the a.c. voltage between terminals 1 and 2 to the point

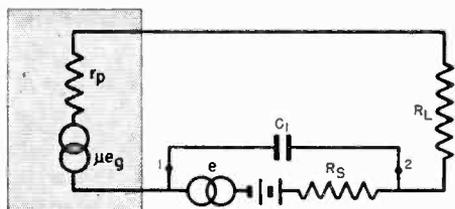


FIG. 16. Stray a.c. voltages from the power supply do not affect the rest of the circuit if the by-pass condenser  $C_1$  is in good condition.

where very little undesired current will flow through  $R_L$  and  $r_p$ .

► These reasons for using a by-pass condenser across the power supply in the plate circuit also apply to the grid circuit, where a by-pass condenser generally is used across the d.c. bias supply. Condenser  $C_1$  in Fig. 13 does not interfere with the d.c. bias supply, but does prevent a signal voltage drop across the supply; it also prevents any a.c. voltage in the bias supply from affecting the grid circuit.

► Thus, by-pass condensers across operating voltage supplies serve three important purposes:

1. They prevent an undesirable loss of signal.
2. They prevent undesirable coupling between stages.
3. They prevent undesired signals from the power supply from getting into the amplifying stage.

► Should a by-pass condenser become defective, the stage operation is certain

to be upset. If the condenser open-circuits (perhaps because a lead pulls away from a foil plate), then the condenser is in effect "not there." We will consequently get a loss of gain, undesirable coupling between stages, and perhaps hum or noise voltages from the power supply. On the other hand, if the condenser short-circuits, it will provide an unwanted path for d.c. This "short" would act like a direct wire connection between terminals 1 and 2 of Fig. 16, so it would remove all d.c. plate voltage and cause a "dead" stage.

### SELF-BIAS

The amplifier circuit in Fig. 13 shows a C battery as the grid bias supply. However, if the radio is equipped

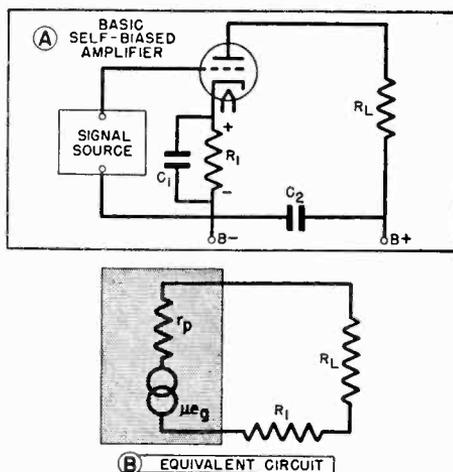


FIG. 17. The tube can furnish its own C bias when this circuit is used. Plate current moving from B+ to the cathode causes a voltage drop across  $R_1$ . As this voltage is between the grid and the cathode, it acts as the bias.

with a power pack so it can operate from a power line (as most sets are today), obviously we don't want to have to use batteries too. While it is possible to obtain bias voltages from the power pack, it is not always convenient to do so, particularly if the pack must supply a number of different stages.

The circuit shown in Fig. 17A makes it unnecessary to use any external source to furnish the proper d.c. bias voltage for the tube. In this circuit, the electron flow in the plate circuit is from the cathode of the tube to the plate, through  $R_L$ , through the plate power supply ( $B+$  to  $B-$ ), and then through resistor  $R_1$  back to the cathode. Thus, *the d.c. plate current must flow through  $R_1$ .*

This current flow through  $R_1$  produces a d.c. voltage drop across it. The polarity of this voltage drop will be as shown, because electrons enter the end of  $R_1$  connected to  $B-$ ; this end of resistor  $R_1$  is, therefore, more negative than the other end.

This voltage is between the grid and cathode, because the grid return from the signal source connects to the negative end of  $R_1$ , while the positive end of  $R_1$  is connected to the cathode. In other words, the d.c. voltage across  $R_1$  makes the grid more negative than the cathode. This is exactly what a C battery or other bias supply does.

Effectively, with this circuit, the tube biases itself—since its own d.c. plate current furnishes the bias voltage, which in turn sets the plate current! That's why such an arrangement is called *self-bias*.

► Service hint: the proper value of resistor  $R_1$  is rather easy to determine, as the normal d.c. plate current and the required grid bias voltage are known from the tube chart characteristics. By dividing the grid bias voltage by the plate current (in amperes), we get the value of  $R_1$ .

► If we did not use by-pass condenser  $C_1$ , there would be an a.c. voltage drop across  $R_1$  just as there was across  $R_S$  in Fig. 15. (The equivalent circuit in Fig. 17B shows this more clearly.) Such a drop is undesirable both because it will reduce the output across  $R_L$  and because it will be introduced into the grid circuit (since the voltage across  $R_1$  acts on the grid). Therefore,  $R_1$  is usually by-passed by a condenser such as  $C_1$  in Fig. 17A.

If by-pass condenser  $C_1$  open-circuits, the stage gain will be reduced. If it short-circuits, the bias voltage will be removed; this will cause a high d.c. plate current and operation on the wrong part of the characteristic curve. Compare this with what you just learned about the effects of an open or a short in the by-pass condenser across the plate supply. *Notice — although the two condensers serve much the same purposes, defects in them may have quite different effects on the set.*

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## Typical Amplifying Circuits

While all amplifiers are basically alike, there are differences between them — differences in the methods of coupling into the grid circuit, in the signal source used in the grid circuit, and in the types of plate loads. The exact circuit depends upon the use to which the tube is put; whether, for example, it is to amplify low-frequency intelligence signals or high-frequency

carrier signals, or is to be used in television or in sound systems. Suppose we now briefly introduce you to some typical amplifiers, which you will study fully in your next few lessons.

### RESISTANCE-CAPACITANCE COUPLING

Fig. 18 shows an amplifier circuit that is frequently found in audio or

video frequency amplifiers (intelligence signal amplifiers similar to *E* and *N* of Fig. 1).

Resistors and condensers are used to couple (connect) this stage to the stages before it and after it, so it is called a "resistance-capacitance coupled amplifier." As you can see, the d.c. plate voltage is obtained from the B supply in the usual way. The stage is self-biased by resistor  $R_2$ . By-pass condensers  $C_2$  and  $C_3$  have already been explained.

When an input signal is applied to terminals 1 and 2, the a.c. portion of this signal will pass through condenser  $C_1$  and develop a voltage across resistor  $R_1$ . If there is also a d.c. voltage between terminals 1 and 2, condenser  $C_1$  will prevent this d.c. component from entering the stage. Thus,  $C_1$  and  $R_1$  form a filter, blocking any d.c. voltage which may be between terminals 1 and 2, but passing on the a.c. signal. (The d.c. blocking is necessary to prevent interference with the bias—the d.c. voltage from the previous stage could force the grid highly negative or highly positive, and thus cause operation at some undesirable point on the tube characteristic.)

The varying a.c. signal voltage across  $R_1$  is alternately added to and subtracted from the bias voltage produced by  $R_2$ . This produces an a.c. variation in plate current, which in turn produces an amplified copy of the signal voltage across the load resistor  $R_3$ .

The voltage across  $R_3$  is passed through another filter,  $C_4$ - $R_4$ , which passes on only the a.c. signal and blocks the d.c. voltage in the plate circuit from the following stage.

Notice that condensers  $C_1$  and  $C_4$  serve to isolate the stage. On the input side,  $C_1$  prevents any external d.c. voltage from interfering with the grid

circuit and, on the output side,  $C_4$  prevents the plate operating voltage from causing an undesirable action in any following stage or device.

► In case you're not sure how an a.c. voltage gets across  $R_4$  from  $R_3$ , notice the ground connections. You can trace a complete circuit (for a.c. only) from  $R_3$  to  $C_4$ ,  $C_4$  to  $R_4$ ,  $R_4$  to ground, ground to  $C_3$ , and from  $C_3$  back to  $R_3$ .

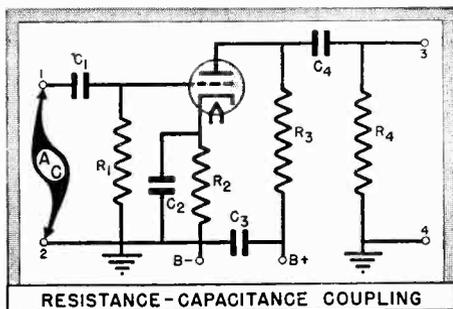


FIG. 18. This basic circuit is the most commonly used type of audio amplifier.

Thus, any a.c. voltage appearing across  $R_3$  is applied to  $R_4$  through condensers  $C_3$  and  $C_4$ . In effect, these condensers place  $R_4$  in parallel with  $R_3$  insofar as a.c. is concerned.

Very many radio circuits are completed in this manner through ground connections, so remember—when you are tracing a circuit, don't stop when you meet a ground connection; trace from the other end of the source until you come to ground again. You can consider that all grounded points are connected just as if wires were run between them.

► However, just because ground points are connected together, don't make the mistake of thinking of currents as wandering about aimlessly. Currents *only* flow in *complete* circuits because of the voltages in those circuits. Thus, in Fig. 18, the current flow from  $R_4$  will be through  $C_3$ ; it will *not* flow through  $R_1$  as there is no path

from the grid end of  $R_1$  to the  $B+$  end of  $R_3$ .

### LOW-FREQUENCY (AUDIO) TRANSFORMER COUPLING

Another amplifying circuit is shown in Fig. 19. Here, resistor  $R_1$  furnishes the self-bias and  $C_1$  is the by-pass condenser preventing any appreciable signal voltage drop across this resistor. Condenser  $C_2$  by-passes the B supply.

The grid circuit is completed through the secondary  $S$  of transformer  $T_1$ , while the plate circuit of the tube is completed through the primary  $P$  of transformer  $T_2$ .

When an a.c. signal is applied to terminals 1 and 2, the resulting current flow through primary  $P$  of transformer  $T_1$  sets up a varying flux which induces an a.c. voltage in the secondary  $S$  of

former  $T_1$ . This current flow reduces a flux change which induces *another* (but *similar*) voltage in the secondary winding. This acts as the input signal for the tube, which varies the plate current of the tube. In turn, this plate current flow through the primary of transformer  $T_2$  causes a flux change which induces a *final voltage* in the secondary  $S$  of this transformer. Thus, while the *original voltage* applied to terminals 1 and 2 gets no farther than the primary of transformer  $T_1$ , successive "carbon copies" are produced by each transformer and by the tube; finally, an amplified copy of the original signal can be obtained from terminals 3 and 4.

► The transformers act as "isolators" like the condensers  $C_1$  and  $C_4$  of Fig. 18, in that d.c. has no influence on their secondary circuits. Basically, therefore, we use d.c. to get proper tube operation, but then we usually separate the signal from the d.c. before passing it along.

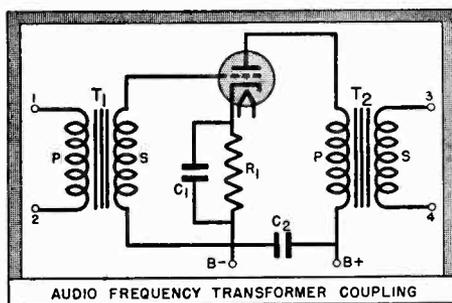


FIG. 19. Transformer coupling is employed when high gain is desired, as you will learn later.

this transformer. This a.c. voltage is between the grid and cathode of the tube, and produces a varying plate current.

The plate current variation must pass through the primary  $P$  of transformer  $T_2$ . This produces a flux change in this transformer, and so induces a voltage in the secondary  $S$  which can be applied to a reproducer system or to another stage.

► Notice the voltage *changes* which take place. One voltage causes a current to flow in the primary of trans-

### R.F. AMPLIFIER STAGES

The low-frequency amplifier shown in Fig. 19 uses laminated iron-core transformers, which cannot be used at frequencies higher than about 15,000 cycles per second. R.F. carrier frequencies are far higher than this, so air-core or powdered-iron core transformers must be used in r.f. amplifiers. A typical circuit for an r.f. amplifier stage is shown in Fig. 20.

It is customary to tune radio frequency transformers so that the amplifier stage will give both amplification and selectivity. Condensers  $C_1$  and  $C_4$  in Fig. 20 are variable condensers which are used to tune  $L_2$  and  $L_4$  (respectively) to some desired radio frequency. Let's see what effect the addition of these condensers has on operation of the circuit.

An r.f. voltage applied to terminals 1 and 2 causes a current flow in  $L_1$  which sets up a varying flux. This induces in coil  $L_2$  a voltage which acts as if it were in series with this coil. Hence, the tuned circuit  $L_2-C_1$  is series resonant (has minimum impedance) for the frequency to which the circuit is tuned.

If the voltage induced in  $L_2$  is of the resonant frequency, it will undergo

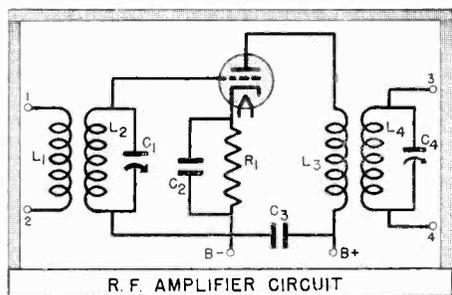


FIG. 20. This tuned amplifier circuit is known as a r.f. stage.

resonant voltage step-up and appear across  $C_1$  in amplified form\*; if it is not of the resonant frequency, it will appear across  $C_1$ , but will not be much amplified by the action of the resonant circuit. Thus, if the signal applied to terminals 1 and 2 consists of a mixture of several frequencies, only the resonant-frequency voltage will appear across  $C_1$  in greatly amplified form (though voltages with frequencies near

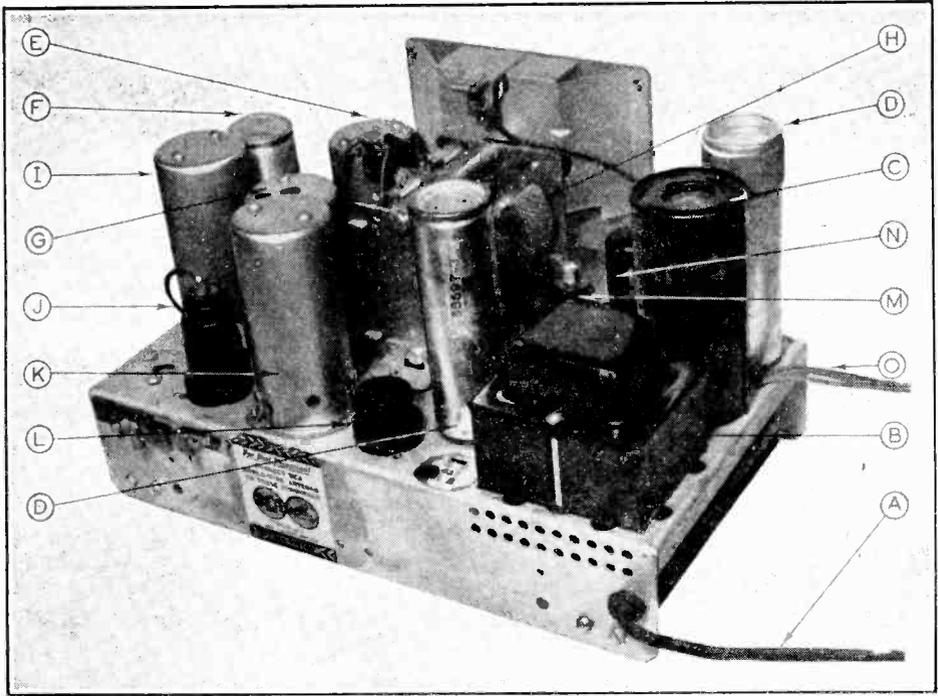
\*Remember, the action of a series resonant circuit in stepping up a voltage is as follows: The coil and condenser together have a resonant frequency at which their reactances cancel each other; when the voltage induced in the coil is at this resonant frequency, a large a.c. current flows in the resonant circuit. This a.c. current naturally produces a large a.c. voltage drop across the coil and another across the condenser, as each one *alone* has a high a.c. reactance. The resulting voltage across either the coil or the condenser is therefore many times greater than the original induced voltage.

the resonant frequency may be slightly amplified).

The r.f. voltage developed across  $C_1$  is between the grid and the cathode, so it causes a varying plate current through  $L_3$ , which in turn induces a voltage in  $L_4$ . Again there is a series resonant circuit ( $L_4-C_4$ ); and again the resonant-frequency voltage will appear across the condenser ( $C_4$ ) in amplified form. This voltage can be passed on to the next stage for further amplification. Thus, this r.f. amplifier has *variable* gain: it amplifies the resonant frequency signal much more than other signals. We can (and, in fact, do) use such a circuit as a *selector* as well as an amplifier. When we tune its two resonant circuits to some desired frequency, signals of that frequency will receive far more amplification than will signals of other frequencies which may be present in the input to the amplifier. In other words, the circuit will *select* signals of the desired frequency.

► Fig. 20 looks very much like Fig. 19. However, the use of air-core transformers (instead of laminated iron-core transformers) and the fact that tuning condensers are used tell us that this is an r.f. circuit exclusively; air-core transformers and tuning condensers are not used in a.f. stages.

► An interesting variation of this circuit can be had by using  $C_4$  to tune  $L_3$  instead of  $L_4$ . This would form a parallel resonant circuit in the plate circuit which would offer maximum impedance at its resonant frequency but far less impedance at other frequencies. In other words, at the resonant frequency the load resistance is a maximum—which, as you have learned, means that the stage gain will be highest at this particular frequency. As a result, the greatest possible amount of the signal voltage will be



Here is a photograph of the chassis of a six-tube all-wave superheterodyne. The various stages can be recognized in two ways—by the types of tubes used and by their associated signal sources and loads. (You will learn more about this in other lessons.) In this figure, the following parts are identified: A, 110 volt a.c. line cord; B, power transformer; C, rectifier tube; D, filter condensers; E, antenna coil; F, frequency converter tube; G, oscillator coil; H, tuning condensers; I, first i.f. transformer; J, i.f. amplifier tube; K, second i.f. transformer; L, second detector (called the demodulator) tube; M, first audio tube; N, power amplifier tube; O, loudspeaker cable.

developed across  $L_3$  at the resonant frequency. Again we have a *selective* circuit which provides maximum gain at the resonant frequency, but less gain at other frequencies.

► Naturally, the coil and condenser combination in each of these circuits must be properly chosen so that their L-C values will give resonance at the desired frequency. It is possible to

change the frequency to which the circuit is resonant (or, as radio men say, to “tune” the circuit) by varying the tuning condensers  $C_1$  and  $C_4$  in step.

► We have given you this brief introduction to amplifiers to show you how similar they are, even though they are designed to perform different jobs. You are going to study the details of these circuits in the next few lessons.

# Tubes as Signal Generators and as Signal Mixers

The amplifying stages we have discussed so far have been purely relaying devices. Each has taken a signal from the signal source, made an enlarged copy of it, and passed it along to another stage. The signal itself has not been changed, except that it has been made larger.

There are stages which perform other functions instead of amplifying, or in addition to amplifying. We now will discuss several of these briefly, so you will see the important parts they play.

## THE OSCILLATOR

The radio stage we call an oscillator is the stage in which an r.f. signal is generated. This signal is an a.c. voltage which varies between positive and negative values just like the a.c. voltage in your power line. In fact, the only differences between power line voltages and r.f. voltages are in their voltage values and frequencies.

Essentially, here is how an oscillator works. As you know, an a.c. grid voltage in an amplifier stage causes an a.c. plate current. This current produces an a.c. voltage drop across the load which is an amplified version of the grid signal voltage. To make such an amplifier into an oscillator, all we need do is to feed part or all of this a.c. load voltage back into the grid circuit with the proper phase. Then the a.c. load voltage will act as a grid signal, which in turn will cause more a.c. load voltage, and so on. An oscillator, in other words, is simply an amplifier which supplies its own grid signal from its plate circuit.

A typical oscillator circuit is shown in Fig. 21. Here, the a.c. plate current

flowing through the feedback coil  $L_T$  produces an a.c. flux. This flux links with  $L_1$ , inducing a corresponding a.c. voltage in it (much as if  $L_T$  were the primary and  $L_1$  the secondary of a transformer). The grid voltage thus developed is fed to the grid (through the parallel  $R_g-C_g$  combination, the function of which we will discuss shortly) and there produces a continuation of the plate current variations, which in turn keeps the grid voltage changing! Obviously, this action can keep on indefinitely, once it gets started. But how does it start?

Because of the  $L_1-C_1$  circuit, all we need do is to cause some change in plate current. The act of turning on the circuit (applying operating voltages) is sufficient. The rise in plate current as the tube warms up will cause a pulse to be induced in  $L_1$  from  $L_T$ . From here, the resonant circuit takes charge and controls the oscillator.

The initial pulse of energy causes a current flow which charges the condenser  $C_1$ . Then, when no further energy is induced,  $C_1$  discharges through  $L_1$ . Electrons flow from the condenser with a rush at first, then, as the condenser voltage drops, the discharge current decreases. In other

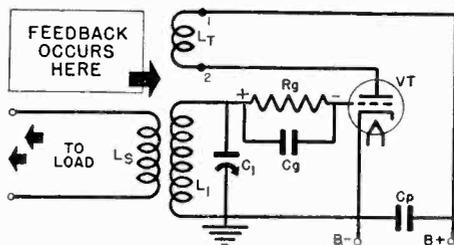
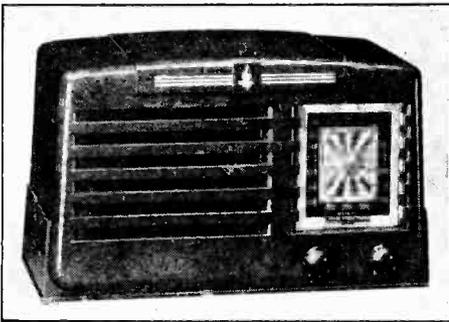


FIG. 21. A basic oscillator circuit.

words, the condenser discharge produces a varying current through the coil. And, as you know, a coil opposes changes in the current flow through it. Therefore, when the condenser tries to discharge through the coil, energy is stored in the magnetic field of the coil. When the electron movement has caused the condenser voltage to reduce to zero, the current flow would tend to stop. However, now, the coil field collapses and keeps the current flowing in the same direction as it was going. This results in the condenser being charged up again, this time with the opposite polarity.



Courtesy Emerson Radio and Photo. Corp.

A typical midget receiver. It uses the same stages as a large console type. The main differences are: 1, a different power supply; 2, different tubes; 3, components smaller physically.

When the magnetic field is completely collapsed, the condenser will try again to make current flow through the circuit, this time in the opposite direction. Thus, if a pulse of energy is fed into the circuit, an oscillatory (back and forth) electron flow will take place in the  $L_1-C_1$  circuit.

This oscillating current flow keeps reversing the polarity with which condenser  $C_1$  is charged. In fact, it follows a sine-wave pattern, with a frequency which is equal to the resonant frequency of the  $L_1-C_1$  circuit. Thus, our resonant circuit is able to convert the initial pulse of energy into a sine-wave

voltage.

If there were no resistance in the circuit, current oscillations would keep on indefinitely. However, any practical circuit always has resistance in the coil and leads. Current flow through this resistance produces power losses ( $I^2R$ ) which rapidly reduce the amplitude of the current oscillations.

This is where the tube comes in. The sine-wave voltage across  $C_1$  controls the plate current, which in turn induces pulses of energy from  $L_T$  into  $L_1$  at just the right time to keep oscillations going. This feedback makes up for the power lost in the resistance of the resonant circuit, so the oscillations are maintained.

► We can make use of the a.c. power produced in the oscillator circuit  $L_1-C_1$  by placing a coil  $L_S$  near coil  $L_1$ . A sine-wave voltage will then be induced in  $L_S$ , which we can feed to other circuits, where it can be used as an input signal.

**Automatic Bias.** Perhaps the easiest way to see why  $R_g$  and  $C_g$  are used in the grid circuit is to imagine what would happen if they were not present and the  $C_1$  voltage were fed directly to the grid. Then, when the a.c. voltage across the condenser went through its positive alternation, the grid would become highly positive and a large plate current would flow. In fact, under these conditions, a high-power tube could actually pass enough current to melt the tube elements.  $R_g$  and  $C_g$  supply a bias voltage which keeps the grid voltage low enough to prevent this action.

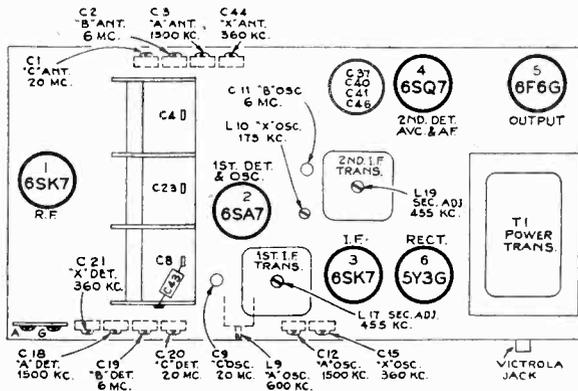
When the grid is driven positive by the voltage across  $C_1$ , grid current will flow because the grid will attract some of the electrons emitted by the cathode. This grid current flow develops a voltage across  $R_g$  with the polarity shown in Fig. 21. This voltage charges up

condenser  $C_E$ .

When the  $L_1-C_1$  circuit drives the grid negative, the grid electron flow stops, but condenser  $C_E$  now partially discharges through  $R_E$ . This discharge current replaces the grid current and maintains the voltage across  $R_E$  practically constant. Then, on the next positive swing,  $C_E$  is recharged, and events repeat themselves. This is called an *automatic bias*. It is another form of self-bias; here the *grid* current flow produces the bias, while in those circuits we studied earlier the *plate* current flow produced the bias.

(1) as the generator of the carrier wave in a transmitter; (2) as the local oscillator in a superheterodyne receiver; and (3) as a test signal generator used by servicemen. These are all r.f. oscillators. Audio oscillators are used mostly in test and measuring equipment.

► This is by no means the only oscillator circuit. You'll find many others in later lessons. But while these oscillators are different in some respects—some to the extent that they produce sawtooth or square waves instead of sine waves—they are alike in being



Many manufacturers furnish complete service information on their receivers in the form of service sheets and manuals. This layout of the top of a chassis is typical of the kind of information you can expect. This layout gives: 1, the positions of the trimmer condensers used for adjusting the circuits to track together; 2, the types of tubes; 3, the functions of the tubes. Thus: 1, a 6SK7 r.f. amplifier; 2, a 6SA7 first detector and oscillator combined (the frequency converter); 3, a 6SK7 i.f. amplifier tube; 4, a 6SQ7 second detector (demodulator) and first a.f. tube; 5, a 6F6G output tube; 6, a 5Y3G rectifier tube. The positions of the stages are different in each radio, but there will always be relatively short signal paths from stage to stage.

**Oscillator Uses.** With the oscillator we've described, we could produce sine-wave voltages of almost any frequency we desire—from audio to radio frequencies—merely by using the proper values of inductance and capacitance in the tuned circuit. The chief radio uses of oscillators are:

amplifiers which supply their own grid signals.

### MODULATED AMPLIFIERS

Now that we know how a carrier signal can be generated, let us see how we can modulate it with an intelligence signal. (This is accomplished in sec-

tion *C* in Fig. 1.) At this point in your Course, we won't give a detailed discussion of the circuits used for modulation, but you should learn the essentials of the process. And, the method used determines the kind of demodulator necessary in a receiver. There are two systems now in wide use; we shall describe both briefly.

► In one important method of modulation, the amplitude (the strength) of the carrier is varied by the intelligence signal. This method—called amplitude modulation (or a.m.)—is used in the standard broadcast stations, in the majority of commercial stations, and for the video television signal.

► Another method is to have the intelligence signal vary the frequency of the carrier. This *frequency modulation* (f.m.) is used for the sound portion of a television program, and for broadcasts on the very high frequencies. In particular, it is used for some police radio and other special services where its unique freedom from atmospheric and man-made interference makes frequency modulation highly desirable.

**Amplitude Modulation.** Fig. 22A represents an unmodulated carrier which has a constant amplitude  $N$ . We want this carrier to be varied in amplitude by the audio signal shown in Fig. 22B. In other words, we want to produce an r.f. signal like that shown in Fig. 22C, where the amplitude of the carrier varies as the audio signal varies. (Actually there are hundreds of r.f. cycles to each modulation cycle. We can't show them all in this drawing.)

A simple way to get this action is to feed the carrier into the grid circuit of an r.f. amplifier, and to arrange the amplifier so that the intelligence signal can vary its plate voltage. With no intelligence signal, the plate current

pulses follow the r.f. variations; the amplitudes of these pulses are fixed by the operating voltages.

When an intelligence signal is applied, its positive alternations will add to the d.c. plate voltage, while its negative alternations subtract from the d.c. voltage. Thus, the plate voltage is varied up and down. You know that the amount of plate current depends on the plate voltage, so you can see that this variation forces the plate current to vary. Thus, the r.f. pulses are made larger and smaller, accord-

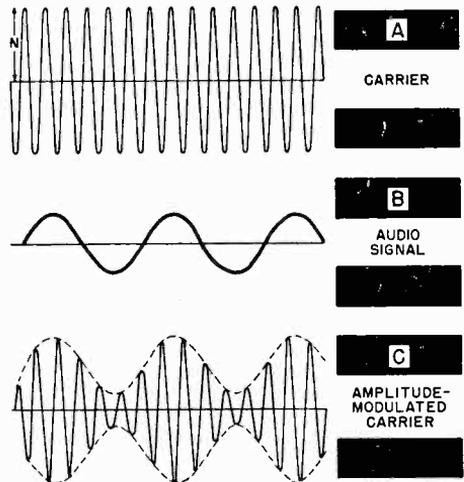


FIG. 22. When the carrier at A is modulated by the audio signal, B, the result can be represented by C.

ing to the intelligence signal variations.

**Frequency Modulation.** In frequency modulation, the amplitude of the carrier is unchanged, but its frequency is varied. This is quite different from amplitude modulation.

The carrier signal is just like the carrier used in an a.m. system, but the "carrier" frequency is known as the *resting* frequency; it is the frequency radiated when there is no modulation.

Suppose we modulate the resting signal with an intelligence signal. Our

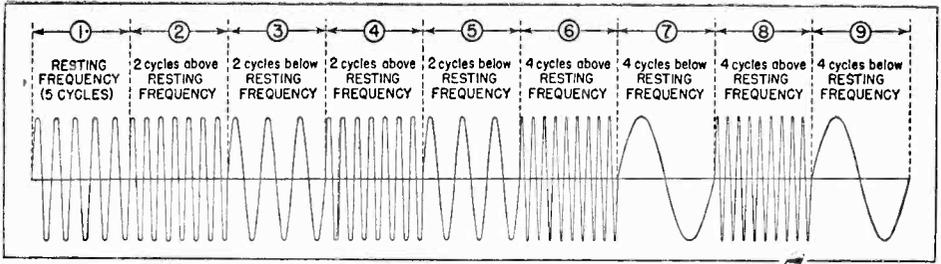


FIG. 23. A fundamental representation of a frequency-modulated wave.

f.m. circuit then produces an output which consists of a signal of constant amplitude that varies in frequency. The frequency increases when the intelligence signal cycle is positive, and decreases when it is negative; the amount of increase or decrease from the normal value depends upon the amplitude of the intelligence signal (the greater the amplitude, the greater the frequency change).

Fig. 23 shows this process in a simplified form. The resting frequency is represented at 1. When modulated by an intelligence signal of medium-strength, the frequency changes up and down as shown at 2, 3, 4, and 5. A stronger intelligence signal causes a wider frequency change, as shown by 6, 7, 8, and 9. (These represent peak values.)

► To sum up: in an f.m. system, the *rate* at which the carrier frequency changes corresponds to the intelligence frequency, while the *amount* of carrier frequency change depends on the amplitude of the intelligence signal.

► The actual circuits used for frequency modulation contain a number of stages and are rather elaborate. However, their fundamental operation can be shown by Fig. 24.

This is the circuit of an oscillator which generates a carrier signal at a frequency determined by the tuned circuit  $L_1-C_1$ . Condenser  $C_1$  is attached to a microphone so its capacitance can

be varied by movements of the microphone diaphragm. When no sound waves are striking the microphone diaphragm, the condenser has a normal or resting value, and the oscillator will produce an r.f. carrier at a frequency known as the "resting" frequency.

When sound waves strike the microphone  $M$ , the diaphragm is moved back and forth. This varies the capacity of  $C_1$  and so varies the resonant frequency of the  $L_1-C_1$  circuit. This, in turn, varies the output frequency of the oscillator—in other words, gives us a frequency-modulated carrier. This is a crude example, but it does show the essentials of this system of modulation.

### FREQUENCY CONVERTERS

The frequency converter, which combines two different radio frequen-

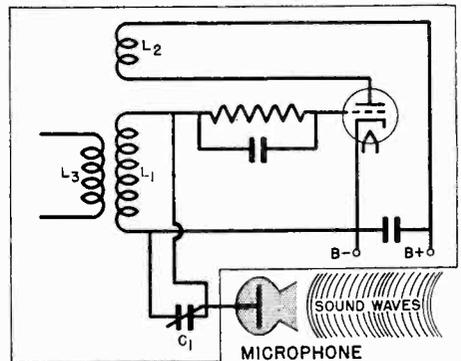


FIG. 24. This simple circuit shows one possible way of frequency-modulating an oscillator. Of course, practical circuits are far more complex.

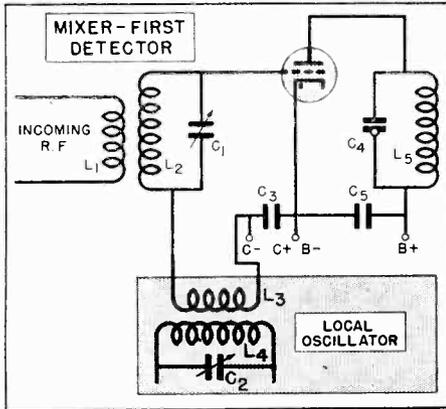


FIG. 25. In the mixer-first detector stage, the incoming signal is mixed and combined with the local oscillator signal. The result is a new carrier frequency known as the intermediate frequency.

cies to form a third frequency, is a very important section of a superheterodyne receiver. You will recall that this is the section which takes the incoming signal and combines it with a signal from a local oscillator so that they produce an intermediate frequency (i.f.).

This frequency conversion takes place in a mixer-first detector circuit like that shown in Fig. 25. The name

of the stage comes from the fact that two r.f. voltages (the incoming carrier and the output of the local oscillator) are fed into the grid circuit, where they *mix*, and from the fact that the C bias voltage is adjusted so it causes plate current cut-off, which makes this a special form of class B amplifier called a *detector*.

If the tube were a perfect amplifier, its plate current would be just a mixture of the signals fed to the grid of the tube. However, by using the characteristics of a detector (to be fully explained elsewhere), we find that the plate current pulsations contain not only the two original signals but also frequencies equal to the sum of, and the difference between, these two. This difference-frequency component is the i.f. we want, so we tune the resonant circuit  $L_5-C_4$  in Fig. 25 to this difference frequency, thereby making the stage output consist almost solely of i.f. voltage. This same action occurs on both a.m. and f.m. signals, because the modulations are just transferred to the new i.f. carrier.

## The Diode in Radio Systems

While all the circuits we have discussed so far use tubes with control grids, diode tubes (which have only a cathode and a plate) are also used widely in radio—particularly as rectifiers in power packs and as detectors for separating the intelligence signal from the i.f. signal. Let's touch briefly on each of these important uses.

### POWER PACK CIRCUITS

The power pack of an a.c.-operated radio or television apparatus is used to convert the a.c. line voltage to the var-

ious a.c. and d.c. voltages required by the apparatus.

The power pack in the average superheterodyne receiver (as well as in the average transmitter) has four important components:

1. *The power transformer*, which changes the a.c. line voltage to a higher a.c. value for the rectifier tube, and to lower a.c. values for the filaments of all tubes in the apparatus.

2. *The rectifier tube*, which converts the stepped-up a.c. voltage into a pulsating d.c. voltage.

3. *The filter section*, which smooths out or filters the variations in the pulsating d.c. voltage.

4. *The voltage divider*, which divides the resulting d.c. voltage into the various values required by the grids and plates of individual tubes.

► A basic power pack, similar to those used in superheterodyne receivers, is shown in Fig. 26. Let us learn a little more about the purpose of each part.

windings are used for each voltage.

Thus, when 110 volts a.c. are supplied to primary  $P_1$  in Fig. 26, secondary  $S_1$  provides 440 volts a.c.; secondary  $S_2$  provides 5 volts a.c. for the diode rectifier tube filament; secondary  $S_3$  provides 6.3 volts a.c. across terminals  $x$  and  $y$  for the filaments of all other tubes in the receiver.

**The Rectifier Tube.** The simplest and easiest way of converting a.c. into

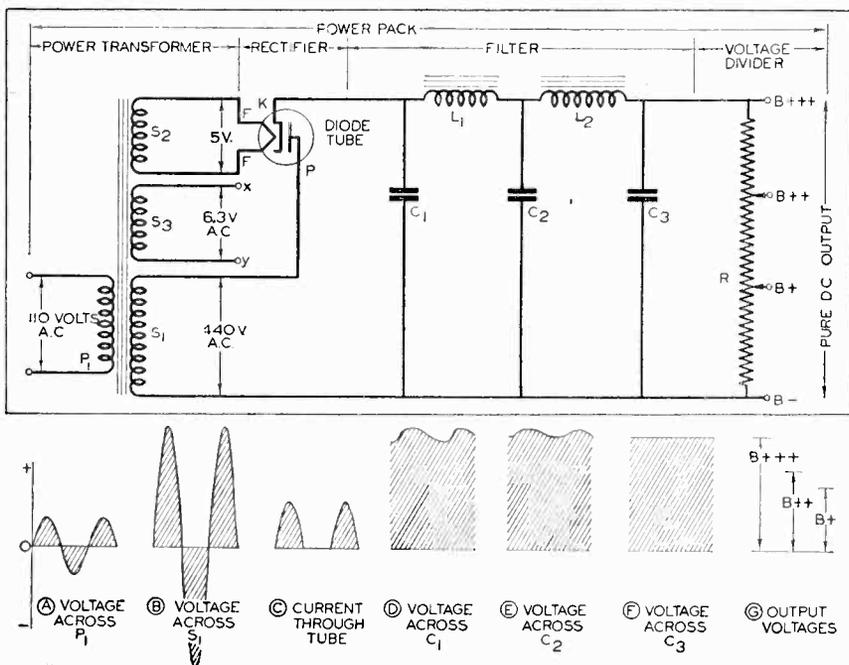


FIG. 26. A simple power supply using a diode tube as a rectifier.

(There is a complete lesson on power supplies later in your Course.)

**The Power Transformer.** The voltage induced in each secondary winding will depend upon the number of turns in the winding. If the secondary has more turns than the primary, its voltage will be higher than the primary voltage; if the secondary has fewer turns than the primary, its voltage will be lower. If different voltages are required, separate secondary

pulsating d.c. is by means of a diode, as you learned in a previous lesson. This tube allows electrons to flow from its cathode  $K$  to its plate  $P$  when the a.c. voltage makes the plate positive, but completely blocks the flow of electrons when the a.c. voltage reverses polarity. Thus, if the voltage across  $P_1$  is represented by graph  $A$  and the voltage across  $S_1$  is represented by  $B$ , the current through the diode tube is represented by  $C$ .

**The Filter Section.** The plate current flowing through the diode rectifier tube is varying or pulsating; as you have learned, this means that it contains a d.c. component plus one or more a.c. components. Since we want only the d.c. for our tube plates and grids, we use a low-pass filter to remove the a.c. components. The chokes offer little opposition to d.c., but  $L_1$ - $C_2$  and  $L_2$ - $C_3$  form voltage dividers for a.c., in which most of the a.c. is dropped across the high-reactance chokes rather than across the low-reactance condensers. The exact manner of filtering, and the special action of  $C_1$  will be covered fully in later lessons on power supplies. However, notice that graphs *D*, *E*, and *F* show that the result is a progressive reduction in a.c., leaving a relatively pure d.c. voltage across  $C_3$ .

**The Voltage Divider.** The output voltage at  $C_3$  may be too high for many of the tubes, so a resistor, having taps or terminals along it, is sometimes connected across the output of the filter to divide the d.c. voltage. The arrangement is shown in Fig. 26 and the voltage division is represented at *G*. It is customary, when a power pack has several output voltages, to mark B+ on the circuit diagram at the terminal which has the lowest positive voltage; the next higher voltage is indicated by B++, the next by B+++ , and so on.

## AMPLITUDE DEMODULATORS

A typical a.m. demodulator stage is shown in Fig. 27A. This corresponds to the demodulator stage *M* in Fig. 1, and is the place where the intelligence signal is separated from the i.f. carrier. (You will recall that the intelligence signal was transferred from the original r.f. carrier to the i.f. carrier in the mixer-first detector stage.)

In an amplitude-modulated system,

the modulated i.f. fed into the demodulator stage has a form like that in Fig. 27B. The envelope of the high-frequency voltage — that is, the dotted lines drawn through the peaks of the i.f. voltage—represents the intelligence signal we want. There are actually two envelopes in this diagram, one drawn through the positive peaks and the other through the negative peaks. We want either one but not

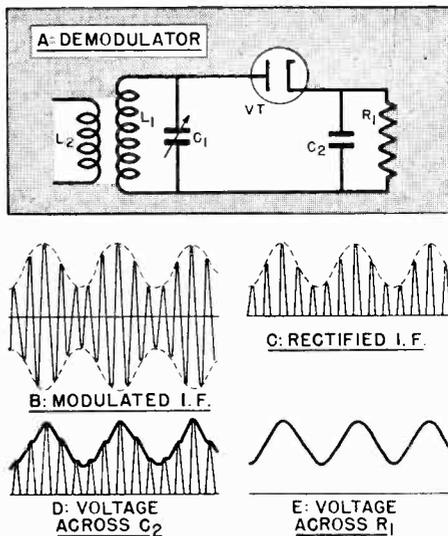


FIG. 27. The diode is also commonly used as a demodulator.

both, since the two are just opposite. If we attempted to send both into some reproducer they would cancel one another and produce no net effect.

Therefore, we apply the modulated i.f. to the diode circuit shown in Fig. 27A. The diode does not permit current flow during the negative alternations of the voltage, so the output of the tube consists of only half of the modulated i.f. (Fig. 27C).

Remember, now—what we actually want is just the *peak voltage* of these half-cycle pulses. Therefore, we feed the diode output into condenser  $C_2$ . During each half-cycle that the tube

conducts, condenser  $C_2$  charges rapidly to the full peak voltage of the pulse; during the rest of the cycle, when the tube is not conducting,  $C_2$  discharges through resistor  $R_1$ . However,  $R_1$  is made so large that  $C_2$  is able to discharge only a small amount before the next pulse comes along and charges it up again. Therefore, the voltage across  $C_2$  (Fig. 27D) remains at or very near the peak voltage of the modulated i.f. pulses all the time. Since  $C_2$  is in parallel with resistor  $R_1$ , the voltage across  $R_1$  (Fig. 27E) also remains just about at the modulation peaks—in other words, the voltage across  $R_1$  is the intelligence signal we want.

## FREQUENCY DISCRIMINATORS

The “detector” or “demodulator” for a frequency modulated signal has another function, in addition to that just described. As you learned earlier, frequency modulation changes the amplitude variations of the original intelligence signal into a frequency change. This produces a signal which varies in frequency but has a constant amplitude (Fig. 23). We must convert the frequency variation back into an amplitude change before we can reclaim the original intelligence signal. The processes of conversion and demodulation both occur in a single stage known as a *frequency discriminator*.

Before we study the circuit, let's review one more fact about resonant circuits.

**Resonance Curves.** You know that an L-C resonant circuit responds best to a signal that has the frequency to which it is resonant—in fact, it actually “amplifies” or steps up such a signal. However, this does *not* mean that a resonant circuit excludes signals of other frequencies. It responds *best* to the frequency to which it is tuned,

but it will respond somewhat to nearby frequencies, with its response becoming less and less the farther away from resonance we go.

This fact is shown by what is called a “resonance curve,” which shows the response of the circuit to the resonant frequency and to other nearby frequencies. A typical curve is shown in Fig. 28. The scale at the left represents the response of a circuit which is tuned to 100 kc. This is indicated by the fact that this is the highest point on the curve. In this case, 100 kc. signals would receive a step-up of 30. Other frequencies are also stepped up, though not as much: frequencies of 90 and 110 kc. are stepped up by a factor of 15, while frequencies of 80 and 120 kc. are stepped up only about 3 each.

Let's see how we can use this variable step-up effect of a resonant circuit

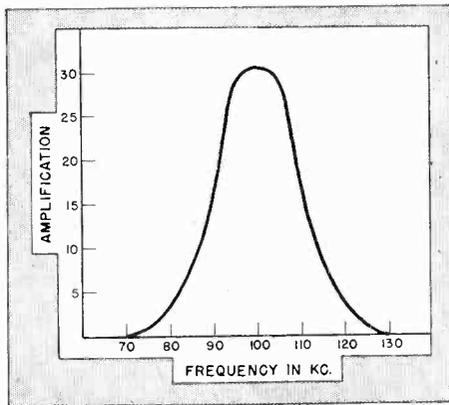


FIG. 28. The response curve of a resonant circuit.

to convert frequency variations into amplitude variations.

**Frequency Discrimination.** One kind of frequency discriminator circuit is shown in Fig. 29A. The circuit  $L_1-C_1$  is tuned to the i.f. “carrier” or resting signal, but responds well to frequencies on both sides of this value. The input signal induces voltages in both  $L_2$  and

$L_3$ , thus causing electrons to flow through diodes  $VT_1$  and  $VT_2$ .

Since electrons move only from the cathode to the plate, electrons flowing through  $VT_1$  come upward through resistor  $R_1$ , producing across this resistor a voltage drop with the polarity shown. Similarly, electrons flowing through  $VT_2$  move downward through resistor  $R_2$ ; the voltage drop across this resistor is therefore of opposite polarity from that across resistor  $R_1$ .

The resonant circuit  $M$  ( $L_2-C_2$ ) is

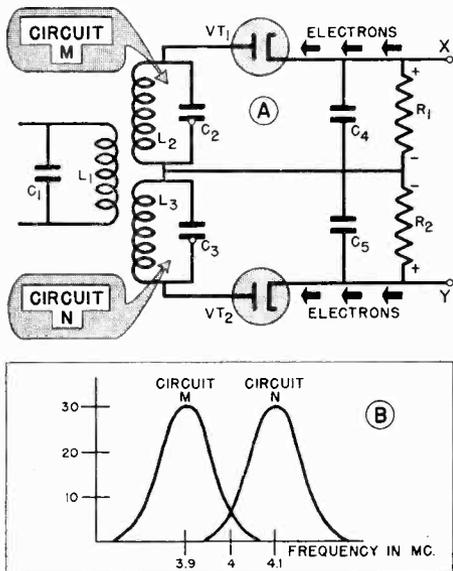


FIG. 29. One type of f.m. discriminator.

tuned to a frequency somewhat lower than  $L_1-C_1$ , while  $N$  ( $L_3-C_3$ ) is tuned to a frequency somewhat higher. If the frequency of 4 mc. in Fig. 29B represents the  $L_1-C_1$  resonance frequency, then 3.9 mc. may represent the resonance frequency of  $M$ , and 4.1 mc. that of  $N$ . Hence, neither  $M$  nor  $N$  is tuned to favor the center frequency, although both pass it along to a certain extent.

The circuits are adjusted so that, at the resting frequency, the voltages ap-

plied to  $VT_1$  and  $VT_2$  are equal, and equal currents flow through  $R_1$  and  $R_2$ . As these are equal resistances, their voltage drops are equal, and, since their voltage drops are *opposite in polarity* (see Fig. 29A), there is NO VOLTAGE between terminals X and Y when only the resting frequency is fed into this stage.

► Now let us suppose the signal frequency is modulated so that it swings from 4 to 3.9, back through 4 to 4.1, then back toward 3.9, etc.

As the frequency approaches 3.9 more voltage will be developed by  $M$ . This causes larger voltages to be applied to  $VT_1$ ; larger currents result and produce larger voltages across  $R_1$ . At the same time, the signal is moving away from the resonant frequency of  $N$ , so less voltage is being applied to  $VT_2$ . The resulting smaller current produces less voltage across  $R_2$ . Hence, for this swing, the voltage across  $R_1$  increases while that across  $R_2$  decreases. The two voltages no longer cancel one another, so a *net voltage* (equal to the difference between them) appears between terminals X and Y.

For example, suppose the  $R_1$  and  $R_2$  voltages are initially 50 volts each, and cancel exactly. On the swing just described, the  $R_1$  voltage may go up to 90 volts and the  $R_2$  voltage may go down to 10 volts. The difference is 90 minus 10 or 80 volts, which will appear between X and Y. Terminal X will be positive, as the polarity will be that of the voltage across resistor  $R_1$ .

► When the swing reverses toward 4.1 mc. exactly the reverse action occurs. Now,  $N$  develops the larger voltage,  $VT_2$  passes the larger current, and the drop across  $R_2$  is larger than that across  $R_1$ . This reverses the polarity of the difference voltage existing across X and Y, since the polarity now is that of the voltage across  $R_2$ .

**Summary:** A signal, shifting in frequency, is introduced into this stage. The frequency variations produce a varying voltage across terminals  $X$  and  $Y$ . Small frequency changes produce small voltages, as frequencies near 4 mc. do not "climb" as high on the  $L_2-C_2$  and  $L_3-C_3$  resonance curves. Larger changes, out to the limits set by the resonance points (3.9 and 4.1 in Fig. 29B), produce larger voltages. Thus, we have a circuit which produces an amplitude variation from a varying frequency signal.

This same circuit also gives us de-

modulation; in this, it is like the diode detector we just discussed for amplitude-modulated systems. The diodes  $VT_1$  and  $VT_2$  rectify the r.f. signal, while condensers  $C_4$  and  $C_5$  reproduce the modulation peaks. Thus, the original intelligence signal appears across terminals  $X$  and  $Y$ .

► *Remember, we don't expect you to grasp all the details of operation of these circuits from the brief descriptions we've given! This has been just a preview of what you'll be studying in your coming lessons—we are introducing you to practical radio stages.*

---

## A Complete Superheterodyne Receiver Circuit

To show you how the stages you have studied are used in a modern radio receiver, we have sketched a basic amplitude-modulation type superheterodyne circuit in Fig. 30. (This same circuit would do for f.m. if the second detector were changed to a discriminator circuit and the resonant circuits were arranged for the proper frequencies. A television receiver is also similar, but there would be differences in the tuned circuits and in the output stage, as well as in the reproducer.) Let's go through its operation briefly, as a summary of what you've learned in this lesson.

► The antenna system intercepts many different modulated r.f. signals, causing modulated r.f. currents to flow in coil  $L_1$ . A number of signals are induced in coil  $L_2$ , but only the signal at the resonant frequency of  $L_2-C_2$  will undergo the full resonance step-up.

Thus, the resonant circuit serves as a preselector which partially separates the desired signal from all others. The

desired signal and such others as get through this resonant circuit then are applied to tube  $VT_1$ . This tube is a class A amplifier, and its plate current variations through coil  $L_3$  induce a voltage in  $L_4$ . The whole stage acts as a combination preselector and r.f. amplifier.

Resonant circuit  $L_4-C_4$  is tuned to the same frequency as  $L_2-C_2$  and serves to separate signals further. Again resonance step-up occurs in this circuit at the desired frequency. (While other frequencies may be getting through, they are not being stepped up as much and so are rapidly being "swamped" or drowned out by the desired signal.)

► At the same time, tube  $VT_2$  in an oscillator circuit is producing a frequency which is higher in value than the incoming carrier frequency. The oscillator tuning condenser  $C_{11}$ , which is operated by the same shaft as tuning condensers  $C_2$  and  $C_4$ , is adjusted so that the oscillator frequency always

differs from the desired incoming signal by a chosen amount—the i.f. value. The coil  $L_9$  provides the necessary feed-back action in this oscillatory circuit. Then, coil  $L_{10}$  couples the oscillating circuit  $L_{11}-C_{11}$  to the input or grid-cathode circuit of tube  $VT_3$ .

► The oscillator and desired input signals are mixed in this mixer-first detector stage, producing the modulated i.f. signal (as well as signals of various other frequencies) in the plate circuit of  $VT_3$ . The resonant circuit  $L_5-C_5$  is tuned to the intermediate frequency,



Courtesy Philco

A typical portable receiver. Once again the standard superheterodyne circuit is used, this time with battery-operated tubes.

and so maximum voltage is induced in coil  $L_6$  at this frequency.

Resonant circuit  $L_6-C_6$  is also tuned to the i.f. value. It assists in separating this particular intermediate frequency from among any others that may be produced by the mixer-first detector tube.

► The i.f. amplifier tube  $VT_4$  is a voltage amplifier, delivering maximum signals to the parallel resonant circuit  $L_7-C_7$ , which is also tuned to the i.f. value; thus, maximum i.f. signals are transferred to  $L_8-C_8$ .

By now, the combined selective effects of all these circuits should have made the desired signal so much larger than any other signals that little or no interference can occur. We get most of our selectivity in the i.f. amplifier; earlier selector circuits are used principally to keep down certain kinds of interference caused by signals far removed from the one desired.

► The signal is now applied to the second detector; here it is rectified by diode  $VT_5$ , which, together with condenser  $C_{14}$ , causes the intelligence signal to appear across resistor  $R_4$ .

► The intelligence signal variations across  $R_4$  are then passed through  $C_{23}$ , appearing across resistor  $R_5$  in the grid circuit of  $VT_6$ . This voltage amplifier then develops an amplified intelligence signal across  $R_7$ , which is transferred to the grid of  $VT_7$  through  $C_{24}$ . This last tube is arranged to deliver power to the reproducer, which is a loudspeaker in this diagram. (Of course, if this were a television set, the reproducer would be a television image-reconstructor tube.)

► A power supply circuit using tube  $VT_8$  is also shown in this diagram. As you see, this power supply is exactly like the one previously discussed in this lesson.

Fig. 30 thus shows all the basic circuits you will find in a superheterodyne receiver intended for an amplitude modulated signal. Of course, *this is a simplified diagram*; actual sets usually have variations from these basic circuits which give certain desired actions. Furthermore, multi-element tubes are generally used in place of tubes  $VT_1$ ,  $VT_3$ , and  $VT_4$ , and sometimes in place of tube  $VT_7$ . (In fact, unless screen grid or pentode tubes are used as  $VT_1$  and  $VT_4$ , these stages would have to be modified to prevent undesired oscillations from occurring

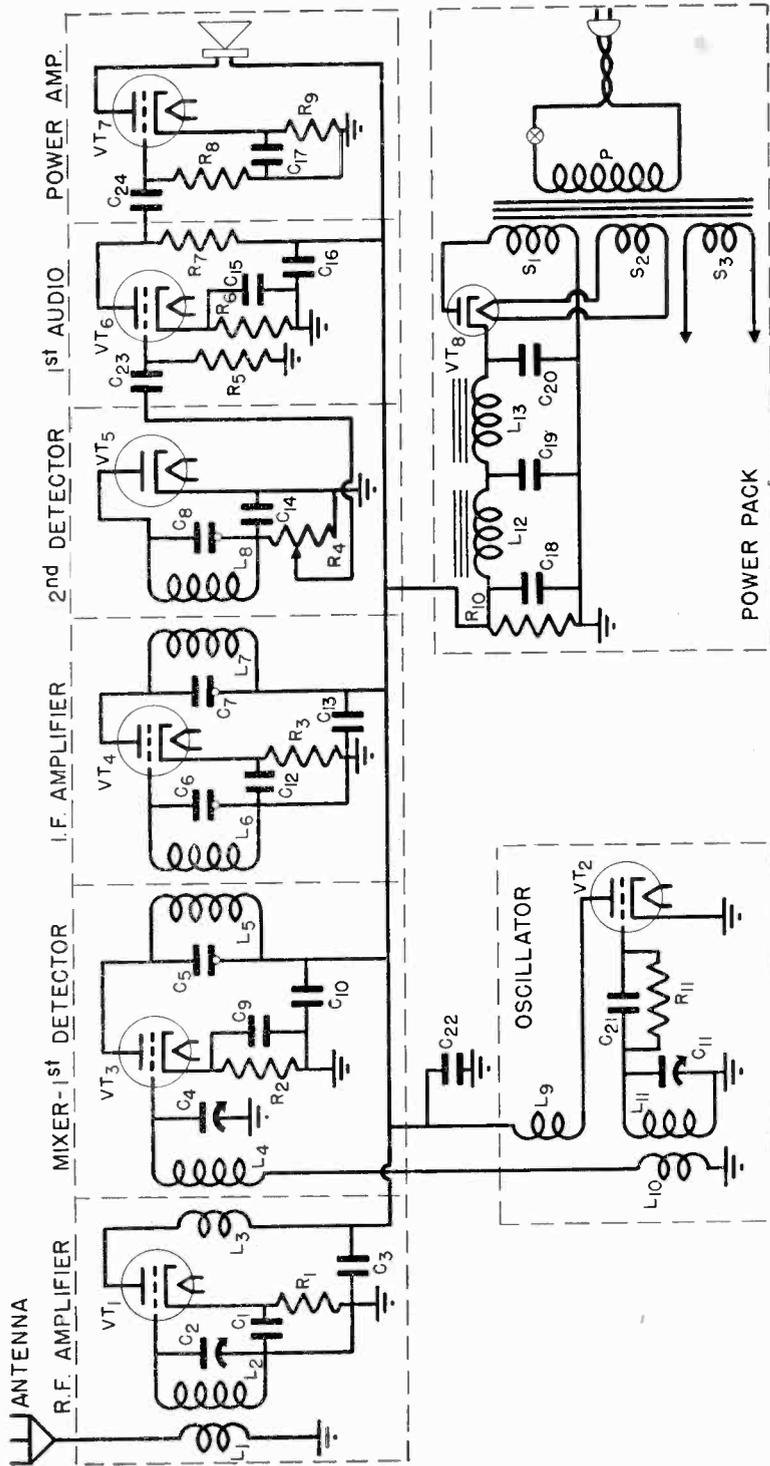


FIG. 30. This is a simplified, basic circuit of a superheterodyne receiver. Of course, the actual circuit would use different tubes and would be somewhat more complex, as you will see later. Also, sometimes the r.f. stage might not be used, and more than one i.f. stage frequently is found. An a.c.-d.c. or battery receiver would use a different power supply. It is important to realize, however, that regardless of the "extras" on the receiver, all superheterodynes can be reduced to a basic circuit like this one. As you read the text, notice how this schematic diagram is drawn to show the orderly progress of the signal from the antenna to the reproducer. However, schematic diagrams are intended to show only the ELECTRICAL connections, not the PHYSICAL positions of parts. Compare this diagram with some of the layout sketches shown elsewhere in this lesson.

within them.) *However, this diagram shows you the relationships between stages and prepares you for the detailed discussions of separate stages that are coming in your next lessons.*

## REVIEWS AND PREVIEWS

Looking back over the lessons you have studied, you will realize that we have gone over the complete radio system more than once. The first time, in an elementary form, we showed how certain radio *parts*—coils, condensers, resistors, and tubes — were found throughout the radio system.

We studied these parts individually, then in small groups and simple circuits. Finally, in this lesson, we have put these parts together into groups that we call *stages*. Now that you have learned the names and purposes of radio stages, and know the order in which radio signals progress from one to another, you are ready to learn the details of operation of each of them.

► Now, let's look at the lessons just ahead. Your next one will be on iron-

core choke coils and iron-core transformers. You have already been introduced to these devices, but they have a number of important characteristics which have much to do with the workings of intelligence frequency amplifiers and power supplies. Once these characteristics are understood, you will plunge right into a study of power supplies, and from there into other stages. Soon, you will have a complete "how and why-it-works" understanding of radio equipment.

Remember, once you understand fully how stages and sections work when *normal*, you will know what to expect when trouble occurs. As you go along, you will see that certain breakdowns produce certain easily recognizable effects. For example, the output of the radio device may be distorted in a special manner, or there may be noise or hum, etc., caused by certain particular breakdowns. This is vital knowledge—the kind that leads you right to the trouble, so you are going to find future lessons more and more interesting and valuable!

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THE N. R. I. COURSE PREPARES YOU TO BECOME A  
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# Lesson Questions

Be sure to number your Answer Sheet 10FR-4.

Place your Student Number on *every* Answer Sheet.

*Send in your set of answers for this lesson immediately after you finish them, as instructed in the Study Schedule. This will give you the greatest possible benefit from our speedy personal grading service.*

1. Briefly state the fundamental principle on which the superheterodyne receiver operates.
2. Which one of the following amplifier classes duplicates the *entire* grid signal in its plate current variations: 1, class A; 2, class B; or 3, class C?
3. What important tube rating tells *directly* how much the grid controls the plate current?
4. When does the plate load in a tube amplifier absorb *maximum power*?
5. Name the three important purposes served by the by-pass condenser  $C_2$  in Fig. 13.
6. If the by-pass condenser  $C_1$  in Fig. 17A short-circuits, does the bias voltage: 1, increase; 2, drop to zero; or 3, remain the same?
7. In the oscillator circuit shown in Fig. 21, which two parts develop the automatic C bias?
8. What are the four important components of an average power pack?
9. Which component of the power pack converts a.c. to pulsating d.c.?
10. What two processes occur in the frequency discriminator stage of an f.m. receiver?

## ENTHUSIASM

Starting work in a new field of endeavor just naturally arouses an intense and eager interest in what you are doing—an enthusiasm which not only makes study and work a pleasure, but also betters the chances for success.

My students have enthusiasm for their Course of training because it is preparing them for a definite goal—an independent business or a good job. They are continually finding, in the lessons studied, explanations for mysteries which they have encountered. Discovering immediate uses for fundamental facts keeps their enthusiasm high.

When enthusiasm is aroused in an *ambitious* man, he reacts like a thoroughbred race horse, giving all the speed and effort in him. If you try to arouse enthusiasm in a mule, however, all he will do is kick!

*A real and lasting enthusiasm for radio will make your study and work as pleasant as play, and will make your life much happier.*

J. E. SMITH

**HOW IRON-CORE COILS AND  
TRANSFORMERS OPERATE  
IN RADIO CIRCUITS**

11FR-2

**NATIONAL RADIO INSTITUTE**

ESTABLISHED 1914

**WASHINGTON, D. C.**



# STUDY SCHEDULE NO. 11

For each study step, read the assigned pages first at your usual speed. Reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind, then answer the Lesson Questions for that step. Study each other step in this same way.

- 1. Magnetic Circuits . . . . . Pages 1-8  
Magnetic circuits can be directly compared to electric circuits; this makes their study easy and interesting. Many of the facts presented here are review, but study them carefully so that you can fully understand how iron cores affect the operation of magnetic devices in which they are used. Answer Lesson Questions 1, 2, 3, 4 and 5.
- 2. Iron-Core Chokes . . . . . Pages 9-11  
Here you learn how air gaps in the core prevent saturation and make it possible to use a choke coil in circuits containing large values of d.c.
- 3. Iron-Core Transformer Fundamentals . . . . . Pages 11-17  
A transformer transfers POWER from one circuit to another. This is a very important fact—it explains why the currents and voltages have the definite relationship you will find described. Be sure to study carefully the practical section on the power transformer. Answer Lesson Question 6.
- 4. Interstage Audio Transformers . . . . . Pages 18-20  
This audio transformer is used for voltage amplification, as well as a coupling device between stages. However, as you will see, fidelity requirements are such that a transformer gain of 3-to-1 is considered good. Answer Lesson Question 7.
- 5. Impedance Matching Transformers . . . . . Pages 21-25  
Transformers handling appreciable amounts of power must be designed for efficiency. Hence, they are designed to match impedances so that a maximum power transfer is possible. In receivers, the output transformer used to match the speaker to the power output tube is the best example. Answer Lesson Question 8.
- 6. Special Transformers—Identifying Types . . . . . Pages 26-28  
After a short section introducing unique transformers, there is a very practical section on identifying transformers and choke coils. Answer Lesson Questions 9 and 10.
- 7. Mail your Answers for this Lesson to N.R.I. for Grading.
- 8. Start Studying the Next Lesson

# HOW IRON-CORE COILS AND TRANSFORMERS OPERATE IN RADIO CIRCUITS

## Magnetic Circuits

AT the low frequencies handled in power supplies and in audio amplifiers, coils and transformers must have high inductances to offer the necessary reactance values. The only practical way of getting such high inductance values is to use cores of magnetic materials, such as iron or alloy steel. (These cores are all commonly known as "iron cores.")

However, the use of an iron core causes inductance devices to have special characteristics that have much to do with the operation of stages in which they are used. For example, the iron cores of audio transformers limit the frequency response or fidelity of audio amplifiers; the iron cores of power transformers prevent them from transferring power with equal efficiency when operated from power lines of different frequencies. It is, therefore, necessary for us to understand the characteristics of these parts so that we may understand the radio stages in which they are used.

► In this lesson, we shall first review a few of the fundamentals about inductances that you learned in earlier lessons. Then, we will consider the general characteristics of magnetic circuits in an iron core. Finally, we will discuss choke coils, power transformers, and audio transformers in detail. In so doing, we will prepare the way for your study of low-frequency stages and power supplies.

### MAGNETIC CIRCUIT FACTS

Magnetic circuits have certain things in common with electric circuits, as you learned in an earlier lesson. A simple electric circuit is shown in Fig. 1A: it consists of a source of electromotive force, and a resistance  $R$  which opposes the flow of current through the electric circuit. Voltage  $E$  and resistance  $R$  together determine the amount of current that will flow. A magnetic circuit likewise has a source, which produces magnetic

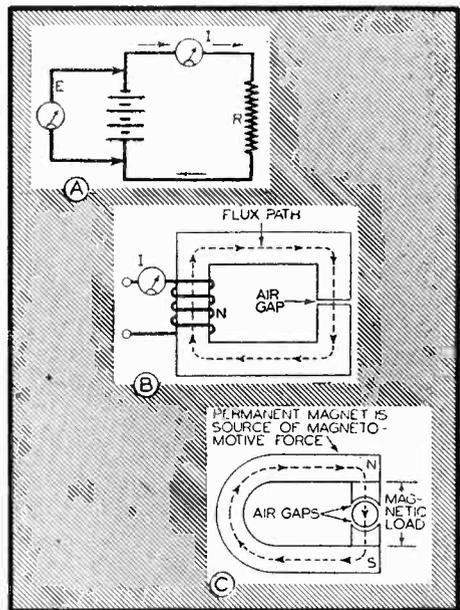


FIG. 1. A comparison between electrical and magnetic circuits proves their actions to be quite similar.

flux, and also has an opposition to the flux. The source of flux is called the *magnetomotive force* (abbreviated m.m.f.), while the *opposition effect* of the magnetic circuit is called its *reluctance*.

One example of a magnetic circuit is shown in Fig. 1B. Here the electric current  $I$ , flowing through a coil of  $N$  turns, creates the magnetomotive force that produces the magnetic flux; the iron core in series with the air gap forms the magnetic circuit that offers a certain opposition or reluctance to this magnetic flux. Actually, the greatest part of the reluctance occurs in the air gap, for iron offers little opposition to magnetic flux. Thus, the iron acts as a "conductor" for the magnetic lines, just as the connecting wires in Fig. 1A act as electrical conductors.

Another example of a magnetic circuit is given in Fig. 1C. The magnetomotive force is produced here by a permanent magnet, and the controlling reluctance in the magnetic circuit is that offered by the two air gaps associated with the cylindrical iron core. Incidentally, this magnetic circuit is typical of those found in direct-current meters.

Thus, you can see that electromotive force (voltage) in an electric circuit is comparable to magnetomotive force in a magnetic circuit, resistance is comparable to reluctance, and current is comparable to magnetic flux.

The same general circuit rules are followed also: increasing the magnetomotive force increases the magnetic flux; decreasing the magnetomotive force decreases the flux. Increasing the *reluctance* of the magnetic circuit by changing some part of its path decreases the magnetic flux; decreasing the reluctance of a magnetic circuit

permits the flux to increase.

### How is Magnetic Flux Measured?

The amount of magnetic flux in a magnetic circuit cannot be measured in the direct fashion we use to measure current in electric circuits. Instruments have been devised that will measure flux, but they are not simple to use; further, a great many measurements usually must be made to discover the total flux flowing, since a single measurement shows only the amount in a small cross-section of the circuit. Fortunately, you need not worry about flux measurements in practical radio work; that is something we can leave to the research scientist. However, the *units* in which flux and the two other magnetic quantities—magnetomotive force and reluctance—are measured are of interest to us, for we shall meet them constantly in discussions of magnetic circuits. Let's take a few moments to see what these units are.

**Units of Force.** You know that an increase either of the number of turns in a coil or of the current through the coil will increase the flux-producing ability (magnetomotive force) of the coil. This magnetomotive force therefore can be expressed in terms of *ampere-turns*.

The *ampere-turn* is the practical unit of magnetomotive force for coils, as it provides a direct means of measuring the magnetomotive force. However, this unit cannot be applied to permanent magnets. For this reason, scientists use the *Gilbert*, a unit of magnetomotive force that applies directly to permanent magnets as well as to coils. To change ampere-turns to gilberts, simply multiply the number of ampere-turns by 1.26; to change gilberts to ampere-turns, multiply the m.m.f. in gilberts by 0.80.

► The magnetomotive force is the

*total force acting throughout* the magnetic circuit. To express the force developed over a portion of the length of the path, a term giving the *magnetomotive force per unit length* is also used. This may be given as gilberts per centimeter. This unit of measurement is called the *magnetic force*, or *magnetizing force*, to distinguish it from the *magnetomotive force*.

**Units of Flux.** The magnetic line of force (used to represent flux) is a unit originally conceived by the great scientist Faraday, and it is retained to this day because of its convenience. The technical name for a magnetic line of force is the *maxwell*. A larger unit, the *kilomaxwell*, is equal to 1000 lines of force or to one *kiloline*.

► *Flux density* indicates how many lines of flux pass through a given unit of area. Flux density therefore can be expressed in terms of lines per square inch, maxwells per square inch, kilolines per square inch or kilomaxwells per square inch. If the unit of area is one square centimeter (metric units) another unit is used; a flux density of one line per square centimeter is known as a *gauss*. A kilogauss is equal to 1000 gauss or 1000 lines of flux per square centimeter.

**Units of Reluctance.** Now we can define the magnetic opposition to the existence of flux in a magnetic circuit. If a circuit has a magnetomotive force of one gilbert and this sets up a flux of one maxwell (one line of force), then scientists say that the magnetic circuit has a reluctance of one *oersted*.

Let us see what controls the reluctance of a magnetic path. Turning back once more to electric circuits for comparison, you will remember that the resistance of the circuit can be increased either by increasing the length of wire path, by decreasing the cross-

sectional area of the wire path, or by making the current flow through wire that has a high resistivity, such as nichrome wire. Similarly, in a magnetic circuit, the reluctance of the path for flux is increased *by increasing the length of the path, by decreasing the cross-sectional area of the path, or by using materials for the path that have higher reluctivity* (higher reluctance per unit volume). Under normal operating conditions, air has much more reluctivity than iron or steel. Different types of iron vary greatly in reluctivity; cast iron is very high and wrought iron is low. ► The comparison to electrical circuits is so close that Kirchhoff's laws for magnetic circuits are much like those for electrical circuits. These laws are:

1. All the flux flowing to a point in a magnetic circuit equals all the flux flowing away from that point.

2. The sum of the reluctance drops in a magnetic circuit must equal the sum of the magnetomotive forces acting in that circuit.

**Permeance of Magnetic Circuits.** In electrical circuits, we express the ability of the circuit to conduct current as its conductance. This rating is expressed in mhos, and is obtained by dividing the number *one* by the circuit resistance in ohms. Similarly, the ability of a magnetic circuit to allow magnetic lines to exist is called its *permeance*. The number *one* divided by the reluctance in oersteds gives the permeance.

## MAGNETIC SATURATION

You know that doubling the source voltage will double the current in an *electric* circuit that contains only resistance. However, you will recall that saturation occurs in a tube, which prevents the current from being exactly

proportional to the voltage when a tube is used. Similarly, it is not always true that doubling the magnetomotive force in a magnetic circuit will double the magnetic flux. Except for air and certain non-magnetic metals, the reluctance of a magnetic path varies with the magnetomotive force acting upon it.

The two curves in Fig. 2 illustrate the difference in action of electric and magnetic circuits. Fig. 2A shows the relation between the voltage applied to a resistor and the current that flows through it. As you would expect, the resultant graph is a straight line.

Fig. 2B is the curve for a magnetic circuit containing iron. This curve shows the relation between the magnetomotive force (m.m.f.) applied to a reluctance and the magnetic flux that flows through it. (The magnetomotive force, the reluctance, and the flux correspond, respectively, to the

in an iron-core circuit, increases in m.m.f. beyond the point of greatest curvature do not produce proportionate increases in flux—in fact, when the horizontal part of the curve is reached, the magnetomotive force can be in-

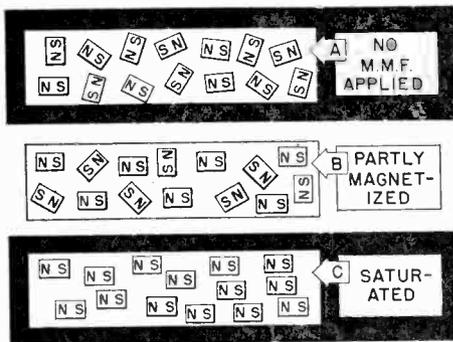


FIG. 3. How the tiny magnets line up in a magnetic material and add their forces to the total applied.

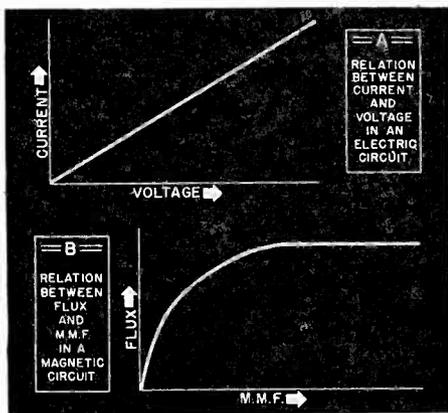


FIG. 2. These curves show how a magnetic circuit differs from a single resistive circuit in that saturation occurs in the magnetic circuit.

voltage, the resistance, and the current in the electric circuit.) Notice—the graph is not a straight line. Instead, it is a curve that gradually becomes horizontal as the magnetomotive force is increased. In other words,

increased greatly without producing more than a small increase in flux.

This condition, in which an increase of the magnetomotive force in a circuit produces little change in the flux, is called “saturation,” and is similar to the saturation action that occurs in tubes. (You will recall that an increase of the filament temperature beyond a certain point does not increase the emission.) Magnetic saturation occurs only in magnetic circuits that contain *magnetic* materials, such as iron or steel; an air-core coil cannot be saturated.

► Scientists explain the phenomenon of magnetic saturation in a very simple way. They say that iron, or any other magnetic material, is made up of millions of tiny individual magnets, which ordinarily are arranged in a random fashion throughout the material as shown in Fig. 3A. When a magnetomotive force is applied to the circuit containing this magnetic material, some of these little atomic

magnets line up parallel to the magnetic lines of force (Fig. 3B), with their North poles all pointing in the same direction. The flux in the circuit then consists of the flux caused by the applied m.m.f., plus the much greater flux caused by the individual magnets; the total circuit flux, therefore, increases tremendously over what it would be in an air-core coil for the same amount of magnetomotive force. If we keep increasing the amount of m.m.f. applied to the circuit, more and more of the individual magnets line up, and the flux continues to increase rapidly. Soon, however, all the individual magnets are lined up (Fig. 3C); they have then added as much to the total flux in the circuit as they can. Then, any further increase in the applied m.m.f. adds to the total flux merely the relatively small amount that an equivalent increase in m.m.f. would add in air. The curve of flux plotted against m.m.f. then becomes almost flat, and we say that saturation has been reached.

► The m.m.f. that must be applied to cause saturation depends on the size, material, and construction of the core. Very often, when we are using an iron-core choke or transformer, we do not want saturation to occur when a normal amount of current flows through the coil. For this reason, transformers and choke coils are often made with air gaps in their cores. An air gap increases the reluctance of the core, so a relatively large current must flow through the coil to cause saturation. (We will say more about air gaps later in this lesson.) Large cores also help prevent saturation, for the larger the core, the more flux it can carry before saturation occurs.

### B-H CURVES

Engineers use graphs—known as

“B-H curves”—much like those in Fig. 4 to show the magnetic characteristics of various materials. These curves are not exactly the same as the curve in Fig. 2B, because the units are different. Flux density—that is, the number of *flux lines per square centimeter* of the material—is plotted along the vertical scale, and magnetizing force (*m.m.f. per unit length*) is plotted along the horizontal scale. Flux density is assigned the symbol  $B$  while *magnetizing force* is given the symbol  $H$ .

In these curves,  $B$  and  $H$  are used

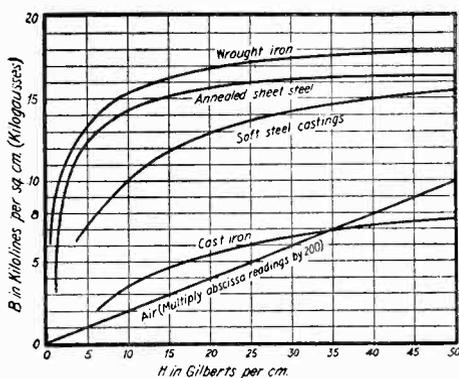


FIG. 4. B-H curves are useful to engineers because they show how various materials differ in their characteristics.

instead of flux and m.m.f. so that the information can be made to apply to cores of any size and also so that the curves will apply only to a particular material (not to an entire magnetic circuit, which may contain an air gap or other materials). These curves make it possible to find the characteristics of the material itself in such a way that the information can be applied to *any* core using that material, whereas the curve shown in Fig. 2 would apply only to a *particular* core or magnetic circuit. ( $B$  and  $H$  are also commonly expressed in English units—flux per square inch and m.m.f.

per cubic inch—as well as in the metric (centimeter) units used in Fig. 4.).

Notice that for air the abscissa (horizontal or  $H$  scale) readings must be multiplied by 200. Thus, to get a  $B$  reading of 5, there must be 5000  $H$  units ( $25 \times 200$ ) for air; this is because air is so much poorer a magnetic conducting material than either iron or steel. The curve for air applies also to non-magnetic materials like brass, copper, wood, bakelite, and fiber. Furthermore, this graph shows that flux density increases uniformly with the magnetic force only for air and non-magnetic materials; all magnetic materials become saturated when subjected to large magnetizing forces. ► Very likely you will never use  $B$ - $H$  curves in your work, as they are intended primarily for the designer of iron-core devices. They are shown here to familiarize you with them, for you will frequently find them mentioned in technical articles.

### IRON CORE LOSSES

So far, we have discussed what might be called “d.c. magnetic circuits”—circuits in which the magnetizing force is supplied by a direct current of steady value flowing through a coil. In radio, however, we are concerned principally with the effects of alternating current flow through coils. As you may suspect, the behavior of coils is considerably more complicated when a.c. is passed through them.

For example, there are certain magnetic losses that every iron-core device has to some extent when a.c. is passed through it—*hysteresis* (pronounced *hiss-ter-E-sis*) losses, *eddy current losses*, and *flux leakage losses*. (This last loss occurs also in d.c.-operated devices, but is much more

important, as you will see, when the device is a.c.-operated.) All of them reduce the efficiency or effectiveness of the device in which they occur, and all can be minimized (but not eliminated altogether) by proper design. As a knowledge of what causes these losses will help you to select low-loss devices for circuits that require their use, let's take a moment to see why these losses occur.

▼ **Hysteresis Losses.** If we send a current through a coil that has an unmagnetized iron core, then reduce the current to zero, the core of the coil will remain magnetized to a certain extent. (This, in fact, is the way permanent magnets are made.) The core is then said to have “residual magnetism.” If we want the core to be completely unmagnetized, we have to do more than reduce the coil current to zero—we must send a small current through the coil in the reverse direction. In other words, we must use up electrical energy to eliminate the residual magnetism of the core.

The reason the core has residual magnetism is that the tiny magnets of which it is composed, once they have been lined up by the magnetic field of the coil current, exert forces on one another that tend to make them stay lined up. Thus, when we reduce the current through the coil to zero, some of these magnets remain lined up and cause some flux to continue to pass through the core. We have to send flux through the core in the reverse direction to turn the magnets from their lined-up positions, thus restoring the core to its original unmagnetized condition.

The property of iron (or other magnetic materials) of retaining magnetism after the magnetizing force is removed is called *hysteresis*. The electrical energy that must be used to

bring the iron back to its original non-magnetic state represents the *hysteresis loss*.

► Any iron-core device fed with a.c. power will waste part of the power in this manner. As you know, the a.c. current in the coil causes the flux to act first in one direction, then in the other. A certain amount of power is converted into heat in restoring the core to its non-magnetic condition each half cycle, so that the next half cycle of flux change may follow.

We cannot eliminate hysteresis loss altogether, but proper choice of materials will minimize it. For example, hard steel retains its magnetism,

link, and the direction in which this induced voltage will act (force a current to flow) is at right angles to the direction of the flux lines.

Now, let's consider the core commonly used in chokes and transformers. For the moment, let's consider the core to be solid, as shown in Fig. 5A. When the flux follows the indicated paths, the iron in the core acts as if it were made up of a great number of parallel rings at right angles to the flux path (that is, *across* the core—see the drawing). The varying flux in the core induces a voltage in each ring; this voltage sends through the ring a current that is

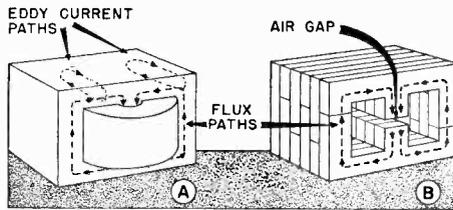


FIG. 5. Notice the large eddy current paths in the solid core. These are reduced in the laminated core.

which is why it is used for permanent magnets. It is, then, a poor material for a transformer or choke core, as a great deal of power would have to be wasted in overcoming hysteresis. On the other hand, soft iron and annealed silicon steel retain very little of their magnetism, so cores made from these materials would have far less hysteresis loss. These are the materials that are generally used in transformer cores.

**Eddy Current Losses.** From your fundamental study of inductances, you know that when a varying current flows through a coil, a varying flux is produced in the coil. This flux variation will induce a voltage in any metallic conductor with which it can

known as an *eddy current*.

This eddy current produces two effects. First, it sets up a flux in the opposite direction to the original magnetic flux and thus reduces the amount of change in the total flux through the core. As a result, the inductance of the iron core coil is lowered.

Even more important—you know that iron has considerable electrical resistance. Therefore, the eddy current flowing in each ring produces a power loss ( $I^2R$ ). This power loss is more important than the loss in inductance, because it, together with the hysteresis loss, wastes power that otherwise could be put to useful work.

► Eddy current losses can be reduced considerably in low-frequency iron-

core devices by constructing the core from thin sheets called *laminations*, as shown in Fig. 5B. (Here the coil is not shown, but it would be in the same position as in Fig. 5A.) These sheets are annealed (softened) in such a way that their surfaces are coated with an iron-oxide scale that is a fairly good insulator. Often the surfaces of the laminations are varnished in addition, to give even better insulation. Insulated in this way, each thin lamination acts like a separate core, and the eddy current rings that form can be no wider than a single lamination. The rings thus become much smaller than in a solid core; less flux can pass through each ring (there is less flux linkage) and therefore less voltage is induced in each ring to cause a power loss.

► Both eddy current and hysteresis losses increase greatly with frequency; this is the reason why even laminated iron cores are of no value for radio frequency purposes. At these frequencies, the only magnetic cores that may be used are made of finely pow-

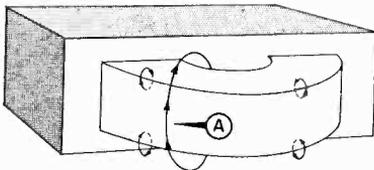


FIG. 6. Leakage lines of force usually fail to link with the entire winding.

dered iron or steel mixed with a clay or plastic binder. The iron or steel particles are so separated that losses are kept low, yet there is enough of this material in the flux path to reduce the reluctance to the point where the inductance is fairly high.

**Flux Leakage Losses.** Since iron is a much better magnetic conductor

than air, most of the flux produced by a coil will flow through the iron core rather than through the surrounding air. However, as indicated in Fig. 6, a certain amount of the flux always takes the high-reluctance air path. (You may compare this to the action of electrical current through a circuit made up of a high and a low resistance in parallel. Most current will flow through the low resistance, but some will go through the high resistance.) Furthermore, as the core approaches saturation, more flux takes these air leakage paths, for the reluctance of a saturated core goes up very rapidly and soon approaches that of the air path.

Where the flux path through the air links the entire coil winding (or windings, when there are more than one) as shown by line A in Fig. 6, complete flux linkage occurs just as if the flux had passed through the core. However, the other air paths in Fig. 6 do not completely link the coil (or coils). The flux that passes through these paths is called "leakage flux," because it "leaks" out of the desired path. Any leakage from the primary winding of a transformer causes less energy to be transferred to the secondary, because *primary leakage flux* does not link with the secondary, so it cannot induce a voltage into the secondary.

This leakage flux is undesirable, both because it reduces efficiency and because it can create interference by inducing voltages in nearby circuits. Design engineers minimize it by designing the iron core to operate well below saturation. Often, also, the coil and core are covered with a soft steel casing so leakage flux will flow through this casing and back to the iron core instead of wandering off through the air.

# Iron-Core Chokes

As you know, the reactance of a coil depends upon both its inductance and the frequency of the applied voltage. If the frequency is low, the inductance of the coil must be high for it to have a fairly high reactance. When we want a high-reactance choke coil in a low-frequency amplifier or a power supply, we must use an iron-core choke, because this is the only kind of choke that can give us the necessary high inductance and still be reasonable in size.

But, while the use of an iron core in a choke solves the problem of getting high reactance at low frequencies, it introduces other problems. In particular, it introduces the problem of saturation—a very real difficulty, because in most practical choke coil applications, d.c. as well as a.c. flows through the choke coil. The d.c. value is frequently high enough to cause saturation or near-saturation of the choke core.

You learned earlier in this lesson that iron-core choke coils very often have an air gap in the core. Now, such a gap increases the reluctance (decreases the flux) in the core considerably, since air has a very much higher reluctance than magnetic materials. Perhaps it seems strange to you that chokes should be built with iron cores to increase the flux through them, only to have the flux reduced by the addition of the air gap, but there is a good reason.

► As you know, the inductance of a coil depends upon the amount of *change* of flux within the coil, rather than upon the total flux present. (Voltages are induced only by variations in flux.) If the flux change for a given change in current is high, the inductance is high; if the flux change

is low, the inductance is low.

► We can use the solid-line curve in Fig. 7 to show what may happen when a mixture of d.c. and a.c. flows through a coil that has no air gap.

Let us first assume that a small amount of d.c. flows, and produces the amount of core flux indicated by point *I* on the curve. Then, let us presume that enough a.c. flows to cause a variation in the flux from point *A* to point *B* on the curve.

Let us now draw a horizontal line from point *A* toward point *C*, and draw a vertical line from point *B* to meet it, as shown in Fig. 7. The length of the horizontal line *A-C* is a measure of the amount of a.c. current change, while the vertical line *B-C* represents the amount of change in the core flux produced by this current change. The longer the vertical line (*B-C*) for a particular line length *A-C*, the greater the amount of flux change for the particular current variation, and hence the greater the inductance.

► Now let's suppose that the d.c. cur-

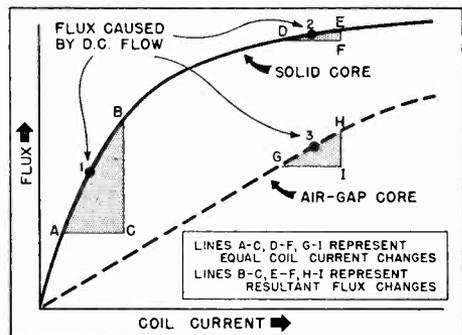


FIG. 7. An air gap reduces the inductance somewhat, but, as a compensation, a magnetic circuit containing an air gap undergoes less change in inductance with d.c. variations; also, a higher current can be applied before saturation occurs. The dotted curve applies to the air-gap core.

rent flow is much greater than that for point 1, so that it produces the coil flux represented by point 2, up in the saturation region of the curve. Let's suppose we have the same amount of a.c. The a.c. will now vary the core flux from point *D* to point *E* on the curve. That is, if we draw horizontal and vertical lines as before, we find that the line *D-F* is the same length as the line *A-C*, so it represents the same amount of a.c. variation. However, when we compare the length of the line *E-F* with the line *B-C*, we find that the *E-F* line is very much shorter. You remember that this vertical line represents the amount of flux change caused by the a.c. Therefore, the increased d.c. flow has, by almost saturating the core, reduced the amount of core flux change that the a.c. can cause. And, since the inductance of the coil depends on the amount of flux change that the a.c. causes, the inductance of the coil has decreased.

► Naturally, we would prefer to operate at point 1, where the coil has its greatest inductance. Unfortunately, the d.c. current in most radio circuits either changes considerably from time to time or is much too large to permit operation at the desired point. Either condition is bad: the first will give us an inductance that varies over a wide range; the second, an inductance that is too low. Therefore, we introduce an air gap into the core to minimize saturation effects, even though we thereby sacrifice some inductance.

► The air gap, as we said, reduces the amount of flux in the core for a given amount of current. This changes the shape of the current-flux curve. For example, an air gap introduced into the core for which the solid-line curve of Fig. 7 is drawn may produce the

current-flux curve shown as a *dotted line* in Fig. 7. Notice—this curve is much more nearly linear (straight) than the solid-line curve that represents the current-flux relationship in a solid core. This means that we will approach the ideal condition far more closely when we operate along the dotted curve, for equal changes in coil current will produce approximately equal changes in flux as long as the d.c. does not cause operation beyond the straight portion of the curve. Since this curve has a longer straight portion, a much higher value of d.c. is permitted.

Let's use these curves to compare the inductance of the air-gap coil with that of the solid-core coil. Say we send through the air-gap coil the same amount of d.c. that caused operation at point 2 on the solid curve. As you see from Fig. 7, this amount of d.c. produces operation at point 3 on the dotted (air-gap) curve. If we use the same size a.c. current variation (*G-I*) that we used for operation on the solid curve, we produce a flux variation *H-I*. This flux variation is considerably greater than the flux variation *E-F* produced by the same current variation at point 2 on the solid curve. Therefore, our air-gap coil has a greater inductance than the solid-core coil has when this amount of d.c. flows through it.

However, the flux change *B-C* produced in the solid-core coil when operated at point 1 is considerably larger than the flux change produced in the air-gap coil by an equal current change. The air-gap coil, then, has less inductance than the solid-core coil when the latter is operated at point 1. In other words, a solid-core choke may have more or less inductance than a similar choke using an air-gap core, depending upon

the amount of d.c. sent through the coil. On the other hand, the air-gap choke will have a fairly constant inductance even with wide variations in the d.c. going through it.

It is quite important that the air gap remain exactly as designed. The space cut into the laminations may be very small. To prevent changes in air-gap spacing, it usually is filled with some non-magnetic material, such as copper or brass strips, cardboard, or paper.

► Even with an air gap, it is possible to saturate the core and reduce the inductance of the coil if we allow enough d.c. to flow through it. For this reason, a normal d.c. flow (often called the *polarizing current*) is specified for chokes by the manufacturer. This flow cannot be exceeded much if the rated inductance of the choke is to be secured. Incidentally, if the d.c. flow is *less* than the specified value, the inductance will be *greater* than its rated value. When speaking of the inductance of an iron-core coil, always remember that this value is obtained only at a specified current value.

► Since a choke coil is always operated on a.c. or on pulsating d.c. (a combination of a.c. and d.c.), it has all the iron-core losses we have previ-

ously discussed. It must be designed to keep hysteresis and eddy current losses low, because it wastes power otherwise. This power produces considerable heat, which may eventually damage the insulation on the coil wire and ruin the choke.

**Swinging Choke.** While most choke coils are designed with an air gap of sufficient size to avoid saturation within their normal operating current ranges, there is a special type that deliberately uses saturation to *obtain* a change in inductance. Since the inductance varies or "swings" with changes in d.c. flow, it is called a *swinging choke*.

We won't go into details of the operation of this choke here, as you will find it fully described in later lessons on power supplies. Briefly, it is used as a kind of voltage regulator in power supplies, it changes in inductance (and therefore in reactance) when the current drawn from the power supply varies, and this changed reactance acts to keep the output voltage of the power supply fairly constant.

This choke is made with a very small, carefully chosen air gap, and is designed for a particular *range* of current values.

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## Iron-Core Transformer Fundamentals

The ability of a varying magnetic field to induce a voltage in any conductor with which it links makes it possible to transfer power from one circuit to another without direct wire connections. Furthermore, by arranging the linkage properly, we can get a step-up or step-down of voltage and can match impedances.

► Suppose we have a coil so placed

that the varying magnetic field produced by another coil links with it. A voltage is induced in the first coil, the amount depending on the flux linkage. This, in turn, depends on the amount of varying flux and on the number of turns of wire on the first coil. (For convenience, the winding connected to the *source* is called the primary, while the other winding, in-

ductively linked to the primary, is called the *secondary*.)

The amount of flux is determined by the design of the primary winding and the primary current. Assuming a fixed current and a particular primary design, the flux *linkage* in the secondary depends on the number of turns of the secondary winding and on the coupling. (With an iron core, the coupling usually is close to unity, so we can assume perfect coupling for now). The more turns, the greater the flux linkage and the greater the voltage induced in the secondary. We can make the secondary voltage higher or lower than the primary voltage by choosing the proper number of turns for the secondary winding. There is, in fact, a very simple relationship between these two voltages:

**The ratio of primary to secondary voltage is the same as the ratio of the turns on these windings.**

► Let us take as an example a transformer, having no losses, with 100 turns of wire in its primary. The secondary of this transformer has 300 turns of wire on it. The ratio of turns on the primary to the turns on the secondary is 1-to-3 (100-to-300), so for every volt applied to the primary there will be three volts from the secondary. Thus, the voltage supplied by the secondary will be three times that fed into the primary. If we apply 10 volts to the primary, the secondary will supply 30 volts. If the primary voltage is 110 volts, the secondary voltage will be 330 volts. This is known as a “step-up” transformer, because the transformer steps up, or increases, the voltage.

(In a practical transformer, there would be many more than 100 turns on the primary and 300 turns on the secondary. However, the *ratio* of the turns may be 1 to 3. The actual

number of primary turns depends on the intended use. In selecting the number of turns, the primary voltage and frequency, the type of circuit, the size of the core, and the secondary voltage and current requirements must all be considered.)

► This action can be reversed by using fewer turns on the secondary than on the primary. If our transformer has 10 turns on the secondary and 100 turns on the primary, the ratio of primary turns to secondary turns is 10-to-1 (100-to-10). The secondary, having fewer turns than the primary, will have a lower voltage; in fact, the secondary output voltage will be one-tenth the voltage applied to the primary. Therefore, if we apply 10 volts to the primary we will get 1 volt from the secondary, while 110 volts fed into the primary will give an 11-volt output from the secondary. Such a transformer is known as a “step-down” transformer, because it steps down, or reduces, the applied voltage.

Thus, a transformer can deliver a higher or a lower voltage from a source to another circuit.

► Here is a practical hint about turns ratios. It is common practice to give the ratio with the larger number first, then to state whether it is a step-up or a step-down ratio. For example, a transformer rated as a 5-to-1 *step-up* type delivers 5 times the primary voltage from the secondary, while a 5-to-1 *step-down* transformer delivers 1/5 the primary voltage from its secondary.

## POWER TRANSFER

The relationship between primary voltage and secondary voltage is only one of the things you need to know to understand transformer action. We will now show that the relationship between primary and secondary *cur-*

rents is equally as important.

A transformer must always be considered as a *power transferring device*. Hence, the current flow in the primary winding depends on the power taken by the load. We'll use the circuit in Fig. 8 as an example. Here we have a 5-to-1 step-up transformer, which raises the 10-volt source value to 50 volts across  $R$ . Let us suppose  $R$  has a value of 25 ohms. The secondary current  $I_s$  will be 2 amperes ( $I_s = E_s \div R$ ), and the power dissipated will be 100 watts ( $P = I_s \times E_s = 2 \times 50$ ).

To supply the load, the transformer primary must draw enough current to take 100 watts from the 10-volt

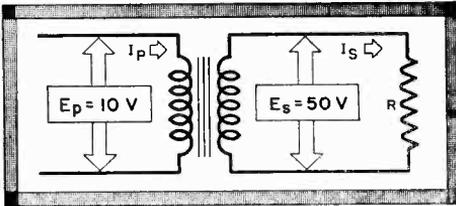


FIG. 8. A step-up transformer.

source. Hence, the primary current  $I_p$  must be 10 amperes ( $I_p = P \div E_p = 100 \div 10$ ).

Compare the primary voltage and current with the secondary voltage and current. The *voltage* is stepped-up from 10 volts to 50 volts, a step-up ratio of 5-to-1. On the other hand, the *current* is stepped-down from 10 amps. to 2 amps., which is a step-down ratio of 1-to-5.

► Similarly, when we have a step-down transformer, the secondary voltage is less than that of the primary, but the secondary current is greater. For example, suppose we apply 110 volts to the primary of a 10-to-1 step-down transformer; the secondary delivers 11 volts ( $110 \div 10$ ). Say the secondary power demand is 550 watts;

the secondary current will be:

$$I_s = P \div E_s = 550 \div 11 = 50 \text{ amperes.}$$

Then, the primary current will be:

$$I_p = P \div E_p = 550 \div 110 = 5 \text{ amperes.}$$

Thus, the voltage goes down from 110 to 11 volts (10-to-1) while the current comes up from 5 to 50 amperes (1-to-10).

► These facts are true because a transformer must transfer *power* from one circuit to another, and *it takes from the source the amount of power required by the load*.

The *name* of the transformer (step-up or step-down) comes from what happens to the *voltage*—not to the current. Thus: if the *secondary voltage is higher than the primary voltage*, the transformer is a *step-up* type; if the *secondary voltage is lower than the primary voltage*, the transformer is a *step-down* type. The **CURRENT** relationships are opposite; that is, in the step-up transformer, the secondary current is less than the primary current, whereas in a step-down transformer, the secondary current is greater than the primary current. As a summary:

**If the voltage is stepped up, the current is stepped down; if the voltage is stepped down, the current is stepped up; and the product of the current and the voltage (current multiplied by the voltage) is the same for both the primary and the secondary.**

**Losses.** Of course, we don't actually get a 100% transfer of power. You should keep in mind that there are losses in all transformers caused by eddy currents and hysteresis in the core, which waste power in heat. Also, there are losses in the coils caused by the flow of current through the resistance of the windings. You may see

this loss referred to as  $I^2R$  loss or "copper" loss. This also takes the form of heat, and varies as the load is varied: the greater the load, the greater the power loss.

► These losses explain why 100% transfer of power from the primary to the secondary cannot be obtained. Actually, the primary source must furnish more power than is taken by the secondary load, because it must supply a total power equal to the load demand plus the losses. As a result, the ratio of primary to secondary currents will not be *exactly* in the ratio of the turns. However, in practice, you can assume the ratios to be nearly correct, as power transfers of 95% to 97% are obtainable in iron-core devices.

Although a 3% to 5% loss seems small, it does limit the amount of power we can transfer through the transformer. The power dissipated in heat increases greatly as the load is increased, so that eventually a point is reached at which the transformer becomes overheated and subject to breakdown. Power transformers are rated according to power-handling ability as well as to voltages and currents delivered. The larger the wire used for the windings (to reduce resistance) and the larger the iron core (to reduce hysteresis and eddy current losses), the more power a transformer can handle before breakdown occurs. This is why transformers having high power ratings are so large and heavy. The weight of a transformer is a good indication of its power-handling ability.

## POWER TRANSFORMERS

The power transformer in a radio receiver is an excellent example of both step-up and step-down transformer actions.

Homes in this country usually are wired so that power outlets deliver 110 to 115 volts a.c. It is highly desirable to obtain all voltages required for a radio receiver from this power line; this means that we must have low a.c. voltages for the filaments and heaters of the tubes, and high a.c. voltages that can be converted into high d.c. voltages for the various tube electrodes. The line voltage value must be stepped-down (decreased) in one case and stepped-up (increased) in the other case. A single power transformer is used for this purpose.

While we won't go into the design of a power transformer, it is interesting to know that the engineer considers many factors. Cost, weight, and efficiency are of particular importance. He must choose first a suitable core material, and then must choose a core size that will give him the required characteristics in a minimum of space and weight. There must be a compromise between core characteristics and the number of primary turns. For example, when a transformer is not loaded (nothing is connected to its secondary winding) the flux changes in the core must induce a back e.m.f. in the primary nearly equal to the applied voltage. Otherwise, the primary current (which depends on the difference between the applied voltage and the back e.m.f.) would be excessive. For a given flux density in the core, a given core area, and power line frequency, the back e.m.f. will depend on the turns on the primary.

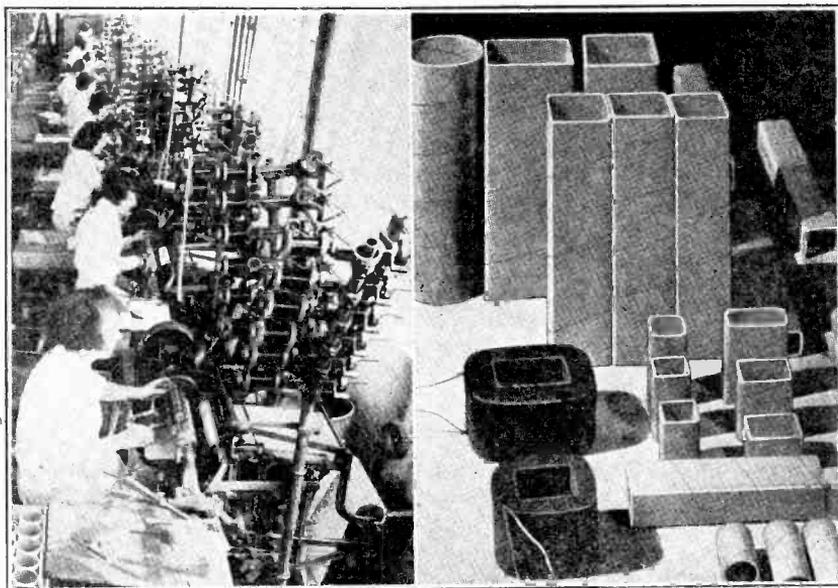
Therefore, the designer must make a compromise. If he increases the core size, the number of turns in the primary can be made less, but the cost of the transformer, its size, and its weight all increase. Increasing the turns reduces the required core size,

but there is a minimum core size too—if the core is too small, saturation may occur before the required flux density is produced.

After selecting a core, the designer computes the number of primary turns that will allow only enough primary current to flow to keep the flux density in the core at the desired value when no load is on the secondary. He then divides this number of turns by the applied line volt-

primary for each volt applied to the primary winding.

After finding this ratio for the primary, the number of turns required on the secondary winding to secure a certain voltage can be determined by multiplying this desired voltage by the turns-per-volt ratio of the primary. For example, if the primary has a ratio of 10 turns-per-volt, and 500 volts is desired from the secondary winding, the secondary winding



*Courtesy Universal Winding Co.*

At the left—this view shows the automatic coil-winding machinery used in the manufacture of chokes and transformers. Six or more coils are wound on a single spiral-wrapped paper form like one of those shown at the right; when winding is completed and the operator has anchored the lead wires, the machine automatically cuts the coils apart. Operators perform only the starting and finishing operations; the machines do the rest, placing layers of insulating paper between each layer of a winding and stopping automatically when the correct number of turns has been wound. Samples of finished coils produced by machines like these can be seen in the foreground in the right-hand photo. The iron core is stacked inside the coil (or coils) to complete the unit. Then, the choke or transformer may be dipped in a moisture-excluding sealing compound and may be enclosed in an iron shield can.

age to find the *turns-per-volt* ratio of the primary. For example, he may find that he has wound 575 turns in his primary winding for a line voltage of 115 volts. Dividing 575 by 115, he finds that he has five turns on the pri-

will require 500 times 10, or 5000 turns. The turns-per-volt ratio of a power transformer varies considerably with the design of the transformer, since, as we have indicated, it depends on the size of the core, the core ma-

terial, the power line frequency, and the desired flux density.

**Turns Ratio.** Although designers of power transformers think in terms of turns-per-volt, you will undoubtedly follow the practice of other transformer users and think of power transformers only in terms of what they can deliver. In other words, you are interested solely in the voltages that may be obtained from each secondary and in the respective current ratings of the secondary windings.

As you know, the output voltage is dependent upon the turns ratio—how many times more turns there are on one winding than on another. The turns ratio times the primary voltage will give you the secondary voltage (provided, of course, you know whether you have a step-up or step-down transformer action in that particular winding). Remember, there will be a different turns ratio for each secondary delivering a different voltage.

► Should it ever prove necessary, you can measure the turns ratio of a power transformer by connecting it to an a.c. source of known value and measuring the voltage across each winding. The larger voltage divided by the smaller voltage gives you the turns ratio. The winding giving the higher voltage will have the more turns of wire.

► *A word of caution*—before you connect a power transformer to a voltage source, be sure that the windings will handle that voltage safely, and be sure the frequency of the power source is correct for the power transformer. Otherwise, the current flow may be excessive and the transformer may heat up so much it will be seriously damaged. (You can connect a winding to a source producing a

voltage *lower* than its rated value, but never to a voltage source that is considerably higher than its rated voltage.)

*Therefore, the first thing you always want to know about a transformer is its primary voltage rating and the frequency for which it is intended.* The operating voltages for a power transformer are given by the manufacturer. Most American power transformers are designed for use on 110-115 volt a.c. power lines, though some are designed for 220-230 volt power lines. (You are most apt to find these latter in sets intended for export to foreign countries, where these power line voltages are more common.) Occasionally, too, you may find a transformer especially designed for some unusual power line voltage.

Transformers are usually designed for 25- or 60-cycle operation. Design compromises allow 25-cycle power transformers to operate on 25- and 40-cycle power lines, while 60-cycle transformers will operate on 50- and 60-cycle power lines. However, a 60-cycle transformer will overheat if it is connected to a 25-cycle power line, whereas a 25-cycle power transformer will tend to be overly efficient, giving higher-than-rated output voltages, if connected to a 60-cycle power line.

Never connect a power transformer to a *D.C.* source. On d.c., only the primary *resistance* limits current flow so the current flow is excessive. The transformer is certain to be damaged or destroyed.

► After making sure the transformer will work properly on the available power line, you will next want to know the voltage and current ratings of the various secondaries. When we take up power supplies, you will learn that a number of different secondary *voltages* are required. The values

needed depend principally on the operating voltages required by the tubes used in the receiver and on the manner in which the tubes are connected (series or parallel connections between filaments).

Each secondary winding will have its own *current* rating, which is the *maximum* current that can be taken from the winding for long periods of time without overheating the transformer. Of course, less current than this can be drawn if desired. Thus, if a filament winding is rated at 2.5 amperes, it can deliver safely any current value up to 2.5 amperes. The current rating needed depends on the number and types of tubes to be supplied.

It is unnecessary to calculate the power handled by a transformer in a receiver as long as you make sure its individual windings have the proper voltage and current ratings. Remember, though, that when a power transformer has several secondary windings, the primary winding supplies the power required for all of them, plus the power wasted in the eddy current, hysteresis, and  $I^2R$  (or copper) losses of the transformer itself.

#### REQUIREMENTS OF AUDIO-FREQUENCY TRANSFORMERS

A power transformer is designed to work at one frequency—that of the power line for which it is intended. A transformer designed to handle audio frequencies must handle a *band* of frequencies instead of a single frequency, and it is supposed to operate over this band without discrimination—that is, it should deliver a voltage proportional to the turns ratio, regardless of frequency, within its designed range. As you will soon see,

these requirements are far more difficult to meet than those imposed on power transformers.

**Audio Frequencies.** The a.c. voltages handled by audio stages correspond to sound waves, and so have the same frequencies as ordinary sound waves. However, although sound frequencies range from 30 cycles per second to about 20,000 cycles per second, good *voice* reproduction requires only frequencies from about 300 to 3000 cycles, and good musical reproduction requires only frequencies from about 60 to 8000 cycles. Thus, an audio transformer does not have to handle the entire range of sound frequencies. In fact, in ordinary receivers it has to handle only frequencies from about 100 cycles to near 5000 cycles, for cost limitations prevent such receivers from having a wider range. However, even this limited range creates a good many problems. We'll take these up in just a moment, but, before we do, let's see what are the two main types of audio transformers.

► Audio transformers are classified according to whether they are used to give voltage gain or to match impedances. The types designed for voltage gain are called interstage transformers, as they are generally used between two vacuum tube stages. Those intended to handle audio power are usually called impedance-matching transformers. In radio receivers, the most common use for such an audio transformer is to connect the power amplifier tube to the loudspeaker. (It is common practice to call this transformer the output transformer.)

Now, let's take up each type in detail and see what problems they will give you in servicing a receiver.

# Interstage Audio Transformers

Interstage transformers are normally designed to have a step-up ratio between the primary and secondary windings, so that there will be a gain in signal voltage. These transformers are connected between the source and some voltage-operated circuit that does not require power, so the secondary current flow is negligible. A typical "load" would be a grid circuit, where only voltage is needed for operation. In effect, then, the secondary winding feeds into an open circuit (or, as radio men say, is not "loaded").

To provide a signal voltage step-up, the interstage transformer should have the highest possible turns ratio. However, the requirements imposed by the wide audio frequency band limits the amount of the turns ratio. Some, at a sacrifice of fidelity, go as high as 5-to-1, but most stay down around 2-to-1 or 3-to-1. Compare this with power transformers, which in high voltage work can easily have a 30-to-1 step-up ratio! Let's see what factors set these limits.

## PRIMARY INDUCTANCE

A typical interstage transformer connection is shown in Fig. 9. Since no power is transferred, the transformer, by itself, could give us a primary to secondary voltage step-up that would be almost constant over a rather wide band of frequencies. However, there are several factors that limit the frequency response of the whole circuit.

One of these is the fact that the a.c. plate voltage ( $\mu e_g$ ) of tube  $VT_1$  divides between the a.c. plate resistance  $r_p$  of the tube and the reactance of the primary of the transformer. The plate resistance is almost con-

stant at audio frequencies, but the primary winding, since it is an inductance, has a reactance that increases with frequency. Therefore, the proportion of the a.c. plate voltage that appears across the primary depends on the frequency; more appears across it at high frequencies, less at low frequencies. The only way we can keep this from causing serious distortion is to make the reactance of the primary fairly high with respect to the a.c. resistance of the tube at all frequencies for which the system is designed. Then we will always have a reasonable proportion of the signal voltage appearing across the primary. There will be less at low frequencies than at high frequencies, but there will be a fair amount at all frequencies in the design range.

This means that we must have a high-inductance primary—one with

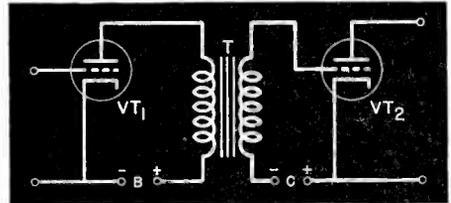


FIG. 9. Typical connections for an interstage transformer.

many turns. The higher the plate resistance of the tube, the more inductance we need in the primary to make its reactance comparable to the plate resistance. As a practical matter, since the amount of inductance we can have in the primary is limited, we have to use a tube that has a low plate resistance; otherwise, the primary inductance would have to be

impossibly large. Generally speaking, only triodes have sufficiently low plate resistances; transformers are used rarely with other tubes.

► The inductance of the primary can be made high by using a large iron core or by using many turns of wire on the primary. However, the use of a large core increases the cost, weight, and size of the transformer greatly. All these factors must be considered in radio equipment, so the necessary high inductance is usually obtained by using a large number of primary turns. There are limits here, too, when voltage gain is desired. If the primary has many turns, a high step-up turns ratio would mean thousands

tiny capacity exists between the turns of wire on a transformer winding, and a somewhat larger capacity exists between layers of wire when they are wound like thread on a spool. These capacities add up; the more turns and the more layers, the greater the capacity. This winding capacity is called *distributed capacity*, from the manner in which it is distributed throughout the transformer.

The distributed capacities can be lumped together and represented by a single capacity across each transformer winding, as shown in Fig. 10. In addition,  $C_P$  and  $C_S$  can be used to represent any stray capacities between the circuit wires, as well as in-

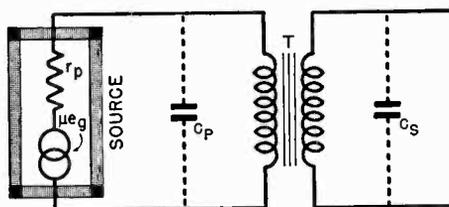


FIG. 10. Notice how the primary distributed capacity  $C_P$  acts as a voltage divider with  $r_p$ .

of secondary turns, requiring a great amount of wire and a large core. Thus, a transformer designed to have a good low-frequency response and a high turns ratio at the same time would have to be very large and expensive.

### DISTRIBUTED CAPACITY

At times, it may seem worth the price to get a transformer with a high turns ratio. But the very fact that there are a large number of turns introduces another factor that limits the frequency range such a transformer can handle.

As you know, a capacity exists between any two conductors that are separated by an insulator. A very

terrestrial tube capacities.

As you can see from Fig. 10, the primary capacity  $C_p$  forms a voltage divider with the plate resistance  $r_p$ . The reactance of a capacity goes down as the frequency goes up. Therefore, as higher audio frequencies are fed into the circuit, more and more of the source voltage will be dropped in the plate resistance of the tube instead of appearing across the primary of the transformer  $T$ . In fact, at very high frequencies,  $C_p$  can actually act as a short-circuit across the primary.

► Careful design can remove some of the undesirable effects of  $C_p$ . For example, this capacity and the inductance of the primary winding can be so arranged that they form a parallel

resonant circuit that has a very high impedance at its resonant frequency. If this is done, practically all the a.c. voltage  $\mu e_g$  that is at or near the resonant frequency will appear across the primary instead of being dropped in the tube resistance. However, at frequencies above this resonant point, the circuit will cut off sharply.

But even the best design cannot eliminate distributed capacity altogether; and, unfortunately, it is always true that attempts to increase the inductance by winding on more turns will increase the distributed capacity also. In other words, we can increase the gain of a transformer only by decreasing its fidelity (its ability to transfer all frequencies equally well). Again, we must compromise. We can have a transformer that has high gain and low fidelity, or low gain and high fidelity, or medium gain and medium fidelity, but we can very seldom get both high gain and high fidelity.

### SATURATION

When a transformer primary is connected into the plate circuit of a vacuum tube, we introduce the problem of saturation. The d.c. plate current of the tube must flow through this primary; if this current is too large, it will produce core saturation and reduce the primary inductance greatly. For this reason, the amount of permissible d.c. current flow is specified for each transformer, and the air gap of the core is fixed for this value.

### SUMMARY

Obviously, an interstage trans-

former is carefully designed. You can appreciate why the high-fidelity types are bulky and very expensive—they have to be wound in special ways to control the distributed capacity and get the proper inductance. Even with special design, it is impractical to have a high turns ratio and have a high fidelity response at the same time, because there are limits to the amount of correction that is economical. Therefore, all high-fidelity interstage transformers will have a *low* step-up turns ratio; usually it will be less than 3-to-1.

Today, high-gain tubes are available, and the increased gain from these tubes has made the voltage amplification of the interstage transformer unnecessary. This fact, plus the shortcomings of interstage transformers, have together practically eliminated this style of transformer from the average receiver. Other coupling methods offering higher fidelity are used instead. However, interstage transformers still are used where gain is important; where some special circuit action is desired; and where weight and cost are not vital factors.

► In your work, you will have occasion to replace these transformers as they become defective. Do not put in just *any* replacement; you must choose a transformer having characteristics similar to the original. Fortunately, transformers are standardized to a great extent so it is possible to get a duplicate or an acceptable substitute from the listings of supply catalogs. In these, you will find transformers listed according to the model number of the sets or other equipment in which they can be used; thus, it is easy to select the proper replacement.

# Impedance-Matching A. F. Transformers

The entire object of amplification in a radio is to get enough signal power to operate the output device—whether it be a loudspeaker, a relay, a recorder, or something else. As the power needed is relatively small (1 to 20 watts in the average radio), a single power output stage is all that is needed, provided we can amplify the signal voltage enough to operate this output stage. For this reason, voltage amplifiers are used ahead of the output stage to deliver to it the required grid voltage. (The output stage may use two tubes, as you will learn later, but only one stage of power

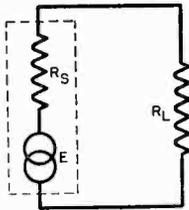


FIG. 11. This basic circuit is used to prove the necessity for impedance matching when the maximum power output is wanted.

amplification is common today.)

► In the power stage, a reasonable efficiency must be obtained so that the output actually delivered will be sufficiently large without requiring too much B-supply power. We can get the greatest transfer of power to the load when the load impedance equals the tube a.c. plate resistance, but it rarely happens that the actual load (the loudspeaker in a receiver) comes anywhere near the tube plate resistance value. However, we can use an impedance-matching transformer to make the two values *appear* alike so that the maximum power output can be obtained.

► Before we learn how this is done,

let's review the need for impedance matching by considering a few examples.

Suppose we have the circuit shown in Fig. 11, where the source resistance is  $R_S$  and the load is  $R_L$ . Let's assume that  $E$  is 12 volts and  $R_S$  is 2 ohms, then let us vary  $R_L$ .

**Case 1.** First, let's make  $R_L$  equal 1 ohm. Then:

$$I = \frac{E}{R_S + R_L} = \frac{12}{2 + 1} = \frac{12}{3} = 4 \text{ amps.}$$

The voltage across  $R_L$  is then:

$$E_{RL} = I \times R_L = 4 \times 1 = 4 \text{ volts.}$$

Now, the power developed in  $R_L$  is:

$$P_{RL} = I \times E_{RL} = 4 \times 4 = 16 \text{ watts.}$$

Thus, when  $R_L$  is 1 ohm, 16 watts will be delivered to it under the assumed conditions.

**Case 2.** Let us now make  $R_L$  equal 2 ohms and go through the same procedure to find the power output.

$$I = \frac{E}{R_S + R_L} = \frac{12}{2 + 2} = \frac{12}{4} = 3 \text{ amps.}$$

$$E_{RL} = I \times R_L = 3 \times 2 = 6 \text{ volts.}$$

$$P_{RL} = I \times E_{RL} = 3 \times 6 = 18 \text{ watts.}$$

This is more power than that of Case 1.

**Case 3.** Let us now make  $R_L$  equal 3 ohms and find the power output.

$$I = \frac{E}{R_S + R_L} = \frac{12}{2 + 3} = \frac{12}{5} = 2.4 \text{ amps.}$$

$$E_{RL} = I \times R_L = 2.4 \times 3 = 7.2 \text{ volts.}$$

$$P_{RL} = I \times E_{RL} = 2.4 \times 7.2 = 17.28 \text{ watts.}$$

► We can now compare the three conditions: Case 1 has  $R_L$  less than  $R_S$ ; Case 2 has  $R_L$  equal to  $R_S$ ; and Case 3 has  $R_L$  greater than  $R_S$ . The circuit current drops as  $R_L$  is increased, while the voltage across  $R_L$  rises as  $R_L$  is increased, as we would expect. Notice the power, however. It is *greatest*

when  $R_L$  equals  $R_s$ , and is less than this *maximum* if  $R_L$  is made *either larger or smaller*. Thus, we get a rule: **The maximum power output is obtained when the source and load impedances are matched or equal.**

As a further example of the need for matching, suppose we have a circuit like that shown in Fig. 12A, with a load connected directly into the plate circuit of a tube. When this circuit is drawn in the equivalent form

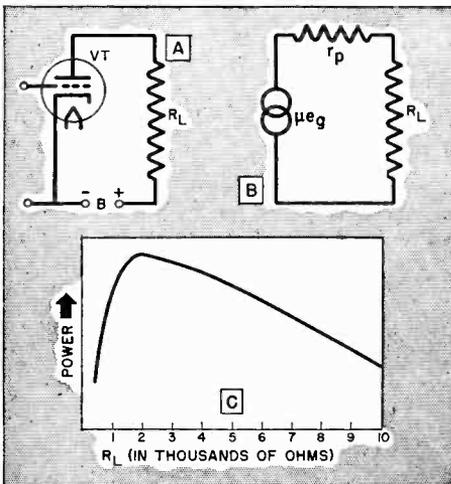


FIG. 12. When the load equals the 2000-ohm plate resistance, maximum power output is obtained.

shown in Fig. 12B,  $R_L$  represents the load resistance, the generator  $\mu e_g$  represents the a.c. plate signal voltage, and  $r_p$  represents the plate resistance of the tube. Let us suppose  $r_p$  equals 2000 ohms and that  $R_L$  is variable, and let's see how the power absorbed by  $R_L$  varies as its resistance is increased from, say, 500 ohms to 10,000 ohms.

The curve in Fig. 12C shows the power developed across the load under these conditions. Notice again that the maximum power is developed when the load resistance and the plate

resistance of the tube are equal (or *matched*). If they are not exactly equal, considerable power still will be developed across the load—but if they are far different, particularly if the load resistance is much less than the plate resistance, then the power developed across the load will be reduced greatly.

► This at once introduces a problem, as the load is rarely equal to the source value in radio circuits. For example, suppose we have the following problem: A triode output tube having a plate impedance of 6400 ohms is to supply power to a 4-ohm speaker voice coil. Obviously, from what you've just learned, these impedances are so poorly matched that very little power will be developed across the speaker coil. We can, however, apply maximum power to the speaker if we couple it to the tube through an impedance-matching transformer.

To see why a transformer can match (that is, equalize) impedances, let us go back to the previously explained power transfer effect of a transformer. Let us suppose the 4-ohm voice coil has 1 ampere of current flowing through it. Its voltage will then be 4 volts and the power it absorbs will be 4 watts. The primary of the impedance-matching transformer must take this power from the source.

Turning now to the tube, we can assume that its a.c. plate voltage will be high. In fact, taking a typical example, the a.c. grid voltage  $e_g$  may well be as much as 30 volts, so if the tube  $\mu$  is 6, there could be as much as 180 volts ( $30 \times 6$ ) in the plate circuit. Clearly, we need a step-down transformer, as the load only requires 4 volts.

Let's assume we have a step-down transformer with a ratio of 40 to 1.

(A step-down transformer can have a far larger turns ratio than a step-up type as it requires fewer secondary turns—not more.) As you know, the ratio of primary to secondary voltage depends on the turns ratio, so, with a 40-to-1 transformer and a requirement of 4 volts across the secondary, the primary voltage must be at least  $4 \times 40$ , or 160 volts. Under this condition, the load is absorbing 4 watts (at least) from the source. We can now find the primary current ( $I_p = P \div E$ ). This gives us  $4 \div 160$ , or .025 ampere.

► Now, knowing the voltage across the primary and the current that must pass, we can find the primary impedance. From Ohm's Law,  $Z = E_p \div I_p = 160 \div .025 = 6400$  ohms. At the beginning, we assumed a tube impedance of 6400 ohms, so the tube and the effective primary value are matched or equal. Thus, when the connections are as shown in Fig. 13, the tube "sees" the 4-ohm loudspeaker through a 40-to-1 transformer as a load of 6400 ohms, so it delivers the maximum power to the transformer for transfer to the speaker.

► Of course, we assumed the proper transformer in our example. If we use some other turns ratio or a different load value, we will have a different primary matching value. Thus, if we use a 30-to-1 transformer with a 4-ohm voice coil, we would have a primary impedance of 3600 ohms, while a 6-ohm voice coil with the 40-to-1 transformer would offer an impedance of 9600 ohms to the tube. Thus, regardless of the tube's plate impedance or the load impedance, they can be matched if a transformer of the correct turns ratio is selected.

► In your service work, you won't have to figure the turns ratio needed to match a tube to a load, as the

manufacturers list transformers for the tubes and speakers (or other loads) with which they will work properly. You can find the proper replacement transformer just by looking up the right tube and load. However, if you are interested in figures, the footnote \* below gives the method of calculating the turns ratio.

Radio men often call the effective impedance of a primary when a load is connected to the secondary the "reflected impedance" of the load. Thus, in our example, the 4-ohm load caused the primary of a 40-to-1 transformer to have an effective impedance of

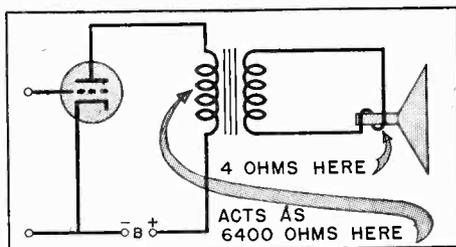


FIG. 13. The impedance-matching transformer causes the load to act in its primary circuit as the required impedance value.

6400 ohms. This 6400-ohm figure is called the "reflected impedance" of

\* Dividing the source impedance by the load impedance gives a number that is the same as that found by multiplying the turns ratio by itself. That is,  $N \times N$ , or  $N^2 = Z_A \div Z_L$ . From this, the turns ratio is equal to the square root of the result of dividing the source impedance by the load impedance. Expressed as a formula:

$$N = \sqrt{\frac{Z_A}{Z_L}} \quad \text{where}$$

$Z_A$  is the source impedance (connected to the primary),  $Z_L$  is the load impedance (connected to the secondary), and  $N$  is the ratio of primary-to-secondary turns.

Using the figures of our example, we have:

$$\sqrt{\frac{6400}{4}} = \sqrt{1600} = 40, \text{ so}$$

the 40-to-1 ratio is the proper one for this particular tube and speaker.

the 4-ohm load. It is equal to the load impedance multiplied by the square of the turns ratio.

► As you will learn in another lesson, an exact match of power tube to load will not give the greatest *undistorted* power output. In fact, the maximum *undistorted* power is obtained when a triode tube feeds into a load *twice* its plate impedance, while with a pentode, a value near one-seventh its impedance is the best value. Transformer lists take these facts into consideration; when you select a replacement from a list, it will give maximum undistorted power for the particular tube and load for which it is intended.

### FIDELITY FACTORS

The fidelity, or frequency response, of a stage containing an impedance-matching transformer is affected by three factors with which the transformer itself is directly concerned. These are *the design of the primary, flux leakage, and distributed capacity.*

### PRIMARY INDUCTANCE

High primary inductance is just as important in an impedance-matching transformer as it is in an interstage transformer. This will be easier to understand by referring to Fig. 14,

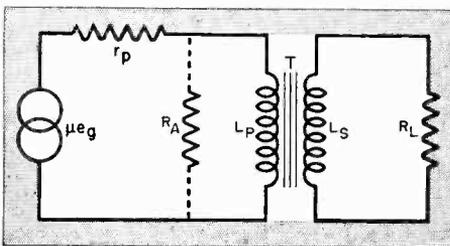


FIG. 14. The load "reflects" in the primary circuit as  $R_A$ . The primary winding must have a high reactance at the desired frequencies so that it will not reduce the effective load value.

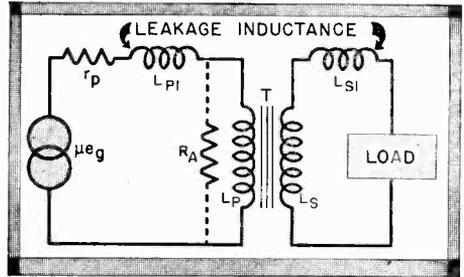


FIG. 15. The turns not used to actually transfer power are represented here as leakage inductances.

which is the equivalent circuit for Fig. 13. In effect, the transformer "reflects" the value of  $R_L$  into the plate circuit as  $R_A$ —a value which is equal to  $R_L$  times the square of the turns ratio of the transformer.

As you will notice,  $R_A$  is effectively in parallel with  $L_P$ . When the primary reactance is very high compared to  $R_A$ , the parallel combination effectively has the value of  $R_A$ , so the tube feeds into the impedance for which the circuit is designed. We then say that the reactance of  $L_P$  has "disappeared," because it has no effect on the circuit.

However, as you know, inductive reactance goes down when frequency is decreased. Therefore, at low frequencies, it is quite possible for the reactance of  $L_P$  to be a rather low value compared with  $R_A$ . When this happens, the effective load is reduced; then the tube a.c. resistance and the load are no longer well matched, and more of the signal energy is lost in  $r_p$ .

To prevent this from happening, the primary inductance must be designed to have a reactance that is five to ten times the tube a.c. plate resistance. Then, when the secondary of the transformer is properly loaded, the primary reactance "disappears" even at low frequencies and is replaced by the reflected load  $R_A$ .

## LEAKAGE INDUCTANCE

The fact that power is transferred in an impedance-matching transformer means that a high primary current flows (compared to that in interstage transformers), and, in turn, this means a higher flux density in the core. This condition tends to produce core saturation, which increases the leakage flux. We have already described leakage flux as the flux taking an air path that fails to link with all (or a part) of a winding. The turns that produce this leakage flux are really not a part of the transformer, for they do not assist the transformer action. We can therefore consider each transformer winding to be made up of two parts, as shown in Fig. 15. One section of each winding represents the turns actually engaged in producing flux linkage ( $L_P$  and  $L_S$ ), and the other represents the inductance producing the leakage flux ( $L_{P1}$  and  $L_{S1}$ ). The latter are called leakage inductances, and are considered to be in the positions shown in Fig. 15.

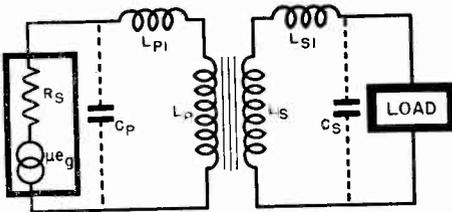


FIG. 16. The distributed capacity  $C_P$  can be made to resonate with the primary, which improves the high-frequency response up to the resonant frequency. However, above this point, there is a sharp drop in response.

The reactance of  $L_{P1}$  naturally reduces the amount of signal voltage that will be developed across  $R_A$ . Furthermore, since the reactance of  $L_{P1}$  increases with frequency, the voltage across  $R_A$  will decrease as the fre-

quency is increased. In the secondary circuit, the reactance of  $L_{S1}$  reduces the voltage applied to the load.

The effects of these two leakage inductances, then, are to reduce the useful signal voltage in the circuit and

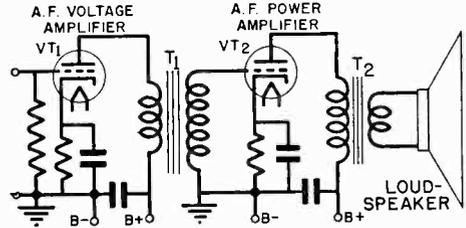


FIG. 17. Both an interstage and an impedance-matching transformer are shown in use here.

also to create distortion (since high-frequency voltages are reduced more than low-frequency voltages).

## DISTRIBUTED CAPACITY

An impedance-matching transformer has distributed capacity, just like an interstage transformer. As shown by Fig. 16, the distributed primary capacity  $C_P$  forms a parallel resonant circuit with the primary and its leakage inductance. Similarly, the distributed secondary capacity forms a series resonant circuit with the secondary and its leakage inductance. However, in a step-down transformer with a high turns ratio (which is what most impedance - matching transformers are), secondary leakage inductance and secondary distributed capacity are so small that they can be ignored.

► Sometimes both interstage and impedance-matching transformers are used in the same apparatus. For example, in Fig. 17 transformer  $T_1$  steps up voltage while  $T_2$  matches the loud-speaker voice coil to the plate resistance of  $VT_2$ .

# Special Transformers—Identifying Types

There are one or two special types of transformers which are of interest. We will introduce them briefly here—their uses will be covered in another lesson.

## CENTER-TAPPED TRANSFORMERS

Fig. 18A shows the polarity that may exist, at a given instant, across the secondary of an ordinary audio transformer. This polarity will be

both a positive and a negative voltage—that is, *two* voltages  $180^\circ$  out of phase with each other—from the transformer *at the same instant of time*.

You will see this principle used many times in radio work. It is used in power supplies to obtain full-wave rectification and in audio amplifiers to drive a push-pull stage (a type of amplifier) from a single tube stage.

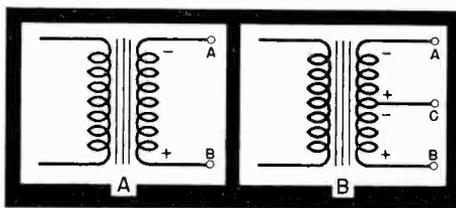


FIG. 18. Many transformers have a center tap on the secondary as at B. This provides a two-phase output that is necessary for the operation of certain radio stages.

reversed on alternate half cycles. As there is but one output voltage, technicians say that this is a *single phase* output.

In Fig. 18B, a center tap has been added to the secondary. The addition of the center tap does not change the voltage induced in the secondary by the primary, but does give us another point from which to measure. Thus, by measuring from this center tap to either of the original secondary terminals, we will get half the total secondary voltage. And notice—when point A is negative with respect to point C, point B will be positive with respect to point C. The next instant, the polarities will reverse, but they will remain *opposite* to each other at all times. Thus, if we use the center-tap C as a reference point, we can get

We will say more about these transformers when we study these stages.

## AUTO-TRANSFORMERS

The schematic diagram of a special type of transformer known as an *auto transformer* is shown in Fig. 19. (*Auto* means “self,” the name coming from the fact that a single winding is used as a transformer.) You will notice there is a tap on the winding: primary voltage is applied between this tap and one end of the winding, and secondary voltage is taken off between the two ends of the winding. The number of turns in the entire winding divided by the number of turns connected across the input, or primary, terminals gives the voltage step-up ratio of this transformer. An auto-transformer also may be used as a

step-down transformer by connecting the source voltage to the ends of the winding and taking the output from the tap on the coil and one end of the winding.

While an auto transformer has certain advantages (principally low cost) to recommend it, there are many circuits in which it cannot be used because of the common connection between the source and the load.

### IDENTIFYING IRON-CORE CHOKES AND TRANSFORMERS IN RADIO RECEIVERS

**Power Transformers.** When the receiver is an a.c. type using a power transformer, you generally can assume that the largest iron-core device is the power transformer, for it must handle all the filament power and the electrode power required by the tubes in the receiver. (However, remember that there are many receivers, principally a.c.-d.c. sets, that do not use a power transformer.)

When you must be sure of your identification, remember that a power transformer has a great many terminals or leads. You will find wires running from it to the filament terminals of each tube socket. You will find two wires running from the power transformer to the plates of the rectifier tube (assuming it is a full-wave rectifier tube). Two more wires will run directly to the line cord, one connecting through the on-off power switch of the receiver.

Power transformers have a primary winding and several secondaries—a high-voltage secondary winding, and one or more filament supply secondary windings. The low-voltage filament supply secondaries will have the *lowest* resistance because they handle high currents. The primary winding will also be low in resistance, but not as

low as the filament secondaries, because it has many more turns of wire than they do.

The *high-voltage* secondary winding of a radio receiver power transformer can always be identified with an ohmmeter, for it will have the *highest resistance* of all windings on the core. This winding usually will have a center-tap terminal located between the two outer terminals of the winding; naturally the resistance between this mid-tap and an outer terminal will be *about* half that between the two outer terminals. (While the tap

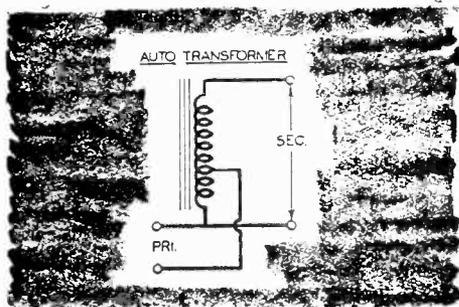


FIG. 19. An auto transformer has a single winding which serves both as the primary and the secondary.

is located so that the number of *turns* on each section of the winding is equal, the half of the winding that is closer to the iron core will use turns of smaller diameter. This requires less wire, so this half of the winding will have a lower resistance than the outer half of the winding.)

**Iron-Core Chokes.** You can distinguish iron-core chokes from iron-core transformers in radio receivers by counting the terminals (or leads). Iron-core *chokes* will have two or three terminals (or leads), while *transformers* have at least four leads. (Auto transformers may have only three leads, but are seldom found in radio receivers.)

Iron-core chokes used in radio receivers are of two types: those designed to handle a.f. signal currents, and those designed to filter the rectified current in the power pack. Signal current chokes are rare in the average modern receiver, so it is safe to assume any iron-core choke is a filter choke, unless you find a *direct wire* connection from this choke to the grid or plate of a signal circuit tube.

**Audio Transformers.** Audio transformers are usually the hardest to identify. It is safe to assume that a receiver that uses a dynamic loudspeaker always has an output transformer or impedance-matching transformer between the loudspeaker and the output stage. Generally, this transformer will be mounted on the frame of the loudspeaker, but it may be mounted on the chassis.

Since the impedance of the average loudspeaker is much lower than the a.c. plate resistance of an output tube, a step-down ratio is required in the output transformer. The primary winding therefore will have a much higher resistance than the secondary winding (which connects to the loudspeaker).

Most other a.f. transformers you encounter in receivers will have voltage step-up ratios, and can be identified through their connections to the grids and the plates of amplifier tubes. The primary winding of an interstage audio transformer will have a lower resistance than the secondary winding. Transformers of this type are heavier and usually much larger than output transformers, because of the larger secondary.

► These are merely general suggestions; for further information on a particular receiver, refer to the schematic circuit diagram and service instructions for that receiver.

**Looking Ahead.** Since you now have mastered the important fundamental features of most of the parts used in radio equipment, and have acquired a good idea of how these parts perform in vacuum tube circuits, you are ready to study in detail the uses for these vacuum tube circuits in radio apparatus. In the next lesson you will find a wealth of interesting and highly practical information on the power supply systems used in radio.

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THE N. R. I. COURSE PREPARES YOU TO BECOME A  
**RADIOTRICIAN & TELETRICIAN**  
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# Lesson Questions

Be sure to number your Answer Sheet 11FR-2.

Place your Student Number on every Answer Sheet.

Send in your set of answers for this lesson immediately after you finish them, as instructed in the Study Schedule. This will give you the greatest possible benefit from our speedy personal grading service.

1. What is the opposition effect in a magnetic circuit called?
2. What is the *practical unit* of magnetomotive force for coils?
3. What condition is said to exist in an iron core when increases in magnetomotive force produce little or no increase in magnetic flux?
4. How can eddy current losses be reduced considerably in low-frequency iron-core devices?
5. Does primary leakage flux induce any voltage in the secondary winding of a transformer?
6. Suppose a circuit uses a *step-down* transformer. Will the primary CURRENT be: 1, the same as; 2, larger than; 3, smaller than the secondary CURRENT?
7. Would you expect a high-fidelity interstage transformer to have: 1, a high; or 2, a low step-up turns ratio?
8. Is the maximum power transferred when the load impedance is: 1, equal to; 2, greater than; or 3, less than the source impedance?
9. When checking the various windings of a radio receiver **power** transformer with an ohmmeter, which winding will have the highest resistance?
10. Although the number of turns on the two halves of a high-voltage winding of a power transformer are equal, are the resistances of the two halves exactly the same?

## HOW TO CONCENTRATE

The secret of rapid progress with any course of study lies in being able to concentrate. If your mind wanders from study while you are alone in a quiet room, try moving to a noisy room, or try tuning in an all-musical program on the radio. Stay away from loud conversation, though.

If noise definitely bothers you, however, do your studying in a quiet location. If you feel too sleepy to concentrate, the room may be too hot. Open the windows, and put on a coat if necessary, for it is easier to study in a cool room. Sponge your face, neck and eyes with cold water.

If you have difficulty in understanding a subject, write down an explanation of it in your own words, or try outlining the subject and studying your outline. Underlining of important words as you study is another aid to learning.

It will pay, too, to experiment until you find the conditions that are most favorable for concentration when you study. Also, try different studying techniques, until you find the one that gives you complete mastery of a subject in the shortest possible time. The faster you master each lesson, the sooner you'll complete the Course and be ready for a good job in radio.

J. E. SMITH

**HOW OPERATING VOLTAGES ARE  
OBTAINED FROM AN A.C.  
POWER LINE**

12FR-2

**NATIONAL RADIO INSTITUTE**

ESTABLISHED 1914

**WASHINGTON, D. C.**



# STUDY SCHEDULE NO. 12

For each study step, read the assigned pages first at your usual speed. Reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind, then answer the Lesson Questions for that step. Study each other step in this same way.

1. Power Supply Requirements . . . . . Pages 1-3  
All radio apparatus can be divided into signal circuits and power supply circuits. Here you learn what these circuits are intended to do, then you study the requirements for proper power supply operation. These are very important, because many radio troubles result from a defect causing improper power supply operation. Answer Lesson Questions 1 and 2.
2. Filament Power . . . . . Pages 3-7  
Tube filaments must be heated enough to give proper electron emission. Excess filament voltage will ruin the tube, while too little does not give proper operation. When a. c. is used, special precautions are necessary to avoid hum-producing variations in the plate current. Answer Lesson Question 3.
3. Obtaining D. C. from A. C. . . . . Pages 7-11  
After settling the filament problem, we must supply the plate and grid elements with pure d. c. power. The line voltage is stepped up and then rectified, so a pulsating d. c. is obtained. Half-wave rectifiers will work, but full-wave types are more efficient, as is proved here. Notice the importance of balanced tubes—hum will result from improperly matched full-wave rectifiers. Answer Lesson Question 4.
4. Taking out the Ripple . . . . . Pages 12-20  
The pulsating d. c. from the rectifier is a combination of d. c. and a. c., so the a. c. must be removed for proper radio performance. The standard filter is a coil-condenser combination, arranged so most of the a. c. is dropped in the coil. The choke input and the condenser input types differ in their operation, d. c. voltage output and effects on the rectifier. This section should be studied and restudied. Refer back to it later on, as many very important facts are presented here. Answer Lesson Questions 5, 6 and 7.
5. Dividing the Voltage . . . . . Pages 20-25  
The filter delivers the pure d. c. needed, but the voltage is a maximum and usually too high, particularly for many of the grid circuits. By using Kirchhoff's and Ohm's Laws, it is easy to figure the resistor values needed to drop the voltage to the required amount. Also, you learn how bleeder currents prevent load changes from reacting on the power supply and thus affecting the voltages delivered to other stages. Answer Lesson Questions 8 and 9.
6. Facts About Rectifier Tubes . . . . . Pages 25-28  
There are two general types of rectifiers—vacuum types and the mercury vapor types. The difference in regulation and current handling ability should be carefully noticed, so the wrong tube won't ever be used. Answer Lesson Question 10.
7. Mail your Answers for this Lesson to N. R. I. for Grading.
8. Start Studying the Next Lesson.

# HOW OPERATING VOLTAGES ARE OBTAINED FROM AN A. C. POWER LINE

## Power Supply Requirements

WHEN you become an expert at tracing circuits, whether you are working on a receiver, a transmitter, or some sort of electronic control, you will automatically divide the device into two parts:

1. *The signal circuits.*
2. *The power supply circuits.*

These two circuits can always be considered separately. In fact, special filters are almost always used to keep them from reacting on one another.

One very important fact you must remember about any modern radio receiver is that it uses the tiny amount of electric power picked up by the antenna to excite a tube or tubes. These tubes in turn control a relatively large amount of power furnished by the batteries or other power source. The circuits which *control* the power are called the signal circuits—the circuits which *furnish* the power are called the power supply circuits.

Both kinds of circuits use resistors, condensers, coils, transformers and tubes. You've already studied the basic actions of these important radio parts. Now you're ready to learn all the facts about their many uses in radio circuits. It's this knowledge, which you'll gain from this and succeeding lessons of your N.R.I. Course, that will make you an expert technician—able to operate and repair all types of radio equipment.

A great many of your later lessons

will be devoted to the signal circuits of modern radios. In this lesson, you're going to learn all about the equally important power supply circuits, without which no radio could operate. You should study this lesson very carefully, for the information it gives you will be useful all through your radio career.

### WHAT THE POWER SUPPLY MUST DO

A radio tube works best when the voltages applied to it have proper polarities and values. The plate, control grid, screen grid and suppressor grid electrodes all require different operating voltages with respect to the cathode of the tube. As you've already learned, these voltages must make the screen and plate *positive* with respect to the cathode, and must make the control grid and suppressor grid *negative* with respect to the cathode. (In some cases the suppressor grid is connected directly to the cathode, and then there is no difference in potential between them.) The filament of the tube must be heated so that electrons will be emitted. You must apply just the right voltage to the filament to heat it properly.

► Thus, one important requirement we must make of the power supply is that it furnish the proper operating voltages for the various tube circuits. Another is that the power supply must not interfere with the operation of any stage by introducing undesirable

voltages. Only the signal voltage applied at the input of the tube should control the output of the stage. We do not want supply voltages to have a ripple component which might act just like a signal and make the loudspeaker produce noise or hum. For this reason, the power supply must be a pure d.c. power source with a minimum of variation or ripple.

Finally, we do not want the power supply to transmit signals from one stage to another. You can easily see that signals straying through the power supply would be as bad as having variations in the power supply.

Let's repeat these three important things required of the power supply, in order to fix them in your mind:

1. It must provide the correct amount of voltage to each tube electrode. Each voltage must have the correct polarity. (As you will soon learn, these voltages may have to be values between 1 or 2 volts and several hundred volts.)

2. It must furnish as near a pure d.c. voltage as possible to those electrodes where variation would produce distortion, interference or unwanted signals.

3. It must not transmit signals from one stage to another. This means that the power supply circuits and the signal circuits can be, and are, electrically separated from each other—which, as you've learned, is the reason we can study them individually.

► Of course, these requirements must be met by any power supply which may be used. Now, let's see what kinds of power supplies have been developed.

## BATTERY SUPPLIES

Batteries meet all the requirements of a power supply—cells can be added to get the required voltage; a pure d.c. is delivered; and their low internal resistance causes little signal volt-

age drop, thus minimizing unwanted signal transfer between stages.

► In fact, batteries are such perfect sources of d.c. power that engineers frequently draw battery symbols in their diagrams to show that a d.c. supply is wanted.

► A simple battery supply circuit is shown in Fig. 1A. The signal circuit parts which you would ordinarily expect to find in the control grid and plate supply leads have been omitted to simplify this drawing. These parts

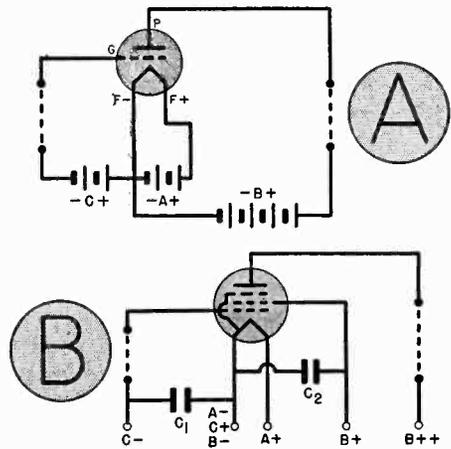


FIG. 1. Power supply connections for a triode and a pentode are shown in A and B respectively. Bypass condensers  $C_1$  and  $C_2$  are used to keep stray signals out of the power supply.

—which might be resistors, coils or transformer windings—would be connected between the points which are connected by the dotted lines.

In Fig. 1A, we have an A battery supplying power to operate the filament, a B battery to operate the plate circuit and a C battery to supply the necessary grid bias. These batteries are usually made up from dry cells, although a storage battery or an air-cell battery may be used as an A supply.

► Although batteries have these very desirable characteristics, they are ex-

pensive, bulky, and must be changed or recharged at frequent intervals. Therefore, as soon as radio became popular, many experts searched for ways to eliminate batteries by converting a.c. from the power line wall outlet into continuous or d.c. power. At first, they tried to get pure d.c. for all tube elements, including filaments. However, the high-current, low-voltage supply needed for filaments demanded special, expensive equipment—so an a.c. type tube, whose filament could use a.c. directly, was developed.

High d.c. voltage is easily produced from the a.c. power line, particularly if the current demand is not too high. Once we have a high d.c. voltage, it is a simple matter to divide it so that the grid and plate elements receive the proper amount of voltage.

► Let us now see how the power supplies work which use power transformers to operate from standard a.c. power lines. First, we will take up filament voltage supplies, then study the rectifiers, filters and voltage dividers used to obtain power for the other tube elements.

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## Filament Power

A d.c. voltage applied to a tube filament will neither cause changes in the plate current wave form nor introduce hum. However, when a.c. is applied, we must take special precautions to make sure the filament heat does not vary and to prevent the a.c. from affecting the grid bias and plate voltage. Let's see what these precautions are for a tube which uses a filament as the cathode or electron emitter.

### FILAMENT TYPE TUBES

**Getting Constant Heat.** When a.c. is fed to the filament, the current rises to a peak twice for each cycle, once on the positive and once on the negative alternation. The heating effect is the same on either alternation. Thus, the filament will be subjected to a maximum heating current 120 times a second if standard 60 c.p.s. (cycles per second) current is used. If the filament is very thin, as it is in most battery-operated tubes, it will become hot, then cool off, 120 times a second. Since, as you recall, the

number of electrons emitted from a filament depends on how hot the filament gets, this varying filament heat will produce a varying electron flow from the filament to the plate. This pulsating plate current, varying 120 times a second, will produce a loud hum.

We can prevent the filament heat from varying by using a filament so thick that, once it is warmed up, it tends to stay at a constant heat. This is exactly how a.c. filament tubes are made.

► You can now see why battery type tubes are not satisfactory when their filaments are powered from an a.c. source—and also why battery receivers using tubes with thin, quickly-heated filaments go into operation faster than a.c. receivers using heavy filaments which take a longer time to heat.

► A heavy, heat-holding filament solves only one of the problems involved in a.c. filament operation. Still other precautions must be taken to prevent an a.c. filament supply from producing a.c. ripple in the plate current.

**Getting Constant Emission.** Now let us see how an a.c. filament voltage can affect the grid voltage. Look at Fig. 1A. Here we can measure the grid voltage by touching the negative probe of a voltmeter to the grid *G* and the positive probe to  $-A$ . Since  $+C$  connects to  $-A$ , the voltmeter is connected across the C battery and measures the C battery voltage.

Suppose, however, that we connect the positive voltmeter probe to  $+A$ . Will we measure the same grid-filament voltage as before? No—because now the A battery voltage will add to the C battery voltage, and we will actually measure the C battery voltage plus the A battery voltage.

Since the grid is more negative with respect to the right-hand side of the filament than to the left-hand side, the number of electrons leaving the right-hand side of the filament is considerably less than the number leaving the left-hand side.\* We don't care about this when d.c. is used on the filament, because the total emission remains constant, and the plate current is not varied by the filament voltage.

When a.c. is used to heat a filament, we cannot connect  $+C$  to either side of the filament. Fig 2, where such a connection is made, shows you what would happen. Here the plate has been omitted for simplicity, and the

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\* When d.c. filament voltage is employed, this results in extra wear on the side of the filament emitting most of the electrons. This effect is unimportant in low-power receiving tubes, but in transmitting tubes where the plate current may be very high, the connections to the filament leads of large tubes are disconnected and reversed about once a week in order that both sides of the filament may receive equal wear. When the filaments of low-power tubes are fed with d.c., as in Fig. 1A, the  $+C$  and  $-B$  connections of the electrode voltages may be connected either to  $-A$  or  $+A$  filament terminals; the  $-A$  terminal connection has become standard for most circuits.

filament is shown as a straight resistance wire because it has resistance. This representation will help you to picture the voltage drop that exists across the filament when filament current flows through it. The C battery furnishes 3 volts d.c., while 6 volts a.c. are used to heat the filament. At the particular instant illustrated, the a.c. voltage gives the filament the polarity shown.

Starting with Fig. 2A, the voltage between the grid and point *a* on the filament is  $-3$  volts. As we move along the filament, the grid-filament voltage becomes progressively greater as the voltage drop along the filament resistance is added to the bias voltage. Between the grid and *b* it is  $-4\frac{1}{2}$  volts; at *c* (the center of the filament) it is  $-6$  volts, and at *d*, where all the filament voltage drop is added to the bias voltage, it is  $-9$  volts. Naturally, most of the electrons reaching the plate come from the section between *a* and *c*, because the grid is less negative with respect to this part of the filament.

If the supply current is 60-cycle a.c., the filament voltage will reverse polarity so that in  $1/120$  of a second we will have the condition illustrated in Fig. 2B. The voltages between the grid and various points on the filament at this instant are indicated in the diagram. As you see, the filament voltage drop is now *subtracted* from the 3-volt grid bias voltage. This creates an over-all decrease in grid-filament voltage—and so considerably increases the total number of electrons reaching the plate, as compared with the number which reached it during the previous half cycle.

In another  $1/120$  of a second, the conditions are again as shown in Fig. 2A. As a result of these changes in grid-filament voltage, the plate cur-

rent alternately increases and decreases. This causes hum.

It makes no difference to the plate from which part of the filament electrons come. The important thing is that the *total number* remain constant (except, of course, when a signal voltage is applied between the control grid and filament).

We can get constant electron emission by connecting  $+C$  to the center of the filament, as shown in Fig. 2C. The voltages between the grid and

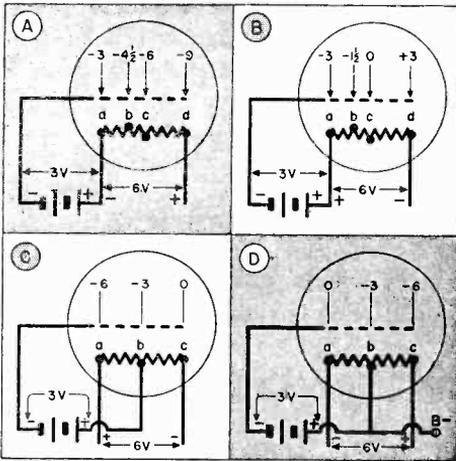


FIG. 2. When a.c. is used to heat the filament, a center tap is necessary to prevent hum.

points *a*, *b* and *c* on the filament when the supply voltage has the indicated polarity are shown directly above the grid symbol. For simplicity, suppose that from point *a* we get 2 electrons, from point *b* we get 8 electrons, and from point *c* we get 14 electrons. (Actually, of course, the number of electrons is far greater.) This makes a total of 24 electrons reaching the plate.

When the filament polarity reverses 1/120 of a second later, the grid-to-filament voltages at *a*, *b*, and *c* will be those shown in Fig. 2D. Now from *a* the plate gets 14 electrons, from *b* it gets 8 electrons, and from

*c*, 2 electrons. In all, the plate receives  $14+8+2$  or 24 electrons, the same number it did in 2C. Thus, a center-tapped filament allows us to maintain a constant flow of electrons to the plate throughout the a.c. filament cycles, even though the amount from any one spot on the filament is varying. (The signal voltage will, of course, still vary the plate current, because it changes the voltage between the grid and all points on the filament at one time.)

► The plate-to-filament voltage would vary just like the grid-to-filament voltage if we connected *B*— to one end of the filament. We can prevent this variation (which would of course cause a varying plate current) in exactly the same way—by also connecting *B*— to the center tap of the filament, as shown in Fig. 2D.

**Getting a Center Tap.** You see that a grid and plate connection to the center of the filament will eliminate a.c. variations in the plate current. While it is perfectly possible to build a tube with a filament center tap, this is not necessary. We need only connect  $+C$  and  $-B$  to a center-tapped resistor across the filament (see Fig. 3A).

Let's see why. In Fig. 3B, we have a tube filament connected between points *A* and *B*. As you've learned, each point along the filament will have a different voltage with respect to one end of the filament. For instance, with 6 volts applied and measuring from point *A*, there might be 1 volt at *c*, 4 volts at *d* and 5 volts at *e*.

If we connect a resistor in parallel with the tube filament between *F* and *G*, the same total voltage will be across this resistor—and there will be differences in voltage between point *F* and other points on the resistor. There will be a point on the resistor where

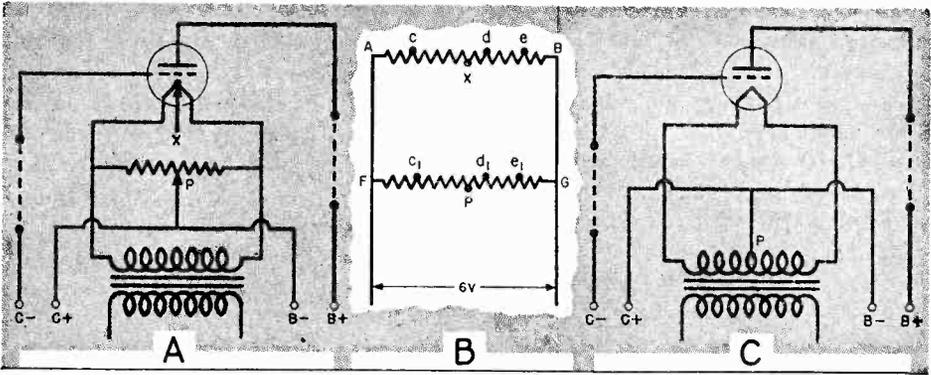


FIG. 3. Practical ways of getting the center tap.

this difference is 1 volt. This is marked  $c_1$  to correspond to the same point on the tube filament. Similarly, a 4-volt difference will be found at point  $d_1$ , and a 5-volt difference at  $e_1$ . Points  $F$  and  $A$  are at the same potential, since they're connected together by a wire.

For every point on the tube filament, there will be a corresponding point on the resistor which will have the same potential with respect to terminal  $A$ . Thus, center tap  $P$  on the resistor has the same potential with respect to terminal  $A$  as has point  $X$ , the center of the filament. Since  $P$  and  $X$  are both in the same circuit and are both at the same potential, we get precisely the same electrical effects by connecting an outside circuit to point  $P$  as we would by connecting it to point  $X$ .

► Frequently, a potentiometer (variable resistor) is used across a tube filament instead of the fixed-tap resistor, as shown in Fig. 3A. Then, if the electronic emission from the filament is not the same over its entire length because of imperfect manufacture, we can find a point on the resistor which corresponds to the emission center of the filament by moving point  $P$  to the right or left. Hum disappears or is reduced to a minimum when the correct adjustment is

made. Usually tubes are manufactured with sufficient care so that a movable tap on the resistor is not required.

► A center-tapped resistor is not always used, since it is less expensive to tap the electrical center of the filament winding on the power transformer instead (Fig. 3C). Here point  $P$  is the  $+C$  and  $-B$  terminal for this tube stage.

### HEATER TYPE TUBES

Variations in voltage between the grid and the electron emitter may also be eliminated by using a cathode or electron emitter that is not connected to the heating source. A heater type

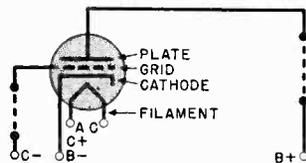


FIG. 4. The elements of a heater type tube.

tube has its filament surrounded by a sleeve coated with electron-emitting chemicals. This sleeve is the cathode, and emits electrons; the filament (whether fed with d.c. or a.c.) only furnishes heat to the cathode. Emission is kept constant by making the filament and cathode heavy enough to

hold an even temperature. The connections for such a tube are shown in Fig. 4. Since no a.c. is dropped in or applied to the indirectly heated cathode, there is no need for center-tapped resistors across the filament supply of these tubes. (Sometimes there is a center tap on the filament transformer. This lead is generally grounded to the chassis to reduce leakage and stray fields and thus further reduce the possibility of hum.)

Heater type tubes have replaced filament type almost altogether in radio receivers using a.c. power supplies. However, filament type tubes are still widely used in transmitters, in power packs, and in some output stages where a slight amount of hum does not matter.

### FILAMENT CONNECTIONS

The tubes commonly used in a.c. operated radio receivers need filament voltages of 1.5, 2.5, 5, or 6.3 volts. These voltages are furnished by low-voltage windings on the power transformer.

Ordinarily, tubes needing the same filament voltage have their filaments wired in parallel and connected to the same filament winding on the power transformer, as shown in Fig. 5A. When circuit requirements make it necessary to separate filaments of tubes needing the same filament voltage, an extra winding will be provided on the power transformer secondary for the filaments which are to be

operated as a separate circuit (see Fig. 5B).

► If tubes with different filament voltage ratings are used in the same piece of radio equipment, separate filament secondary windings must be provided for each group, as shown in

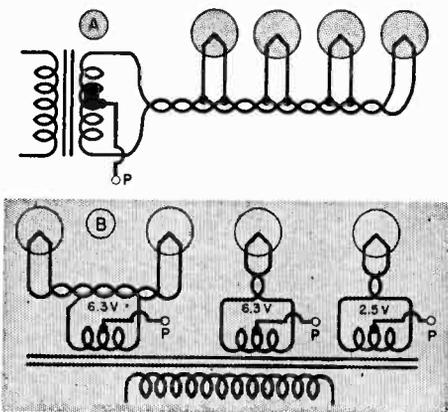


FIG. 5. Filaments must be connected according to their voltage and current needs.

Fig. 5B. A 2.5-volt tube, for example, cannot be operated from a 6.3-volt source, for the filament of the 2.5-volt tube will draw too much current and burn out almost at once. That's something to remember in your service work—*always be careful to operate a tube on the correct filament voltage.*

Circuits 5A and 5B may be used with either filament or heater type tubes. With filament tubes, the center tap *P* must be used as the *C+* and *B-* connection.

## Obtaining D. C. from A. C.

You have learned that the filament can be supplied with a.c. directly if we use a cathode or indirectly heated type of tube, or use a tube with a heavy filament and make the proper

connections. But we must use d.c. for the plate and grid supply.

To operate the average radio receiver and amplifier properly, 110-volt, 60 c.p.s. power must be converted

to about 400 volts d.c. without an appreciable ripple.\* Theoretically, you cannot take out all the ripple, but practically, you can make it so small that it produces no noticeable hum in the loudspeaker or other output device.

Fig. 6 shows how the power pack works. The 110 volts from the power line are first raised to a high a.c. voltage by a voltage step-up transformer. Next, the a.c. is passed

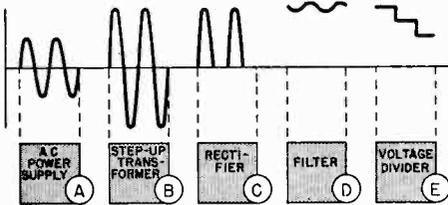


FIG. 6. We start with a.c. and get d.c.

through a device which we call a rectifier, which allows current to flow only in one direction. A tube is generally used. A rectifier tube produces pulsating direct current, varying from a peak (the largest) value to zero. To remove the variation, the electric power is passed through an electrical filter consisting of condensers and choke coils, which smooths out the variation or ripple, leaving pure d.c. Finally, the high d.c. supply voltage is divided as required. The changes from a.c. to divided d.c. are portrayed in Fig. 6. At A, B, C, D and E, the wave shape of the output voltage of each section is shown.

You know already that a transformer with more turns on its secondary than on its primary will step up a voltage according to its turns ratio. The power transformer in the average receiver steps up the power line voltage to as much as 350 or 400 volts—ample voltage for the tubes

\* By ripple, we mean a small a.c. voltage superimposed on (combined with) the d.c.

used in radio receivers, amplifiers and control circuits. Of course, this a.c. voltage cannot be applied to the receiving tube electrodes, since the electrodes would be alternately positive and negative. We must rectify this a.c. supply into a d.c. voltage.

## HALF-WAVE RECTIFIERS

Figs. 7A and 7B show the basic operation of a half-wave rectifier. In these circuits, the single resistor  $R_L$  represents all the tubes and circuits of a radio. We can use this representation because each tube draws current, and thus acts as a load on the power supply. The combined resistance of these various loads is equal to the supply voltage divided by the total current drawn. Our resistor  $R_L$  has the same ohmic value as the combined resistance of the tubes, and so has exactly the same effect as the tubes on the power supply. This resistor is called the load resistance.

Alternating voltage comes from transformer  $T$ . We want the voltage supplied to the load  $R_L$  always to have the polarity shown in Fig. 7A—in other words, we want to cut out the half cycles of the supply voltage which would reverse the polarity of the voltage across  $R_L$ .

We can cut out these unwanted half cycles of supply voltage by using

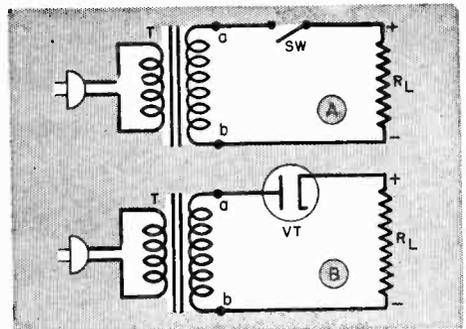


FIG. 7. A rectifier tube acts like an electronic switch.

some sort of switch that will close each time transformer terminal *a* is positive, and open each time this terminal is negative, with respect to terminal *b*. This can be a mechanical switch\* (*SW* in Fig. 7A), but most radios use an electronic switch.

Fig. 7B shows a diode tube *VT* which acts as an electronic switch. When terminal *a* of the transformer becomes positive, the plate of the tube is made positive, and electrons flow from the cathode to the plate. (This corresponds to switch *SW* in Fig. 7A being closed.) The flow ceases when the voltage at terminal *a* drops to zero. During the next alternation the terminal polarity reverses, making the plate negative with respect to the cathode. Now no electrons flow to the plate, and our electronic switch is open.

If the voltage across terminals *a* and *b* changes at the rate of 60 cycles per second, the voltage across  $R_L$  as the result of the rectifying action of tube *VT* will be a series of half-wave pulses (60 every second), as shown in Fig. 8A.

This voltage across  $R_L$  is neither pure d.c. nor pure a.c., but rather a combination of both. (We started with pure a.c. and now have the desired d.c. mixed with some a.c.; this is called a pulsating d.c.) Analysis of this voltage by a special laboratory instrument called a harmonic analyzer will show us that it consists of the d.c. component (or part) shown in Fig. 8B, the 60-cycle ripple (or a.c. voltage) shown in *C*, and the ripples of higher frequency and smaller amplitude shown in *D* and *E*.

Now, we want only the pure d.c. component of this voltage shown in

Fig. 8B, because the a.c. components shown in *C*, *D* and *E* would cause hum if we applied them to our receiver. As you will learn later in this lesson, these a.c. components can be removed by sending the rectified voltage pulses shown in Fig. 8A through an a.c. filter. This removes everything but the pure d.c., which is then applied to load resistor  $R_L$ .

You can see that the d.c. component in Fig. 8B is not as great in magnitude as the peaks of the pulsating d.c. in Fig. 8A. This means that not all of the rectified voltage is available as

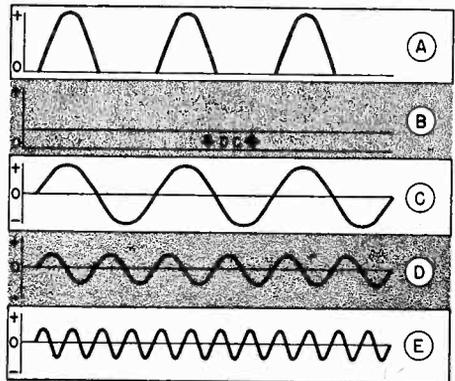


FIG. 8. The output of a half-wave rectifier.

d.c. for the load. Let's see how we can get more d.c. output.

## FULL-WAVE RECTIFIERS

When we use the half-wave rectifier shown in Fig. 7, only one-half of the *supply* wave is rectified. (This does not mean that we are wasting power, for, when the tube is not passing current, none is taken from the source.) If we put both halves of the supply wave to work by using a full-wave rectifier, we can get a higher rectified voltage. In other words, we get a larger d.c. component from full-wave rectification than from half-wave.

Further, in the half-wave rectifier, the lowest ripple frequency is equal to

\* Later, when auto radios are studied, you will see how an electromechanical switch called a vibrator is sometimes used to close and open power supply circuits automatically.

the frequency of the supply line, while the lowest ripple frequency of a full-wave rectifier is twice the frequency of the line (120 cycles for a 60-cycle line). This higher ripple frequency may be smoothed out with smaller, less expensive parts.

Full-wave rectification uses both halves of the a.c. cycle, as the name implies. By making a change in the transformer and by using a dual tube rectifier, we can make current from both halves of the cycle flow in only one direction through the load in our former half-wave rectifier circuit.

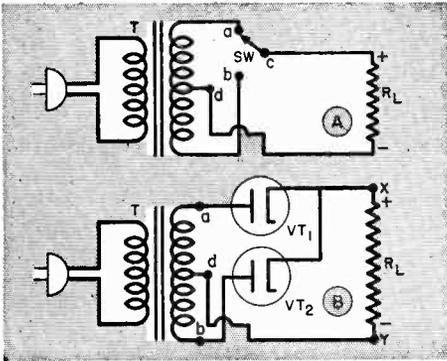


FIG. 9. A full-wave rectifier is a double switch.

The transformer change consists of providing the secondary with a center tap which is used as the reference terminal. When one end of the winding is positive with respect to this center tap, the other end will be negative, and vice versa.

Going back to our idea of a rectifier tube as an electronic switch, our full-wave rectifier circuit is really that shown in Fig. 9A. Here we have a switch arm *c* which throws to whichever transformer terminal (*a* or *b*) is positive with respect to center tap *d*. When *a* is positive with respect to center tap *d*, the switch closes *a-c* so the voltage between *a* and *d* is applied to the load.

When the a.c. cycle reverses, mak-

ing *a* negative and *b* positive, the switch changes connection and we apply voltage *b-d* to the load. Now let's substitute two tubes for the switch, as shown in Fig. 9B, and see how the circuit works.

When point *a* is positive with respect to *d*, electrons flow from the center tap *d* to load terminal *y*, to *x*, to the cathode of tube *VT*<sub>1</sub>, across to the plate and to point *a*. The electrons cannot go from *d* to *b* because they cannot pass through *VT*<sub>2</sub> when point *b* is negative with respect to *d*. Therefore the path taken through *VT*<sub>1</sub> is from *d*→*y*→*x*→cathode→plate→*a*. On the next half cycle, when *b* is positive with respect to *d*, the path is through tube *VT*<sub>2</sub>—that is *d*→*y*→*x*→cathode→plate→*b*.

You see at once that the tubes alternate in passing electrons through the load *R*<sub>L</sub>. Also, the actual voltage rectified is half the voltage across *a* and *b*. In other words, the a.c. voltage across the entire transformer secondary in Fig. 9B must be twice as

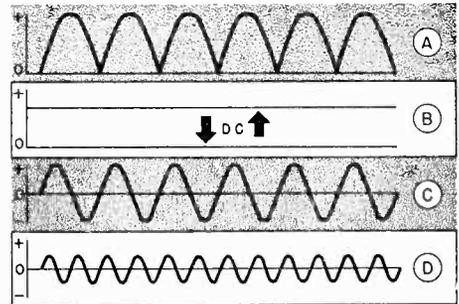


FIG. 10. The output of a full-wave rectifier.

much as in Fig. 7B to realize about the same peak rectified voltage, because only half of the total secondary is being used at a time.

The pulsating output current through the load has the wave form shown in Fig. 10A. Compare this with Fig. 8A and you will see that there are twice as many pulses, with

no gaps between pulses. This makes the d.c. average of this wave about twice as much as that of a half-rectified wave (compare Fig. 10B with 8B). Of course, we must get rid of the ripple frequencies shown in Figs. 10C and 10D before we can use this d.c. output for the load.

Full-wave rectifiers are so practical that they are almost universally used in radio receivers. Instead of two separate tubes as shown in Fig. 9B, a twin diode tube is generally used, as shown in Fig. 11. This tube has two separate diode rectifiers in a single glass or metal envelope. A rectifier tube of the cathode type is shown in Fig. 11A, while a filament type is shown in Fig. 11B. In filament types, the positive load terminal must connect to the filament. The connection may be made to either *filament* lead.\*

(Perhaps you wonder why we connect directly to the filament instead of to a center tap on the filament winding. While it is true that the direct filament connection creates a certain amount of a.c. in the rectifier output, this is removed along with the ripple frequencies which are present in any rectifier output. The direct filament connection therefore does no harm.)

\*The filament voltage serves no other purpose than to make the filament hot enough to emit electrons and does not affect the B supply voltage in any way. Neither does the B supply voltage have any effect on the filament voltage. The two circuits are quite independent and are simply joined together at one point. We want to impress on you the fact that there is no interaction between the circuits, for students (forgetting that a circuit must be complete before current can flow) often ask why the high d.c. voltage does not burn out the filament. As you can see, the d.c. is not applied across the two filament leads and hence cannot cause an excessive current to flow through the filament.

The two diode tube sections (or separate tubes, if they are used) should be "balanced"—that is, have the same emission characteristics. Furthermore, each diode should be supplied with the same amount of a.c. voltage to rectify. Both conditions make the

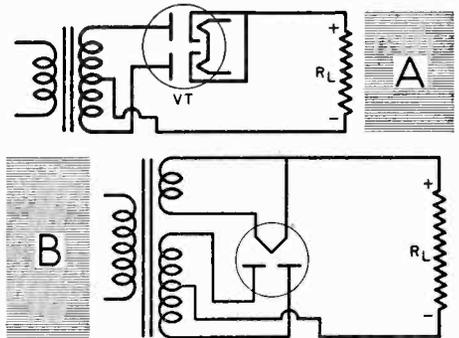


FIG. 11. Twin diode tubes are usually used for full-wave rectification.

filtering job easier. Should one tube pass much more current than the other, the output wave form will become more like that of the half-wave rectifier in Fig. 8A than that in Fig. 10A. In other words, the frequency of the main ripple will be cut in half; since the filter is not designed to suppress this lower frequency properly hum will be produced.

► The transformer is designed to supply approximately equal voltages to the diodes. Very little trouble can occur here. Usually an unbalance is caused by the emission of the two tube sections changing to unequal values. When you are servicing a power supply, check both sections of the rectifier tube in a tube tester and then compare the readings. If the tube sections are greatly unbalanced, you should install a new tube.

# Taking Out The Ripple

As you have already seen, the output of a rectifier is made up of d.c. and a number of a.c. components. We want only the d.c. to pass through the load, so we must "filter out" the a.c. in some manner. A combination of an iron-core choke and a condenser is most widely used for this purpose. Let's see how they work.

## THE L-C FILTER

Fig. 12A shows the power supply circuit we have studied so far. For simplicity,  $S$  is used to represent the transformer-rectifier section, supplying both a.c. and d.c. to load  $R_L$ . If we introduce a choke coil  $L$  in the circuit as shown in Fig. 12B, the d.c. and all the a.c. components must flow from the rectifier circuit through  $L$  and  $R_L$ . Both a.c. and d.c. voltage drops will exist across choke coil  $L$ .

Now, you learned in an earlier lesson that a choke coil may have low d.c. resistance and high a.c. reactance. Because of its low ohmic resistance, negligible d.c. voltage will be lost in  $L$ , and most of the d.c. supplied from the rectifier will be applied to the load  $R_L$ . But if the reactance of choke  $L$  is made large compared to the resistance of  $R_L$ , most of the a.c. voltage will be dropped in the choke. If we select an inductance which will drop a large amount of the fundamental a.c. frequency, even greater voltage drops will exist for the higher frequency components—because the reactance of the choke will increase with frequency.

We could make the inductance of the choke coil so large that only a negligible amount of ripple voltage would appear across the load, but such a coil would be much too expensive and bulky. Instead, we can use an easily obtained coil and introduce a

condenser into the circuit, as shown in Fig. 12C. The condenser  $C$  is chosen so that its reactance is low compared to the resistance of  $R_L$ , so it acts as an a.c. short circuit across the load. As the a.c. circuit in  $B$  includes the reactance of  $L$  and resistance of  $R_L$ , the low-reactance condenser in  $C$  also reduces the total a.c. circuit impedance, causing a greater a.c. flow through the choke and thus increasing the ripple voltage drop across the choke.

As far as a.c. is concerned, you can think of this circuit as containing a low-reactance condenser in series with the high-reactance choke. Since the same current flows through both, there will be a large voltage drop across the high-reactance choke, and just a small voltage across the condenser. As the load  $R_L$  is in parallel with the condenser, it also has only a small a.c. voltage across it. The filtering is just as good with this arrangement as if the choke inductance had been increased.

**Another Explanation.** There is another way of looking at a choke coil-condenser section such as this. The action of the coil is very simple. As you know, it always attempts to prevent a change of current—by stor-

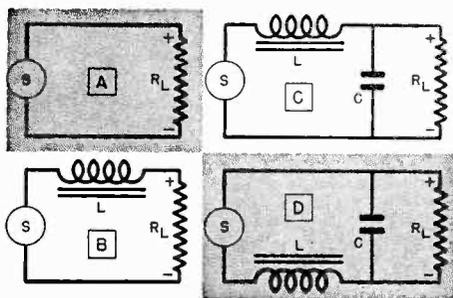


FIG. 12. The choke input filter is formed by using a choke and condenser.

ing energy in the magnetic field when the current is increasing, and releasing energy when the current decreases. It tries to hold the current constant, so the effect of the coil by itself is to "smooth out" the current variations.

The condenser charges as the current pulse from the rectifier increases. When the pulse passes its peak and starts to decrease, the condenser discharges into the circuit. This means the condenser helps make up for the drop in current from the rectifier. Thus, like the coil, the condenser smooths out current variations; the combination of the coil and condenser naturally does this smoothing out better than either of them alone. The effect is that of cutting off the current peaks and filling in the valleys, so only an average d.c. current remains. As these  $L$ - $C$  filters forcibly remove the a.c. component, they are known as "brute force" filters.

**Summary.** To sum up, the d.c. component of the rectifier output is divided between the *choke d.c. resistance* and the *load resistance*. Since the choke coil has a low d.c. resistance, most of the d.c. voltage appears across the load where it is wanted.

The a.c. components are divided between the *high choke reactance* and the *low condenser reactance*, so practically all the a.c. is dropped in the choke coil, leaving little a.c. across the condenser. Since the condenser and load are in parallel, this means that there will be little a.c. ripple across the load to cause hum.

► There is another important point we want you to see clearly. In Fig. 12C we have a series a.c. circuit consisting of the source  $S$ ,  $L$  and  $C$ . As in any series circuit, the current will be the same at any point in the circuit. Therefore, if we move  $L$  to the position shown in Fig. 12D, the same current will flow through it as in Fig.

12C. If the same current flows through  $L$ , the same amount of voltage will be dropped across it and the change in position of the choke does not affect the a.c. voltage division. Hence, it does not matter whether the choke is placed in the negative side or in the positive side of the filter circuit; the same amount of filtering is obtained in either case.

## MULTI-SECTION FILTERS

As you've just learned, a filter is a voltage divider. As far as the ripple is concerned, the divider consists of inductive reactance  $X_L$  and capacitive reactance  $X_C$ . The voltage divides in almost exact proportion to the reactance values, so the ripple reduction factor (ripple input divided by ripple output) equals  $X_L \div X_C$ .

Now suppose that we have one filter which reduces the ripple 100 times, or from 100 volts to 1 volt. This means that the reactance of  $X_L$  is 100 times as great as that of  $X_C$ . Now suppose we feed this remaining 1-volt ripple into a second filter using the same values of  $L$  and  $C$  as the first. Again the filter will reduce the ripple fed to it 100 times—from 1 volt to .01 volt. The *total* reduction is now from 100 volts to .01 volt, which is  $100 \div .01$  or 10,000 times.

Clearly, the reduction obtained from two filter sections is their reduction factor product ( $100 \times 100$  is 10,000). To get a reduction of 10,000 times in a single filter, the choke reactance would have to be 10,000 times the condenser reactance and the expense would be prohibitive. But we can get such a reduction easily, using inexpensive parts of reasonable size, just by using two filter sections as shown in Fig. 13. Filters hooked up this way are called "cascade" filters.

Cascade filters are almost always used in transmitter power packs, where

fewer sections would mean very expensive parts. Sometimes as many as three sections in cascade are used in transmitters, high-fidelity amplifiers and special equipment. In home receivers, a single-section filter, using modern high-capacity electrolytic condensers and a good choke coil, is satisfactory in most cases.

### TUNED FILTERS

You've learned that the larger the reactance  $L$  is, compared to  $C$ , for the

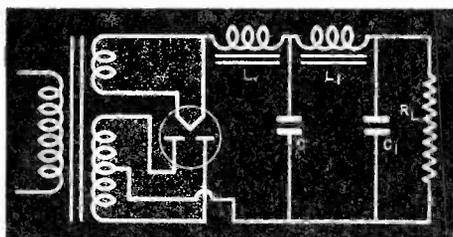


FIG. 13. A cascade filter.

ripple frequency we wish to suppress, the greater will be the ripple reduction. Now, a parallel resonant circuit at resonance has a greater resistance than the reactance of either the inductance or capacity used in it. We can combine these facts to produce a more efficient filter by using a circuit like that shown in Fig. 14A. Here  $L$  has been replaced by a parallel resonant circuit  $L_1-C_1$  through which the d.c. and all a.c. components must pass.

When  $L_1$  and  $C_1$  are tuned to a definite frequency, this circuit will offer a very large impedance to current of that frequency, much larger than  $L_1$  alone. As a result, when the a.c. ripple divides between the resonant circuit and  $C_2$ , most of the a.c. is dropped across the tuned circuit because of its high impedance at the ripple frequency.

For d.c., the resistance of the resonant circuit is simply the resistance of the wire with which  $L_1$  is wound. This

resistance is low, so most of the d.c. appears as a charge across  $C_2$ , from which it is passed on through the next filter section to the load.

A fairly large value of  $L_1$  should be chosen so the capacity of  $C_1$  will be small. At frequencies above resonance, the condenser reactance is lower than that of the choke, and the higher ripple frequencies are by-passed around the choke. By keeping the capacity of  $C_1$  small, this effect is reduced, but it is desirable to have another filter section  $L_2-C_2$  to be sure to eliminate the higher ripple frequencies.

► Now, a series resonant circuit has a low impedance at the resonant frequency. In a filter, the shunting condenser should have low reactance. This suggests that, instead of a parallel resonant circuit in place of  $L$ , we might use a series resonant  $L-C$  circuit in place of  $C$ . Such a circuit is shown in Fig. 14B.

When  $L_2$  and  $C_1$  are in resonance with the lowest ripple frequency, they offer very little opposition to that component. Therefore, when the a.c. ripple divides between  $L_1$  and the resonant circuit, most of the ripple is dropped across  $L_1$ . This circuit is effective at the frequency for which it is designed (the lowest ripple frequency), but at higher frequencies, resonance does not occur and the circuit acts like an inductance. Thus, all component ripple voltages higher in frequency divide between  $L_1$  and  $L_2$ , creating a high-frequency ripple across  $L_2$ . It is for this reason that a tuned filter must always be followed by another untuned section ( $L_3-C_2$  in Fig. 14B).

For 60-cycle, half-wave rectifiers, the circuit  $L_1-C_1$  in Fig. 14A or  $L_2-C_1$  in Fig. 14B would be made to resonate to 60 c.p.s.; for a full-wave rectifier, these circuits should resonate to 120

c.p.s. As you've learned, these are the lowest ripple frequencies found in these rectifiers.

► You may find a radio receiver or amplifier, using a tuned filter, which has a hum that seems to defy correction. This is often caused by a change in the inductance value of the iron-core choke (shift in the laminations from rough handling), or perhaps the filter circuit is not quite tuned to the proper frequency. To remedy this, try varying  $C_1$ . Small condensers added in parallel will gradually increase the capacity. Should the condenser already be too large, use a smaller one in place of it and add others in parallel until minimum hum is obtained.

► If condensers  $C_2$  and  $C_3$  in a power supply like that shown in Fig. 14A are small paper condensers, quite often the best way to make repairs is to eliminate the tuned filter altogether by removing its tuning condenser  $C_1$ . Then you must substitute high-capacity electrolytic condensers for the small paper filter condensers. For the circuit of Fig. 14B, remove  $L_2$  and replace  $C_1$  and  $C_2$  with higher capacity electrolytics. The results with the new filters will, in many cases, be better than those with the original tuned filters in these cases.

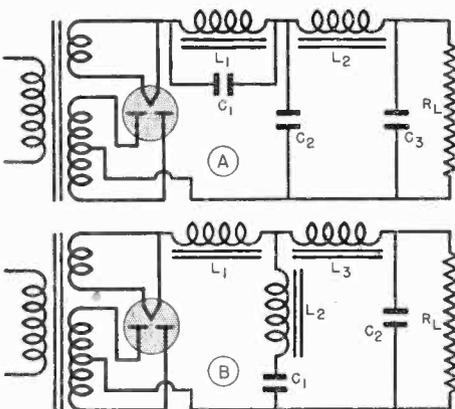


FIG. 14. Examples of tuned filters.

► In each of the filters we have so far studied, the output of the rectifier has been fed into a choke. This is known as a "choke input type" filter. Voltage division has been used to cause the d.c. component of the rectifier output to appear across the load, and the energy in the a.c. components has been deliberately wasted. Now we will see how some of this energy may be captured and put to use.

### CONDENSER INPUT FILTERS

Suppose we connect a condenser  $C$  across  $R_L$  in the half-wave rectifier

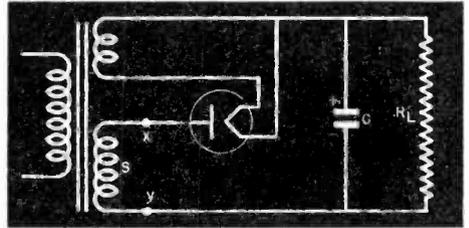


FIG. 15. A condenser alone forms a partial filter, delivering a much higher d.c. voltage providing the current demand is low.

circuit shown in Fig. 15. What will happen? As before, the rectifier will try to produce an output wave form like that shown in Fig. 16A. Notice we say "try"—actually something entirely different will be produced.

When the a.c. voltage makes terminal  $x$  positive with respect to terminal  $y$  of transformer secondary  $S$ , the tube will pass current. As the tube current increases, condenser  $C$  is charged up until the condenser voltage builds up approximately to the peak \* of the a.c. voltage.

After reaching its peak value, the a.c. voltage begins to decrease and the plate current drops. Now the con-

\* The peak value is the maximum positive or negative value of an a.c. cycle. It is 1.41 times the r.m.s. value which is read by an a.c. voltmeter. Thus, the peak is 423 volts when the r.m.s. value is 300 volts.

denser is charged to a high voltage. This charge cannot escape quickly, so when the source voltage drops, it leaves the condenser voltage higher than the source voltage. The condenser voltage is in series with the supply voltage in winding *S*, between the tube cathode and plate, and (because of the polarity of the condenser) opposes this supply voltage. As soon as the condenser voltage is greater than the supply voltage, the plate of the tube is made negative. This cuts off the current flow through the tube.

The electrons stored in the condenser now leak off through the load  $R_L$ . The condenser thus acts as a source of current for the load, and gradually discharges in the process. When the a.c. voltage again becomes positive, the tube conducts as soon as the supply voltage becomes higher than the voltage of the partly discharged condenser. The condenser is then charged again to the peak voltage of the supply, and the cycle already described repeats.

This action is shown in Fig. 16*B*, in which the heavy line shows the voltage impressed across the load. As the a.c. voltage builds up from zero to the peak value, the condenser charges. When the a.c. decreases from the peak toward zero, it drops below the condenser voltage, and the tube no longer conducts. Gradual discharge of the condenser brings its voltage down by the time the next pulse rises from zero. Then, the condenser is recharged and current comes from the tube. As the source voltage goes from the peak toward zero, the condenser again takes over.

The important thing for you to understand is that the condenser acts as a source for the load for a longer part of the cycle than does the tube.

The gradual reduction in voltage across the condenser, shown by the

slanting lines between the peaks, is caused by the discharge of the condenser through load  $R_L$ . If the load resistance is made lower in ohmic value, the condenser will lose its charge and voltage more rapidly. When the condenser is charging, only the tube resistance and the transformer impedance limit the charging current, so the condenser reaches full charge almost at once.

The time in seconds it takes to drain a condenser of 63% of its original

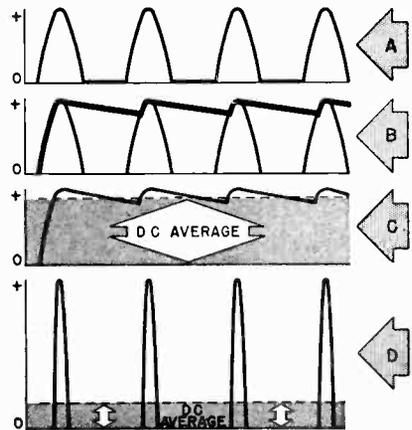


FIG. 16. The rectifier output when using a condenser input filter.

charge equals the condenser capacity in mfd. multiplied by the load resistance in megohms. Thus, if  $R_L$  in Fig. 15 has a resistance of 5000 ohms (.005 megohm) and  $C$  a capacity of 20 mfd., it will take  $.005 \times 20$  or  $1/10$  of a second to drain the condenser. But the condenser is being recharged 60 times a second (if the power line is a 60-cycle supply) and therefore never fully discharges. The wave form of the voltage across the condenser is shown in Fig. 16*C*. This wave consists of a d.c. component and small a.c. ripple components, just like other pulsating currents. But the d.c. average is almost equal to the peak of the a.c. cycle—instead of being only

about  $3/10$  of this value as it was for the ordinary half-wave rectifier shown in Fig. 7B. Clearly, the use of this condenser gives a much higher d.c. output for the same a.c. voltage input.

Furthermore, the a.c. ripple is far less, so the condenser by itself is a partial filter. A regular L-C filter is used between the condenser and load to remove the remaining ripple. A typical power pack using this filter system is shown in Fig. 17. This filter is known as the "condenser input" type, because the tube output goes to  $C_1$  first. You will find condenser input filters in most radio receivers.

► But there is a price to pay for the increased d.c. output and lowered ripple obtained from the condenser input filter. The rectifier tube only supplies current for a small period of time, yet this current, by charging the condenser, must operate the receiver continuously. This means the current peaks must be very high and sharp.

The current into the condenser and load from the tube is shown in Fig. 16D. The average d.c. is low, because the rectified current pulses exist for short time periods, yet it must equal the current required by the load.

Therefore, the rectifier tube works very hard for short periods of time. It is important that the load be kept at a reasonable value; otherwise, this terrific rush of current can damage the tube. The tube must be rated for not only the average current but also this peak current.

Where the load resistance is so low that the condenser discharges considerably, the peak current required may be too high for any standard tube. If so, a choke input filter must be used. You will find choke input filters in the power supplies of transmitters, in many public address systems, and in some radio receivers where high power is required.

## PRACTICAL FILTER FACTS

The performances of the two main types of filters—choke input and condenser input—are most easily compared by means of curves. Fig. 18 shows a family of curves for a type 5V4G tube using an input filter condenser.

These curves show you how the voltage at the *input* of the filter varies with load current for different input capacitances. (The *output* of the filter will be equal to the input to the filter minus the drop in the filter choke. Since this drop depends on

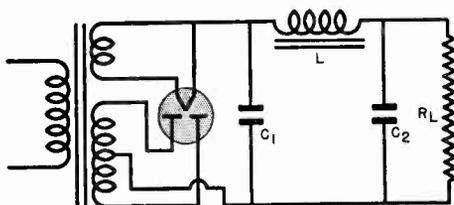


FIG. 17. A complete condenser input filter.

the d.c. resistance of the choke, and therefore will probably be different for every filter, the best way to compare performances of two different filters is to measure their *input* voltages. This is what we do in Figs. 18 and 19.)

To use the graph in Fig. 18, locate the curve you want. The one at the top marked *400 RMS Volts Per Plate*,  $C = 8 \text{ MFD}$ , will do as an example. *400 RMS Volts Per Plate* means that this curve was made with 400 volts a.c. applied to each plate of the full-wave rectifier tube. The input condenser has a value of 8 mfd., shown by the notation  $C = 8 \text{ MFD}$ . Now let us see how much input voltage to the filter is obtained when the load is 40 ma. Locate 40 ma. at the bottom of the graph, move straight up to the curve and then straight over to the left of the graph. This brings us to about 538 volts, the input voltage to

the filter when 40 ma. are drawn by the load.

We started with 400 volts and we are getting over 500! But the 400 volts is an r.m.s. value, so the a.c. *peak* is about 564 volts. Our value of 538 volts d.c. is thus close to the peak value.

Using the same curve, you will find the input voltage drops to 490 volts at a current of 120 ma. This shows that "loading" the circuit (increasing the load current) reduces the input voltage considerably. This drop is caused mostly by the fact that the condenser had to lose a great deal of charge to supply the higher current. If a smaller input condenser is used, the drop with increased load current becomes even greater (as shown by the curve marked 4 mfd), because the condenser loses its charge even more rapidly.

► Now let's use the curves in Fig. 19 to compare the effect of load current changes on the filter input voltage for both choke and condenser input. The condenser curves are the solid lines; the choke curves are dotted.

You see at once that the d.c. voltage at the input of the filter is considerably less for choke input than for condenser input. This is shown by the fact that the condenser input curves for a definite a.c. plate voltage are higher upon the graph. Thus, at a current of 100 ma., we can get about 455 volts d.c. from a 400-volt a.c. input to a condenser input filter, while we can get only about 330 volts when this same 400 volts a.c. is used with a choke input filter.

To put this another way—with a condenser input filter, 400 volts a.c. must be applied to each rectifier plate to obtain 400 volts d.c. with a load of 180 ma. If you use a choke input, you will need about 500 volts a.c. per plate to get the same results. Thus,

for a fixed d.c. filter input, a choke input filter needs a transformer capable of producing a higher a.c. rectifier plate voltage than is needed with a condenser input filter.

► Should the input filter condenser open up (lose its capacity), the d.c. voltage would be greatly reduced, because the circuit then acts effectively as a choke input filter (as if the condenser were not there.)

► The condenser input filter thus has advantage over the choke input type. However, while the voltage drops rapidly as the load current increases in a condenser input filter, it changes very little when a choke input is used. This change in voltage as the load current changes is called regulation.\* The better regulation obtained with a choke input makes this arrangement especially valuable when the load current goes through wide variations. Such variations occur particularly in transmitters and in high-power audio amplifiers used in some receivers and in public address systems.

## SWINGING CHOKE

The dotted line curves shown in Fig. 19 were made with an input choke having a definite inductance in henrys. If some other value choke is used, the shape of the curve will be approximately the same, but it will be higher up on the graph if the inductance is less, and lower (reducing the filter input voltage) if its inductance is greater.

In other words, the smaller the value of the input choke in henrys, the greater the input voltage to the filter, because this permits the following condenser to act more like an input condenser. As you've just learned, a condenser input filter always has a

\* Briefly, regulation is the change in voltage which occurs when the load current changes. The regulation is said to be good when the change in voltage is small.

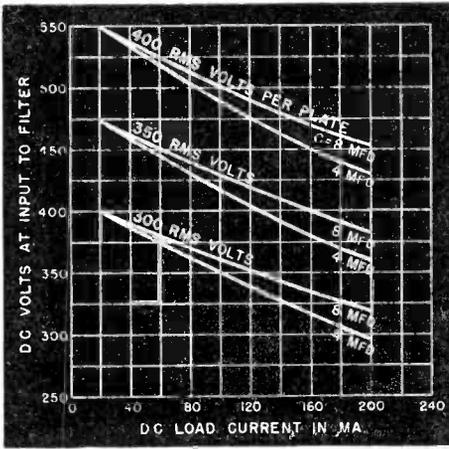


FIG. 18. These curves show that larger input filter condensers give higher output voltages, for large load currents.

higher input voltage than a choke input filter for the same rectifier plate voltage.

Now, as you learned a short while ago, the *output* of a filter (that is, the voltage furnished to the load) equals the *input* to the filter minus the d.c. voltage drop within the filter. Since the resistance of the filter is constant, this d.c. voltage drop in the filter naturally rises as more current is drawn through the filter. Also, there are d.c. voltage drops in the rectifier tube and transformer. This means that the output voltage of an ordinary filter will drop as more current is drawn by the load.

But if the input choke inductance can be made variable, so it can be decreased when the load current increases, the input voltage will rise and make up for the d.c. drops inside the power pack. Thus, the *output* voltage will remain almost constant.

By using a choke of special design, we can obtain a variable choke inductance. The air gap of this choke is purposely made small so that the core may be easily saturated. As you learned from your study of iron-core

coils, saturation occurs when increases in current no longer cause increases in flux. When this condition is approached, the inductance of the choke decreases—which is just what we want to maintain constant output voltage.

Thus, by designing the choke so it will approach core saturation as the load current increases, we get a variable input choke inductance which gives us a more constant load voltage. Since the inductance of such a choke increases and decreases, it is called a *swinging choke*. You won't often find one in radio receivers, but they are common in public address systems, transmitters and electronic control devices.

Of course, the swinging choke gives a variable amount of ripple reduction, so it is used only as an input choke. Another standard filter section is always used after this choke, forming a circuit like Fig. 13.

► Saturation will occur in any iron-core choke if d.c. in excess of its rated value flows through it. Leakage in an output filter condenser (such as  $C_2$  in

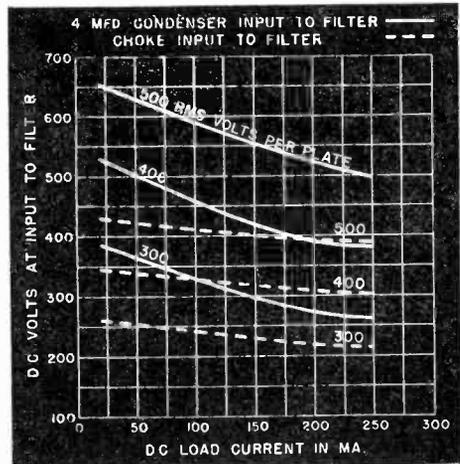


FIG. 19. Comparing the output of choke and condenser input filters, we find the condenser input type delivers a higher d.c. voltage with poorer regulation.

Fig. 17) can let enough current flow through filter choke  $L$  to saturate it. The reduced inductance of  $L$  destroys

the filter action and permits the ripple voltage to pass on to the load, causing hum.

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## Dividing the Voltage

The high d.c. voltage delivered by the power pack filter is used to supply B and C voltages to various tubes. Often these tubes require different plate, screen grid, suppressor grid, cathode and other voltages. Let us now see how these voltages are obtained from the output of the filter.

### D.C. Circuit Laws to Remember.

In order to understand d.c. voltage distribution circuits, you must remember these important circuit facts:

- 1. The currents flowing to a terminal equal the currents flowing away from that terminal (Kirchhoff's Current Law).
- 2. In any *complete* circuit, the supply voltage equals the sum of all the voltage drops (Kirchhoff's Voltage Law).
- 3. The voltage drop across any device equals the current times the resistance (Ohm's Law).
- 4. The polarity of a voltage drop can be determined by remembering that the terminal of a device at which electrons enter is negative with respect to the terminal at which they leave.
- 5. Points connected together by a resistor, coil, or wire through which d.c. current is not flowing are at the same d.c. potential—there is no voltage drop between them.
- 6. "Voltage" always means the voltage *difference* between two points. Thus, when you speak of plate voltage you mean the voltage difference between the plate and cathode, and when you speak of grid voltage you mean the voltage difference between the grid and cathode.

- 7. A point can be negative with respect to one point and at the same time be positive with respect to some other. Thus, in a tube circuit, the cathode is negative with respect to the plate and, at the same time, positive with respect to the control grid.
- 8. If two points are grounded, there is no d.c. voltage difference between them. They act as if they were wired together directly.

### BASIC DIVIDERS

If the power supply furnishes exactly the right amount of voltage for the load, we can connect the load directly across the d.c. supply terminals. Since the voltage is correct, the right current will flow. We need only to be sure the supply can furnish all the current required.

But in any radio device there will be many loads requiring only fractional parts of the supply voltage, and not all of these loads need the same amount of voltage. How can we divide the power supply voltage to meet all requirements?

**Series Voltage Drops.** If the power source furnishes more voltage than is needed by the load, the correct load voltage may be obtained by placing a resistor between the power supply and load, as shown in Fig. 20A. The supply voltage will divide between the load resistor  $R_L$  and the voltage-dropping resistor  $R_1$ . If we choose the correct value of  $R_1$ , just enough voltage will be dropped across it to leave the right amount of voltage for  $R_L$ . We can figure the value

of  $R_1$  by using Ohm's Law, which says that  $R_1 = V \div I$ . Here  $I$  is the current which is to flow through the load \* and  $V$  is the excess voltage we wish to drop across  $R_1$ .

Remember—the series dropping resistor may be placed on either side of the load or, if necessary, we can split the series resistor, placing part on each side of the load. The same current flows through each part of a series circuit, so the same voltage drops will occur as long as the total resistance is the same.

► Suppose we connect a number of similar loads in parallel, as in Fig. 20B. Now the current through  $R_1$  is the sum of the load currents, so the value of  $R_1$  must be figured by dividing the voltage drop which is to occur

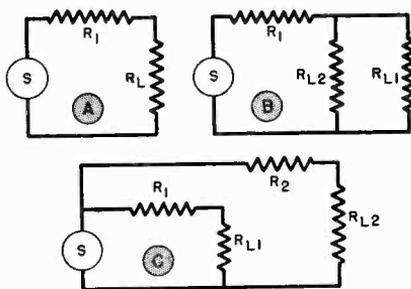


FIG. 20. Series resistors used to drop the voltage.

across it by the sum of the load currents. This connection can be used when the voltages required by the loads are the same—even though different currents may be required.

► When different voltages are needed—for example, to supply a plate and a screen grid—the method shown in Fig. 20C may be used. Resistors  $R_1$  and  $R_2$  are figured for the load each is used with.

Both Fig. 20B and 20C can be extended to any number of loads. This voltage dividing method is known as

“series voltage dropping,” because the dropping resistor is in series and is the only extra part used.

**Bleeders.** Frequently, the load current varies over wide limits. This produces a varying voltage drop across the series resistor, thus changing the division of voltage between the resistor and the load. We can reduce this variation or eliminate it entirely by drawing off or, as technicians say, “bleeding” extra current through the series resistor. A resistor known as a “bleeder” is connected across the load to draw the extra current.

The circuit looks like Fig. 20B, in which  $R_{L2}$  will act as a bleeder across  $R_{L1}$ . By choosing bleeder  $R_{L2}$  so it draws far more current than does  $R_{L1}$ , the voltage drop across  $R_1$  will be caused mainly by the bleeder current flowing through it. Then, small variations in the value of  $R_{L1}$  can cause only very little change in the total current through  $R_1$ , so the voltage drop across  $R_1$  remains practically constant. This means the voltage across  $R_{L1}$  is practically constant no matter what current  $R_{L1}$  draws (within limits, of course), so the voltage regulation at  $R_{L1}$  is very good. The bleeder resistor must be designed to draw a great deal more current than the variations expected in the load current, yet not draw more current than the supply can furnish. Of course, if the load current does not change or the change does not matter, no bleeder is needed—a series dropping resistor is sufficient.

**Voltage Dividers.** Instead of using a separate series resistor and a separate bleeder resistor, we can use a single resistor, tapped to form a voltage divider as shown in Fig. 21A. Section  $R_1$  is the series section, while  $R_2$  is the bleeder section.

Additional taps may be provided on  $R_1$  to supply various loads (see Fig.

\* Since this is a series circuit, the load current also flows through  $R_1$ .

21B). Here the bleeder current  $I_B$  through bleeder resistor  $R_2$  flows through all sections of the voltage divider. In section  $R_4$ , currents  $I_B + I_3$  flow; in  $R_3$ , currents  $I_B + I_3 + I_2$  flow, and in  $R_1$ , we have  $I_B + I_3 + I_2 + I_1$  flowing. In each section, the value of the resistor in ohms equals the voltage across the resistor divided by the current flowing through it.

All possible combinations of loads, series dropping resistors and bleeders have not been shown, but Figs. 20 and 21 show the basic methods of voltage division. Reference to them will help you solve any power supply voltage divider network, *provided you understand Kirchhoff's Current and Voltage Laws.*

WE HAVE STRESSED THESE LAWS BEFORE, BUT THEY ARE SO IMPORTANT IN PRACTICAL RADIO WORK THAT WE REPEAT THEM AGAIN:

1. The sum of the currents flowing to a point is equal to the current flowing away from that point. Thus, in Fig. 21B,  $I_B + I_3$  equals the current flowing through  $R_4$ .

2. The sum of the voltage drops in a complete circuit equals the source voltage which produced them. For example, in Fig. 21A the supply voltage divides between  $R_1$  and the parallel resistors  $R_2$  and  $R_L$ . The same voltage exists across  $R_2$  and  $R_L$ , and this voltage added to that across  $R_1$  equals the source voltage  $S$ . Similarly, in Fig. 21B, the sum of the voltages across  $R_1$  and  $R_{L1}$  equals the source voltage. Also, the sum of the voltages across  $R_{L2}$ ,  $R_3$ , and  $R_1$  equals the source, and so on.

### PRACTICAL DIVIDER CIRCUITS

Fig. 22 gives an example of series dropping resistors in a practical cir-

cuit. Plate current flowing through resistor  $R$  causes a voltage drop across  $R$  which reduces the supply voltage to the correct amount for the plate. In the same way, screen current through  $R_1$  causes a voltage drop across this resistor which reduces the voltage to the desired screen value. (The resistance of each resistor, as you know, is calculated by dividing the required voltage drop by the electrode current through the resistor.)

► The cathode current (which is the sum of the plate and screen currents)

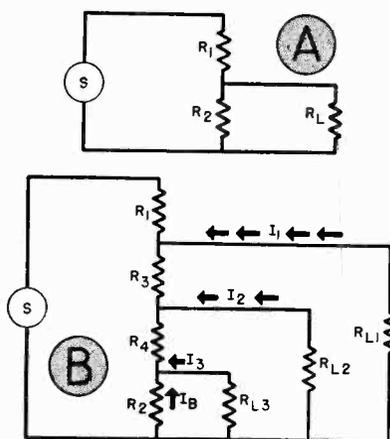


FIG. 21. Examples of voltage dividers.

flows through  $R_2$ , dropping only a few volts across it because of the low ohmic value of  $R_2$ . The polarity\* of the voltage drop is as shown. Since the grid connects to the negative side of the drop and the cathode to the positive side, this voltage is between the grid and cathode, and so biases the tube. For this reason,  $R_2$  is known as the cathode resistor, or as a C bias resistor, or simply as a bias resistor.

\* To determine the polarity of a voltage drop, remember: THE END OF A RESISTOR (OR OTHER PART) AT WHICH ELECTRONS ENTER IS ALWAYS NEGATIVE WITH RESPECT TO THE END AT WHICH THEY LEAVE. Tubes serve as signposts for electron direction, since electrons always move from cathode to plate.

► If a number of tubes require the same screen voltages, the screens are joined together and the combined screen currents flow through  $R_1$ . Since the combined currents through  $R_1$  will now be greater, the ohmic value of  $R_1$  must be decreased if the same screen voltage is to be maintained. The value of  $R_1$  in ohms will be equal to the required voltage drop divided by the combined currents flowing through it.

► Where several tubes require equal C voltages, the cathodes could be joined together. The ohmic value of  $R_2$  would then equal the desired C voltage divided by the combined cathode currents. Ordinarily, however, separate C bias resistors are used for each tube to prevent a possible variation in the current of one tube from affecting the bias of the others.

► To produce equal plate voltages for several tubes, we connect the plate supply circuits in parallel. Resistor

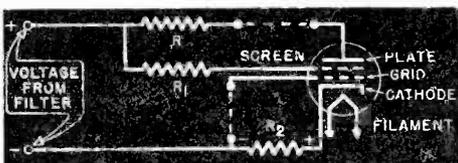


FIG. 22. A practical example showing series voltage-dropping resistors.

$R$  is then figured by dividing the voltage drop required across  $R$  by the sum of the plate currents which flow through it. Often the voltage from the power supply is correct for the tube plates; then resistor  $R$  is not needed.

► If there are other tubes which do not require the same plate, screen and grid voltages, each of these tubes will have its own plate, screen and bias resistors. Even where several tubes use the same voltages, each one may have its own series resistor to prevent variations in one tube from affecting

the operating voltages of the others.

► Fig. 23 shows a typical bleeder voltage divider. Screen grid voltage reduction is produced by  $R_1$ , which carries both the bleeder current drawn by  $R_2$  and the screen grid current of the tube. The bleeder serves to stabilize or hold steady the screen grid voltage. Should the bleeder open, however, only the screen grid current would flow through  $R_1$ , so its voltage drop would be less and more

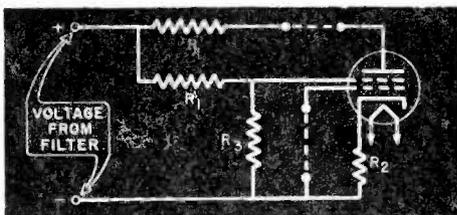


FIG. 23. A typical bleeder circuit.

voltage would be applied to the screen grid.

► A practical application of the voltage divider is shown in Fig. 24. Here, resistor  $R_2$  is the bleeder. Series resistor  $R_3$  provides the correct screen voltage, and the C bias voltage is produced across  $R_1$ . Notice that both point 2 on the voltage divider and the cathode of the tube are grounded. The cathode and bleeder current electrons come from the minus supply terminal and flow through  $R_1$  to point 2. Here they divide, the cathode electrons flowing through the chassis to the cathode, and the bleeder electrons continuing on through  $R_2$ .

► Fig. 25 shows three common ways of getting bias voltage from the power pack. In Fig. 25A, resistors  $R_1$  and  $R_2$  (this may be a single tapped resistor) are placed in the power supply B—lead. Their action is similar to that of  $R_1$  in Fig. 24, except that the cathode currents of all the tubes now flow through the resistors. Grids of tubes which need the full voltage for

bias are connected to  $C-$  — while grids of tubes needing less bias are connected to  $C-$ .

You recall that a d.c. voltage drop exists across the choke and that the choke can be in the negative side of the circuit. Fig. 25B shows how we can use this drop to supply bias. Here, the drop across the inductance is higher than needed for bias, so the inductance is tapped. This is common practice when the speaker field\* furnishes the filter inductance, as the drop may be 100 volts or so and the

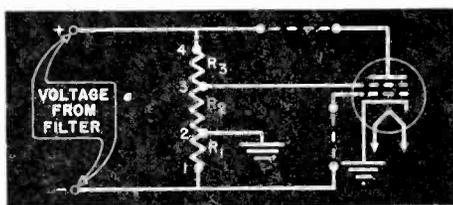


FIG. 24. A complete voltage divider.

required bias is usually 50 volts or less. This scheme is mostly used for power output tube bias.

Another way of using the voltage drop across the inductance is shown in Fig. 25C. Here a voltage divider is arranged across the inductance, with the resistances figured to give the desired voltage between  $B-$  and each tap. Since no current is taken by grid circuits, these resistors can be high values—in fact, they must be to prevent the resistors from interfering with the action of the inductance.

\*The speaker field is a large inductance, requiring a d.c. current flow to establish a fixed magnetic field for proper loudspeaker operation. Since it has sufficient inductance, it can be used as the choke coil in a filter. The normal d.c. current of the set then provides the necessary field excitation. The d.c. resistance of a field is higher than that of the average choke coil, but proper power transformer design makes up for the extra d.c. voltage drop. The method described here makes use of this drop for bias purposes, so the single part is used as a speaker field, a choke coil and a source of bias.

## VOLTAGE REGULATOR TUBES

Sometimes the voltage must be held very closely to the design value. Bleeder resistors are not perfect—in order to hold the voltage within small limits, the bleeder must draw a current many times higher than the possible variations drawn by the load. Often the power supply cannot furnish such high currents.

Further, if a line voltage change makes the power pack output change, the bleeder is helpless. As you can readily see by examining any of the bleeder circuits shown, the voltage across the bleeder will rise or fall as the source voltage rises or falls.

For these reasons, a voltage regulator tube (Fig. 26) is better than a bleeder for keeping voltage constant. This regulator tube  $VR$  is a gas-filled tube without a filament. When suf-

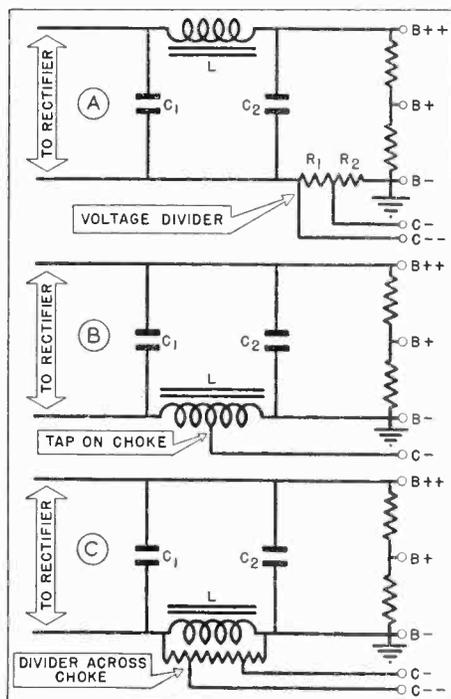


FIG. 25. Three ways of getting bias from the power pack.

ficient voltage is applied to the tube, the gas ionizes and becomes conductive.

Unlike a bleeder, this tube does not have constant resistance. Its internal resistance depends upon the number of gas ions between the elements of the tube. When the voltage from the filter rises, the number of ions increases; the tube resistance drops, and the tube draws more current. When the voltage drops, the ionization is less; the tube resistance rises, and the current flow drops.

This means that a rise in the filter voltage will cause an increased current flow through  $VR$ , which in turn means a larger drop across  $R$ , as this current flows through  $R$ .

This drop across  $R$  automatically compensates for the increase in source voltage, so the voltage applied to tube  $VT_1$  remains very nearly constant. Similarly, when the load current decreases, the  $IR$  drop decreases and the voltage across  $VR$  increases. This causes a sharply increased current through  $VR$ , restoring the drop across

$R$  to about its original value.

Should the opposite condition occur, then  $VR$  draws much less current, allowing the voltage to rise.

With a resistor  $R$  of proper size, this regulating circuit will keep the

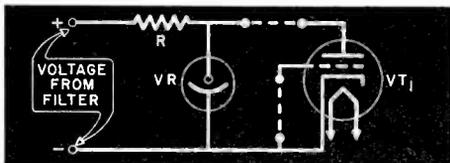


FIG. 26. Using a voltage regulator tube.

voltage supplied to  $VT_1$  from varying more than 1 or 2 volts under normal circumstances. The current taken by the  $VR$  tube ranges from a few ma. to about 30 ma., which is a reasonably small amount of current.

Voltage regulator tubes rated at 75, 90 and 150 volts are available, known as the  $VR75$ ,  $VR90$  and  $VR150$  types. By using several of these tubes in series, or by using a voltage divider across a tube, higher and lower voltages can be controlled.

## Facts About Rectifier Tubes

Not every radio receiver or amplifier uses the same type of rectifier tube. Just as tube manufacturers make many tubes with different characteristics which can be used for detectors, amplifiers and power output purposes, so do they make many types of rectifier tubes from which the radio designer can choose the one best suited to a particular power supply.

### VACUUM RECTIFIERS

Ordinary rectifier tubes are evacuated just like other receiving and transmitting tubes. Hence, they are called "vacuum rectifiers."

You have already seen curves in Figs. 18 and 19 showing the input voltage available from choke and condenser input filters. These curves were taken with a particular type of rectifier tube; they may be quite different for another type, because of the resistance between the plate and cathode of the rectifier tube. This resistance is between the transformer source and the load. When the tube is supplying current, a voltage drop exists in it. Variations in the load current requirement cause a varying tube voltage drop—and hence a variation in the output voltage, since the tube drop

is subtracted from the source voltage.

If the cathode and plate are spaced close together, the tube resistance will be low, and the voltage drop across the plate-cathode of the tube will be low even with high load currents. Naturally, even a fairly large current variation can't cause much total change in the voltage across a low resistance, so this tube construction gives improved regulation. But the a.c. voltage which can be applied to the plate of such a tube is limited, since too much voltage will cause a flash-over between the plate and cathode and ruin the tube. Let's look into this matter of flash-over a little further.

**Peak Inverse Voltage.** A rectifier tube should pass electrons from the cathode to the plate when the plate is positive. But what happens when the cathode is positive with respect to the plate, a condition which exists once every cycle of the a.c. supply?

If the voltage is too high, a current will arc (spark) between the elements or between the leads coming through the glass "pinch" (or seal) of the tube. This either breaks the seal, letting air into the tube, or destroys the emission qualities of the cathode. The tube is ruined in either case.

The maximum a.c. voltage that you can connect to a rectifier tube without causing an arc when the plate is negative and the cathode is positive is called the *peak inverse voltage* rating of the tube.

Fig. 27 shows a single diode connected through a load  $R$  to a high-voltage secondary. Once each half cycle the tube plate is positive (+), while the cathode ( $K$ ) is negative (-), and once each half cycle the plate is negative (-) and the cathode is positive (+). When the diode passes electrons, the voltage across the tube is lower than  $V_{a.c.}$  because of the

voltage drop in the load  $R$ . But when the inverse condition exists (that is, when the plate is negative), no current flows through the tube. Then the voltage  $V_{d.c.}$  stored in the condenser  $C$ , added to the peak  $V_{a.c.}$  across the high-voltage secondary, is applied between the cathode and plate. If the load current is low, the voltage across  $C$  may nearly equal the peak of  $V_{a.c.}$  (The peak value is 1.41 times the value measured by an a.c. voltmeter.) Therefore, the maximum inverse voltage which is applied between the plate and cathode is  $V_{a.c.} \times 1.41 \times 2$ —twice the peak voltage of the transformer secondary. This value must not exceed the peak inverse voltage rating of the tube.

In the case of a full-wave rectifier, the peak inverse voltage will be applied first between one plate and its cathode, and then between the other plate and its cathode. Therefore,

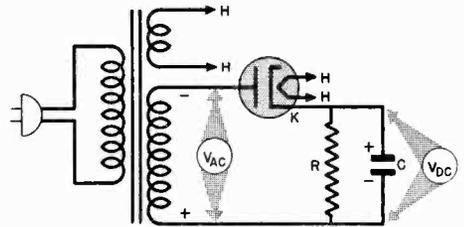


FIG. 27. The peak inverse voltage is the sum of the d.c. output and the peak a.c. input.

both diodes of the full-wave rectifier tube must have the same peak inverse voltage rating.

With a choke input filter in place of the condenser, the voltage  $V_{d.c.}$  is considerably reduced. This means that when an input choke is used, a larger a.c. voltage may be applied to the rectifier plates without danger of exceeding the manufacturer's inverse peak voltage rating.

Thus, in choosing a rectifier tube, we must consider not only its d.c.

output current and voltage, but also its peak inverse voltage rating and the peak current which it can safely pass.

## MERCURY VAPOR RECTIFIERS

Where we need both high voltage and good regulation, we cannot use a vacuum type tube because the close spacing necessary for good regulation limits the peak inverse voltage rating. Instead, the mercury vapor tube is used. This tube is like a vacuum type with wide plate-to-filament spacing, but contains a small ball of mercury. When the tube heats up, a quantity of mercury vapor is formed and at the same time, an electron cloud forms around the filament.

As electrons speed toward the plate, they strike mercury atoms in the vapor and knock out free electrons, leaving positive gas ions. The extra electrons move toward the plate along with the original electrons. The positive ions move toward the filament getting into the electron cloud, where they combine with electrons and partially destroy space charge effects. This lets more electrons reach the plate from the filament, because fewer are repelled by the cloud back to the filament.

When the load requires more current, more electrons pass through the tube. This increases the collisions with mercury atoms, thus producing more free electrons and further reducing the space charge. Effectively, this is the same as reducing the plate-to-cathode resistance. The increased current through the smaller tube resistance gives about the same tube voltage drop as before, so the output voltage remains almost constant. In fact, the tube drop stays at about 15 volts for all values of current which the tube can safely handle. This constant voltage drop means the d.c. output does

not vary with load current changes, so regulation is excellent.

The heavy ions must not strike the filament as they can destroy its emission. Hence, the peak current must be kept within the tube rating and the electron cloud must form before the tube passes current. Since the mercury vapor tubes used in receivers have quick-heating filaments, this is no problem. When large mercury vapor rectifier tubes are used in transmitters and high power public address systems, a special time-delay relay is

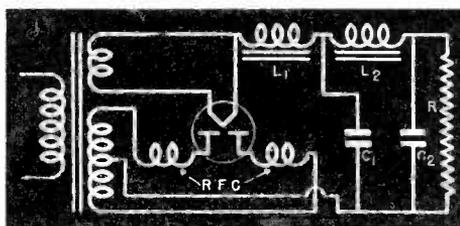


FIG. 28. The mercury vapor tube requires r.f. chokes in the plate leads to keep down interference.

used to turn on the plate power after the filament power is applied and the tube has warmed up.

A typical full-wave mercury vapor rectifier circuit is shown in Fig. 28. In operation, the rectifier tube emits a blue glow which quickly distinguishes it from the vacuum type rectifier tube (which emits no glow when in good condition\*). The filter circuit has a choke input, for a condenser

\* When a vacuum tube rectifier has been overloaded by excess current, usually caused by a shorted filter condenser, gas is often formed in the envelope. Then there will be a blue glow which at first can be seen inside the space between the plate and cathode. Later, this glow will increase as more gas molecules are released from the metallic structure of the tube. The glow may spread over the entire interior of the tube. Frequently, the excess current causes the rectifier plates to become red hot. The glow in a vacuum type tube is steady and is not usually as brilliant as the flickering glow of a mercury vapor rectifier.

input would cause sharp peak current rushes which would ruin the tube. To improve regulation, the input choke is usually a swinging choke.

Sudden current changes in the rectifier circuit may set up radio frequency (high-frequency a.c.) currents which will interfere with the signals in the signal circuits. For this reason, small r.f. chokes *RFC* are used in each plate circuit. The tube may have to be shielded to prevent stray fields from getting into the signal circuits. Shielding should not be used unless necessary, as it prevents the heat generated by the tube from escaping and so eventually causes tube destruction. A fuse in the power line is recommended; then, if a short circuit or sudden overload causes excessive tube current, the fuse will burn out before the tube cathode is damaged.

### BRIDGE RECTIFIER CIRCUIT

The bridge rectifier circuit is a circuit which uses four diodes to get the effect of full-wave rectification from a half-wave transformer. Since the saving in the cost of the transformer is more than the tube cost only in high-power equipment, bridge rectifier circuits are usually found only in transmitting and measuring circuits. Full-wave rectifiers are much more common in radio sets.

A typical bridge circuit is shown in Fig. 29. You will find it easy to remember this circuit by following the electron flow. For example, when terminal *x* is negative with respect to terminal *y* of transformer secondary *S*<sub>1</sub>, electrons flow from the supply terminal *x* of *S*<sub>1</sub> to the rectifier terminal *a*, and seek a path through the bridge circuit to terminal *y*. Since the only electron path through a normal tube is from the cathode to the plate, the

electrons cannot go through tube *VT*<sub>4</sub>; instead, they pass through tube *VT*<sub>1</sub> to point *b*. The only electron path from point *b* is through the load resistor *R* to point *c*, because the path through tube *VT*<sub>2</sub> is blocked. From point *c*, the electrons travel through tube *VT*<sub>3</sub> to point *d* and then to the supply terminal *y*. This electron path is shown by the solid black arrows.

When the cycle reverses, electrons leave the supply terminal *y* and take the *y*→*d*→*b*→*c*→*a*→*x* path, indicated by the white arrows.

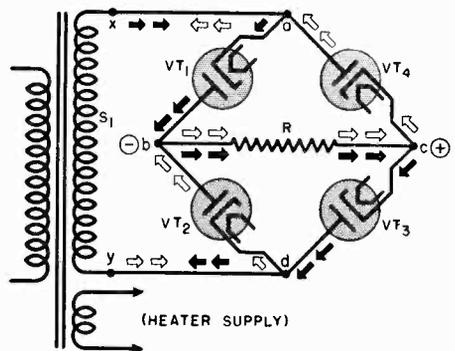


FIG. 29. A bridge rectifier circuit.

As you see, electrons flow from point *b* to point *c* no matter which terminal of the supply they leave. Thus this circuit gives us full-wave rectification between points *b* and *c* without a center tap on the transformer and with a transformer source voltage equal to that fed only one plate of the usual full-wave circuit. Points *b* and *c* form the output terminals of the rectifier; since electrons flow from *b* to *c*, *b* is the negative (−) and *c* is the positive (+) terminal.

Of course, a standard filter circuit is connected between the rectifier output terminals and the load. A voltage divider can be used too, if one is needed.

# Lesson Questions

Be sure to number your Answer Sheet 12FR-2.

Place your Student Number on every Answer Sheet.

*Send in your set of answers for this lesson immediately after you finish them, as instructed in the Study Schedule. This will give you the greatest possible benefit from our speedy personal grading service.*

1. Into what two parts may the circuits of a radio receiver be automatically divided?
2. Briefly give the three important requirements of a power supply.
3. What would you expect to happen to a 2.5 volt filament tube, if it were placed in a 6.3 volt tube socket?
4. What type of current exists at the output of the rectifier tube: (1) *a.c.*; (2) *pure d.c.* or (3) *pulsating d.c.*?
5. What two parts are most widely used in an a.c. filter?
6. What happens to the d.c. voltage when an input filter condenser opens up (loses its capacity)?
7. As far as the filtering action is concerned does it matter whether the filter choke is placed in the positive or the negative side of the circuit?
8. Why is extra current drawn through a voltage divider by a bleeder resistor?
9. Suppose the bleeder resistor opens in the circuit shown in Fig. 23. Would you expect the screen grid voltage to: (a) *increase*; (b) *decrease*, or (c) *remain the same*?
10. Would you expect to find: (1) *a choke input*; or (2) *a condenser input filter* used with a mercury vapor rectifier tube?

## CASHING IN ON DISCONTENT

Discontent is a good thing—if it makes you want to do something worth while. If you had not been discontented, you would never have enrolled for the N.R.L. Course.

Practically every one is discontented. But some of us are “floored” by discontent. We develop into complainers. We find fault with anything and everything. We end up as sour and dismal failures.

Those of us who are wise use our discontent as fuel for endeavor. We keep striving toward a goal we have set for ourselves. We are happy in our work. We face defeat, and we come out the victors.

At this minute you may be discontented with many things—your progress with your Course, your earning ability, yourself.

Make that discontent pay you dividends. Don't let it throw you down. If you do, you may never be able to get up again. Keep striving to remove the cause of your discontent. Remember that it's always darkest before the dawn. And a real N.R.I. man works hardest and accomplishes most when he is face to face with the greatest discouragements.

J. E. SMITH.

**SPECIAL POWER SUPPLIES  
FOR RADIO EQUIPMENT**

13FR-2

**NATIONAL RADIO INSTITUTE**

ESTABLISHED 1914

**WASHINGTON, D. C.**



# STUDY SCHEDULE NO. 13

For each study step, read the assigned pages first at your usual speed. Reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind, then answer the Lesson Questions for that step. Study each other step in this same way.

1. **The Universal A. C.-D. C. Power Supply** . . . . . Pages 1-8  
The universal power supply is very commonly encountered, as it is used in most midget receivers. This supply has the advantage of small size and low cost. Also, it will operate from any 110-volt power line, whether a. c. or d. c. Special methods are used to obtain the needed operating voltages, even to the development of special tubes for such receivers. Answer Lesson Questions 1, 2, 3 and 4.
2. **Voltage Doublers** . . . . . Pages 8-12  
When voltages are needed which are higher than the universal supply can deliver but a power transformer is not wanted, a voltage doubler can be used. These circuits are able to step up the voltage as long as the current demand is low. Both the half-wave and full-wave types are found in some of the larger table model receivers and in other radio and electronic equipment. Voltage doublers will work only from an a. c. power line, as they depend on the phase reversal on alternate half cycles for operation. Answer Lesson Question 5.
3. **Vibrator Power Supplies** . . . . . Pages 12-19  
The vibrator made auto sets practical, and led to the development of other mobile equipment. This unit removed the requirement for bulky, expensive batteries or an expensive motor generator, by making it possible to obtain standard voltages from a single low-voltage battery which is already available in the car. Here you learn how the mechanical switch changes d. c. into a pulsating current which can be stepped up by a transformer and treated like a. c. A tube rectifier can be used, or another vibrator section can be made to convert the high-voltage a. c. to d. c. again. A vibrator power supply comes closest to duplicating the standard a. c. supply, in regard to the voltage and current output. Answer Lesson Question 6.
4. **Modern Battery Receivers** . . . . . Pages 19-25  
A truly portable receiver must run from a self-contained power supply. Batteries are ideal, particularly since modern low-drain tubes operate from small light-weight cells. These low-current tubes have also made it possible to obtain filament power from the ordinary universal power pack, as well as plate and grid voltages. Thus, by using a small a. c.-d. c. power supply as well as batteries, the three-way portable can be run from power lines when available, thus saving the batteries. Answer Lesson Questions 7, 8 and 9.
5. **Special Power Supplies** . . . . . Pages 26-28  
There are instances where a motor-generator unit is needed to furnish power for some piece of standard or special equipment. Also, you may be called on to convert from one power supply type to another, due to differences in available power or for some special operation. The methods are discussed here, so you can see just what is possible and the best way of getting the desired operation. Answer Lesson Question 10.
6. **Mail Your Answers for this Lesson to N. R. I. for Grading.**
7. **Start Studying the Next Lesson.**

# SPECIAL POWER SUPPLIES FOR RADIO EQUIPMENT

## The Universal A. C.-D. C. Power Supply

THE standard a.c. power supply you've studied so far uses a power transformer to step up the line voltage. The stepped-up voltage is then applied to a rectifier tube, which in turn feeds the load through a filter. This standard power supply is nearly always used when high voltages or large currents are needed.

► But sometimes it is impossible or undesirable to use a transformer in the power supply. In some sections of our largest cities, for example, the power lines furnish d.c. instead of a.c. Of course, a transformer can't work on d.c.

Then, too, some places have 25-cycle current instead of the usual 60-cycle. This requires a special, more expensive transformer.

There are also cases where a power transformer is unnecessary—modern midget receivers, for example. These sets will operate reasonably well on just the line voltage. Further, they are so small that there is no room in them for a bulky transformer. And finally, they are built to sell at very low cost—which is possible only if the expense of a power transformer is eliminated.

► Now we are going to study a universal a.c.-d.c. power supply which contains no power transformer. It is designed to operate from any a.c. or d.c. power line, and is entirely satisfactory for small sets and low-power apparatus. Let's first see how the B and C supplies are obtained, then go into the question of filament supply a little farther on.

### The B-C Power Pack

A simple universal power supply is shown in Fig. 1. As you see, no power transformer is used—the power line connects directly to the circuit. The half-wave rectifier tube *VT* is followed by a standard condenser input filter which connects to the load.

This circuit works just like the standard half-wave rectifier with a condenser-input filter that you've already studied. That is, when the source voltage makes point 2 positive with respect to point 1, the tube conducts; electrons flow from point 1 through load  $R_L$ , and  $C_1$  is charged to the polarity shown. When the voltage reverses, making point 1 positive with respect to point 2, the tube no longer

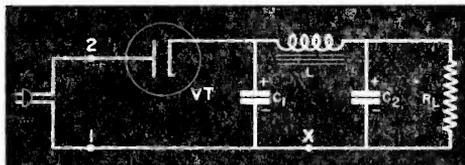


FIG. 1. This universal power supply uses no power transformer.

conducts;  $C_1$  partially discharges through load  $R_L$ , maintaining an electron flow through it. Thus, the only difference between this circuit and the standard half-wave rectifier circuit is that we've managed to eliminate the power transformer. (Remember, the circuit is completed between points 1 and 2 by the power line leading to a generator.)

Of course, the maximum possible voltage across the load in this circuit is the peak value of the power line

voltage. As most power lines deliver about 115 volts,\* the peak value will be about 165 volts. Actually, when current is taken from the power pack, the drop in the tube usually brings the total voltage across the load down to about 100 volts. Obviously, from what you know about filters, a condenser-input filter is necessary in a universal receiver to get a higher-output voltage than would be delivered by a choke-input filter.

► As in the standard power supply, the choke coil in Fig. 1 can be placed at point X (between the negative leads of the filter condensers), and the voltage drop across it can then be used to furnish bias voltages.

◁ **Circuit Variations.** Many receivers use dynamic loudspeakers, which have a field coil that must be energized by d.c. from the power sup-

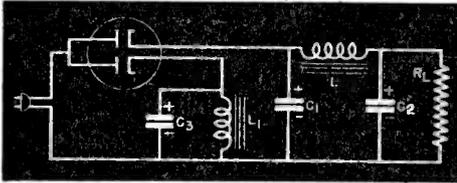


FIG. 2. The speaker field is energized from a separate diode here.

ply, in order for the speaker to operate. If the coil has a low resistance (500 ohms or less), we can use it in place of the filter choke in Fig. 1 and thus make the coil do double duty.

► But most speakers used in a.c.-d.c. sets have a field coil resistance of around 2500 ohms. A coil with such high resistance can't be used as a filter choke in our universal circuit,

\* Standard power lines are rated at 110 volts, 115 volts or 117 volts. Actually, these ratings mean the same kind of power line, as the power line voltage may be any value between 100 volts and 120 volts, depending on the load on the line. The peak value is about 1.4 times the r.m.s. value for which the line is rated.

because there would be too much d.c. voltage dropped across it, and therefore too little output from the filter. When we use such a speaker, we have to put a regular choke in our filter section, and find some other way of connecting the speaker coil to the power supply to give the coil the current it needs.

A circuit which does this is shown in Fig. 2. Here, a dual diode tube is employed. (Separate diodes could be used, if desired.) One diode is used as a half-wave rectifier for the load  $R_L$ ; the other acts as a rectifier to supply the speaker field, marked  $L_1$ , which acts as the load for this tube section. The filter condenser  $C_3$  provides sufficient filtering for the field. The other filter  $C_1$ - $L$ - $C_2$  is a standard condenser-input filter, used to deliver pure d.c. to the load  $R_L$ .

► As you know, the input filter condenser draws a rather high current from the rectifier tube until the condenser is fully charged. As a result, in the circuit shown in Fig. 2, each of the rectifier tube sections undergoes considerable strain when the circuit is first turned on. Fig. 3 shows how the circuit can be changed to relieve some of this strain and save one filter condenser at the same time.

Here the two cathodes of the rectifier tube sections are tied together, as are the plates. This lets us use the current capacity of the two diodes (which are now in parallel) to charge condenser  $C_1$ , thus reducing the current each tube section must carry. The speaker field  $L_1$ , in parallel with  $C_1$ , is connected directly across the output of the rectifier. Enough filtering is produced by  $C_1$  for the field supply.  $C_1$  thus takes the place of condenser  $C_3$  in Fig. 2, and at the same time acts as the input filter condenser for the filter  $C_1$ - $L$ - $C_2$ .

► In very inexpensive receivers, the filter choke may be replaced by a

cheaper, but less efficient resistor. Fig. 4 shows a typical circuit. If the resistor  $R_1$  could be made large enough so that the ratio of its resistance to the reactance of condenser  $C_2$  was about the same as that of the choke reactance to condenser reactance, the filtering obtained would be approximately the same. However, a resistor this large cannot be used, because the d.c. voltage drop across it would be so great that little voltage would be left for the load. A smaller resistor must be used, hence, the filtering is not as good as that obtained with the average choke.

The resistor can be made larger, improving the filtering, if we can reduce the d.c. current flow through the resistor. As most of the d.c. required is for the plate of the output tube, it seems logical to divert this flow. By connecting the output tube plate circuit between terminal  $X$  and  $B-$  in Fig. 4, this plate current does not go through  $R_1$ .

This method has the disadvantage that the only filtering given the out-

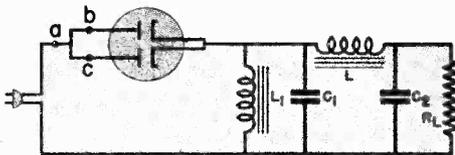


FIG. 3. The diode elements are in parallel, so have a greater current safety factor.

put tube plate supply is that furnished by  $C_1$ , so a considerable a.c. ripple is fed to the output tube. However, since there is no amplification between the output tube plate and the loudspeaker, and since the output transformer and speaker will seldom respond to low frequencies in these inexpensive receivers, this ripple will not cause an objectionable hum.

If the screen grid of the output tube or any electrode of any other tube is connected to  $X$ , too much hum will be

produced. Therefore, all the rest of the tube electrodes are connected between  $B+$  and  $B-$ , so they will benefit from the additional filtering of  $R_1$  and  $C_2$ .

**Regulation.** The filter systems of Figs. 1, 2, 3 and 4 are condenser-input filters. As you know, their regulation

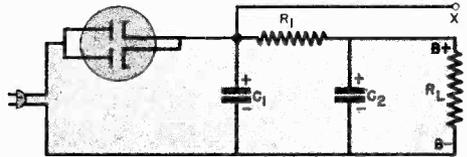


FIG. 4. Resistor  $R_1$  is used in place of a choke coil, making a less efficient but still useful filter.

will depend on the amount of current taken by the load and upon the amount that condenser  $C_1$  is discharged in between pulses of current from the rectifier. Just as in the standard power pack, the less the load current demand, or the larger the input condenser, the nearer the output voltage will approach the peak of the a.c. rectifier plate voltage.

Fig. 5 shows the regulation obtained with different sizes of filter condensers. You will notice that the d.c. voltage is high when low currents are drawn, but drops off very rapidly as the current demand is increased. It is easy to see that the condensers must be high-capacity types to give a reasonable output.

**Tube Protection.** As you already know, a large input condenser draws a high peak current from the rectifier tube. When a transformer is used, the resistance of the high-voltage winding limits the current through the rectifier tube to some extent; but the power line current in Fig. 3 is limited only by the fuses in the house wiring, which may have ratings as high as 25 amperes.

This is no protection at all for the rectifier tube. Many of these tubes

were ruined by excess peak current\* before receiver manufacturers hit upon the idea of placing a resistor in series with the plates of the tubes. A single resistor of from 20 to 30 ohms may be placed in series with both plates at point *a* of Fig. 3, or a 50-ohm resistor may be placed in series with each plate at points *b* and *c*.

Now, even if condenser *C*<sub>1</sub> is considerably discharged, the maximum current flow is limited by the series resistor plus the tube resistance to a value less than 2 amperes. Even so, an overload lasting for a considerable time (which might be caused by a defective condenser) will either burn out the protective resistor or destroy the emission of the tube.

### Filament Supply

So far, you have learned how to eliminate the power transformer and still obtain about 100 volts for plate and C bias supplies. Now how shall we supply the filaments?

Early a.c.-d.c. receivers used cathode type 6.3-volt a.c. tubes, obtaining the required filament voltage from the power line by using a resistor to drop the line voltage. Let's see how this can be done.

Suppose we have five tubes rated at 6.3 volts. If we connect these tubes in parallel, we will need 6.3 volts to operate them, and the total current demand will be the sum of all the filament currents. Presuming these five tubes have filaments rated at .3 ampere each, a current of 1.5 amperes

\* Excess current through the tube will make the cathode lose its electron-emitting ability. It may even cause the plate to sag and touch the cathode. Usually, however, the metal strip inside the tube which is used to connect the cathode to the lead from the tube base acts somewhat like a fuse, burning out if the current is much greater than 2.5 amperes. This metal strip cannot be made heavy, because it would then conduct too much heat away from the cathode of the tube.

must be taken from the power line ( $5 \times .3 = 1.5$ ). If we use a series resistor to drop the voltage from the line voltage value of 115 volts to the desired 6.3 volts, about 108 volts must be dropped in the series resistor. At a

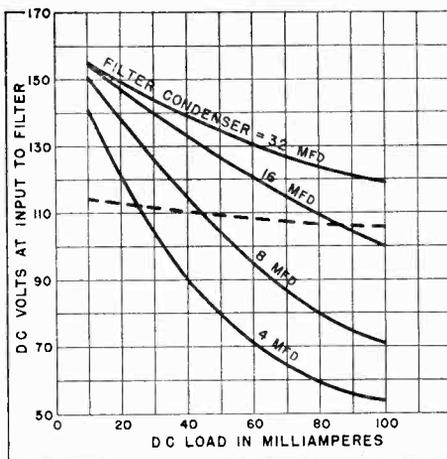


FIG. 5. Regulation curves for the universal power supply.

current of 1.5 amperes, the power dissipated in this resistor is about 162 watts. (Watts = current  $\times$  voltage,  $1.5 \times 108 = 162$  watts.)

This is entirely too much power to waste. We will waste much less by arranging the tube filaments in series, as shown in Fig. 6.

Since the same current flows through all the parts of a series circuit, the tube filaments shown in Fig. 6 must have the same current rating. Let's assume again that this is .3 ampere, and that the tubes are 6.3-volt tubes. The voltage drop across the tubes will be the sum of their voltage ratings, so there will be about 31 volts dropped across the tubes. As a result, only about 84 volts must be dropped across the resistor *R*. At a current of .3 ampere, the power loss in the resistor is now only 25 watts. Obviously, this represents a great saving in power.

As soon as a.c.-d.c. circuits proved

practical, tube manufacturers began to make tubes with higher filament voltage ratings. The ratings jumped to 12, then to 25 volts. In particular, the rectifier and power output tube voltages were increased.

Since the current ratings of these tubes were kept at the same value as the available 6-volt tubes, the new tubes and the 6-volt tubes could be connected in series. As you know, voltage drops add in a series circuit—so using tubes with higher filament voltage drops means that less voltage must be dropped by resistor  $R$ . Consequently, less power is wasted in this resistor.

► When the tubes are in series, resistor  $R$  can be an ordinary 35- or 50-watt resistor, as long as it has the proper resistance value.\* However, a regular 35- to 50-watt resistor is rather bulky; further, dissipating this amount

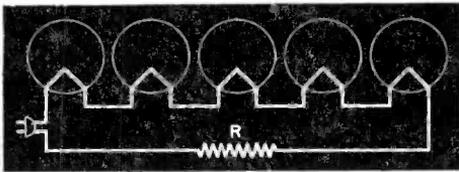


FIG. 6. The resistor drops the voltage to the amount required by the series-connected filaments.

of heat underneath the chassis may dry up the electrolytic filter condensers.

\* You may have to figure the value of resistor  $R$  in some repair jobs. To do this, add the voltage drop across the tube filaments. Subtract this sum from 117 volts, which is considered the line voltage value. The remainder is the voltage which is dropped across the resistor  $R$ . Now look up one of the tubes in a tube chart to find what the filament current is. Divide the resistor voltage drop by this current value and you will get the resistance value. The wattage rating of the resistor will be the resistor voltage drop multiplied by the current through the resistor. To be safe, choose a resistor with a higher wattage rating than this figured value.

**Ballast Tubes.** To avoid excessive heat under the chassis, the resistor may be placed in a tube known as a ballast tube or resistance tube. This

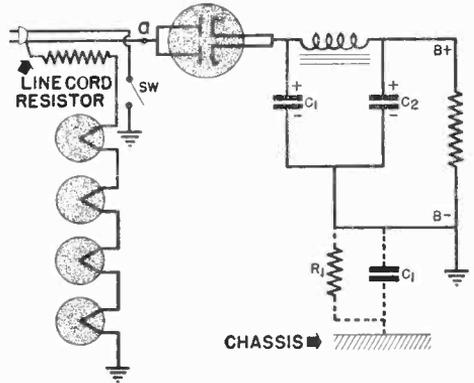


FIG. 7. Using a line cord resistor. In practical circuits of this type, you may find a protective resistor in series with the rectifier plates; also, there may be a speaker field connected across input filter condenser  $C_1$ .

is a metal or glass tube with a regular tube base, which has only a resistance element in it. The resistance element is wired in series with the filaments of the other tubes, so the circuit is the same as Fig. 6. A ballast tube dissipates heat like all other tubes, from the top of the chassis, and so causes no heat under the chassis. Tubes with various resistance values may be obtained to suit the circuit.

**Line-Cord Resistors.** Another common voltage-dropping resistor is the Cordohm or line-cord resistor. This is a line cord having three (or more) wires. A typical circuit using a Cordohm is shown in Fig 7. Two of the wires form the regular a.c. power cord going to the B power pack. The third wire is a resistance unit of the proper size for the particular tube combination it is used with. This resistor value is figured the same way as an ordinary resistor, when a replacement is necessary. Wattage ratings need not worry you, however, as

these cords can safely dissipate 30 watts.

► Since this resistor unit runs the entire length of the cord, the cord will become warm. This may alarm some of your customers; if so, explain to them that the cord contains a resistor which is supposed to dissipate heat outside the set chassis. For this reason, such cords should always be stretched out to their full length, never bundled up inside the radio. Also, these cords should never be cut to shorten them, as this reduces the resistance, even if the proper connections are made.

### Eliminating the Series Resistor.

In many of the more recent receivers, new types of tubes are used which have filament voltage ratings that add up to the line voltage total. A typical circuit is shown in Fig. 8. Here the 35Z5 rectifier tube requires 35 volts for its filament. The 50L6 output tube requires 50 volts, and the remaining tubes require 12 volts each. (Notice that the first part of the tube number shows the approximate filament voltage needed for the tube. This is standard practice in numbering the newer tubes.) The total is about equal to the line voltage, so no series resistor is needed.

### Grounding Precautions

In Figs. 7 and 8, switch SW (the ON-OFF switch of the set) opens the filament and B supply circuits simultaneously by opening one side of the power line. Notice that one end of the switch, one end of the filament string and the B— terminal are shown going to ground symbols. *This means that these three terminals are all connected together.* For convenience, each point is often connected to the chassis; since the chassis is metal and a good conductor, this is exactly the same electrically as connecting the three wires directly.

Since one wire of the house power line is grounded and this ground is connected to the chassis by the cord of the receiver, no set wired like this should ever be connected to a ground by a separate wire. If it is, and the power cord plug happens to be inserted in the wall socket so the *ungrounded* wire of the house line is connected to

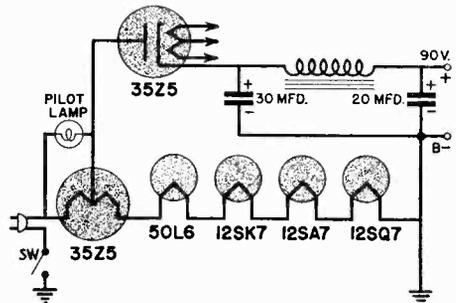


FIG. 8. The tube filaments have ratings adding up to the line voltage, so no series resistor is needed. (The 35Z5 tube filament is shown separately from the diode here for convenience; both are parts of a single tube.)

the chassis, the house line will be shorted through the set. Of course, this will blow out the power line fuses. **NEVER USE A GROUND CONNECTION ON AN A.C.-D.C. RECEIVER** unless a terminal is provided by the set manufacturer. And, for your personal safety, remember that the set chassis will be 115 volts above ground if the plug happens to be in the socket upside down. If it is, you will get a shock if you touch the chassis while you are standing on a concrete floor or other ground.

► In many a.c.-d.c. receivers, the chassis is not an electrical part of the circuit. In such sets, points marked with a ground symbol on diagrams are connected by a wire (often called a "bus"). The chassis may be connected to this ground bus through a condenser and resistor ( $C_1$  and  $R_1$  in Fig. 7). The chassis is indicated by the special symbol shown in this figure

when it is not used as a part of the circuit.

### Dial Illumination

A pilot lamp is used in most radios to light the dial and show when the receiver is turned on. Let's see how the lamp is wired into the circuit.

The obvious way is to use a pilot lamp with the same current rating as the tube filaments, and connect it in series with them. Unfortunately, this method won't work. When the radio is first turned on, the tube filaments are cold and have lower-than-normal resistance. It takes several seconds for them to warm up. On the other hand, the pilot lamp warms almost immediately. Thus, when the radio is first turned on, the low filament resistance allows a higher-than-normal current to flow for a few seconds—which would burn out the pilot lamp.

Instead, we put a small resistor in series with the filaments, and connect the lamp across it. The lamp usually has a lower current rating than the tube filaments. The shunting resistor across the pilot lamp carries the difference in current between the pilot lamp rating and the current flowing in the circuit. For example, if the pilot lamp is rated at .15 ampere, and .3 ampere is flowing, the shunt resistor must carry the difference between the two, or .15 ampere.

The resistor has such a value that the voltage drop across the resistor and lamp is 4.25 volts\* when the normal filament current flows. The drop is somewhat higher during the first rush of current, but now does not exceed the 6.3-volt rating of the pilot lamp. This means the lamp does not run at full brilliance except during the filament warm-up period. However, it

\* The resistor value can be found by dividing 4.25 volts by the current it must carry. This current is the difference between the filament current and the pilot lamp rating.

gives enough light—and will not be burned out by the high initial current.

► When a series filament resistor, ballast tube or line cord is used, a tap on the resistor element may provide a shunt resistance, as shown in Fig. 9. If desired, the section  $R_1$  can be a separate resistor.

► In Fig. 8, the 35Z5 filament in the filament string has a tap on it, so a pilot lamp can be connected across a part of the tube filament, which acts as the resistor element in shunt with the pilot lamp.

Connecting the plate of the 35Z5 to this filament tap, as shown in the diagram, makes it unnecessary to use a protective series plate resistor. This puts the parallel combination of the pilot lamp and the tube filament shunting section in series with the

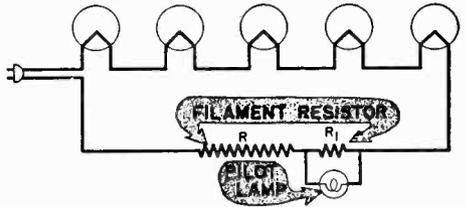


FIG. 9. The pilot lamp can be connected across a tapped section of the filament resistor.

plate of the rectifier tube; if too much current flows, the pilot lamp will burn out. Should the overload continue, this section of the tube filament may open. But usually, if the set is turned off quickly after the pilot lamp burns out, the pilot lamp will be the only part damaged. Before replacing the bulb, the filter condenser should be checked to be sure the overload is not the result of a defect in it. Incidentally, in any a.c.-d.c. receiver, a burned-out pilot lamp should be replaced at once. Never operate the receiver with the pilot lamp burned out, because another part may eventually be damaged.

► The most common pilot lamps for a.c.-d.c. sets are 6.3-volt bulbs rated at .15 or .25 ampere. The .15-ampere bulb has a brown colored glass bead supporting its filament, while the .25-ampere bulb has a blue bead. Naturally, you should always replace a lamp with another of the same rating.

► You will probably often be asked by owners of a.c.-d.c. sets why the pilot lamp flares up brilliantly when the set is just turned on, then grows considerably dimmer. The brilliance at the start is caused by the initial rush of filament current through the pilot lamp. Then when the tubes warm up, the current through the pilot lamp decreases, making the lamp less brilliant.

The pilot lamp shown in Fig. 8 will go through this same action, then become brighter again. As in Fig. 9, the first brilliant flare is caused by the extra current flow in the filament circuit. The pilot lamp dies down until it practically goes out as the filaments warm up. Then when the tubes draw plate current from the B power supply, the rectifier plate current flows through the pilot lamp and again lights it up.

### Operation on D. C. Power Lines

Any a.c.-d.c. receiver can be used on a d.c. power line. All you need do is place the power plug in the wall socket in such a way that the plate of the rectifier tube is positive. Then current will flow continuously through the rectifier tube. The tube just passes on current, and does not act as a rectifier at all; however, it does act as a safety device for the electrolytic condensers. As you know, an electrolytic condenser will break down and become a low-resistance conductor if voltage of the wrong polarity is impressed across it. The rectifier tube prevents this from happening, because it will not pass current if the power plug is installed with the wrong polarity.

The dotted line in Fig. 5 shows the regulation obtained when d.c. voltage is applied to the input of an a.c.-d.c. receiver. Since the rectifier tube passes the needed d.c. average all the time, the filter condensers do not discharge, and the output voltage is practically independent of the effect of the filter circuit. Instead, it depends primarily on the resistance of the rectifier tube and filter choke. The only use for the filter circuit now is to remove any stray ripple voltages which may be on the d.c. power line.

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## Voltage Doublers

Sometimes, a higher voltage than is obtainable from the usual universal a.c.-d.c. power pack is wanted, without using an expensive transformer. This is made possible by another transformerless circuit called a voltage doubler. Let us see how it works.

### Full-Wave Voltage Doubler

If we can charge two condensers, then put them in series so that their

voltages add together, we will get twice the charging voltage from the series combination. This is the theory of the full-wave voltage doubler circuit shown in Fig. 10.

This circuit resembles a half-wave rectifier circuit using a condenser-input filter, except that it has an extra condenser and an extra rectifier. The easiest way to understand its operation is to trace the flow of electrons.

When the a.c. cycle makes point 2 negative with respect to point 1, electrons will flow from point 2 to point 1. How do they get there? There are two possible paths, through  $C_1$ - $VT_1$

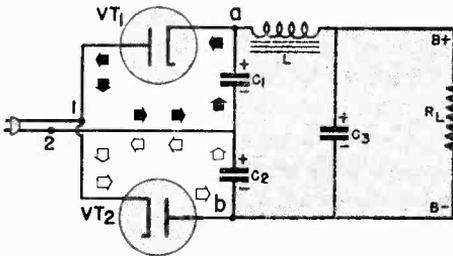


FIG. 10. This circuit doubles the line voltage without using a transformer.

and through  $C_2$ - $VT_2$ . However, electrons cannot flow from a plate to a cathode, so the  $C_2$ - $VT_2$  path from point 2 is blocked when point 2 is negative. Electrons flow instead into one plate of condenser  $C_1$ , and an equal number of electrons are forced out of the other plate. This charges condenser  $C_1$  with the polarity shown. The electrons leaving  $C_1$  flow through  $VT_1$  to point 1. The circuit is shown by the black arrows.

When the a.c. line voltage polarity reverses, electrons leave point 1 and, finding their way blocked by tube  $VT_1$ , pass through  $VT_2$  and charge  $C_2$ . Those electrons driven out of  $C_2$  during the charging process flow to point 2. This half cycle is shown by the outline arrows.

Notice that condensers  $C_1$  and  $C_2$  are alternately charged from the power line. Also notice the polarity of the voltages across the condensers. Since they are in series, the voltage between terminals  $a$  and  $b$  is approximately the sum of the condenser voltages. Of course, one condenser is being discharged by the load while the other is charging, so we don't get quite double the peak of the line voltage. The actual voltage depends

upon the amount of load and the size of the condensers.

This circuit is known as a full-wave voltage doubler because both halves of the a.c. cycle are rectified. Hence, 120 cycles is the lowest ripple frequency which must be filtered by  $L$ - $C_s$  if the power line furnishes 60-cycle current.

► Fig. 11 shows the regulation characteristics of a full-wave voltage doubler in which condensers  $C_1$  and  $C_2$  are of equal value. Curves are shown for values of 5, 10, 20 and 40 microfarads for each condenser.

You can see that increasing the load current results in considerable voltage drop. In other words, the regulation is poor. The larger the capacity of the condenser, the less the drop in d.c. voltage for increased load current—but even at best, the regulation of this circuit is like that of a universal

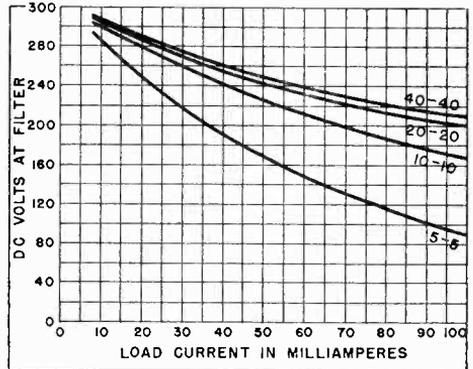


FIG. 11. The regulation of a voltage doubler is like that of the universal power supply.

a.c.-d.c. power pack, and is considerably worse than that of a transformer type power pack. It can be used satisfactorily only where the load current is not liable to vary greatly or where high currents are not required.

**A Practical Circuit.** Of course, Fig. 10 is a simplified circuit. A typical practical circuit, using a dual-

diode tube, is shown in Fig. 12. In its important details it is the same as the circuit shown in Fig. 10.

Let's trace the electron path through Fig. 12. When point 2 (near the power plug) is negative with respect to point 1, the path is from point 2 to point c, through  $C_1$  to cathode  $K_1$ , and from  $P_1$  through resistor  $R_1$ , to point 1. ( $R_1$  and  $R_2$  are protective resistors in series with the tube's plates.) On the next half cycle, the path is from point 1 to cathode  $K_2$ , to  $P_2$ , through  $R_2$  to point b, through condenser  $C_2$  to point c, and thus back to point 2. Therefore, the circuit works exactly like Fig. 10.

In addition, the filament circuits are drawn in. You can trace the cir-

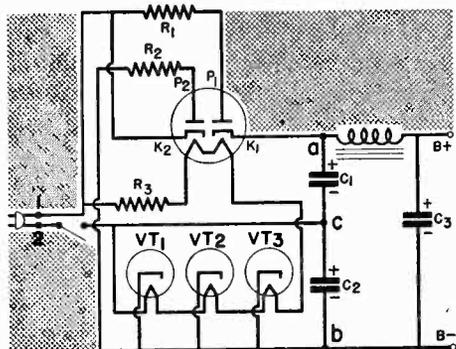


FIG. 12. A practical full-wave voltage doubler.

cuit from point 1 through resistor  $R_3$ , through the filament of the rectifier tube and in turn through tubes  $VT_3$ ,  $VT_2$  and  $VT_1$ , back to point 2.

► Notice that the cathodes of tubes  $VT_1$ ,  $VT_2$  and  $VT_3$  all connect to  $B-$ , as is normal in radio receiver circuits, but the filaments are connected directly across the power line. The filament of tube  $VT_1$ , which connects to point 2, also connects to point c (between the filter condensers). Therefore, the voltage difference between the cathode of this tube (which connects to point b) and the filament (which connects to point c) is the

voltage across condenser  $C_2$ . This means there is a d.c. voltage between the cathode and filament of this tube which is approximately equal to the line voltage. Modern tubes have excellent insulation between the filament and cathode, so there is little danger of this voltage breaking down the tube between these elements, but a certain amount of leakage between them may develop. Since the filament of  $VT_1$  is positive with respect to the cathode, it may attract some electrons from the cathode, so there can be a current flow through the space within the tube between the cathode and filament. The a.c. voltage across the filament will vary this leakage current and thereby create a certain amount of hum. As you will learn later, tube  $VT_1$  is usually the first audio or detector tube, where not even a slight hum can be allowed. Careful tube selection for a minimum amount of leakage is necessary in these cases. Voltages exist between the cathodes and filaments of the other tubes, but leakage currents in these tubes are not so important.

Now let's look into another voltage-doubler circuit, which will solve some of this leakage current difficulty.

### The Half-Wave Voltage Doubler

Like the full-wave voltage doubler, the half-wave voltage doubler uses a condenser as a voltage source—but as a source of voltage for a rectifier tube. Fig. 13 shows the basic circuit.

Suppose point 2 is negative with respect to point 1. Electrons will flow from point 2 into condenser  $C_1$ , forcing other electrons out; these go to point 3, then through  $VT_1$  to point 1. (Of course, there is no path for these electrons through  $VT_2$ .) This charges  $C_1$  almost to the peak line voltage, with the polarity shown. The electron path is shown by the outline arrows.

On the next half cycle, point 1 is negative with respect to point 2. Now electrons cannot travel through tube  $VT_1$ ; instead, they move from point 1 through condenser  $C_2$ , tube  $VT_2$  and

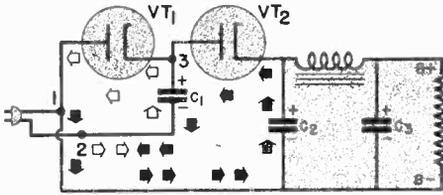


FIG. 13. The half-wave voltage doubler.

condenser  $C_1$  to point 2, as shown by the black arrows. (This time, the path through  $VT_1$  is blocked.)

Now just how does the voltage double? On the last half cycle, remember that point 1 was negative and point 2 positive. Since point 2 connects to the negative terminal of condenser  $C_1$ , the voltage stored in  $C_1$  adds to the line voltage, so that the voltage between points 1 and 3 is approximately double the line voltage. It is not exactly twice the line voltage, because  $C_1$  is being slowly discharged by the load on this half cycle, but it is not far below double voltage if the load has a high resistance. This voltage is applied to tube  $VT_2$ , so condenser  $C_2$  is charged to almost twice the line voltage (and, of course, almost twice the line voltage appears between  $B+$  and  $B-$ ).

Since the current passes through rectifier tube  $VT_2$  on only every other half cycle, this is a half-wave voltage doubler as far as condenser  $C_2$  is concerned. Because of the half-wave action, the output voltage is somewhat less and the regulation is worse than a full-wave doubler.

► Notice that the  $B-$  terminal connects to one side of the power line in Fig. 13. This means that when the filament string is connected across the power line, there will be no d.c. volt-

age difference between the first tube cathode and its filament, and therefore no leakage current. We have thus improved upon the full-wave voltage doubler in one way, but have had to sacrifice some regulation to do it.

A complete half-wave voltage-doubler circuit is shown in Fig. 14. The action is identical to that of Fig. 13 as far as the B supply is concerned. Protective resistors  $R_1$  and  $R_2$  are used in series with the plates; also the pilot lamp  $P$  and its shunting section of the filament are between condenser  $C_1$  and terminal 2—which is another protective feature. The current flow

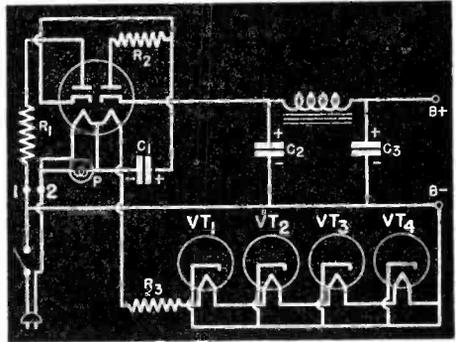


FIG. 14. A typical commercial half-wave voltage doubler.

for the B supply helps to keep the pilot lamp lighted.

The filament circuit includes a resistor  $R_3$  to drop the voltage to the required amount for the tube filaments. The pilot lamp could be placed across a tapped section of  $R_3$ , but this is unnecessary because of the pilot lamp tap on the rectifier filament.

### Voltage Multipliers

You have seen how the line voltage may be doubled by using charged condensers and rectifier tubes. This method need not stop here—we could have voltage triplers, voltage quadruplers, etc. Of course, every time

the voltage is increased another step, the regulation becomes worse and the available current becomes less. For this reason, voltage doublers are the practical limit for most radio purposes. However, Fig. 15 shows how a voltage tripler can be made by adding an ordinary half-wave rectifier to a voltage doubler.

When terminal 1 is positive with respect to terminal 2, there will be an electron path from terminal 2 through condenser  $C_1$  and tube  $VT_1$ . At the same time there will be another path through tube  $VT_3$  and  $C_3$  back to terminal 1. Thus condensers  $C_1$  and  $C_3$  are charged on this half cycle.

On the next half cycle, electrons will move from terminal 1 into  $C_2$ , forcing other electrons through  $VT_2$  and  $C_1$ . As in any half-wave voltage doubler,  $C_2$  is charged on this half

cycle to double the line voltage. This voltage is then added to that of  $C_3$ , so that three times the line voltage exists across  $C_2$ - $C_3$ .

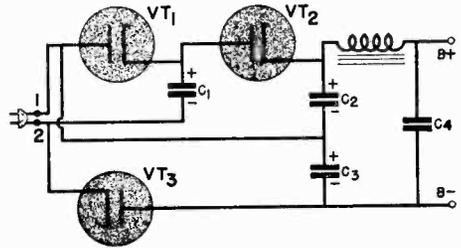


FIG. 15. A basic voltage tripler.

► A very important fact is that only a.c. sources can be used with voltage doublers, triplers, etc. No output can be obtained if a d.c. power line is connected to any of these voltage multipliers, because the polarity of a d.c. power line does not reverse.

## Vibrator Power Supplies

So far, we have discussed power supplies which depend on a.c. or d.c. power lines. Power lines are not always available, however, so we must also consider other sources of power. Batteries are one such source, and can be used for portable or mobile equipment and for farm receivers, as will be discussed later.

In many locations, certain batteries are already available. On mobile devices such as automobiles, trucks, boats and airplanes, storage batteries of 6, 12, or 24 volts are used to operate the ignition and starting equipment; in many farm homes, 32-volt batteries are used for lighting purposes. These batteries furnish d.c. voltages which are adequate for filament supply but are not high enough to be used efficiently as plate and

screen grid voltage supplies. We could use additional batteries, as was done in early installations, but means have now been discovered for raising the voltages of these batteries to a higher and more efficient value. Let us see how this can be done.

► Anyone who has not studied the fundamentals of radio might say, "use a step-up transformer," but you know that a battery delivers only d.c. Since steady d.c. provides no changes in flux linkage, there will be no voltage induced in the secondary if a battery is connected to the primary of a transformer. However, if d.c. flows through the primary in pulses, there will be changes in flux linkage and an induced secondary voltage will result. You will recall from your study of coils that when d.c. is first applied, the

current rapidly builds up from zero to its maximum value and then flows at a steady rate. The flux linkages change as long as the current changes. Let's see how this idea can be applied to a battery and a transformer.

Suppose we connect a storage battery to the primary of a transformer as in Fig. 16A. When the circuit is first closed, current will flow through the primary; the resultant change in flux linkage will induce a large voltage across the secondary, as shown by 1-2 in Fig. 16B. The primary current will rise rapidly until it reaches its maximum value (determined by the resistance of the wire with which the transformer primary is wound), then remain constant until the circuit is broken. When the primary current becomes constant, the secondary voltage will drop to zero (2-3) and stay there, since voltage is induced only when there is a change in flux linkage.\* When the primary circuit is opened, the current will cease flowing and the magnetic field will quickly collapse. This will cause another change in flux linkage, this time in the opposite direction, and a secondary voltage pulse will appear, as shown by 4-5-6 in Fig. 16B. This voltage will also drop to zero when changes in flux linkage cease.

► By using a step-up transformer and repeating this opening and closing regularly, we can get a series of alternate pulses from the secondary with high peak voltages.

From this, you can see that d.c. voltage can be stepped up with a transformer and changed into a.c. voltage if the primary circuit of the transformer is opened and closed fast enough with a switch. This allows the d.c. to flow in pulses which rise from zero to maximum and then de-

crease to zero again. Of course, if we could make the d.c. *reverse* in the primary, we would get more a.c. secondary pulsations, closer together—and thus obtain a greater a.c. output average.

### Full-Wave Vibrators

Fig. 17A shows how we can make the d.c. reverse. Notice that the primary is center-tapped. One battery terminal connects to this tap; the other battery terminal connects to a switch arm which connects first to one outside primary lead and then to the other.\*

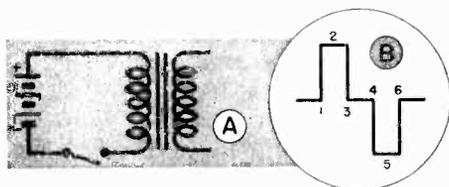


FIG. 16. A. C. can be produced by regularly interrupting a d. c. circuit.

Suppose the switch is thrown from position X to position Y. When the contact at X is broken, the current through  $P_1$  drops to zero, as shown by line 1-2 in Fig. 17B. During the time it takes the switch arm to travel to position Y, the primary current remains zero. Then, as the switch arm makes contact at Y, we have current flow 3-4 through  $P_2$ .

Since current flows in opposite directions through the windings  $P_1$  and  $P_2$ , a *decreasing* current in one produces a flux in the same direction as an *increasing* current in the other; therefore, the flux linkage change produced by current 1-2 decreasing through  $P_1$  is in the same direction as the flux linkage change produced by

\* One battery lead is grounded; it does not matter which one. The ground is shown because one terminal of a storage battery usually connects to the frame of the car or other device with which it is used.

current 3-4 increasing through  $P_2$ . The change 1-2 produces a secondary voltage pulse like  $a-b-c$  in Fig. 17C. Then the change 3-4 produces pulse  $d-e-f$  in the same direction.

Between 4-5 in Fig. 17B, the current through  $P_2$  is constant and there is no change in flux linkage. When the switch arm is thrown from  $Y$  to  $X$ , the current through  $P_2$  drops to zero, as shown at 5-6, and when contact at  $X$  is made, current  $P_1$  rises as shown by 7-8.

The changing flux due to currents 5-6 and 7-8 produces voltage pulses  $g-h-i$  and  $j-k-l$ . Then currents 9-10 and 11-12 produce pulses  $m-n-o$  and  $p-q-r$ . Thus, we now get double pulses in the secondary instead of the single ones of Fig. 16.

► We could apply the secondary voltage shown in Fig. 17C to a rectifier tube, pass the tube output through a filter, and get a fair d.c. voltage output. However, there are several things wrong with these voltage pulses.

One is that the voltage consists of narrow double pulses in each direction, while we'd rather have pulses which are broader and not as high. We can get them by speeding up the switch shown in Fig. 17A, and by moving its contacts closer together, so that very little time elapses while the switch moves from one contact to the other. This means that the flat parts of the curve in Fig. 17B, where the current is zero (2-3, 6-7, etc.) are practically eliminated, so the primary current change will be almost a straight line between points 1 and 4, 5 and 8, 9 and 12, etc. As a result, each pair of secondary pulses will merge together, and there will not be much of a gap between opposite pairs of secondary pulses.

Even so, these pulses are very high, sharp peaks. This makes the filtering job difficult. It also produces sparking at the switch contacts, be-

cause each rapid change of flux in the transformer induces a back-electromotive force (back e.m.f.) in the primary which tends to cause arcing between the contact and the switch arm. This will burn and soon ruin the switch contacts.

**The Buffer Condenser.** The sparking problem is easily solved by connecting a low-capacity condenser (called a buffer condenser) between the secondary terminals.\* Now, when

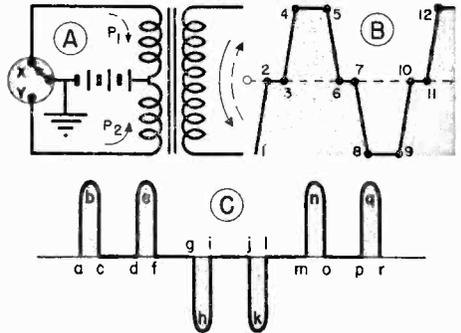


FIG. 17. Double voltage peaks are produced by using a full-wave interrupter.

the secondary voltage starts to rise, the condenser charges. This tends to reduce the sharpness of the peak in the secondary circuit, thus cutting down the back e.m.f. and thereby reducing the sparking at the vibrator contacts. Further, the discharge of the condenser when the secondary voltage begins to reverse helps to give us a secondary voltage more like that shown in Fig. 18—which has a shape approaching that which would be fed to a standard rectifier circuit.

### Practical Vibrators

Fig. 19 shows how modern vibrators operate. The switch arm is a

\*The condenser is chosen to work with a particular transformer and a certain rate of switch closure. Its capacity is quite important. When replacing a defective condenser, use the capacity and working voltage originally used by the manufacturer.

thin, flexible, spring-like metal reed. As it vibrates, it alternately closes contacts *N* and *O*, thus alternately closing the circuit through halves of the primary.

The shaded area shows the "motor" used to drive the reed. This motor is

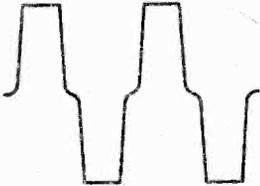


FIG. 18. The output of a vibrator-transformer-buffer condenser combination, which can now be rectified.

somewhat like the buzzer in an electric doorbell. Two types are used—the separate driver type and the shunt type.

**Separate Driver.** The separate driver type is shown in Fig. 19A. The vibrating reed is between *K* and *R*. Contact *M* is a spring contact which touches the reed when the reed is at rest. When the *ON-OFF* switch *S* is closed, electrons flow through contact *M* and electromagnet *L*. This energizes the electromagnet, which pulls up the soft iron bar *K* fastened to the end of the reed. This moves the reed, opening contact *M*, which breaks the circuit through the electromagnet. Since the electromagnet is no longer energized, the reed moves back to its original position, again closing *M*. Current again flows through *L*, and the cycle repeats as long as the battery is connected. The period of vibration of the reed depends on its mechanical characteristics—its length, springiness, etc.

In moving back and forth, the reed closes contacts *N* and *O* alternately. Each closure completes the circuit through one of the primary winding sections. The buffer condenser *C*<sub>1</sub> in

the secondary prevents arcing at contacts *N* and *O*. This does not affect the motor circuit and so does not protect *M*, therefore a separate condenser *C* is used across this contact.

The contacts are large and flat-surfaced, made from a hard grade of tungsten to reduce pitting and burning to a minimum.

**The Shunt Motor.** The shunt motor shown in Fig. 19B does not require a separate contact and is more widely used than the separate driver motor. When switch *S* is closed, electrons flow through the chassis (shown by ground symbols), electromagnet *L*, and primary winding *P*<sub>1</sub>. The electromagnet then pulls the reed upward, closing contact *N* and connecting the battery directly across primary *P*<sub>1</sub>. Contact *N* shorts the coil *L*, and the current now flows through *N* and the reed instead of through coil *L*, so the electromagnet releases; the reed springs back. Its springiness carries it beyond the resting position—this overshooting causes it to close contact

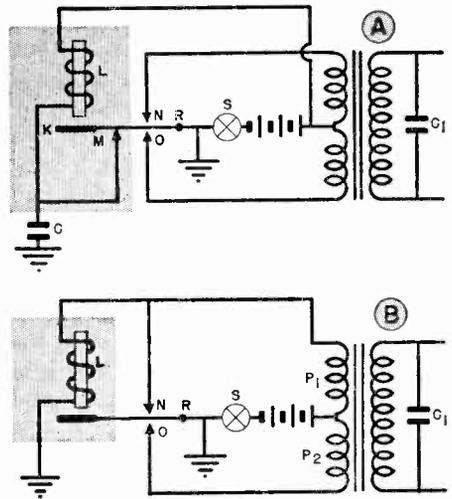


FIG. 19. Separate driver and shunt vibrator motor units. One or the other of these will be found on practically all commercial vibrators.

O. Then current flows through  $P_2$ . When contact  $N$  is broken the current again starts to flow through  $L$ , and soon builds up enough magnetic pull to jerk the reed back and close contact  $N$ . This cycle of action is repeated as long as  $S$  is closed. We always have current flowing through  $P_1$  (it flows through  $P_1$  and  $L$  when  $N$  is open), but this current is much smaller than that when  $N$  is closed and does not have any appreciable effect on the wave form of the secondary voltage.

### Rectification

Now that we know how a high a.c. voltage may be obtained from a d.c.

capacity condensers in the filament and  $B+$  supply leads, prevent this interference (called "hash") from being fed to the other tube circuits. Direct radiation is prevented by complete shielding, as shown by the dotted lines.

► Notice that the two battery leads are marked *A-hot* and *A-ground*. The one marked *A-ground* connects to the grounded side of the battery. In power packs using tube rectifiers, it makes no difference whether  $A+$  or  $A-$  is grounded.

► If the filament current drawn by the rectifier tube can be eliminated, the drain on the battery can be re-

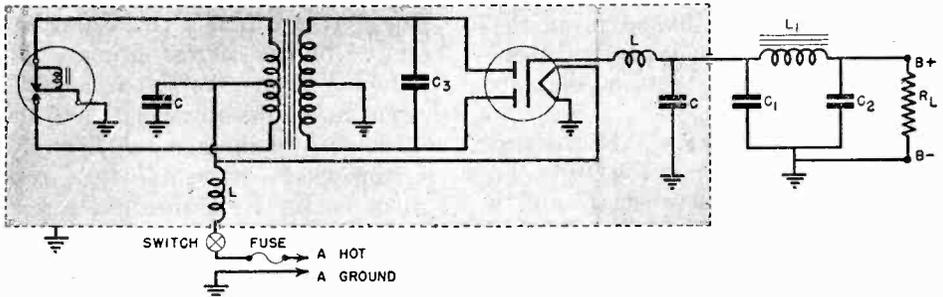


FIG. 20. A complete vibrator power supply using a tube rectifier.

source, let us see how this a.c. voltage is rectified. Fig. 20 shows a typical full-wave rectifier circuit commonly used for this purpose. The only real difference between this and an ordinary power pack is that the filament of the rectifier tube is supplied from a battery. As one battery terminal is grounded, the filament is at the same potential as  $B-$ . This means good insulation is needed between the cathode and heater of the rectifier tube, for as much as 400 volts may exist between them.

Also, the voltage pulses which result from rectification are still rather sharp and the rectifying process generates r.f. noise signals. R.F. filters, consisting of r.f. coils and small ca-

duced. For this reason, cold-cathode rectifier tubes are sometimes used in auto radios. To refresh your memory of these tubes, refer to Fig. 21, which shows the structure of a typical full-wave cold-cathode rectifier. The tube is filled with gas which, on ionization, becomes an excellent conductor.

Rectification is caused by the physical structure of the tube. When one of the pointed plates is positive it can pull many electrons out of the large cathode. But when the cathode is positive with respect to a plate, the cathode can pull only a few electrons from the small surface of the plate; as a consequence, only a comparatively small current flows in the undesired direction. In other words,

the cold-cathode tube is a two-way conductor that conducts far better in one direction than the other.

The ionization of the gas in the tube causes a brilliant purplish glow which continually flickers as current through the tube changes. The cold-cathode tube has a serious disadvantage in that it generates a large amount of r.f. noise signal—much more than the mercury-vapor tube which it resembles. Often this interference is difficult to eliminate because of poor joints at shield contacts to the chassis, changes in the tube characteristics, etc. If you meet this difficulty, the easiest way to solve it is to rewire the circuit for a heater type rectifier. Usually it is only necessary to wire in the filament circuit, retaining the original tube socket.

**Synchronous Vibrators.** Sometimes it is desirable to eliminate the rectifier tube altogether, particularly where space limitations are important. Let us see how we can do so.

Basically, a rectifier tube is just an electronic switch that reverses a circuit connection every time the a.c. cycle reverses, and so keeps current flowing always in the same direction through the circuit. We can replace this electronic switch with a mechanical one—a vibrator—and get the same effect. Of course, this vibrator must be synchronized with the vibrator in the primary circuit so that both of them open and close their circuits at the same time. We can get this synchronization by mounting both sets of vibrator contacts on the same vibrating reed. This also helps to save space, by making two reeds unnecessary.

Fig. 22A shows the fundamental operation of such a circuit. Switch *SW* is synchronized with the vibrator, and replaces the rectifier tube. Notice that more is involved than a simple replacement of the tube by the switch;

the circuit is also changed around, with the rectifying element in the negative side of the output circuit. This is necessary because one side of the battery is grounded, which means that the vibrating reed is also grounded. As the same reed is used in *SW*, if we connected the switch in the positive side of the output (as we do with a tube) we would be grounding *B+*.

When current flows through the primary in the direction shown by the

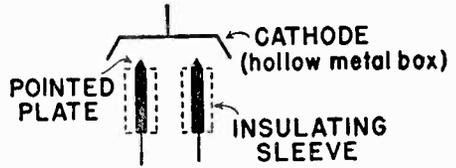


FIG. 21. The elements of a gaseous or cold-cathode rectifier tube.

solid arrow, the polarity of the secondary voltage is that shown by the solid arrows drawn between the secondary terminals. Current flow through the other half of the primary reverses the secondary voltage polarity, as shown by the dotted arrows.

Notice that the center tap *c* on the secondary will be positive with respect to *a* when current flows one way through the primary, and positive with respect to *b* when current flows the other way. Obviously, switch *SW* must connect *a* to ground when *a* is negative with respect to *c*, and *b* to ground when *b* is negative with respect to *c*, in order to maintain the right polarity for the load *R<sub>L</sub>*. Therefore, it must be synchronized with the primary vibrator. This is the reason these types are called “synchronous” vibrators, while the other type is known as the “non-synchronous” type.

► Fig. 22B shows the complete circuit of a practical synchronous vibrator, with both switch units operating from a common vibrator reed.

Either a separate driver or a shunt motor unit may be used, with the shunt type predominating. Notice that the whole unit is enclosed in a shield (dotted lines) and that r.f. filters  $L-C$  are used in both the  $B+$  and filament leads.

► You must be careful about the battery connections when you use one

This reverses the  $B+$  and  $B-$  terminal polarities, which prevents operation of the radio device. Further, this reversed polarity will ruin the electrolytic condensers if it exists for more than a few seconds.

Different polarities exist in various cars, boats, etc., where the batteries are used for ignition and other pur-

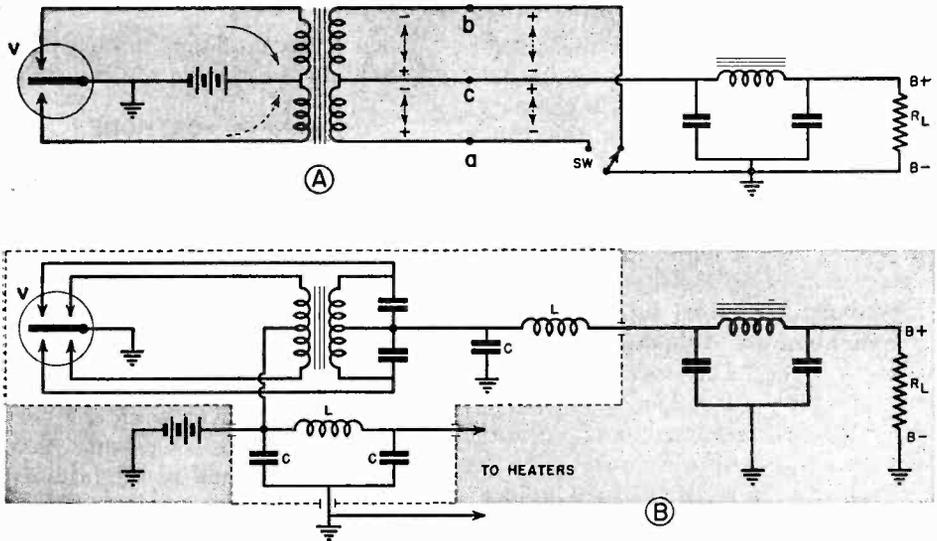


FIG. 22. The elements of a synchronous vibrator which uses mechanical rectification.

of these synchronous vibrators. When the battery polarity is reversed, the direction of current flow in the primary windings reverses, which interchanges the secondary polarities. Battery connections do not matter in a vibrator power pack which uses a rectifying tube—the tube automatically passes current in the proper direction, as whichever plate is made positive will pass current.

The synchronous vibrator pack is put together with a particular polarity in mind. Should the battery polarity be reversed, the secondary contacts would be closing to the *positive* terminals (with respect to  $c$  in Fig. 22A) instead of the negative terminals.

poses. As the battery cannot be reversed, some provision is normally made for correcting this trouble, usually by providing a means of reversing the secondary contacts. A terminal board connection, or a vibrator which can be reversed in its socket is commonly used. Always check the vibrator output after the battery has been connected. If the output has the wrong polarity, reverse the connections by the means provided.

► While vibrator systems are most often found in cars, small boats and planes, they are also used in some farm receivers and may operate from a 2- to 6-volt storage battery or

from a 32-volt farm lighting plant. In the latter case, the receiver tube filaments are generally wired in se-

ries, as in an a.c.-d.c. receiver, to operate directly from the 32-volt supply.

## Modern Battery Receivers

When you get into servicing, you'll find that receivers using batteries for their power supplies are very common; in fact, because of the wide popularity of modern portable radios, there are many more battery-operated sets in use today than there were in the days when batteries were the only source of power available. All these sets use A, B and C batteries—or suitable substitutes—to supply filament, plate and grid operating voltages. Let's look into the various ways these batteries may be used.

### Filament Voltages

The cost and trouble of replacing worn-out batteries has always been one of the main objections to battery-operated receivers. To combat this objection, engineers developed battery tubes which required lower and lower current drains. The first battery tubes, for example, required 5 volts to supply their filaments. Since each tube drew .25 ampere (250 ma.), each filament consumed 1.25 watts. Then 3.3-volt tubes\* which drew .132 ampere (132 ma.) made their appearance, followed by 2-volt, .06-ampere (60 ma.) types. Modern tubes need 1.4 volts and draw only .05 ampere (50 ma.)—a power consumption of only .07 watts, and the future will undoubtedly bring forth further progress in this line.

\*It is common practice to speak of tube types by their filament voltage rating. Thus, we may have 50-volt tubes, 1.4-volt tubes, 6.3-volt tubes, etc.

► The filaments are generally connected in parallel to an A battery which furnishes about the correct voltage. If this voltage is too high, a series resistor is used to cut it down. Fig. 23 shows such a circuit; resistor  $R$  is used to drop the voltage to the correct amount. The difference in voltage between that required by the tubes and that furnished by the battery is the voltage which the

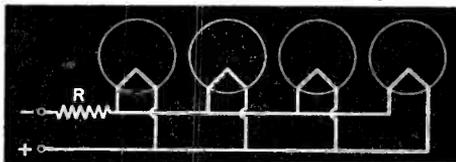


FIG. 23. Battery tube filaments are frequently connected in parallel.

resistor must drop. As the tubes are in parallel,  $R$  must carry a current equal to the sum of the filament currents. The value of  $R$  is then equal to the required voltage drop divided by the total current.

► Sometimes tube filaments requiring the same current are wired in series and supplied by an A battery which will deliver the sum of the tube filament voltages. This connection increases the A voltage required, but reduces the current drain to that used by only one tube, so the wattage (voltage  $\times$  current) furnished by the battery is no greater than for parallel filament operation—and the lower current drain lengthens the life of the battery.

## Grid Bias Voltages

Usually the control grid of a tube requires a bias to make the tube operate properly. This bias voltage normally makes the grid negative with respect to the cathode (or the filament in battery tubes).

**The C Battery.** A typical C battery arrangement for furnishing bias is shown in Fig. 24. This battery is made up of a series-connected group of 1.5-volt dry cells. It may have only two terminals, or it may have a number of taps to supply tubes needing different C bias voltages. Notice that  $C+$  connects to  $A-$ , so the bias is applied between the grids and the negative sides of their filaments. The grids of  $VT_1$  and  $VT_2$  are connected together and have a bias voltage of  $-1.5$  volts; the grid of tube  $VT_3$  is biased 3 volts negative, while  $-4.5$  volts is applied to the grid of tube  $VT_4$ .

Since the grid circuit does not ordinarily draw current, the C battery usually has a long life.

**The Bias Cell.** Various means have been worked out to eliminate the C battery and thus save space in a receiver. A cell known as a "bias cell" is frequently used in battery sets—and also in a.c. receivers where small amounts of bias are needed and hum may develop if other supplies are used.

Essentially, the Mallory grid bias cell is a very small electrochemical cell with a potential of approximately 1 volt. Its current-supplying capacity is less than 1 microampere, but it can furnish voltage indefinitely as long as no current is taken from it. This makes it suitable for the grid circuit, where normally no d.c. is drawn from the bias supply. If strong signals drive the grid positive, the resulting grid current flow will be in such a direction as to recharge the

bias cell and thus help prolong its life.

► The cell consists essentially of an acorn-shaped metal cap or shell filled with an electrolyte, and a black disc which seals the circular end of the shell. The shell is the negative electrode, the black disc the positive electrode. The disc is insulated from the shell and sealed in place by a rubber gasket around its edges.

Suitable mounting brackets have been developed for this cell, usually consisting of a cup in which the cell fits and a spring contact which presses against the black electrode and holds the cell firmly in position. Mounting brackets holding as many as nine cells in series are available for applications requiring more than 1 volt. The cell is connected just like the C battery, except that its small size makes it possible to mount it right in the receiver.

► Bias cells must be checked with a vacuum tube voltmeter which draws no current, since the cell can't supply enough current to operate an ordinary type voltmeter. When replacing a bias cell, slip the old cell out of the mounting and slip in a new one. Make sure the spring arm has enough tension to make firm contact with the black electrode.

**Convection Current Bias.** Another method of biasing a tube depends on the fact that some of the electrons flowing in a tube hit the control grid wires and are collected by the grid. These electrons flow out of the control grid lead, through the external circuit, and back to the cathode. This flow is called a "convection current." If a resistor is placed in the control grid circuit, the electrons flowing through it produce a voltage drop which biases the tube. The end of the resistor connected to

the grid is negative, as the electrons enter this end.

The convection current varies with plate current, increasing with increases in plate current. Even at best, however, this current is so small that it is measured in microamperes, so a large resistor of 10 or 15 megohms is necessary to get enough bias.

Like the bias cell, the convection current method of biasing may be used with any type power supply. It is ordinarily used in a.f. amplifier stages.

**Automatic C Bias.** Another method of biasing the tube without the use of a C battery is shown in Fig. 25. Electrons flow from each tube filament to its plate, then to B+ and through the B supply to B-. In order to get back to the filaments,

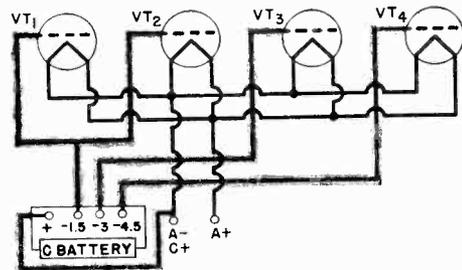


FIG. 24. Grid bias paths from a C battery.

the electrons must go through resistor  $R$  to the chassis, then through the chassis to the grounded terminal of the filament circuit. Therefore, the plate current from all of the tubes flows through resistor  $R$ . This produces a voltage drop across the resistor with the polarity shown. (Remember, the end of the resistor which electrons enter is always negative.) Suppose we bring the grid return of tube  $VT_4$  to the negative side of resistor  $R$ . Then, the drop across the resistor makes the grid negative with respect to its filament—in other words, biases the tube. We can connect any number of grid returns to resistor  $R$ .

Of course, the bias voltage we get depends on the amount of plate current flowing and the value of the resistor in ohms.

This method is called “automatic bias,” because the bias will auto-

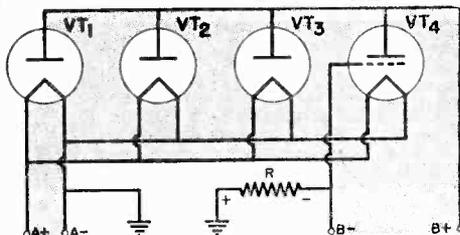


FIG. 25. The voltage drop produced by plate current flow through  $R$  produces an automatic C bias.

matically adjust itself for plate current changes, increasing when plate current increases and decreasing when plate current decreases. Notice that this method makes the B supply furnish the bias voltage.

**Filament Biasing.** One of the important reasons for connecting tube filaments in series is the fact that it is possible to get a bias voltage from the filament voltage drops. Fig. 26 shows how this can be accomplished. Here, the plate and grid signal circuit parts have been omitted for simplicity.

This circuit contains four 1.4-volt tubes, with the filaments in series. A 6-volt A pack is used, so there is a voltage drop of 1.5 volts across each filament, and .05 ampere flows through the complete circuit.

Tube  $VT_1$  has its grid returned to point 1, which is the negative side of its filament. There is no bias on this tube, as there is no voltage difference between the grid and filament.

There is a drop of 1.5 volts between points 1 and 2. Since point 3 is connected to point 2, this same drop exists between 1 and 3. Therefore, if

we connect the grid of  $VT_2$  to point 1 there will be a bias voltage of 1.5 volts between this grid and point 3 on the filament of  $VT_2$ . Similarly, the 1.5-volt drop between points 3 and 4 can be used to bias tube  $VT_3$ .

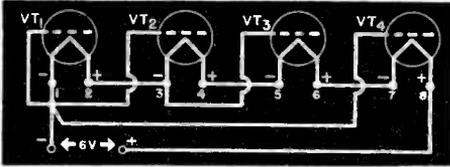


FIG. 26. The d. c. voltage drop across series-connected battery tube filaments can be used for bias too.

The grid of tube  $VT_4$  is connected back to point 1. There is a 1.5-volt drop between 1 and 2, a similar drop between 3 and 4, and a third 1.5-volt drop between terminals 5 and 6, making a total of 4.5 volts between point 1 and point 7 (the filament of  $VT_4$ ). Therefore, tube  $VT_4$  has a bias of 4.5 volts.

### The B Circuit

The B supply circuits in a modern battery receiver are very simple. The  $B+$  supply leads are joined together and are connected to the  $B+$  battery lead. The  $B-$  terminal usually connects to the  $A-$  filament terminal. The screen grid and other positive electrodes may connect to  $B+$  or to a tap on this battery. Thus, the one battery supplies all the needed positive voltages. A series resistor may be used instead of a battery tap to provide screen grid voltages, particularly on the more modern sets. A bleeder is never used in battery sets, because the bleeder current would shorten the life of the B battery.

### Typical Battery Sets

Now let's put together the things you've just learned, and study the

complete battery supply circuit in Fig. 27.

Two switches are "ganged" (connected) together in most battery sets and are used to turn the set ON or OFF. In Fig. 27, switch  $SW$  is the actual ON-OFF switch. When turned OFF, this switch opens the filament circuit and thus stops tube emission, which prevents the flow of plate current. The second switch,  $SW_1$  in Fig. 27, serves to open the B supply circuit to prevent leakage paths from draining the B battery. This is necessary because, in many sets, by-pass condenser  $C_1$  is an electrolytic, used to keep signal currents out of the B battery. Since all electrolytic condensers have a certain amount of leakage, there will be a constant drain on the B battery un-

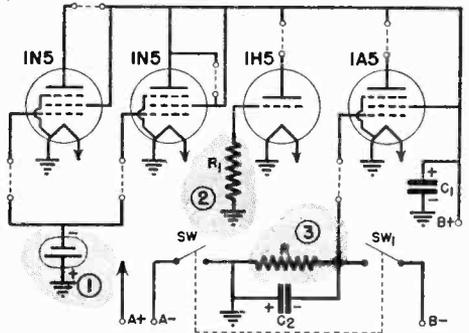


FIG. 27. Three biasing methods are shown here: 1, bias cell; 2, convection current; and 3, automatic bias.

less the B circuit through  $C_1$  is opened by switch  $SW_1$ .

With the switches closed, current flows through the tube filaments, heating them to the point where they can emit electrons. Plate current then flows from  $B-$  through switch  $SW_1$ , resistor  $R$ , and the chassis to the grounded tube filaments. The electrons return to  $B+$  from the various screen and plate circuits.

In passing through resistor  $R$ , the plate and screen currents set up a voltage drop with the polarity marked



allowed to flow through the filaments of tubes other than their own, for this would cause undesirable coupling between stages.

Therefore, we provide condenser paths for the signal currents between the tube filaments and  $B-$ . The size of each condenser must be chosen so that the impedance it offers to the signal current it is supposed to carry will be far less than the resistance offered by any path through the filaments. Since  $B-$  is grounded to the chassis, we ground one end of each condenser to the chassis and connect the other end to the filament of the tube with which it is to be used. Thus, the signal current of tube  $1A7$  passes through condenser  $C$ , that of  $1N5$  through  $C_1$ , and that of  $1T5$  through  $C_2$ . Since  $1T5$  is a power output tube and carries an audio frequency plate current, we must use a high-capacity electrolytic condenser for  $C_2$ . It should be connected with the polarity shown. No condenser path is provided for tube  $1H5$ , because its signal current flows directly to it without having to go through any other tube first. ► Referring back to Fig. 28, you see that the  $1T5$  is biased by grounding its control grid return, thus connecting it to  $A-$  (which is 4.5 volts negative with respect to the  $1T5$  filament). No external bias source is used for the  $1H5$  and  $1N5$  tubes. The  $1A7$  control grid (fourth grid from the filament) is connected through resistor  $R$  to the filament of the  $1H5$ , which connects to the filament of the  $1A7$ .

You can readily see that any voltage drop across resistor  $R$  will make the  $1A7$  control grid negative with respect to its filament. Now, when a signal (an a.c. voltage) is applied between diode plate  $D$  of the  $1H5$  and its filament, the tube acts like an ordinary rectifier for which resistor  $R$  is the load. The amount of rectified voltage across  $R$  depends on the

strength of the signal. The greater the signal voltage, the greater the voltage across  $R$  and the greater the bias on the  $1A7$ . Since this bias controls the gain of the  $1A7$  tube, we have a circuit in which the gain goes down for strong signals and goes up for weak ones—in effect, an automatic sensitivity control or, as radio men call it, an automatic volume control (a.v.c.). You'll study the automatic volume control in great detail later on, but even now you can see the principle on which it works.

### The Three-Way Receiver

The chief use today for batteries is in portable receivers. To save batteries, it is desirable to be able to operate a portable on a power line whenever the set is used where a line is available. In other words, we'd like to have a three-way power supply which can use a.c., d.c. or batteries. Let's see what kind of a circuit we need to get it.

In the receiver shown in Fig. 28, we need 6 volts for the filaments and 90 volts as the  $B$  supply. The latter can easily be obtained from a power line by using the familiar a.c.-d.c. half-wave rectifier supply shown in Fig. 30. Between the  $B+$  and  $B-$  terminals there will be approximately 90 volts, so these terminals may be connected directly to the  $B+$  and  $B-$  receiver leads.

► The lowered filament current demand of modern tubes makes it possible for this rectifier also to supply power for the filaments, something not possible in the early days of radio. Modern tubes only require .05 amp. (50 ma.) for filament operation, so when in series, this is a value well within the rectifier tube limits.

A voltage divider consisting of  $R_2$  and  $R_3$  is used across the rectifier terminals to provide the filament voltage. The 6 volts needed for the  $A$

supply appear across resistor  $R_2$ .  $R_2$  acts as a bleeder, making the A supply voltage more or less independent of tube drain.

The rectifier output is filtered by its own filter and also by the filament by-pass condenser  $C_2$  in Fig. 29. This

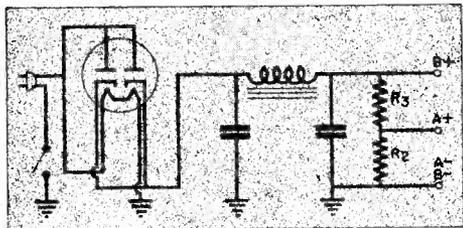


FIG. 30. This universal power supply can be used in battery sets to form a battery—a. c.-d. c. receiver.

condenser acts as a filter for tubes  $1A7$ ,  $1N5$  and  $1H5$ , because it is across the A supply as far as they are concerned. It does not act as a filter for the  $1T5$ , however, because the  $1T5$  comes between the A supply and the points where  $C_2$  bridges across the A supply circuit. Therefore, some hum currents flow through the  $1T5$  filament; since this is the output tube, there is little amplification of the hum before it reaches the loudspeaker, so a small amount of hum is permissible.

► There is an important precaution you should remember when servicing three-way receivers. *Never pull out and reinsert a tube in its socket while the receiver is turned on.*

In receivers which have a bleeder resistor across the A supply (like  $R_2$  in Fig. 30), doing so will usually cause no trouble. But suppose you pull

out a tube in a set having no bleeder across the A supply—what will happen? This breaks the circuit so there will be no filament current through the series resistor  $R_3$ , and the voltage between the A+ and A- terminals rises to practically the B supply voltage. If the pulled tube is any except the  $1T5$ , the condenser  $C_2$  will charge up to this high voltage. When the tube is reinserted, this condenser discharges through the filaments; since  $C_2$  is a large condenser, the discharge current may be great enough to burn out one of the filaments. To be on the safe side, *turn the receiver off while removing and replacing tubes.*

► We will not concern ourselves here with the various switching arrangements used to disconnect the batteries when the set is operated on a.c. or d.c., as these will be covered later. As a matter of fact, the batteries are sometimes left in the circuits at all times, and the output of the power pack is connected right across them. (No bleeders are used in such circuits, since they would drain the batteries. However, the batteries themselves act as bleeders across the power supply.)

This prolongs the life of the batteries, for the power pack sends a reverse current through the batteries at the same time it supplies A, B and C voltages to the tubes. Of course, the current through these dry batteries cannot recharge them, but it does drive off the hydrogen bubbles which form around the carbon electrodes. This reduces the internal battery resistance and helps the batteries last until their active ingredients are used up.

# Special Power Supplies

Sometimes neither power lines nor batteries can supply the power we need. For example, the power line may have the wrong voltage or frequency for the equipment we want to operate, and the device may need far more power than can be readily supplied by batteries. Let's see what special power supplies have been developed to meet such a situation.

## Engine-Driven Generators

One way to get the power we want is to make it ourselves with a gasoline-driven generator. Usually we choose to develop 110-volt, 60 c.p.s.\* power, so standard, easily-obtainable equipment may be employed. Most a.c. equipment is built for this frequency and voltage because they are found throughout the U.S.A., except for the relatively few districts having d.c., 25-cycle or 40-cycle a.c. lines.

▶ A typical gasoline-engine-driven a.c. generator is shown in Fig. 31. This type is used by servicemen in mobile public address installations, which usually require fairly large amounts of power. The unit is entirely self-contained and is furnished with a gasoline storage tank. Various sizes can be obtained, but a 300-watt unit is quite common and satisfies most needs. A 6-volt battery (usually a storage battery) is the only auxiliary equipment. It is needed to excite the electromagnets of the generator and to operate the ignition system of the gasoline engine. In an automobile installation, the car battery may be used. The output of the generator has a simple spark filter

\* The abbreviation "c.p.s." stands for "cycles per second." Radio men frequently shorten this expression to "cycles," calling a 60 c.p.s. line a 60-cycle line.

(coil and condenser) to suppress interference.

## Operating Equipment on Supplies for which They Were Not Designed

In your service work, particularly if the power line in your community is not the standard 60-cycle, 110-volt variety, you may be asked to make a set run on power for which it was not designed. Let's take up some of the most common problems of this sort, and see how you can solve them.

**Adapting to a Line of a Different Frequency.** Most a.c. power lines have frequencies of 25, 40 or 60 c.p.s. Occasionally you will have to install a radio receiver using a power transformer on a line having a frequency other than that for which the receiver was designed. Can a receiver de-

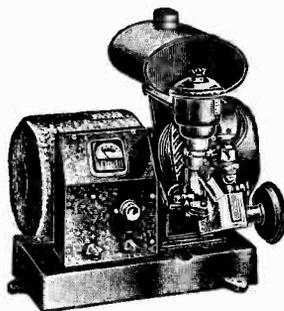


FIG. 31. A gasoline engine-driven generator of the type used in radio and public address work.

signed for one frequency be used on another? Yes and no!

For one thing, a receiver designed for 60 c.p.s. operation will not usually have a ripple filter sufficient to handle 40 to 25 c.p.s. Even more important, transformer equipment designed for low frequencies will work at higher frequencies (within reasonable limits) but the reverse is not true. That is, a 25 or 40 c.p.s. receiver will work with

a slight change on 60 c.p.s., but a 60 c.p.s. receiver should never be run on 25 or 40 c.p.s. lines. They may operate for a while, but in a short time the transformer will burn up.

The reason is that most transformer primaries are built to have a fairly high reactance *at the frequency on which they're designed to operate*; this prevents the primary from drawing large currents from the line. Low-frequency transformers, for example, have large, heavy iron cores to help give them the reactance needed. Now as you have already learned, the reactance of a coil is less the lower the frequency of the voltage applied to it. Therefore, a primary designed for 60-cycle lines will have a much lower reactance when it is used on 25- or 40-cycle lines, and so will draw much more current than it is designed to carry. This current overload will overheat the transformer and eventually burn out its windings.

Therefore, to operate a 60 c.p.s. receiver from a 25 c.p.s. line, the power transformer must be replaced with a 25 c.p.s. transformer having the same output ratings. Then extra condensers must be added in parallel with the filter capacities to increase the amount of filtering. If the pack has tuned filters, they must be retuned to 25 c.p.s. or the tuning parts removed.

► When a 25 c.p.s. or 40 c.p.s. receiver or amplifier is to be operated from a 60 c.p.s. line, the transformer will be more efficient at the higher frequency (because of its large core) and therefore will deliver a higher voltage than normal. If voltages more than 10% above normal are produced, you should use a line regulator (variable resistor) in series with the primary, and adjust the resistor until the voltages are normal. Special resistors for this purpose are available from radio supply stores.

### D.C. Equipment on A.C. Lines.

Often people move from a 110-volt d.c. district to a 110-volt a.c. district. In this case it is best to recommend getting an a.c. receiver, because it will be much better than the d.c. receiver. But if the customer is satisfied with his d.c. receiver (and it is only a d.c. set, not a universal receiver), you could recommend a small a.c. motor-driven d.c. generator. This

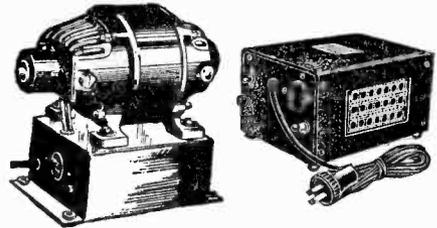


FIG. 32. Either the rotary converter at the left or the inverter at the right can be used to obtain 110-volt, 60-cycle a. c. power from d. c. power lines. The inverter is a high-power vibrator unit, using no rectifier.

equipment is usually more costly than a new receiver, however.

### A.C. Equipment on D.C. Lines.

As a rule, a.c. receivers are so much better than d.c., universal and battery receivers that it is quite common to adapt a good a.c. receiver to 32-, 110- and 220-volt d.c. lines instead of buying a receiver designed for these special voltages. Two procedures are possible. Use a d.c. to a.c. rotary converter (a combination d.c. motor and a.c. generator) or a magnetic vibrator type d.c. to a.c. converter (often called an inverter). These devices are shown in Fig. 32.

For the average receiver, a 100-watt converter or inverter is large enough. Adaptation is simple—just insert the receiver power plug in the receptacle of the converter, then connect the converter to the d.c. line. If a ground terminal is provided on the converter, connect a wire from it to the regular ground. The rotary con-

verter is costly, but has a long life and can be had in any desired power rating for receivers or public address equipment; the magnetic vibrator inverter is comparatively inexpensive, but its power capacity is limited to 200 watts maximum, and the vibrator must be replaced about once a year.

### Power Line Operation of Battery Sets

You may often wish to operate a battery receiver (not a three-way receiver) using 1.4-volt or 2-volt tubes from the a.c. power line. Power packs are available which do not take up any more room than the original bat-

teries; in fact, the batteries may be removed and the power packs substituted for them. A typical power pack is shown in Fig. 33. A full-wave rectifier tube is used with an untuned filter to provide B and C voltages. The B tap of the voltage divider is tapped to provide any voltage normally required, while various values of C voltage are obtained by adjusting the slider on the lower section of the divider.

A copper-sulphide (similar in action to a copper-oxide) type rectifier is used in a full-wave bridge circuit for the A power. Filters in cascade furnish hum-free A voltage.

Copper-sulphide, copper-oxide and a recently developed selenium-iron unit are metal discs, specially prepared to act as rectifiers. These rectifiers have very low voltage drops and can be designed to carry high currents, so are better than receiver-type tubes where low-voltage, high-current power is needed. Vacuum tube rectifiers have an internal drop of 15 or more volts, which may be half or more of the available voltage in a low-voltage circuit and so represents a large power loss. (Tubes are better in high-voltage circuits, where 15 volts may be only a 3 or 4% loss and where disc rectifiers cannot withstand the high peak inverse voltages.)

These units have the characteristic of letting electrons flow in one direction much easier than in the other. The elements are bolted or clamped together in such a manner that all four sections of a bridge make a single unit having four connections.

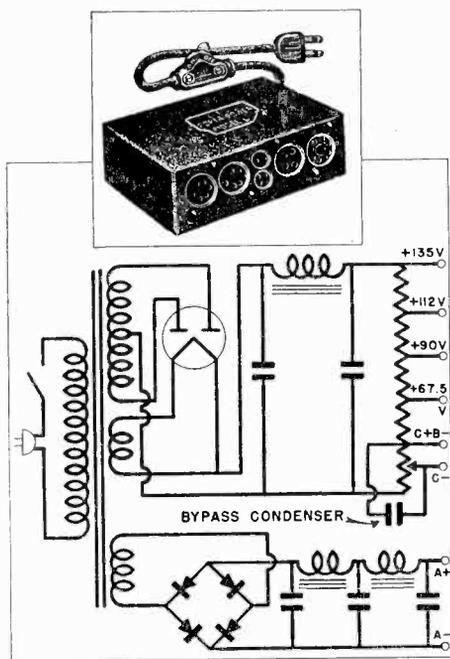


FIG. 33. This compact A-B-C power unit can be used in place of batteries so power-line operation can be obtained.

# Lesson Questions

**Be sure to number your Answer Sheet 13FR-2.**

**Place your Student Number on every Answer Sheet.**

***Send in your set of answers for this lesson immediately after you finish them, as instructed in the Study Schedule. This will give you the greatest possible benefit from our speedy personal grading service.***

1. What is the maximum possible voltage which could be obtained from a universal a.c.-d.c. power supply, operated from an a.c. 115-volt power line, with no load?
2. Why is a condenser-input filter used instead of a choke-input filter, in a universal a.c.-d.c. receiver power supply?
3. What is the purpose of placing a resistor at point *a* in Fig. 3?
4. Suppose you are replacing the pilot lamp in a universal receiver. Is the current rating of the replacement lamp important?
5. Can a transformerless receiver using a voltage doubler be operated from a d.c. power line?
6. In connecting a receiver having a non-synchronous vibrator and tube rectifier, does the polarity of the storage battery matter?
7. Name five methods of getting C bias in a battery receiver.
8. Why is switch  $SW_1$  used to open the B supply circuit when turning off a battery receiver like the one shown in Fig. 27?
9. What type of meter must be used to check a grid bias cell?
10. Why are copper-oxide (or selenium) rectifiers always found in low-voltage, high-current power supplies, instead of vacuum-tube rectifiers?

## FIRST IMPRESSIONS

First impressions mean a lot in this busy world. An applicant for a job has a pretty tough time making the grade if his appearance and first few words do not make a favorable impression on the employment manager. A salesman likewise gets the "cold shoulder" if there is anything about him which annoys the prospect.

With technical material of any kind, however, first impressions can be very treacherous. Oftentimes a simple technical book will contain a number of apparently complicated diagrams, charts, graphs, sketches or tables. Since we glance mostly at illustrations when inspecting a book, we are apt to get a misleading impression. Also, paragraphs, pages or entire lessons may seem difficult during the first reading, but become almost magically clear during the second or third reading.

If first impressions of a required task are favorable, fine and dandy; if unfavorable, don't be discouraged, but wade right into the work and give it a chance to prove that *first impressions don't always count*.

J. E. SMITH.

**HOW SOUNDS AND SCENES ARE  
CONVERTED INTO AND REPRODUCED  
FROM ELECTRICAL SIGNALS**

14FR-2

**NATIONAL RADIO INSTITUTE**

ESTABLISHED 1914

**WASHINGTON, D. C.**



# STUDY SCHEDULE NO. 14

For each study step, read the assigned pages first at your usual speed. Reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind, then answer the Lesson Questions for that step. Study each other step in this same way.

1. Introduction to Acoustics . . . . . Pages 1-7  
Since the entire broadcasting system is created for the purpose of transferring *sound* from one point on the earth to another, you will find this discussion of sound to be highly interesting. This is background information, to help you understand the characteristics of sound as they affect the electrical signal. Answer Lesson Questions 1, 2, 3, 4, and 5.
2. Technical Facts about Sound and Hearing . . . . . Pages 8-13  
Here you will learn the technical differences between noise, speech, and music, and will learn that the human ear has peculiar response characteristics—important to radio systems. Answer Lesson Questions 6 and 7.
3. Sound Pickups and Reproducers . . . . . Pages 14-17  
An introduction to pickups and reproducers, in which you learn how sound is converted into electrical currents, and vice versa.
4. Requirements of an Audio Amplifier . . . . . Pages 18-25  
A practical section showing how an amplifier can distort sound signals. Answer Lesson Question 8.
5. The Fundamentals of Television . . . . . Pages 26-31  
Old and modern ways of converting a scene or picture into electrical signals which can be handled by a radio broadcasting system. Answer Lesson Question 9.
6. Video Amplifier Requirements . . . . . Pages 32-36  
The wide range of frequencies which must be handled in modern television require unique circuits. This section is intended to give you an idea of some of the problems—don't spend too much time on it as you will study television in far greater detail later. Answer Lesson Question 10.
7. Mail your Answers for this Lesson to N.R.I. for Grading.
8. Radio Receiver Troubles—Their cause and Remedy . . . . . Reference Text 14X-1  
Read the instructions on the inside front cover of this text, then just thumb through the book, reading only the headlines, to get acquainted with its contents and learn how to find any desired subject in it. Once you begin actual servicing of radio sets, you'll really appreciate the value of this carefully cross-indexed summary of receiver troubles.
9. Start Studying the Next Lesson.

# HOW SOUNDS AND SCENES ARE CONVERTED INTO AND REPRODUCED FROM ELECTRICAL SIGNALS

## Introduction to Acoustics

**R**ADIO, as you know, is a *means of sending intelligence through space*. Since it requires no man-made paths between sending and receiving points, it is obviously more desirable in many cases than communication systems which send the same intelligence over land wires or cables. Although land wire telephone, telegraph, and picture-transmitting systems are not of particular importance to radio men, they differ fundamentally from radio systems only in the method used for transmitting the intelligence from the sending to the receiving point. In fact, research work carried out by telegraph and telephone companies has contributed much to the improvement of radio apparatus and technique.

► The three forms of intelligence which can be sent over either wire or radio communication systems are: 1, *sound*, such as that used in radio broadcasting, in radiotelephone communication, and in telephone systems; 2, *pictures*, either still or moving, such as those transmitted by radio or land wire television and facsimile systems; 3, *code*, such as the dots and dashes used in radio and wire telegraphy.

Code transmission can be obtained simply by opening and closing a key in the transmitter circuit which, in the case of radio, causes the carrier current to be fed intermittently to the antenna and which, in the case of wire telegraphy, sends pulses of current over the land wires.

On the other hand, sounds and scenes

represent intelligence which must be converted into equivalent electrical signals before being sent over wires or given "rides" through space on radio carriers. At the receiving end, the intelligence signals pass through many electrical circuits before being restored into the original sounds or scenes.

Thus, we must have a device capable of responding to all the characteristics of the sounds or the scenes, so that it can render a faithful electrical reproduction of the desired intelligence. Also, we must have reproducing equipment which can be fed an electrical signal and which will reproduce the original sounds or scenes as faithfully as is economically practical.

Furthermore, amplifiers must be used in both the transmitter and the receiver, in order to increase the voltage and the power of the electrical signal. Since the electrical signal characteristics, in turn, depend upon the sounds or the scenes, we must learn something about the characteristics of sound waves and light waves. First, we will study sound waves.

### SOUND IS A WAVE MOTION

You know that sound can be produced by striking metal objects together, by rustling stiff paper, by vibrating your vocal chords, by plucking a stiff wire or spring, or by any one of thousands of different methods. Yet, though the ways we can create sound appear to be many and varied, they are basically identical. *In every*

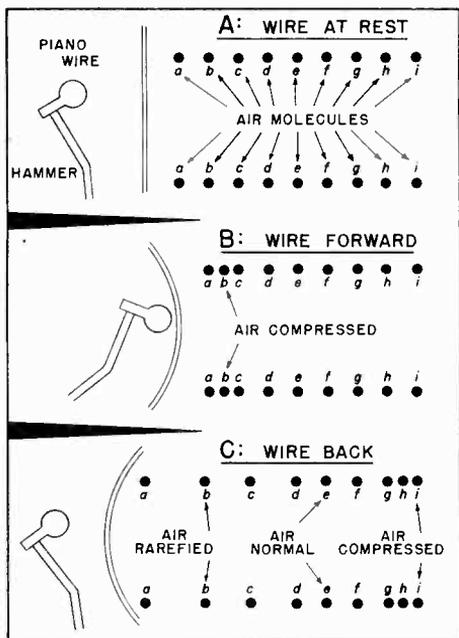


FIG. 1. When a piano key is depressed, a "hammer" strikes a piano wire (actually a group of wires). The wire is set into vibration; this vibration is transferred to the surrounding air molecules. The alternate compressions and rarefactions of the air result in sound waves which travel out from the wire in all directions. (For clarity, only one direction is indicated here.)

case, we create sound only if we produce a vibration. The vibration may be of very short duration, as when two stones are struck together, or it may be prolonged, as when a bell is rung—but, whether short or long, a vibration must exist before sound can be produced.

► However, a vibration alone is not enough to make a sound. The following is a favorite experiment in high-school physics classes: An alarm clock is suspended in a glass jar. The jar is then sealed except for a hose connection leading to an evacuation pump. Provided enough of the air in the jar is removed, no sound is heard when the alarm clock goes off—even though you can actually see the clock hammer

strike the bell. This is an illustration of another basic fact about sound: *no sound is produced by a vibration which occurs in a vacuum*. There must be an elastic medium, capable of conducting the sound vibrations, in contact with the vibrating object before a sound is produced. This medium may be a gas, like air, or a liquid, like water, or a solid, like steel—in fact, it may be anything but a vacuum.

We know, then, that we must have some vibrating object and some medium in contact with the object before we can have a sound. Scientists tell us that the sound itself consists of *a vibration of the medium*. The vibrating object is the *source* of sound, the medium is the substance through which the sound is *transmitted*, but the sound itself is not a physical object like either of these; it is a vibratory *motion*.

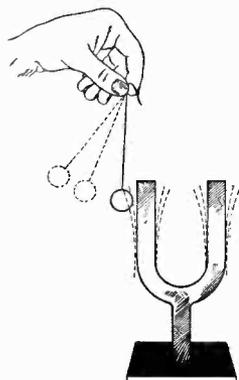
► Let's make this clearer with an example. Suppose we consider what happens when a piano wire is struck by its hammer. Fig. 1A shows a piano wire, its hammer, and a schematic representation of a few of the air molecules which surround the wire. If we strike the piano key, the hammer hits the wire and bows it outward. The moving wire strikes the air molecules *a* in front of it and drives them forward; they then strike molecules *b*, which in turn are driven forward to strike molecules *c*, and so on. In this way, the motion of molecules *a* will be transferred all the way along the line of molecules. In fact, if it were not for the inevitable losses of energy which occur in such an operation, the forward motion of the molecules would be transferred indefinitely.

However, it is only the *motion* which is transferred in this manner; the individual molecules lose most of their energy when they strike the next molecules in line, and soon come to a stop.

In other words, *molecules* do not travel from the source of sound to you—they *pass along energy from one to another*.

Fig. 1B shows the molecules as they would look an instant after the hammer has struck the wire. As you can see, the forward motion of the molecules has produced a compression of the air in the region of molecules *a*, *b*, and *c*, while farther on, where the motion has not yet reached, the molecules are still their normal distances apart (the air is uncompressed).

After it has bowed as far forward as it will go, the piano wire snaps back toward its original position. However,



Here is experimental proof that sound-producing bodies vibrate. The tuning fork will produce sounds when struck and forced into vibration. If a ball is suspended so that it touches the vibrating fork, it will bounce away, proving that the fork is in motion.

because of its elasticity, it does not just come back and stop; instead, it overtravels and bows backward. Its backward motion drives away the air molecules which were originally behind it, leaving a partial vacuum in its path. Molecules *a*, which by this time have stopped their forward motion, now rush back to fill up this vacuum; molecules *b* then come back to fill the space vacated by molecules *a*, molecules *c* come back to take up the space just vacated by molecules *b*, and so on. We now have a backward motion of the air molecules which, like the previous forward motion, will be transferred all along the line of molecules.

Fig. 1C shows the distribution of the molecules when the piano wire has reached the end of its backward swing: molecules *a*, *b*, and *c* are now farther apart than normal, because they are still moving to fill up the vacuum created by the backward motion of the wire; hence, the air in this region is rarefied. (Rarefy means to make less dense; it is the opposite of compress.) Molecules *d*, *e*, and *f* have already gone through their forward motion, and are just starting on their backward motion. Therefore, they are about their normal distances apart, and the air in this region is neither compressed nor

rarefied. Molecules *g*, *h*, and *i* are still going through their forward motions, so the air in their region is compressed. Thus, distant molecules are made to move back and forth in the same manner as those at the source of the sound.

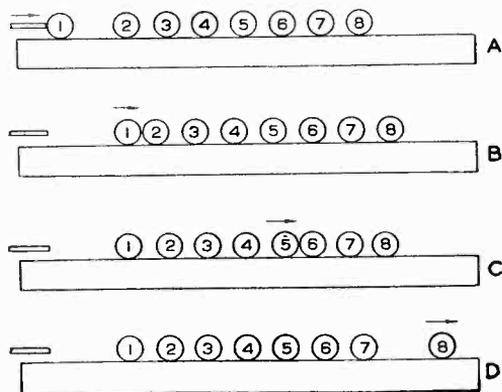
The piano wire will snap back and forth at least several more times before it comes to rest, and each complete back-and-forth motion will produce two motions which will be transferred through the air molecules surrounding the wire—a *forward* motion (which we can call a compression wave) and a *backward* motion (a wave of rarefaction). Remember: the individual molecules move forward only until they strike other molecules, and

move backward only until they are struck by the next wave of forward-moving molecules.

► Let's pretend that the piano wire produces a pure tone having sine-wave characteristics. (Actually it doesn't produce a pure sine wave, but let's see what could happen if this were true.) Then, Fig. 2 illustrates the movements of the molecules after the piano wire has vibrated several times. Instead of picturing just two lines of molecules in this figure, we have drawn vertical lines which represent millions of

ticular instant. The horizontal line represents normal air pressure. At points of rarefaction, the air pressure is below normal; at points of compression, it is above normal. The final result, when we draw a curve through all the points constructed, is a simple sine wave.

► However, whether or not the sound wave is a sine wave, regardless of its form, the air molecules will be alternately compressed and rarefied. The plot of air pressure thus follows the vibrations of the source, so that the plot shows the exact wave shape. This



Another example of wave propagation. When ball 1 is struck by the cue at the left (in A) it moves forward, striking ball 2. Ball 1 then stops rolling as its energy has been imparted to ball 2, which now strikes ball 3, etc. As shown in B, C, and D, the balls move successively as energy is transferred from ball to ball. Sound energy is transmitted from molecule to molecule in much the same way.

molecules; each molecule on a given line is the same perpendicular distance from the piano wire as every other molecule on that line. The lines which are bunched close together therefore represent compression waves, while those spread out represent waves of rarefaction.

Notice the wave form shown directly below the picture of the sound waves. This wave form is constructed from the picture by plotting the air pressure or air density at various distances from the piano wire, at one par-

means that sound is a *variation of pressure* in the transmitting medium.

► To put this important fact another way, we can say that a diaphragm struck by a sound will be alternately pushed in by the compression waves and pulled out by the waves of rarefaction. The pressure on the face of the diaphragm will vary above and below normal.

► Actually, a sound wave is very seldom a pure sine wave. The vibrating piano wire we have already discussed would, in addition to the fundamental

sine wave pictured, also give off several harmonics of the fundamental, and might have an extremely complex wave shape. You will learn more about this a little later in this lesson.

### SPEED OF SOUND WAVES

The speed of sound waves in any solid or liquid material depends upon two things, the *elasticity* of the material and the *density* of the material. You know that a material can be made to stretch or compress by applying force; the amount of compressing or stretching (the decrease or increase in size) is called the strain. *The more force it takes to strain a material a given amount without permanently changing its shape, the greater is the elasticity or springiness of this material.*

Notice—this scientific definition of elasticity is just about the opposite of the way most people use the word. The average man would say, for example, that rubber is more elastic than steel. But according to the scientific defini-

tion of elasticity, steel is much more elastic than rubber, for it takes much more force to compress a piece of steel a given amount than it does to compress a piece of rubber the same amount. Keep the scientific definition clearly in mind, for that is what we mean when we refer to the elasticity of a material. As a general rule, the harder a material, the more elastic it is.

► The density of a material is its weight per unit volume; if bricks of identical size are made up from various materials, the material in the heaviest brick will have the greatest density.

► The more elastic a sound-conducting material is, the faster sound waves can travel through it; the greater the density of a sound-conducting material, the slower will be the speed of sound waves through it. Both elasticity and density must be considered in determining whether one material will transmit sound faster than another. If two material objects have approximately equal densities, you can determine readily which will transmit sound

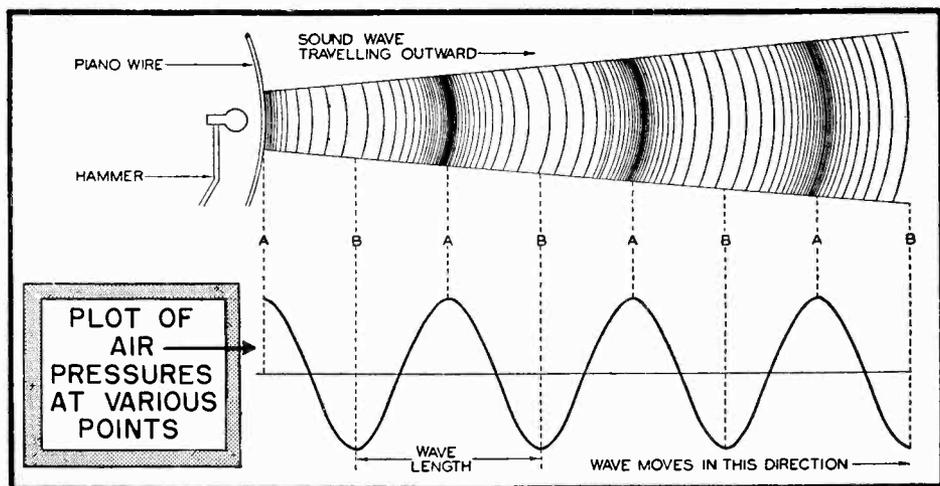


FIG. 2. By plotting the air pressure we would find that the sound waves travel away from the wire as alternate changes in pressure. The actual "plot" will show the wave form of the source. Thus, we get a sine-wave plot if the source produces sine-wave sounds. (A piano wire produces a more complex wave than that shown here.)

waves the faster by comparing their elasticities. For example, sound waves travel faster through aluminum than through soft rubber, because aluminum has a higher elasticity.

The speed of sound waves differs for various materials and even varies considerably with conditions in the same material. With air, for instance, the speed of sound waves increases with barometric pressure and with temperature. Careful laboratory experiments have shown that, at a normal atmospheric pressure and at a temperature of 0° centigrade, the speed of sound waves through air is about 1,089 feet per second. (These measurements are made in still air, since wind naturally affects the speed of sound in air.) Since the speed of electromagnetic waves (radio and light waves) is 186,000 miles per second, it is easy to understand why you can see lightning a long time before you hear the thunder which it produces.

Sound waves travel through water at a rate above 4700 feet per second. Steel is about seven times as dense as water (which would tend to slow up sound waves), but its elasticity is so much greater than water that sound waves will travel through steel at about 16,300 feet per second, almost four times their speed through water.

### REFLECTED, TRANSMITTED AND ABSORBED SOUNDS

In radio and in the allied fields of public address and sound movies, we have a great deal to do with sounds produced in air. Ordinarily, these sounds are produced in rooms or in confined places where walls, ceilings or floors prevent the sound from traveling out in all directions. When sound waves strike a material, they are *reflected from the surface of the material, absorbed by the material, or*

*transmitted through the material;* these three factors are shown in Fig. 3.

**Echoes.** When sound waves are reflected by a flat surface which is quite far away from the source of sound, we may hear *echoes*, which are distinct but weaker reproductions of the original sound. Echoes are heard in many large rooms or in small rooms having curved walls which reflect the sound waves back to their source in a concentrated form. Echoes can be removed from a room by changing its shape or making it smaller; this clearly is a job for the architect.

**Reverberation.** When sound waves are reflected repeatedly back and forth

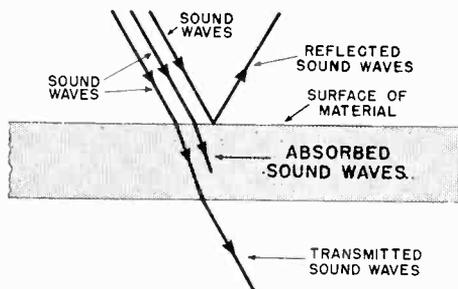


FIG. 3. Three things can happen to a sound wave which hits the surface of a material, as shown here. The kind of material will determine which one (or more) of these effects will occur.

between the walls of a room in a manner similar to that shown in Fig. 4, we have *reverberation*. The waves which reach the listener after reflection (over paths 2, 3, and 4) mix with the waves heard directly (over path 1) to give a "blurred" sound. The blur is produced because the reflected waves take longer than the direct waves to reach the listener, since they have farther to go. Thus, a sound heard over path 1 is followed a fraction of a second later by the same sound heard over path 2, then by the same sound heard over path 3, and finally by the same sound heard over path 4. This means that sounds coming over path 1 will be interfered

with by earlier sounds coming over the other paths. This reduces the understandability of words. If a great many such reflection paths exist, it may take a considerable time for a sound to die out, so that all intelligibility is lost.

The time required for a sound in a room to decrease to one-millionth of its original intensity is called the *reverberation period* of the room. Auditoriums which have reverberation periods of between one and two seconds are considered to have good acoustic qualities. The reverberation period of a room can be reduced by placing sound-absorbing material on the walls and ceiling.

**Sound-Absorbing Materials.** The better the sound-absorbing material used, the less sound will be reflected or transmitted. A soundproof room must have good sound-absorbing surfaces which keep out external sounds and also absorb sounds produced inside. An open window is sometimes considered to be an ideal sound “absorber” because it allows the sound to pass out of the room; on the other hand, it also allows external sound to enter the room, so you can’t soundproof a room by leaving all the windows open! A sound-absorbing material dissipates the sound energy by converting it into heat. There are two ways of doing this: 1, a porous material has tiny “pockets” such that when sound waves enter, their energy is dissipated by being reflected back and forth in the pocket; and 2, soft materials will “give” under sound pressures but do not have the elasticity to bounce back and “throw” the sound wave back as a reflection. Thus, energy is absorbed by the “cushioning” effect of the material. Solid dense materials are good sound transmitters, and therefore poor absorbers; soft, pliable, and porous materials like

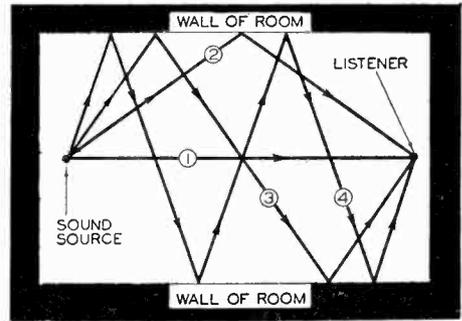


FIG. 4. Reverberations are produced whenever sounds can come to a listener over more than one path. Thus, sound waves reflected from the walls may travel over paths 2, 3, and 4 to the listener. As these paths are longer than the direct path (1), these reflected sounds arrive later than the direct sound waves. If they are of sufficient strength, they may blur the sounds. However, a certain amount of reverberation is desirable when listening to music.

velvet, Celotex, rock wool, cotton, carpet, and porous plaster are good sound-absorbing materials. The thicker the material, the more absorption of sound there will be.

► The fact that an audience absorbs sound waves is recognized by public address technicians; they reduce reverberation and echo effects by using directional loudspeakers which are directed at the audience rather than at the walls of the auditorium. The same thing is done in movie theatres—possibly you have noticed that the sound changes in quality as the size of the audience changes.

► So far, we have shown how sounds are produced, and also indicated some of the problems encountered when sound waves are produced within rooms. Reflection and sound absorption have much to do with the characteristics of the waves which we intend to pick up. Before we go into a study of pickups, however, let us learn more about the wave shapes which we will encounter.

# Technical Facts About Sound and Hearing

A vibrating body, as you already know, produces sound waves. Since these sound waves consist of compressions and rarefactions of the particles in the transmitting medium, sound waves are studied best by measuring the pressure which they exert in the medium. Thus, in Fig. 2, the pressure which the sound waves (produced by a vibrating piano wire) exert at any point varies in exactly the same manner as the motion of the wire. If the diaphragm mechanism shown in Fig. 5A is placed in the path of a sound wave, the variations in sound pressure will bend the diaphragm in and out. This motion will be traced on a moving strip of paper by the link and pencil mechanism. If the sound wave is of a simple sinusoidal form, the tracing on paper will resemble the sine wave shown in Fig. 5B; if it is of a complex form, the tracing may be somewhat like that shown in Fig. 5C.

You will learn shortly that a complex sound wave, like that shown in Fig. 5C, consists of a basic or fundamental frequency and many overtone or harmonic frequencies, each of which can be considered to be a simple sine wave. Any *simple sine wave* has two important characteristics—*amplitude* and *frequency*—whether it is a sine-wave sound traveling in a sound-conducting medium or a sine-wave current flowing in an electrical circuit. *The amplitude of a sound wave determines the loudness of the sound, while its frequency determines its pitch.*

## HARMONICS

Speech, music, and noise are never pure sine-wave sounds; they always consist of a fundamental frequency and many higher frequencies which we call

harmonics or overtones. *The harmonics or overtones give certain characteristic qualities (called "timbre") to a sound.* That is, these combinations of fundamental and harmonic frequencies give to speech and music distinguishing characteristics which, through experience, we are able to interpret. Thus we are able to distinguish between a harmonica, a trumpet, a violin, a flute, and a tenor singer, due to the different harmonic values even though all may be producing the same fundamental frequency.

From the foregoing, you can see that any sound has three characteristics: loudness, pitch, and quality or timbre.

### How Common Sounds "Look."

The wave forms of a number of common sounds are illustrated in Fig. 6. These waves can be seen on the screen of a device known as a cathode ray oscillograph, which you will study later. This device would trace the wave shown at *A*, if you were to say "ah" before a microphone which was suitably connected to the oscillograph. A greatly different wave form, that shown at *B*, results when the same sound "ah" is sung. The key of C nearest the center of a piano keyboard (known to musicians as "middle C") produces the wave form shown at *C*, while street noise gives the very jumbled wave form shown at *D*. You will notice that with the exception of noise, wave forms of sound appear to repeat themselves at regular intervals; the time of one such interval determines the basic pitch or fundamental frequency of the sound.

► Musical tones, whether produced by stringed instruments, wind instruments, or the singing human voice, all consist of a fundamental frequency and

a number of harmonics. For example, one of the C notes on a piano has a fundamental frequency of about 517 cycles per second. This note also contains a second harmonic of 1034 cycles, a third harmonic of 1551 cycles, a fourth harmonic of 2068 cycles, and a fifth harmonic of 2585 cycles. Each harmonic is a pure sine-wave tone.

The amplitudes of these harmonics differ greatly. The second harmonic in

tain instruments, some of the harmonics are missing and, in some cases, a higher harmonic is stronger than a lower harmonic, or it may be even stronger than the fundamental frequency. It is the harmonics which make musical tones pleasing, while their number and amplitude determine the "timbre" or the characteristic which lets us distinguish between instruments.

► Speech differs from music essentially in that it is less melodic, being produced more nearly as a monotone. Speech sounds, like music, contain a fundamental pitch which distinguishes between the voices of children, women, and men, plus many different higher frequencies with certain frequencies predominating to give the characteristic distinctions between the voices of different persons.

► Noise is somewhat harder to define. Essentially, any unpleasant sound can be called a noise, whether it be just an excessively loud sound or whether it be musical notes combined in an unpleasant manner. Ordinarily, however, a noise is considered to be any irregular, distracting or unpleasing sound which would tend to mar musical reproduction or reduce the intelligibility of speech. Graphs of most noise sounds show irregular, sharp peaks, and are of an unsymmetrical nature. Noise contains overtone frequencies which are not necessarily harmonic frequencies—that is, they are not related mathematically in the same manner as are harmonics. Furthermore, instead of having a single well-defined fundamental frequency, the noise pulse usually will consist of a number of frequencies over a rather wide band.

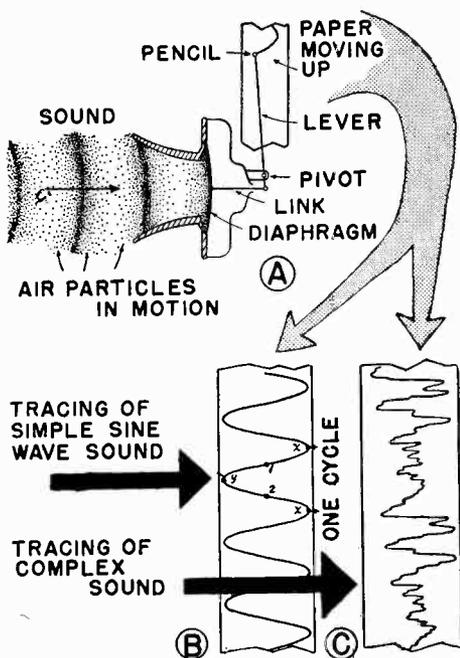


FIG. 5. The simple mechanism at A can be used to trace sound waves, thus proving that sound waves exert a varying pressure on the diaphragm. Typical tracings are shown at B and C.

the above example is 20% of the amplitude of the fundamental, the third harmonic 25% of the fundamental, the fourth harmonic 10%, and the fifth harmonic about 8% of the fundamental. Another instrument producing the same note also may produce the same harmonics, but the amplitude of these harmonics would not be the same as those produced by the piano. In cer-

## FUNDAMENTAL FREQUENCIES

The piano, organ, and harp produce the greatest ranges of *fundamental* fre-

quencies—from about 16 to about 4096 cycles. A baritone singer produces fundamental frequencies between about 80 and 400 cycles; a piccolo can “tweet” between about 500 and 5000 cycles; a ukulele has the limited frequency range of between about 300 and 1000 cycles. Thus, each instrument has its own range of fundamental frequencies as well as harmonics or overtones of these fundamentals. Radio apparatus must handle all these fundamentals and the audible overtones, if it is to give high fidelity reproduction.

### THE HUMAN EAR

The human ear is by no means an ideal sound-interpreting device. One can control his ears, causing them to hear desired sounds and to disregard others. This is helpful in ignoring noise, but the same mechanism serves to fool one into hearing sounds that are not present! Thus, a person familiar with the timbre produced by, let us say, a piano, will automatically “hear” this timbre although a radio may be reproducing only a part of the harmonics necessary for a true reproduction. This extra effort is tiring, however, and soon one does not wish to listen further.

► In addition, human ears actually can become accustomed to some types of distortion and like it. A boomy radio receiver, with an excessive low-frequency response—sounding as if it were in a barrel—gives distorted reproduction, but many persons like this. In fact, some prefer it to high-fidelity reception. To musicians, however, or to others who appreciate high fidelity, any such distortion may be very annoying.

► Radio technicians should understand this queer behavior of the human ear; it is likewise of vital impor-

tance to the designer of radio apparatus. There is no need of spending time and money in reducing distortion when the improvement in fidelity cannot be noticed by the average human ear; nor is there any need to make a receiver respond to frequencies which cannot be heard by the normal human ear.

By studying the responses of thousands of persons to sounds of various

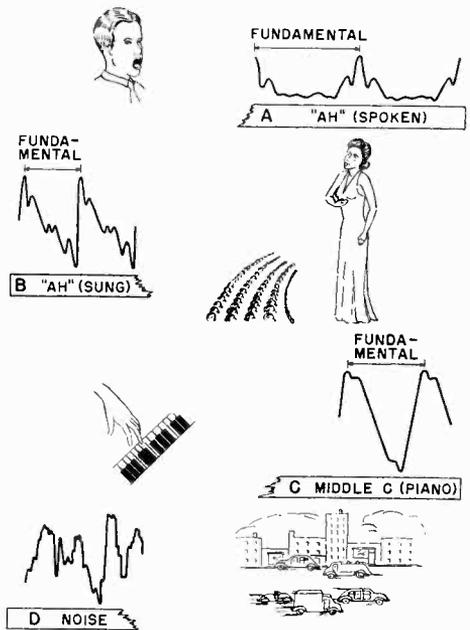


FIG. 6. Sounds can be “seen” as well as heard. By connecting a cathode ray oscilloscope to a microphone, curves like those shown here will be traced on the fluorescent screen of the tube.

frequencies, the Bell Telephone Company has found that the maximum range of frequencies which can be heard by even the best ears extends from 20 cycles to 20,000 cycles; others claim that this range is from 32 to 16,000 cycles. These tests also showed that the human ear is far more sensitive to sounds in the 1000- to 4000-cycle range than to sounds outside this range. Very high sound pressures are

required before the ear can detect very low and very high frequencies in the audible range.

**Sounds Can Cause Pain.** When sound pressures are increased too high, a person stops hearing and actually begins to feel the sounds. At the lower frequencies, the vibration of the air can be felt by all parts of the body, but at the high frequencies, the action is more a sensation of pain in the ear. The response characteristic of the average human ear, obtained by noting when each frequency can just be heard and just be felt, is given in Fig. 7. A pure sine-wave sound which could be varied in frequency from zero to 20,000 cycles was used in this test; the loudness of the sound was measured in terms of the pressure exerted on a flat surface. You will note that at 2000 cycles it takes about .0005\* bars of r.m.s. sound pressure to make the sound audible, and about 1000 bars (2 r.m.s. pounds per square foot) before the sound can be felt. At this frequency, then, the pressure at the threshold (the beginning) of feeling is about 2,000,000 times the pressure at the threshold of audibility. This difference in threshold values decreases for higher and lower frequencies, as you can see in Fig. 7.

**The Decibel.** You might think that a 2000-cycle sound at the threshold of feeling should seem 2,000,000 times

louder to the human ear than a 2000-cycle sound near the threshold of audibility, but this is not exactly the way in which the human ear responds. *The amount by which a pure sine-wave sound must be increased before the change in sound level can be distinguished by the average human ear is called a "decibel."* Doubling the pressure of a sound causes a 6-decibel increase in the ear's sensations. Increasing the pressure gradually to ten times its original value results in twenty sensations of increase in sound, or a 20-decibel (abbreviated 20 db) increase. Increasing the pressure one hundred times results in a 40-db increase; increasing the pressure one thousand times gives a 60-db increase; and an increase of 2,000,000 times corresponds to about 126 db. Thus, you can see that increasing the sound pressure of a 2000-cycle note 2,000,000 times gives the equivalent of 126 separate and distinguishable increases in sound, as far as the human ear is concerned.

► The foregoing statements bring out a number of important characteristics of the human ear. Notice that doubling the pressure of a sound causes a 6-decibel increase in the ear's sensation. This does not mean that doubling the pressure results in a sound six times as loud—it means that, as the pressure is gradually raised, there would be six recognizable increases in the volume. In other words, as the sound is increased gradually, there would be a certain point at which it would be possible for one to recognize that the new pressure produces more sound than the original pressure. Then, taking the new pressure as a basis, a further increase eventually would produce another recognizable volume step. There are six of these steps involved when the pressure is doubled.

This fact at once shows that the hu-

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\*The bar is a term expressing pressure per unit area; one bar equals .002 pound of pressure per square foot of surface. Since sound pressure on any material is varying continually, we must deal with the r.m.s. value of the pressure, just as we deal with r.m.s. values of current and voltage. Sound or acoustical power is proportional to the square of sound pressure, just as electrical power is proportional to the square of voltage (electrical pressure). The acoustic bar is used here; one bar is equivalent to one dyne per square centimeter when the pressure is measured in metric units.

man ear is far more sensitive to pressure changes *when the original volume is low*. In other words, doubling the sound pressure from one bar to two bars results in a 6-decibel increase, and so would a doubling of pressure from one hundred bars to two hundred bars. Therefore, if the original sound level is low, a small change in pressure will produce a recognizable increase in loudness. On the other hand, the ear becomes less sensitive as the sound pressure increases, so that at high levels large changes in pressure are necessary before the ear can detect any change in volume.

into account. The amount of power fed to the loudspeaker determines the sound pressure the speaker will be able to develop and, in turn, the apparent loudness of the sound. Therefore, it is convenient to speak of the audio power in terms of decibels rather than electrical watts. This allows the designer to see at once just how much an increase in electrical power output will increase the sound output.

Again, a reference value is necessary. Any value can be chosen arbitrarily; radio men normally use the power level of 6 milliwatts, and call this the zero level or zero db. Notice—

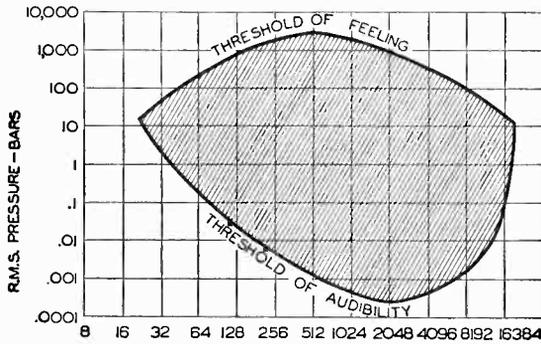


FIG. 7. The response of the average human ear to pure sine-wave sounds of various frequencies is given here. Sound waves having pressures within the shaded area can be heard. The r.m.s. pressure of the sounds is expressed here in "bars"; one bar equals .002 lb. per square inch.

Notice further that the decibel is a *unit of sound change*. It always represents the comparison between one sound level and another. Furthermore, it represents the number of steps in the increase. We cannot say that one sound is just twice as loud as another—but we can say that one sound is a number of steps louder than another.

### POWER LEVELS

Because the ear responds to sound in the peculiar way just described, the radio designer must take this factor

this is a unit used in radio when comparing electrical powers, and does not have any relationship to the acoustic reference level. In fact, we cannot say just how much acoustic power will be produced by this amount of electric power, as this depends on loudspeaker efficiencies, the acoustic treatment of the loudspeaker enclosure and a number of other factors. However, we can reasonably expect that a rise of 5 db in electrical power will produce a rise of 5 db in sound power.

From the foregoing, you can see that

the 6-milliwatt level is purely an arbitrary reference level, to which other electrical powers can be compared. Fig. 8 gives db values for various amounts of power in watts, when this reference level is used. You can see that a power level of 19 watts corresponds to about 35 db, 6 watts to 30 db, 1.9 watts to 25 db, etc.

If the power level is less than the reference value, the output is considered to be a *minus decibel* value. Thus, an output of .002 watts is about 5 db below the zero level (commonly called 5 db down, or minus 5 db, and written -5 db).

From the table, you can see that doubling the *power* results in a 3-db increase; multiplying the power by 10 causes a 10-db increase; multiplying by 100 causes a 20-db increase and so forth. Thus, the decibel increases for *power* changes are different from those caused by *pressure* changes.

► Regardless of this, the response of the human ear to sounds follows the decibel scale. As we have said, the average ear can distinguish a 1-db change in sound if the sound is a pure sine wave. If, however, the sound is complex (speech or music for example) the smallest change in level which the average ear can notice is about 3 db (corresponding to a power ratio of 2 to 1).

From this, you can see that giving a power increase or decrease in terms of decibels lets us tell at once whether the increase or decrease will be noticeable to the ear. Thus, if we consider a complex sound, we have to increase the power three decibels, or double it, before the ear is able to tell that the

sound has been increased at all. Hence, if an amplifier is feeding 6 watts to the loudspeaker, we must go to 12 watts before the increase would be noticeable. In other words, under the same conditions and for the same loudspeaker characteristics and efficiencies, it is not possible to tell the difference

WATTS	DB	WATTS	DB
.00048	-11	.006	0
.00060	-10	.008	1
.00076	-9	.010	2
.00095	-8	.012	3
.0012	-7	.015	4
.0015	-6	.019	5
.0020	-5	.024	6
.0024	-4	.030	7
.0030	-3	.038	8
.0039	-2	.047	9
.0048	-1	.060	10

WATTS	DB	WATTS	DB	WATTS	DB
.076	11	.95	22	11.9	33
.095	12	1.2	23	15.2	34
.12	13	1.5	24	19.0	35
.15	14	1.9	25	23.7	36
.19	15	2.4	26	30.4	37
.24	16	3.0	27	38.0	38
.30	17	3.8	28	47.4	39
.38	18	4.7	29	60.0	40
.47	19	6.0	30	75.9	41
.60	20	7.6	31	94.9	42
.76	21	9.5	32	119	43

FIG. 8. This table shows the relation between electrical power (in watts) and decibels, when the zero db level is set at .006 watt.

between an amplifier producing 6 watts and one producing 7 watts.

► As we will see shortly, the fact that the human ear has this peculiar characteristic is quite helpful since it means that a form of distortion known as frequency distortion is not as noticeable as it would be otherwise.

# Sound Pickups and Reproducers

We have shown that one of the important characteristics of a sound wave is that it exists as a variation in pressure. Therefore, it is obvious that a simple and direct means of converting a sound wave into an electrical current is to use a pressure-actuated pickup unit, so designed that it converts the variations in pressure into electrical current variations.

By the same reasoning, the sound reproducer must change electrical current variations into sound pressure variations.

Let us turn now to typical pickups and reproducers, and learn something of the fundamentals of their operation. At this time, we won't go into all of their characteristics — we'll just see how they operate.

## SOUND PICKUPS

There are a number of different kinds of microphones in use today, but the five most common types are shown in Fig. 9. All of them use some form of diaphragm upon which the sound pressure can act. They differ principally in the method of converting the diaphragm movement into electrical current variations.

**Carbon Microphone.** Fig. 9A gives a simple sketch of the single-button carbon microphone. Part *D*, a very thin aluminum disc or diaphragm (about .001 inch thick) is stretched over a fixed metal ring *R*. The pressure of sound waves on disc *D* moves the disc back and forth, alternately squeezing and loosening carbon particles *C* in the telescoping metal sack or button labeled *K*. Carbon itself is a resistance material, but this device depends upon *contact resistance* for its operation. That is, as the particles are squeezed

together, more of their surfaces will be in contact, so the resistance between them decreases. Then, when the particles are allowed to separate, they make poorer contact so the resistance increases. The result is that the electrical resistance between disc *D* and container *K* varies continually with the motion of the diaphragm. When this single-button microphone is placed in an electrical circuit containing a d.c. voltage *E*, the current passing through the circuit will be varied by this changing resistance which, in turn, varies in accordance with the wave form of the sound. Audio transformer *T* is placed in this circuit to transfer the variations in current to another circuit.

**Condenser Microphone.** Fig. 9B shows a simplified cross-section view of a condenser microphone. The thin aluminum disc or diaphragm marked *D*, mounted on ring *R*, is placed about .001 inch away from the fixed heavy plate *P*, thus forming a simple two-plate air condenser. These two plates are connected into a circuit containing a high voltage d.c. supply *E* and resistor *R*. Varying sound pressures change the distance between *P* and *D*, thus changing the capacity of the condenser. Changing the capacity this way results in a change in the charge stored in the condenser, so that the current through the circuit varies when the microphone picks up sound, and a varying voltage which has the same wave form as the original sound is produced across resistor *R*.

**Dynamic Microphone.** The dynamic microphone shown in Fig. 9C has a thin diaphragm *D* on which is mounted a light-weight coil of wire. This coil moves between the poles of

a permanent magnet when the action of sound moves the diaphragm. As a result, there is induced in the coil a varying voltage whose wave form is a reproduction of the wave form of the sound.

**Velocity Microphone.** The velocity microphone illustrated in Fig. 9D operates on much the same principle as the dynamic microphone, except that here sound waves move a thin crimped metal ribbon *M* back and forth through a magnetic field produced by a permanent magnet. A voltage is induced in this metal ribbon; this induced volt-

phragm *D*, so that movements of the diaphragm caused by sound waves will bend the crystals. Electrons which flow to the flat surfaces of the crystals under this bending are collected by metal collector plates *P*.

► The electrical output of microphones is very small so it is necessary to amplify this output many times. Therefore, at the transmitter, the microphone will feed into an audio amplifier designed to raise the signal power to the point needed for modulation. In public address systems, a similar amplifier is used to increase the

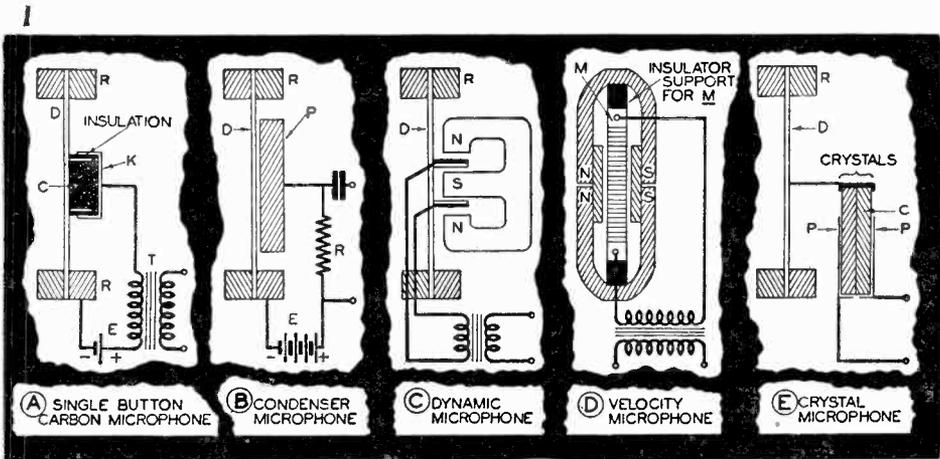


FIG. 9. Simplified diagrams illustrating the operating principles of five different types of microphones. Each converts changes in sound pressure into equivalent changes in electrical signals.

age is stepped up by the transformer.

**Crystal Microphone.** The bending or straining of a Rochelle salt crystal produces charges of opposite sign on the opposite faces of the crystal. The potential difference which exists between these charges is proportional to the strain. This principle is utilized in the crystal microphone illustrated in Fig. 9E, where two square crystals are mounted back to back to increase the electrical action. This square crystal unit *C* is clamped at three of its corners; its free corner is linked to dia-

phragm *D*, so that movements of the diaphragm caused by sound waves will bend the crystals. Electrons which flow to the flat surfaces of the crystals under this bending are collected by metal collector plates *P*.

### SOUND REPRODUCERS

Headphones and loudspeakers are the devices used for sound reproduction. Although there are a number of types of each of these, they operate on the principle of converting electrical current variations into variations in air pressure, corresponding to sound waves.

After the carrier wave has been demodulated at the receiver, there exists

a varying current which we call an a.f. current. This current is the electrical counterpart of the sound wave as it was transmitted and as it has been modified by its passage through the radio up to this point. Usually, there is enough power at the output of the average demodulator to operate a pair of headphones satisfactorily, but an audio amplifier is necessary before a loudspeaker can be operated. Thus, there is a miniature counterpart of the transmitter audio amplifier used in the receiver for the purpose of increasing the audio power to the level required by the loudspeaker.

Essentially, both headphones and loudspeakers operate somewhat on the principle of the dynamic microphone in reverse. They utilize electric signals to set into motion a diaphragm or a cone, alternately compressing and rarefying the air in front of the reproducer to reconstruct the original sound wave.

**Headphones.** In the common headphone unit, illustrated in Fig. 10A, a thin flexible steel diaphragm is placed over the two poles of a horse-shoe magnet, and a coil having many turns of insulated wire is placed around each leg of this magnet. The audio current flowing through the two coils alternately increases and decreases the attraction which the permanent magnet has on the diaphragm, causing the diaphragm to move in and out, thus producing sound waves.

**Magnetic Loudspeaker.** If a high sound output is wanted, the reproducer must be able to displace a large amount of air, and so must have a large diaphragm or cone which can be moved appreciable distances. The balanced armature electromagnetic reproducer shown in Fig. 10B has been used widely for this purpose. In this loudspeaker, a soft steel armature is pivoted be-

tween two sets of *N* and *S* poles which are parts of a powerful permanent magnet. Also surrounding the armature is a solenoid or coil which carries the audio current. This coil makes the ends of the armature alternately of op-

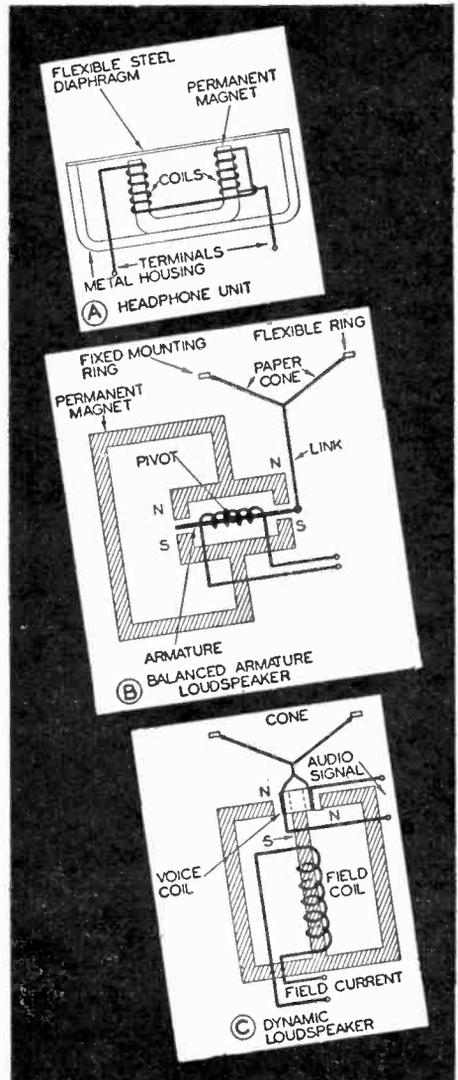


FIG. 10. Simplified diagrams illustrating the operating principles of three different types of sound reproducers. Each converts electrical energy into mechanical energy (motion), which in turn is used to produce sound waves.

posite polarity. The ends of the armature, therefore, alternately move towards and away from the permanent magnet poles. This movement is mechanically relayed to the large paper cone so that the paper cone is pushed in and out, setting the surrounding air into vibration and producing sound which is a reproduction of the original wave form. The outer edge of the cone is designed in such a way that it can be attached to its ring-shaped supporting frame without preventing free movement of the cone in and out under the action of the armature.

**Dynamic Loudspeaker.** The dynamic loudspeaker shown in Fig. 10C is the most widely used sound reproducer, for it is capable of delivering high sound outputs when the cone is used with a baffle.\* Here, a coil of wire is wound on a thin bakelite or paper tube which is attached directly to a paper cone (as shown) or to a metal diaphragm (not shown). The audio signal passing through this coil causes a varying magnetic field. This field in-

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\*A baffle is a large flat surface, a cabinet, or a horn-shaped enclosure which serves to prevent waves from the back of the speaker cone from interfering with those from the front, and also acts to "load" the cone by coupling it to a larger volume of air.

teracts with a fixed field which is produced either by a permanent magnet or by an electro-magnet. The result is that the coil is forced to move in and out of the fixed field, driving the cone and thus moving the air ahead of it.

The amount of movement depends on the field strengths. Hence, the "voice" coil is fed with considerable audio power, and the fixed field is made strong by making the separation between the north and south poles of the magnet as small as possible.

A dynamic loudspeaker can handle from 3 to 50 watts of audio signal power (depending upon its size), while a magnetic loudspeaker can handle from 1 to 3 watts. A headphone unit can handle only about 50 milliwatts.

► You are going to study loudspeakers and headphones in much greater detail in later lessons. However, you can see now that the purpose of the audio amplifier is to increase the power up to the level necessary to operate the loudspeaker. It would appear that the amplifier at both the receiver and the transmitter should be rather simple in its design and operation. Still, this is not altogether true, as this amplifier is largely responsible for the fidelity with which the electrical sound signal is handled.

# Requirements of an Audio Amplifier

All amplifiers may be divided into two broad groups, high-frequency amplifiers and low-frequency amplifiers. High-frequency amplifiers handle the r.f. carrier signals (which may or may not be modulated), while low-frequency amplifiers handle only the intelligence signals. In this lesson we will learn something about the requirements placed on low-frequency amplifiers. Let's cover first the low-frequency amplifier used in sound radio receivers.

*The low-frequency amplifier in a sound radio receiver is called an audio amplifier because the frequencies it handles correspond to audio or sound frequencies. As we have previously mentioned, the audio amplifier is the section in a radio receiver following the second detector or demodulator. A corresponding amplifier is used in the transmitter between the sound pickup device and the modulator, and similar amplifiers are used in public address systems, electronic musical instruments, electrical phonograph amplifiers, and many photoelectric control devices.*

The basic requirement of any low-frequency amplifier is, of course, to increase the signal level. We apply a signal to the grid and obtain a change in the plate current. In turn, this plate current flow through the load produces a varying load voltage. This load voltage variation is larger than the grid voltage variation, provided that the value of the load has been chosen properly.

► We now have our amplified signal, but is it *exactly* like the original grid signal? If we have a sine-wave signal, we want to have an enlarged sine-wave signal across the load—if we start with

a complex signal, we want the output signal to be exactly like it. However, if the output wave is different in any manner from the input signal (except for being amplified), then the signal is being distorted. The sounds we obtain from the loudspeaker will not be exactly like those originating in the broadcasting studio when the amplifier (or the speaker, for that matter) introduces harmonics which were not in the original signal, or when the relative amplitudes of any of the harmonics in the signal are altered.

From this, the manner in which the signal level is increased is of importance, so we must see how it is possible for an amplifier to distort the audio signal. Before studying the three forms of distortion, let's see how complex waves are formed, so that we can see just what may happen to the signal within an amplifier.

## WAVE FORMS

Any wave form differing from a pure sine wave contains one or more harmonics. For example, in Fig. 11, the wave at *C* can be obtained by combining the fundamental frequency *A* with its second harmonic *B*. You can check this fact by combining the amplitudes along the dotted lines. Thus, adding the distance 1-2 of the fundamental (the distance from the curve to the line *R-R*) to 3-4 of the harmonic gives 5-6 on the resultant.

The distance 9-10 on the second harmonic must be subtracted from the distance 7-8 on the fundamental to get the resultant 11-12, because the second harmonic is going through the opposite phase to that of the fundamental wave at this particular moment. Thus, you can see that, when both waves are

on the same side of the reference lines  $R-R$ , their amplitudes add. If they are on opposite sides, the smaller is subtracted from the larger to get the resultant. Incidentally, the resultant wave will be changed in shape if the amplitudes of the fundamental and the harmonic are changed.

► In Fig. 12, we show the result of adding a third harmonic to its fundamental. Again the same procedure is used in combining the fundamental wave  $A$  to the third harmonic  $B$  to get the resultant wave  $C$ . Thus, adding the distance 1-2 to 3-4 gives 5-6. Subtracting 9-10 from 7-8 gives 11-12.

► When other harmonics are added, the same procedures are followed. As shown in Fig. 13, adding a number of harmonics gives another and entirely

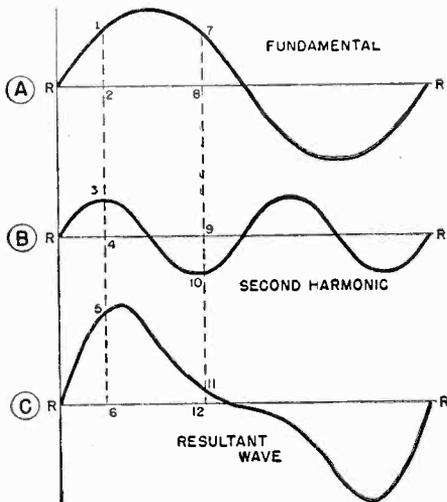


FIG. 11. Here is the result of adding a second harmonic and a fundamental wave. Any wave which is not a sine wave consists of a fundamental and one or more harmonics.

different resultant wave shape. Again the same procedure is followed in that the amplitudes are added and subtracted to give the resultant. Thus, along the dotted line, the amplitude 1-2 is measured, the distance 3-4 is subtracted, the distance 5-6 is added,

the distance 7-8 is zero, leaving the resultant, 9-10.

► From the foregoing, you see that there is an infinite variety of resultant wave forms possible, depending upon the harmonics which are added to the fundamental and upon the amplitudes of the harmonics. (In addition, as we shall learn later, the phase of the harmonics will change the resultant wave shape.)

However, regardless of the shape of the resultant wave, we can always say that any complex wave is a combination of simple sine wave forms. This is the reason that we can study the operation of amplifiers by using simple sine waves—the complex waves which the amplifier must handle are made up from these sine waves.

► As a summary, we know that:

1. Any complex sound or signal consists of a fundamental frequency plus harmonics.
2. If the relative amplitudes of any of these components are altered, we have distortion.
3. If any components are added, which were not in the original signal, we have distortion.

We can now go on to study the kinds of distortion and their relative effects. We will see that practical amplifiers are far from ideal.

## FREQUENCY DISTORTION

If one section or stage of a radio *strengthens certain frequencies more than other frequencies, or removes some frequencies entirely*, the resulting sound will not be like the original. This distortion is known as *frequency distortion*, because certain frequencies are favored more than others.

As an example, suppose we have a 1000-cycle fundamental note with a fifth harmonic at 5000 cycles. If the amplifier responds better to 5000 cycles

than it does to 1000 cycles, the amplitude of the fifth harmonic will be increased more than its fundamental is, which means that the resultant wave shape will be changed. On the other hand, if the amplifier has a lower response at 5000 cycles than at 1000 cycles, then the fifth harmonic amplitude will be reduced, which again means that the resultant wave shape is changed.

You can see that frequency distortion is produced because an amplifier does not have the same gain for all the frequencies it is supposed to handle.

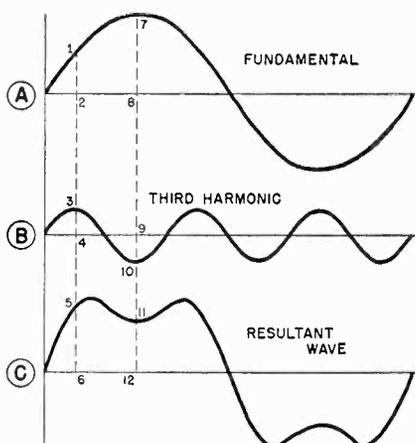


FIG. 12. Adding a third harmonic to a fundamental produces the resultant shown here. Of course, the actual shape of the resultant depends upon the amplitudes of the fundamental and harmonic waves. However, the general shape is correct, and should be compared with the resultant shown in Fig. 11. Notice that odd harmonics (the third, fifth, etc.) produce a resultant with both halves exactly alike, while even harmonics (second, fourth, etc.) produce a resultant which does not have symmetrical halves.

This type of distortion is caused by the circuit parts (coils, condensers and transformers) associated with the vacuum tube in the amplifier stage.

► Earlier, we mentioned that the ear does not notice a power change of less than about 3 db (power ratio of about 2 to 1) when complex sounds are be-

ing heard, which means that frequency distortion is less troublesome than would appear at first glance. It is quite permissible to have radio apparatus amplify some frequencies almost twice as much as others, or to convert some frequencies into sound twice as well as others, for the human ear is not able to notice this amount of distortion. This characteristic of the human ear is very much appreciated by the radio designer, since it is quite difficult for him to make radio apparatus amplify very low and very high frequencies equally as well as in-between frequencies.

Fig. 14 shows a typical response curve for an audio amplifier. Notice that the scale at the left is in decibels, because this permits us to compare the response to the hearing capabilities of the ear.

► When making a curve of this kind, the power output at some reference frequency (usually 1000 cycles) is taken as the standard or reference value. Then, the power output at other frequencies is found. If the amount of output is less than that at the reference frequency, then the db output is less, while greater outputs give increases in the decibel level.

As you will notice from the graph in Fig. 14, the curve is within 3 db of the reference level (24 db) from about 80 cycles to above 4000 cycles. At 5000 cycles there is a peak in the response which is nearly 6 decibels higher than the reference, so the output at frequencies about this value would be noticeably greater.

Above 8000 cycles, the response cuts off sharply, so frequencies above this point are poorly reproduced, if at all.

► When the very high frequencies are cut off by radio apparatus, sound loses some of its fidelity of reproduction. However, not all the high frequencies

up to 20,000 cycles are necessary for understandability or appreciation of music. You can cut off sounds at about 8500 cycles and still get high quality speech and music. Why reproduce frequencies up around 15,000 to 20,000 cycles, when only very young people and trained musicians are able, as a rule, to hear these frequencies at all?

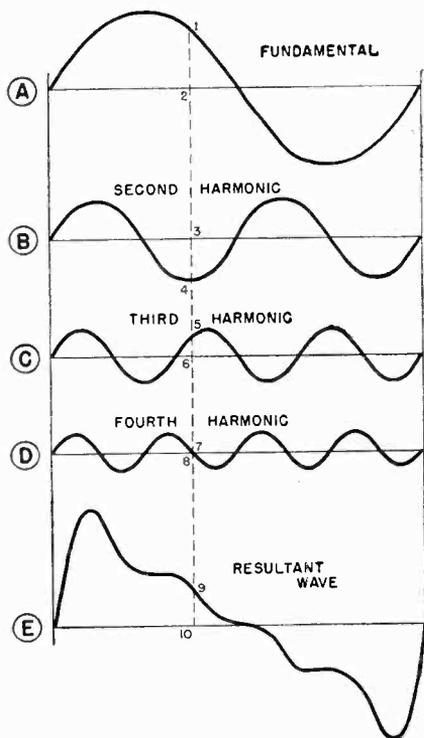


FIG. 13. When a number of harmonics are added, the resultant wave shape depends upon whether the even or the odd ones predominate, and upon the amplitudes of all of them.

With age, one's hearing ability decreases so that elderly people cannot even hear frequencies above 6000 cycles in most cases. Therefore, for all practical purposes, it is not necessary to reproduce these high frequencies.

► Most of the power needed to reproduce music is used to reproduce low-frequency notes, and, since the ear is not sensitive to the lows and highs (low

and high frequencies), the reproduction of sound at a volume much below that of the original makes the music sound unreal. It is for this reason that the so-called low (or bass) boosters are included in radio receivers to amplify excessively these lower frequencies and get effects which sound more natural to the ear. Some of these boosters are automatic compensators, while others are tone controls, adjustable at the will of the customer. Thus, it is quite possible that the customer will even introduce a certain amount of frequency distortion, in order to make the sounds more natural to his ears!

Of course, if too many of the high or low frequencies are cut off, then the frequency distortion becomes objectionable.

### ✓ AMPLITUDE DISTORTION

When a radio circuit or device does not produce current or voltage changes which are *exactly proportional*, at each instant of time, to the changes occurring in the voltage or current values of the incoming signal, we have what is known as *amplitude distortion*.

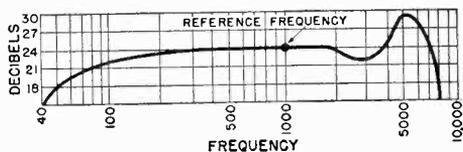


FIG. 14. Here is a typical response curve for an audio amplifier, showing the output over its frequency range. The output is given in db above the .006 watt zero level, so that variations in the response can be judged in terms of their effects on the human ear. Using the db table in Fig. 8, practice converting the db values into watts. Thus, 24 db is equal to 1.5 watts, etc. (Sometimes you will find that a similar curve is used, but the reference frequency output is labelled 0 db. Then values above are written as +db, while those lower are written as -db. That is, when 24 db is called 0, 27 db becomes +3, 30 becomes +6, 21 becomes -3, and 18 becomes -6 db. This allows us to consider the output in terms of the db change above or below the reference level.)

If the original wave is a sine wave, then amplitude distortion will result in the production of harmonics which were not present in the original signal. If the original signal is of a complex nature, additional harmonics are added, or the relative amplitudes of the harmonics are changed by amplitude distortion.

Fig. 15 shows a typical example of amplitude distortion. At A, we have a sine-wave signal. We want the amplifier to increase the amplitude of this signal, as shown at B, without changing its wave shape. However, if we get the amplified signal with the wave shape changed as shown at C, then we have amplitude distortion. As you can see, this wave is not an exact replica

that *even* harmonics, such as the second, fourth, sixth, etc., have been added to the fundamental, as shown in Fig. 17. In other words, the tube's action is approaching that of a half-wave rectifier. From your knowledge of rectifiers, you will remember that this wave form has a d.c. component as shown in Fig. 17. This d.c. current will be added to the normal d.c. current value. This means that, when a signal is applied to an amplifier tube operating on the lower bend of its characteristic, the resulting distortion will cause the d.c. plate current to rise. This is one of the means whereby a serviceman can determine when distortion occurs—he can apply a sine-wave signal and measure the d.c. plate

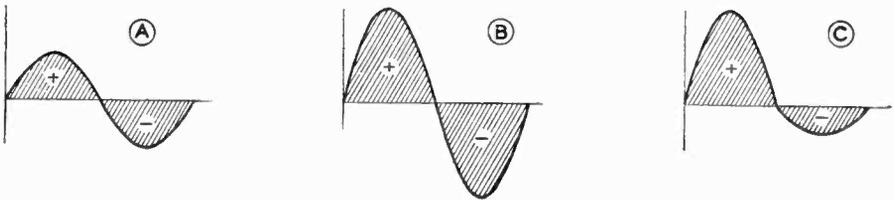


FIG. 15. A typical sine wave is shown at A. When it is amplified, it looks like that shown at B. However, should amplitude distortion occur, it may look like the wave shown at C.

of the original sine-wave signal.

► What could produce this change in wave form? One of the most common sources of amplitude distortion is the operation of a class A amplifier over a curved tube characteristic, as shown in Fig. 16. Notice that the input signal  $e_g$  has more control over the plate current on its positive swing than it does on its negative swing. This results in an output signal (produced by plate current  $i_p$ ) which is not a sine wave—or rather, it is a pure sine wave plus harmonics. Thus, although we started with a sine-wave signal on the grid, we are obtaining a sine wave plus harmonics in the plate circuit.

Analyzing the wave form of the output signal shown in Fig. 16, we find

current before and after the signal is applied. If the stage is a class A amplifier, there should be no change in the d.c. plate current. If there is a change, distortion is taking place.

Should a complex wave be applied to the amplifier having the characteristic shown in Fig. 16, even harmonics will be added to the wave if they are not already present, or the amplitudes of any even harmonics which are present will be increased by this distortion.

► Going further, a similar distortion can occur if the grid is allowed to swing positive. As shown in Fig. 18, there is an upper bend in a tube characteristic; this will also produce distortion. The output wave is like that of Fig. 16 except that it is reversed, or

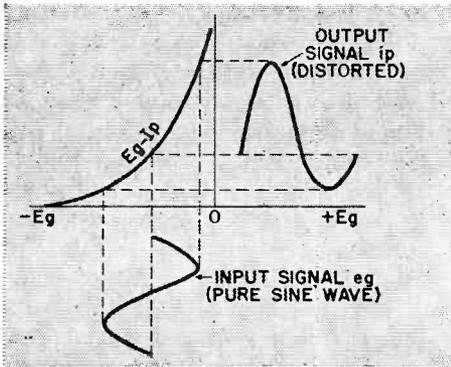


FIG. 16. Amplitude distortion is produced whenever the  $E_g-I_p$  curve of a tube is not linear (straight) in the operating region.

is upside down. It has the same components as are shown in Fig. 17, except that they are shifted in phase  $180^\circ$ . The negative (downward) swings are the larger, so the d.c. component will be of the opposite polarity to that shown in Fig. 17. In this case, there will be a reduction in plate current when the signal is applied.

► Therefore, if the d.c. plate current changes when a signal is applied to a class A stage, distortion is occurring. If the d.c. plate current *rises*, the tube is operating on the lower bend of its characteristic. (This operation is caused by excessive bias or low plate supply voltage.) If the d.c. plate current *falls*, the tube is operating on the upper bend in its characteristic. (Operation here is caused by too little bias or by an excessive plate supply voltage.) Notice the value of *knowledge* to the serviceman. Upon observing that distortion is occurring, he can take a meter reading and, *from his knowledge of circuit actions*, he can reason directly to the probable causes of the trouble!

► There is another kind of amplitude distortion in which *odd* harmonics are added instead of *even* harmonics. This distortion will be produced when the

signal applied to the grid is excessive, so that operation occurs over *both* tube curvatures as shown in Fig. 19. Here, the shape of the output wave indicates that the third, fifth, seventh, and other odd harmonics are being added to the fundamental. This distortion is very noticeable to the ear, but it will not cause a change in the d.c. plate current if both halves of the output wave are exactly alike. The serviceman must use a cathode ray oscilloscope to actually "see" the wave form before he can find the stage producing this diffi-

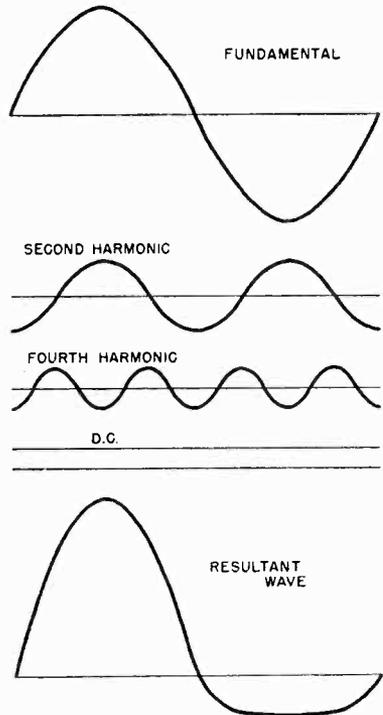


FIG. 17. Here is an analysis of the wave produced by the  $E_g-I_p$  curvature which was shown in Fig. 16. Notice that even harmonics are added, plus a d.c. component. Although we have shown only the second and fourth harmonics, there are actually small amounts of other even harmonics added. A phase shift is also involved in that the harmonics are added  $90^\circ$  out of phase with the fundamental. This phase shift cannot be heard, but the added harmonics are noticeable. (In television, the phase shift is of importance also.)

culty, unless there are other clues in addition to the distortion.

► Incidentally, it is possible to tell from the wave shape whether the distortion is an even or an odd harmonic addition. If the two halves of the resultant wave are *exactly alike*, only odd harmonics have been added. If two halves are not symmetrical, however, then even harmonics have been added. Where both even and odd are added, it is not so easy to determine this unless one or the other happens to predominate.

Amplitude distortion is readily detected by the ear. Strangely, the ear seems more critical of odd harmonic distortion than it is of even harmonic distortion. An odd harmonic distortion causing a change in harmonic levels of more than 5% begins to be noticeable,

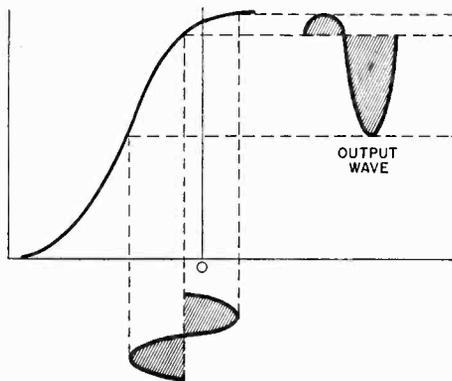


FIG. 18. Amplitude distortion will occur also if the tube is allowed to operate at the upper bend of its characteristic.

while an even harmonic distortion of as much as 10% is relatively unnoticeable.

## PHASE DISTORTION

When a radio circuit or part changes the phase relationship between different frequencies in the signal, we have phase distortion. The distance (phase) between peaks of the different signal

components is changed, with the result that there is a change or distortion in the wave form of the output signal. (Later in this lesson we will show an example of phase distortion.)

The human ear is relatively unconscious of even large amounts of phase distortion, being far more critical of amplitude or frequency distortion. The reason for this is that the ear hears the individual components of the wave rather than the resultant wave. That is, the sound wave is "taken apart" by the ear. Hence, the ear is quite conscious of added or deleted frequencies, and of changes in amplitude. It cannot detect changes in *time* in the arrival until such change approaches the time length of a syllable or of a musical note.

However, phase distortion is far more important in television. As we shall learn later in this lesson, the time delay caused by even a small phase shift in a television signal will be noticed readily by the eye.

## AUDIO AMPLIFIER REQUIREMENTS

From the foregoing, we can begin to see just what the requirements are for an audio frequency amplifier. In order, we may well list them as follows:

**Voltage Amplification.** A certain amount of voltage amplification is necessary in all radio receivers, as the voltage output of the average second detector is too low to operate the power output stage. Therefore, one or more voltage amplifying stages are used between the second detector and the power stage.

**Power Amplification.** Raising the voltage level is not enough—*power* is required to operate a loudspeaker. Therefore, the output stage of the audio amplifier in a radio receiver will

always be a power stage, designed to deliver sufficient power to operate the loudspeaker. Actually, this stage can be said to be converting signal voltages into high signal powers.

As you have learned in earlier lessons, one of the important differences between power stages and voltage amplifying stages lies in the choice of the

In public address systems, where large amounts of signal power are required, medium power stages may be used to drive high power stages. Similarly, in transmitters, the power levels may be quite high. In these two instances, true power amplification does occur, for the final power stage is actually *driven* by a low power stage.

**Distortion.** From the foregoing, you can see that every effort must be made to cause tubes to operate on the straight-line portion of their characteristic curves. Inductance and capacity values must be carefully chosen and controlled to prevent the undesirable formation of low-pass and high-pass filters which would tend to limit the frequency response. Even a matter like the choice of the load value is a compromise between the amount of amplification desired and the fidelity of response wanted.

► In your next lesson, you will learn much more about these important requirements and about how they affect the serviceman and his choice of replacement parts.

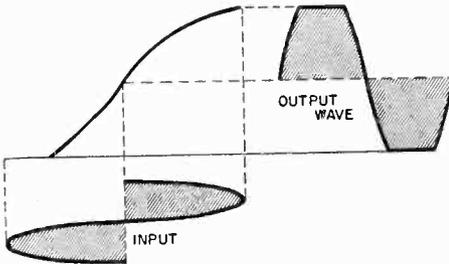


FIG. 19. When the input signal is allowed to become so large that it exceeds the normal operating region, both halves of the output wave will be distorted. This shows that odd harmonics have been added. Strangely, this distortion is far more noticeable than is even harmonic distortion.

load values. However, there are other important differences as we shall soon learn.

# The Fundamentals of Television

Television involves the transmission of intelligence which reaches our brain through our eyes. First, let us consider what the eye sees when it looks at an object. Ordinarily, it looks at reflected light, made up of electromagnetic waves; occasionally, it looks directly at light sources such as electric lamps, a fire, or the sun. The eye sees color because the electromagnetic waves in the visual band have different frequencies, each frequency or group of frequencies giving, through the action of the brain, a color sensation. The human eye serves as a complicated lens (much like the lens in a camera), for it projects these electromagnetic waves on the *retina*, a surface at the back part of the eye. This retina is composed of millions of nerve ends, each of which is connected to the brain. These nerve ends interpret the strength of each electromagnetic wave which hits them (determined by the brightness of the object) and they also interpret the frequency of the wave (the color of the object). *Each nerve end "sees" only a tiny portion of the entire scene*; the brain reconstructs the over-all picture by assembling all the nerve impulses. Thus, the eye breaks up the scene into elements, each of which is transmitted over a separate nerve channel to the brain.

One scientist calls the human eye nature's own television system. The object viewed acts as the transmitter in the system, sending out electromagnetic waves which are picked up by the eye acting as a receiver, and then are relayed to the brain to give us the sensation of seeing.

**A Suggested Television System.** This action of our visual mechanism immediately suggests a method of con-

structing a television system. Why not arrange thousands or millions of tiny electric eyes on a screen to pick up the light waves, and connect these by thousands of wires or radio frequency transmitters to a receiver containing thousands of tiny glow lamps? Each of these would reproduce the amount of light picked up by its corresponding electric eye, so the combination of all the lamps would reproduce the object viewed by the transmitter. Yes, a television system like this has actually been tried for land wire television, but only on a small scale. The scheme was found to work after a fashion, but obviously was far from practical, as entirely too many wires were necessary.

**Practical Television Systems.** The television systems in use today do not attempt to pick up a complete scene and transmit it to a receiver all at once. Instead, television takes advantage of an eye characteristic known as *persistence of vision*—the ability of the eye to retain an impression of an object for a short time after the object has disappeared from view. This makes it possible to send a portion of a scene at a time, so long as the entire scene is transmitted before the eye "forgets" the first of it.

The scene is broken up into elements by *scanning* or by viewing a small portion at a time. Scanning is an operation very like what you do when you read this page. You don't look at the page and attempt to read every word at once. Instead, you read the first line from left to right, swinging quickly back to the left-hand side of the second line, read the second line, go back to the start of the third, and repeat the process until you have taken in every word by itself.

That is just about what a television "camera" does. (This camera is the pickup device of a television system, corresponding to the microphone in a radio system.) In effect, an "eye" in the camera travels over the top edge of the scene from left to right, swings quickly back to the left-hand side, moves down slightly, travels horizontally over the scene again, and repeats the process until the whole scene has been scanned. As you no doubt know, or have guessed, this "eye" is really a light-sensitive surface which converts the light received from the scene into an electric current. This current, which of course varies as different parts of the scene come into view of the scanning eye, is then transmitted by radio

produced when a scene is scanned. Suppose we wish to televise a picture like that in Fig. 20A. After it has been scanned by the camera, transmitted to the receiver, and reproduced on the receiver screen, it will have something of the appearance shown in Figs. 20B and 20C. That is, it will consist of a series of lines; these lines will vary in brightness along their length, and so make up the picture we see. The more lines we have in a given area, the greater the detail of the final picture. Fig. 20C, which has 120 lines, exhibits more detail than Fig. 20B, which has only 60.

Note that, as you move the illustrations in Fig. 20 farther and farther away from you, a point is reached for each illustration where the details seem to blend into a complete and nearly perfect reproduction of the original. This brings out an important fact about television: if a reproduced picture is made larger without increasing the number of lines, the picture will have to be viewed from a greater distance to get a satisfactory eye impression.

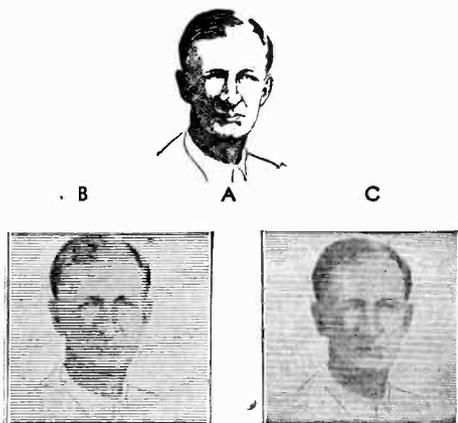


FIG. 20. The drawing at A is reproduced as a series of lines at B and C. Greater detail is obtained by using more lines, as at C.

to the receiver. At the receiver, the process is reversed, and the original scene is traced out line by line.

This is a highly simplified version of how a television system works, but it will serve to show you the basic idea of operation. Right now, the important fact for you to grasp is that a scene is televised "bit by bit," not as a whole.

► Fig. 20 illustrates the general effect

## HOW SCENES ARE SCANNED AND REPRODUCED

Before considering the technical details of breaking up a scene into a number of lines, it will be valuable to get clearer ideas of how a scene is taken apart or *scanned*, and how a scene is reproduced. Only the basic schemes will be considered; naturally, different television experts have different ways of accomplishing the desired results.

### Mechanical Scanning Methods.

Even though mechanical methods of scanning are considered inadequate today, we will consider them first since they are easier to understand. Punch a hole in the center of a small business card with a pin and hold the card up to one of your eyes, so you can look

through the hole. Turn to some object or scene. Notice that you can see only a small part of this scene through the tiny hole. Now move the card horizontally from left to right; you see all the portions of the scene along the line which you are scanning. Move the card back and forth horizontally while shifting it vertically downward a lit-

plete scanning of the entire picture, for each hole on the disc scans one line. If the disc is revolved fast enough, the visual sensation is the same as if the entire picture were being seen at one time.

The exact arrangement of the holes on the scanning disc is shown more clearly in Fig. 21B. The observer views the scene through the mask, a rectangular opening in a piece of black cardboard. As the disc is rotated, each hole moves across the opening in this mask, the outermost hole in the spiral moving across the top of the opening and each succeeding hole moving across the top line down. Finally, when the innermost hole has moved across the bottom of the opening, the outermost hole again scans the top line and the entire scanning process starts over again.

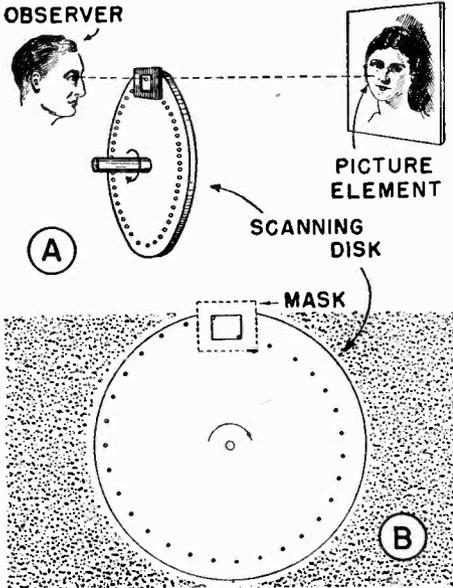


FIG. 21. This diagram shows an elementary mechanical scanning disc system. If the disc is rotated rapidly enough, the observer will be unconscious of its presence, as persistence of vision will allow him to apparently see the entire scene, although he is actually viewing only a tiny spot at a time.

tle at the end of each line, and your eye will scan the entire scene, piece by piece.

**The Scanning Disc.** In place of this crude scanning device, we can use the system shown in Fig. 21A, in which a large number of holes are arranged in a spiral fashion on a rotating disc called the *scanning disc*. This disc really replaces the business card used in our previous example: one complete revolution of the disc gives one com-

**Mechanical Television Transmitters.** If the observer in Fig. 21A is replaced with a light-sensitive cell, this cell will deliver a varying electric current which is at all times proportional to the amount of light reaching the cell, and therefore proportional to the shade of lightness or darkness of the element of the picture being scanned at a particular instant. This arrangement gives us a means of converting a picture or scene into a varying electrical current. This current or picture signal can be amplified and placed on a radio carrier for transmission through space. At the receiver, the carrier can be demodulated and the picture signal amplified sufficiently to operate a picture reproducer.

**Mechanical Television Receivers.** In the early television receivers, the amplified picture signal was fed to a neon glow lamp like that shown in Fig. 22A. This lamp consisted of a wire anode and a rectangular flat metal piece (the same size as the reproduced picture) which served as a cathode.

These elements were enclosed in a gas-filled glass envelope. A red glow of light formed on the plate when sufficient voltage was applied between the electrodes; the intensity of this glow varied with the applied voltage. The amplified picture signal was made to change the applied voltage, thus changing the intensity of the glow.

A pin-hole scanning disc was rotated before the glow lamp in such a way that the holes scanned the glowing plate. The transmitter and the receiver were so synchronized that when the scanning disc at the transmitter

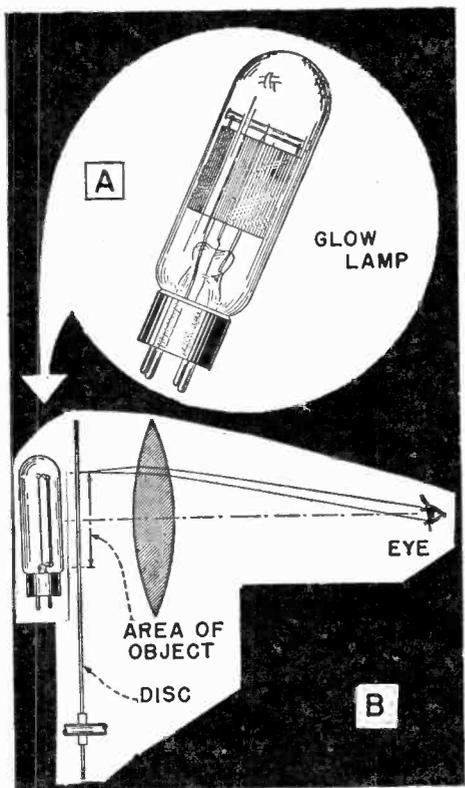


FIG. 22. An early type of mechanical television reproducer. The glow-lamp light depended on the brilliancy of the spot being scanned at the transmitter at that moment. The scanning disc is synchronized (in step) with the transmitter disc so that it arranges the light elements in their proper sequence.



Courtesy Don Lee Broadcasting System

The engineer is holding an electronic pickup tube such as is used in television studios. Scanning is accomplished within this tube by electronic means.

started to scan the top line of the scene, the receiver scanning-disc likewise started to scan the top line. Line by line the scanning discs were kept in step or in synchronism, so that the intensity of the glow lamp at any instant corresponded to the intensity of the light reflected from that same element on the actual scene. The arrangement of the scanning disc and glow lamp are shown in Fig. 22B. The lens shown is a magnifying glass used to enlarge the image to three or four times the size of the glow lamp plate.

**Electronic Television Transmitters.** Although present day methods of scanning and picture reconstruction differ greatly from the method just described, the principle of breaking up a picture into a number of elements which are scanned line after line is still

used. Figure 23 illustrates a modern electronic television camera. The scene is focused on the photoelectric plate by a high-grade camera lens combination. This light-sensitive photoelectric plate consists of millions of tiny light-sensitive spots, each insulated from the others and each scarcely larger than the point of a pin. Under a microscope this plate looks as if it were covered with grains of sand.

When a scene is projected on the photoelectric plate by the lens, the ac-

amount of light reaching that section. Thus, some spots on the plate are more positively charged than others, and we actually have an electronic image of the scene. An electron gun now shoots a fine stream of electrons at the photoelectric plate. Electromagnetic deflecting coils (here designated as "deflecting yoke") shift this electron beam horizontally and vertically, one line at a time, to scan the entire photoelectric plate from top to bottom. When this electron stream strikes a positively

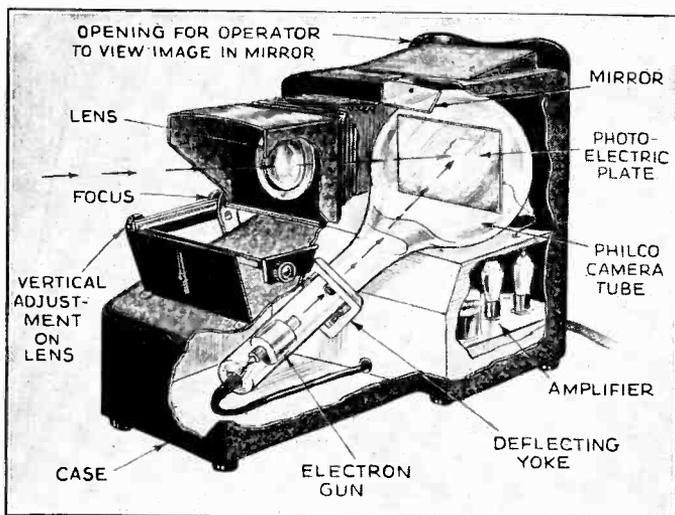


FIG. 23. A cut-away sketch showing the arrangement of parts inside one type of electronic television camera. *Courtesy Radio-Craft*

tion of light drives out electrons from each of the tiny light-sensitive units. These electrons pass through the space in the tube to a conducting surface on the inside of the glass envelope, which is at a high positive voltage and therefore attracts the electrons. The action of light thus leaves the photoelectric plate elements more or less positively charged (because they have lost electrons).

Naturally, the amount of electrons lost from any given section of this photoelectric plate depends upon the

charged surface, that surface recovers its electrons and, in so doing, relays the charge to a flat metal supporting electrode which is back of, but insulated from, the photoelectric plate.

In this manner, an electronic impulse is relayed from each spot which is hit by the electron beam. The size of each impulse corresponds to the amount of light striking the spot, so the sum of all the impulses (sent one at a time) constitutes a picture signal. The supporting electrode collects the picture signal and, after a great deal

of amplification, the picture signal is placed on a carrier wave and transmitted through space, just as in the mechanical television system. In addition, impulses are sent at the beginning of each "frame" or new picture, to keep the image-reconstructing devices in step with the scanning mechanism at the transmitter.

### Electronic Television Receivers.

Figure 24 shows a simplified diagram of a typical electronic picture reconstructor. This employs an electron gun and two sets of electromagnetic deflecting coils. Special oscillators generate the current pulses which flow through these coils; the oscillators are controlled by the synchronizing impulses sent out by the transmitter. A spot of light appears on the special screen at the end of the tube when it is hit by the electron beam produced by the electron gun; the brilliance of the spot increases with the speed of the electrons in the beam and with the number of electrons in the beam.

The picture signal voltage controls the speed and number of the electrons in the beam by means of a special grid electrode, and the deflecting coils control the scanning of the beam across, and up and down, the screen. The combined action is such that while the beam is sweeping across the screen, its intensity is changing continually in accordance with the picture signal, and the effect of "painting" light on the screen is secured. The picture size is controlled here by the size of the screen; a 12-inch diameter tube gives a  $7.5" \times 10"$  picture.

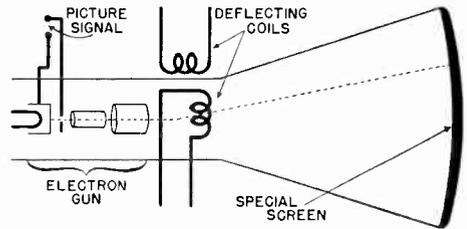


FIG. 24. A simplified diagram of an electronic picture reconstructor tube.

# Video Amplifier Requirements

Television, like sound systems, requires both low- and high-frequency amplifiers. However, the requirements of both are far more exacting than those on the corresponding sound amplifiers. Basically, there must be voltage and power amplification, but the very wide frequency band which must be handled with a minimum of distortion requires special tubes and unique circuits, as we shall point out in later lessons.

## VIDEO FREQUENCIES

The low-frequency television amplifier (called a video amplifier) appears almost misnamed, as it must handle frequencies which are well up in the r.f. range of sound systems. However, as in sound systems, the video signal is only that which was removed from the carrier by the demodulator; the television carrier is far higher in frequency.

First, let's see just what range of frequencies must be handled by the video amplifier. To begin with, we want a picture with the greatest possible amount of detail. People are familiar with the details of photographs and the movies and naturally expect television to be as good. As technical problems are solved, details become better, so that today the fidelity is acceptable. Even so, still greater fidelity is being sought.

► Another problem is *flicker*. To give the sensation of motion, the scene must be scanned over and over, with one scanning rapidly replacing another, just as in the movies. Each scanning is called a "frame" and, the more frames there are per second, the less chance there is for the eye to see them individually—they blend together bet-

ter. If too few are transmitted, a flickering of the image becomes noticeable.

► Greater picture detail can be obtained by increasing the number of lines per frame; increasing the number of frames per second gives less flicker of the reproduced picture. Both of these factors contribute to what is called high-definition (or high-fidelity) reproduction. However, there are definite limits to the number of lines and frames which can be handled.

**Picture Elements.** The maximum frequency of the picture signal current is of great interest to every television engineer, for all television equipment must be designed to handle this frequency. In order to have a way to calculate the maximum frequency of the picture signal current, it is assumed that the picture being scanned consists of a checker-board pattern of black and white squares, with each square being equal in size to one of the sensitized spots on the photoelectric plate of the television camera. (Since each of these sensitized spots is the smallest part, or element, that the camera can "see" of a scene, these spots are usually called "picture elements.")

The signal current is said to go through one cycle each time the electron scanning beam passes over one light and one dark picture element, because the signal current goes through a maximum and a minimum value each time this happens. For our checker-board scene, the picture signal current goes through a cycle each time the scanning beam passes over two consecutive picture elements. To find the maximum frequency of the picture signal current, then, all we have to do is compute the number of picture elements scanned per second and divide

by two (since it takes two elements to make one cycle). Let's go through the simple computations involved and see just how high this maximum frequency may be.

If each picture element is considered to be as high as it is wide, it is easy to compute the number of elements in one complete picture. Assuming a square picture with  $N$  lines, then there will be  $N$  picture elements per line, or  $N$  times  $N$  picture elements in the complete square picture, which is known technically as *one frame*. For example, at present the television standards call for a 525-line picture. Hence, in a square picture there will be 525 times 525 or 275,625 picture elements. For ordinary calculations, 276,000 elements will be sufficiently accurate.

**Aspect Ratio.** The pictures commonly involved in television are not square, however; they are wider than they are high. The width of a picture divided by its height is called the *aspect ratio*. In order to conform to motion picture standards, the aspect ratio has been standardized at 4/3 or 1.33. This means that the number of elements in each line has been increased by the aspect ratio, which we will designate as  $a$ . Now the number of picture elements per frame or picture will be  $N$  times  $N$  times  $a$ . For the example just considered, the total number of elements will therefore be 276,000 times 4/3 or 368,000.

**Frame Frequency.** The number of pictures sent per second is the *frame* (or picture) *frequency*; let us designate it as  $F$ . By multiplying the number of picture elements in a frame by the frame frequency, we get the total number of picture elements per second. The total number of picture elements per second is then  $N \times N \times a \times F$ . Since it takes two picture ele-

ments to make a cycle, we get the maximum number of cycles per second by dividing the preceding formula by 2. The standard frame frequency is 30; in our example, then, we get the maximum frequency involved by multiplying 368,000 by 30 and then dividing by 2; the result is 5,520,000 cycles per second.\*

**Factors Affecting Frequency.** In actual television practice, the picture elements are being scanned only about 85% of the time. The remainder of the time is used for sending line synchronizing impulses. This fact increases our maximum picture frequency because we must crowd our picture elements into 85% (85/100) of a second. We must, therefore, multiply our computed value by 1.17, making the maximum picture frequency 1.17 times 5,520,000 or approximately 6,458,400 cycles.

► Up to this point, our analysis of the maximum frequency has been based upon the assumption that there is always a sharp contrast between adjacent elements of the picture or scene, one dark and the next light. Of course, this is not true in actual practice, for no scene is made up of perfectly arranged checkerboard squares. Several adjacent picture elements in a line may reflect the same or nearly the same amount of light. Also, in most scenes, especially those having action, it is not necessary that slight variations between the shades of adjacent picture elements be transmitted. The average scene thus contains considerably less than the maximum possible number of cycles (changes from light to dark).

This is quite fortunate, for it reduces the maximum frequency required. Tests and experience have

\*The formula now is: Maximum theoretical picture frequency  $f_0 = \frac{1}{2} N \times N \times a \times F$ .

shown that apparatus capable of sending about 60% of the maximum frequency is fairly satisfactory. Since the maximum number of cycles was assumed in our example, we multiply 6,458,400 by .6 and get about 3,900,000 cycles (3.9 megacycles) as the final maximum frequency for a 525-line picture scanned 30 times per second, with the standard aspect ratio of 4/3. Any increase in this frequency up to the

responding to the vertical scanning frequency (the frame frequency). Satisfactory reproduction of slow changes in background illumination requires, however, that frequencies down to at least 10 cycles be passed, so we should consider 10 cycles as the minimum frequency for a practical high-fidelity television system.

► For a 525-line picture having an aspect ratio of 4/3 and a frame frequency of 30, the picture frequency ranges from a minimum of 10 cycles to a maximum of about 3.9 megacycles. Compare this with the frequency range of high fidelity sound, which extends from about 35 to 8500 cycles per second!

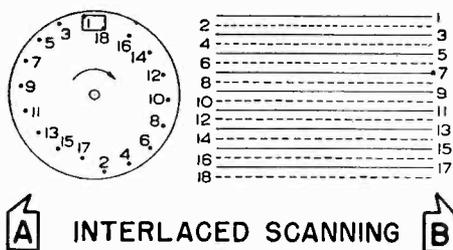


FIG. 25. The double-spiral type of scanning disc at A illustrates the principle of interlaced scanning. The arrangement is such that the successive holes on one spiral skip every other line of the image, and the holes on the other spiral scan the lines missed by the first spiral. The result is shown at B. Whether produced by the double-spiral disc at A or by an electronic television system, the lines of the picture are scanned (and reproduced) in the order 1-3-5-7-9-11-13-15-17-2-4-6-8-10-12-14-16-18.

extreme limit of 4.5 megacycles permitted within a television band gives a definite improvement in picture fidelity.

**The Minimum Frequency.** The upper part of the average outdoor scene (usually the sky) is bright, while the lower part is considerably darker. In scanning such a scene, the picture elements are varying in light intensity at a high level for the upper half of the picture and at a low level for the remainder of the picture, giving one cycle of change from light to dark for each scanning of the picture. Transmitting these changes properly calls for a minimum frequency corre-

**Interlaced Scanning.** Increasing the number of frames per second reduces flicker, but it also steps up the maximum signal frequency. For example, a 525-line picture with an aspect ratio of 4/3 and 30 frames per second gives a maximum frequency of about 3.9 megacycles. At 60 frames per second, the maximum frequency becomes about 7.8 megacycles, which is way beyond the present ability of television apparatus. But scientists were not to be balked; they resorted to interlaced scanning, an old principle of television which was used on many of the scanning disc systems. In these systems, the holes were arranged in two sets of spirals, as shown in Fig. 25A. There was one set of holes scanning or reconstructing every other line, as illustrated in Fig. 25B. Although the number of lines in the picture is not increased, the frame is scanned vertically *twice* for each complete scanning of all the elements. This does not change the maximum signal frequency, but reduces flicker considerably. Each of the "scannings" are called a "field" to distinguish it from the complete image which is called a

frame. There are two fields in each frame, in the system just described. ► This same principle of interlaced scanning has been applied to electronic television methods. It was discovered that at a field frequency of 60, by using an odd number of scanning lines per picture, the picture would automatically interlace. This is why 525 lines is the present standard rather

than 500 lines or 520 lines or any other even number. In television, the phase distortion may tend to shift an impulse from one picture element to the other which definitely will cause distortion.

The effects of phase distortion are illustrated in Fig. 26. When the signal

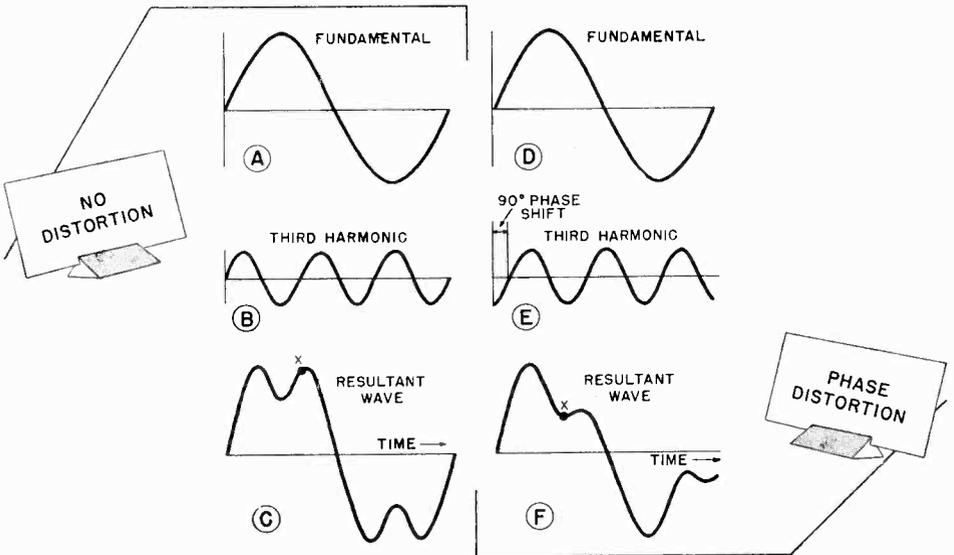


FIG. 26. A comparison between the resultant waves will show that phase distortion will produce a different wave shape. In television, this means the spot brilliancy at the receiver will not correspond to that of the scene at the transmitter.

than 500 lines or 520 lines or any other even number.

### PHASE DISTORTION

Radio apparatus designed for sound signals must handle the required range of frequencies in the sound, and must not introduce either frequency or amplitude distortion. Television amplifiers also have these requirements, as well as one other, that there be negligible phase distortion. With sound sig-

nals, phase distortion is of little consequence because the human ear is concerned only with the frequency of a sound and with its strength, but in television, the phase distortion may tend to shift an impulse from one picture element to the other which definitely will cause distortion. This can easily be proved by comparing corresponding points, such



Courtesy General Electric Co.

A combination television and all-wave receiver. The television reproducer tube is at the upper left and the television receiver controls are in the center. The broadcast-short wave receiver controls are at the upper right. The loudspeaker (used for both the all-wave set and for the television sound program) is behind the grille cloth at the bottom of the cabinet front.

as  $x$ , on the waves  $C$  and  $F$ . Since the amount of light produced on the screen of the image reconstructor tube depends on the amplitude of the wave, this difference in amplitude may easily make the image element far lighter or far darker than the original.

**Looking Ahead.** Now that you have some idea of the requirements placed on low-frequency amplifiers, you are

ready to study practical amplifiers. In your next lesson, you will learn how low-frequency amplifiers are made to meet the requirements you have just studied. Then, in other lessons, you will go on to high-frequency amplifiers, demodulators, and other radio sections and stages. Soon you will have a complete understanding of *all* the sections in modern receivers.

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# Lesson Questions

Be sure to number your Answer Sheet 14FR-2.

Place your Student Number on every Answer Sheet.

Send in your set of answers for this lesson immediately after you finish them, as instructed in the Study Schedule. This will give you the greatest possible benefit from our speedy personal grading service.

1. Name the three forms of intelligence which can be sent over either wire or radio communication systems. *light, sound, radio*
2. Why isn't sound transmitted by a perfect vacuum? *no medium*
3. What is the speed of sound waves through air under normal conditions? *1100 ft/sec*
4. What three things can happen to sound waves which strike a material? *reflection, refraction, absorption*
5. How can the reverberation period of a room be reduced? *use sound absorbing material*
6. What are the two important characteristics of a sound having the form of a simple sine wave? *amplitude, frequency*
7. Can the average human ear detect changes of less than 3 db in the intensity of a complex sound? *no*
8. Which TWO of the following are the most objectionable types of distortion insofar as sound signals are concerned: 1, frequency distortion; 2, amplitude distortion; 3, phase distortion? *2, 3*
9. What characteristic of the human eye allows it to retain an impression of an object for a short time after that object has disappeared?
10. What is meant by the aspect ratio of a television picture? *Persistence of vision  
width divided by height of a picture*

## CONVERSATIONALLY SPEAKING

There is danger in thoughtless conversation. Too many of us say things, unintentionally of course, which hurt, irritate or embarrass a friend—and a word once spoken can never be unspoken. Think before you talk.

Conversation is a give-and-take proposition, and listening is the “take” part. Talk only when you can say something of interest. Otherwise, remain silent. Let your silence be eloquent enough to show that you derive pleasure from listening—that you consider the words of your companion far more valuable than anything you could say. This kind of silence can make just as many friends as good conversation.

Talk about things which will interest and please *your listeners*. Their hobbies, their work, their children and their homes are all good opening topics for conversation. *Don't talk about yourself, your troubles or your work*, unless asked.

Avoid expressing definite opinions on controversial subjects, for they often lead to unpleasant arguments. Ridicule of another person is likewise taboo at all times. If you can't say pleasant things about others, keep quiet. Finally, reserve technical discussions for technically-minded listeners.

J. E. SMITH

**LOW-FREQUENCY AMPLIFIERS  
FOR SOUND AND  
TELEVISION RECEIVERS**

15FR-2

**NATIONAL RADIO INSTITUTE**

ESTABLISHED 1914

**WASHINGTON, D. C.**



# STUDY SCHEDULE NO. 15

For each study step, read the assigned pages first at your usual speed. Reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind, then answer the Lesson Questions for that step. Study each other step in this same way.

- 1. Low-Frequency Voltage Amplification . . . . . Pages 1-8  
The low-frequency amplifier consists of a voltage amplifier and an output stage. In this section you learn about the most popular voltage amplifier—the R-C coupled type—and learn how the fidelity of signal reproduction depends on parts values. Answer Lesson Questions 1 and 2.
- 2. More About Voltage Amplifiers . . . . . Pages 8-13  
Amplitude distortion; how stages are cascaded; and the fidelity of transformer-coupled and impedance-coupled voltage amplifiers, are the important subjects of this section. Answer Lesson Questions 3 and 4.
- 3. Single-ended Power Amplifiers . . . . . Pages 13-19  
Loudspeakers require signals with considerable power. Therefore, in sound receivers, the voltage amplifier supplies signal voltage to the grid of a power output tube. This section describes the differences between voltage amplifier and power stages. Answer Lesson Questions 5 and 6.
- 4. Push-Pull Power Stages . . . . . Pages 19-25  
The push-pull output connection offers higher power and better fidelity than does a single-ended power stage. Study this section carefully; there is much to learn about this important circuit. Answer Lesson Question 7.
- 5. Inverse Feedback . . . . . Pages 25-30  
Much of the distortion that arises in pentode and beam power output tubes can be eliminated by feeding back an out-of-phase signal so as to cancel some of the distortion. Here you learn how inverse feedback circuits of this kind work. Answer Lesson Question 8.
- 6. Phase Inverters . . . . . Pages 31-34  
Phase inverters are tube stages used in place of the input push-pull transformer. These stages reverse the signal phase  $180^\circ$  so that the push-pull grids may be properly driven. Answer Lesson Questions 9 and 10.
- 7. Facts about Low-Frequency Amplifiers . . . . . Pages 34-36  
This is a review section for the lesson. If you do not understand all of it, you should review the proper sections. Later on, this summary will help to refresh your mind on the basic characteristics of low-frequency amplifiers.
- 8. Mail your Answers for this Lesson to NRI for Grading.
- 9. Start Studying the Next Lesson.

# LOW-FREQUENCY AMPLIFIERS FOR SOUND AND TELEVISION RECEIVERS

## Low-Frequency Voltage Amplification

**R**ADIO signals go through several receiver sections to reach the low-frequency amplifier. As a review, you will recall that the signals (picked up by the receiving antenna) consist of radio frequency variations that follow the speech, picture, code, or music intelligence. The desired modulated carrier is selected and amplified by the r.f. and i.f. amplifiers. Next, the signal is fed into the demodulator (also called the detector). The demodulator stage eliminates the carrier, and produces a signal that is the electrical equivalent of the original intelligence—the signal can now be used to reproduce the sounds or picture we want, except that it is too weak to operate the reproducer. The job of the low-frequency amplifier is to increase the voltage or the power of this signal (from the demodulator) sufficiently to operate the reproducer.

**Requirements.** An increase in the signal voltage is all that is required to modulate the electron stream in the image reconstructor tube of a television set. However, the loudspeaker of a sound receiver may require considerable *power*—it is common to find sound receivers with power outputs ranging from 1 watt to as high as 10 or 12 watts. (Public address systems range up to 100 watts and more.)

By proper design, it is possible to obtain the necessary power from one or more “power” tubes, provided sufficient signal voltage can be applied to the control grid of this tube (or tubes). To obtain this signal voltage, the output of the demodulator is fed through

one or more *voltage* amplifier stages. These voltage amplifier stages plus the power output stage make up the complete low-frequency amplifier. (The cathode ray tube or picture reproducer is the output stage in a television receiver.)

Since the low-frequency amplifier deals with the electrical equivalent of the original intelligence, the amplifier must of course create as little distortion as possible. Amplitude and frequency distortion must be controlled in a sound receiver, and in a television set the designer must worry about phase distortion, also.\*

► In this lesson, we shall study the complete low-frequency amplifier. (Future lessons will cover r.f. and i.f. amplifiers and demodulators.) We will begin with voltage amplifiers and then go into power stages.

We find that low-frequency *voltage* amplifiers are classified according to the devices used to couple the stages together. Thus, we have the resistance-capacitance coupled amplifier, the impedance-coupled amplifier, and the transformer-coupled amplifier. Of these the resistance-capacitance type (commonly known as the “resistance-coupled” or as the “R-C” amplifier) is the most widely used, so we will start with it.

---

\*The low-frequency amplifier in a sound receiver is called the *audio amplifier* (because it handles frequencies corresponding to the audible range), and that in a television receiver is called the *video amplifier*.

## THE R-C AMPLIFIER

Fig. 1 shows the details of resistance-capacitance coupling, so named because resistors and condensers serve to couple one tube to another.

You have met the amplifier before, but let's study its operation in more detail. As you have learned, the filaments are supplied with heater power, even though it is customary not to show the filament connections on schematic diagrams of the kind shown in Fig. 1. (This is purely a convenience to avoid complicating the diagram when it is being used to study the signal paths.) The plate supply is indicated by the signs  $B+$  and  $B-$ . Grid bias is obtained from the self-bias resistors  $R_2$  (for  $VT_1$ ) and  $R_5$  (for  $VT_2$ ). The a.c. components of the plate currents are by-passed around these resistors by  $C_2$  and  $C_5$  respectively. That is, there is only a small a.c. voltage drop between the tube cathodes and  $B-$ , because the reactances of the condensers are lower than the resistances of the resistors they by-pass. However, the d.c. current has to flow through the bias resistors, and the resistor values are chosen so that they provide the voltage drop needed to furnish the desired grid bias.

► The input signal voltage  $e$  causes a signal current to flow through  $C_1$  and  $R_1$ . The resulting voltage drop across grid resistor  $R_1$  becomes the grid signal voltage for  $VT_1$ . (The signal current is a.c., so it "passes through" the condenser. Hence, the entire a.c. voltage  $e$  appears across  $R_1$  except for the portion lost in the reactance of the condenser, as we shall soon show.) Tube  $VT_1$  amplifies the signal, so an amplified version appears across plate load resistor  $R_3$ . This load a.c. voltage causes a signal current to pass through condensers  $C_3$  and  $C_4$ , and grid resistor  $R_4$ . The resulting voltage across  $R_4$

becomes input for  $VT_2$ .

Condenser  $C_3$  is called the *coupling condenser* because it "couples" the circuits of  $VT_1$  to those of  $VT_2$ . Condenser  $C_4$  also could be considered as a coupling condenser, but instead it is called a by-pass condenser, because it is primarily selected to by-pass the signal so that the signal current does not flow through the B supply. (This condenser completes the a.c. circuit to ground, so that the signal is directly applied to the grounded end of resistor  $R_4$ .) This completes the passage of the signal through the stage using  $VT_1$ ; it now is acted upon by  $VT_2$ .

**Frequency Band.** Before we can determine the gain of the stage using tube  $VT_1$ , we must consider the range or "band" of frequencies that this amplifier is called upon to pass.

The *use* to which the receiver is put determines the width of the band of frequencies that the low-frequency amplifier must pass. Communications receivers, intended only for the reception of voice or code signals, have to amplify only those frequencies between about 200 and 3000 cycles.\*

Sound receivers for the home must amplify a wider band for musical reproduction. Although sound frequencies range from 30 cycles to above 16,000 cycles, the average inexpensive radio is limited to a frequency range of from about 200 cycles to 5000 or 6000 cycles. Some of the better receivers, including the average f.m. receiver, have ranges of 50 cycles to 8000 cycles. A few high-fidelity a.m. and f.m. receivers have a range of from 30 to 16,000 cycles.

In television, the desired frequency band ranges from 10 cycles to near 4,000,000 cycles. This band is so wide

\*Like most radio men, we use "cycles" for "cycles per second," which is the correct term.

that special compensators are necessary, as we shall soon see.

► Even though the low-frequency amplifier is intended to pass a particular band of frequencies (as we just described), the amount that it amplifies the higher and lower frequencies in this band is different from the amplification given to frequencies in the middle of the band. That is, it is not perfect in its frequency response, and it may not be economical to improve the response beyond a predetermined level. Therefore, the parts are chosen to give a certain response, and this places limits on the replacement values that

come from changes in the load, or from voltage-dividing networks that vary with frequency. Let's see what occurs in our R-C amplifier of Fig. 1.

**Mid-Frequency Response.** First, let's consider the response to frequencies in the middle of the band. If the amplifier is designed to pass (reasonably well) frequencies between 100 cycles and 5000 cycles, then we can call frequencies from 1000 to 3000 cycles the mid frequencies.

When we examine the parts values used in an R-C amplifier in an a.m. or f.m. receiver, we find that the capacity used as  $C_4$  in Fig. 1 is always so large

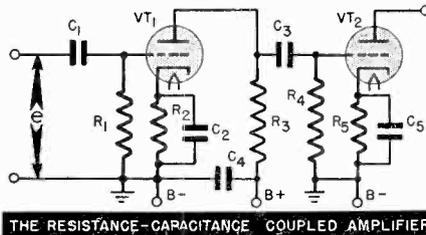


FIG. 1. This R-C coupled amplifier is the basic circuit found in most sound and television low-frequency amplifiers.

can be used. Hence, it is advisable to learn more about how an R-C amplifier (as found in a.m. and f.m. receivers) responds to the middle, low, and high frequencies in its pass band.

## FREQUENCY RESPONSE

You know that the gain of a stage depends on the amplification factor ( $\mu$ ) of the tube, the tube's plate resistance, and the value of the load impedance. The grid voltage is stepped up by the factor  $\mu$ , and the resulting equivalent a.c. plate voltage ( $\mu e_G$ ) is then divided between the plate resistance and the load. Therefore, since  $\mu$  and the plate resistance remain the same almost regardless of frequency, any change in the over-all gain must

that this condenser has negligible reactance to *all* frequencies in the band to be passed. Also,  $C_3$  has little opposition to the mid frequencies, so there is practically no reactance between  $R_3$  and  $R_4$  at these mid frequencies. Therefore, *insofar as a mid-frequency signal is concerned*,  $R_4$  is connected in parallel with  $R_3$  and the combination forms the load for  $VT_1$ . Thus, the equivalent circuit at the mid frequencies is as shown in Fig. 2A.

You will recall that the higher the load resistance is made, the greater the voltage gain will be. In fact, we get almost 90% of the  $\mu$  of the tube as a stage gain when the load resistance is 10 times the a.c. plate resistance  $r_p$ . So you might think that we need only to increase the values of  $R_3$  and  $R_4$  in or-

der to get more gain. That is true, but there are practical limits on the values we can use—we will take these up after we see what happens at the low and high frequencies.

**Low-Frequency Response.** When we go down to low frequencies (those lower than the mid frequencies) we find that the coupling condenser ( $C_3$  in Fig. 1) comes back into the picture, because, as the frequency decreases, the condenser reactance increases. Hence, it begins to act with  $R_4$  as a voltage divider. In other words, as shown by the equivalent circuit of Fig. 2B, the signal voltage developed across plate resistor  $R_3$  is actually fed into a filter  $C_3$ - $R_4$ , which passes higher frequencies better than lower frequencies. As the frequency decreases, the reactance of  $C_3$  increases; a greater proportion of the signal voltage is then dropped across  $C_3$ , and less appears across  $R_4$ . Since  $VT_2$  is operated *only* by the signal voltage across its grid resistor  $R_4$ , the less this voltage is, the lower the  $VT_2$  input signal will be. Therefore, there is less voltage fed to  $VT_2$ , so the over-all grid-to-grid ( $VT_1$  to  $VT_2$ ) gain is less, and the gain drops still more as the frequency is decreased further. Hence, in the low-frequency range, this circuit has frequency distortion (unequal amplification of different frequencies).

**High-Frequency Response.** The equivalent circuit for high frequencies is given in Fig. 2C. Condenser  $C_3$  is no bother because its reactance at high frequencies is negligible. However, we now have a new factor, the capacity  $C_s$ , which represents a number of different tube and stray circuit capacities that we have lumped together for convenience. These capacities do not appear in Fig. 1 (we will explain them shortly), but you can see their result at once. Since the reactance of  $C_s$  de-

creases as frequency is increased, it will act as a shunt across the resistors at higher frequencies. In effect, this reduces the load value at higher frequencies, so more and more of the signal voltage will be lost across  $r_p$  within tube  $VT_1$ . Hence we have frequency distortion at the higher frequencies also, but from another cause.

► We have given the three equivalent circuits together in one drawing so that you can compare them with each other. Notice that the schematic (Fig. 1) shows the electrical connections, *but does not directly indicate how the circuit will act at different frequencies.* Even the equivalent circuits of Fig. 2 indicate only the trend of operation; until parts values are known we cannot tell how serious the frequency distortion may be. As we will now show, *changing the parts values will radically change the response.*

#### EFFECT OF PARTS VALUES

We can continue to assume that  $C_4$  in Fig. 1 is sufficiently high in capacity to have no effect on any frequencies the amplifier is designed to pass. Therefore, we can concentrate on the limits set for the parts  $R_3$ ,  $R_4$ ,  $C_3$ , and  $C_s$  of Figs. 1 and 2.

**Value of  $R_3$ .** The value of the load is one of the factors that determine the stage gain. Since we want the stage to amplify, you might imagine that the value of plate resistor  $R_3$  should be high. It is desirable to have this a high resistance at the middle and low frequencies, but not at the higher frequencies. This is because the higher the resistive load is, the sooner  $C_s$  becomes an effective shunt. This effect is shown by Fig. 3. Here, curve *A* shows the frequency response for a high load resistance, and curve *B* the response for a much lower load resistance (assuming the same value of  $C_s$  in each case).

Naturally, the higher the load resistance is, the higher the amplification at the mid frequencies. Therefore, curve *A* shows fairly high gain at around 2000 cycles—higher than that shown by curve *B*.

However, compare the output at 2000 cycles with that at 5000 cycles for

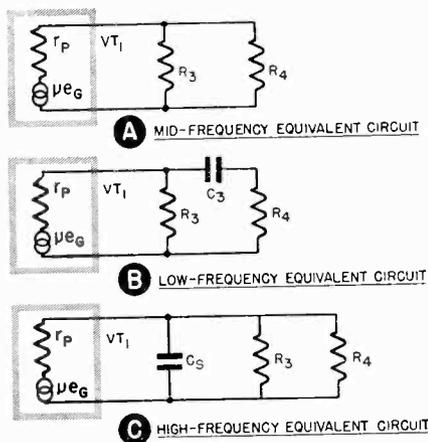


FIG. 2. The R-C amplifier responds differently to the high, middle, and low frequencies in its pass band. These equivalent circuits show how the circuit components affect the response.

curve *A*. There is a severe loss of higher frequencies (frequency distortion) beginning at about 4000 cycles. On the other hand, curve *B* shows a much smaller *difference* between the mid-frequency response and that at 5000 cycles. In other words, the gain shown by curve *B* drops only from 5 to about 3 in the range from 2000 to 5000 cycles, whereas the gain shown by curve *A* changes from 10 to about 3 over the same frequency range.

The ideal amplifier—one having no frequency distortion—would amplify *all* frequencies in its pass band equally. Hence, curve *B* approaches the ideal more closely than does curve *A*. This shows that there is good reason for keeping the effective load for the stage low when high fidelity is desired. (However, this reduces gain, and often

a compromise is necessary to get reasonable gain with good fidelity.)

► In addition, the d.c. plate current of  $VT_1$  passes through  $R_3$ , and this brings up a power supply problem. The drop in  $R_3$  (caused by the d.c. plate current) is subtracted from the B supply voltage. The remainder is the plate-cathode voltage actually supplied to the tube, and we want this voltage to be reasonably high. Since there are limits to the power supply voltage, the value of  $R_3$  must be kept down to avoid too much d.c. voltage drop. The most common values in a.m. and f.m. home receivers range from 50,000 ohms to 250,000 ohms. Therefore, the gain of a practical R-C amplifier stage is not as high as is theoretically obtainable.

**Value of  $R_4$ .** Increasing the resistance of  $R_4$  seems to be another way of

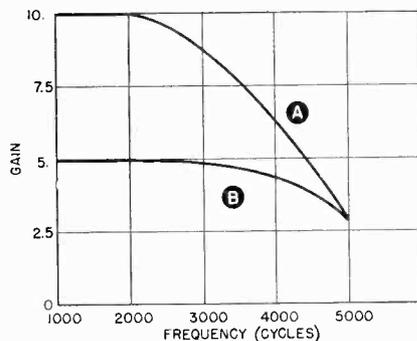


FIG. 3. Using a higher load value gives greater gain (curve *A*), but there is a more rapid drop-off in the high-frequency response.

getting a higher load value, since this resistor is in parallel with  $R_3$ , and in addition, no d.c. is supposed to flow through it. However, if the following tube ( $VT_2$  in Fig. 1) is a power tube, we may encounter gas trouble.

No matter how well a tube is evacuated in manufacture, some gas (air) remains in it. The heat of operation may drive gas molecules out of the

“pores” of the metal elements. When the gas level upsets operating conditions, we say that the tube is “gassy.” This trouble occurs particularly with power tubes because of their high current levels.

When the tube becomes gassy, speeding electrons strike the gas molecules and knock out electrons, leaving positive gas ions. The positive gas ions move toward the cathode, but some “capture” electrons from the grid. This causes an external electron flow to the grid element. This electron flow must pass through  $R_4$ , and therefore, develops a voltage across it. The grid current may be quite small at first. However, if resistor  $R_4$  is very high in value, then even a very small gas current will cause an appreciable drop across it.

The voltage drop that occurs when the tube is gassy has a polarity that makes the grid end of  $R_4$  positive. Hence, the drop opposes the bias voltage, and causes a reduction of the grid bias. Consequently, there is an increase in plate current. Eventually, the higher than normal plate current, and the bombardment of the cathode by the heavy gas ions will destroy the emissive properties of the cathode and thus ruin the tube.

For this reason, there is a definite upper limit to the size  $R_4$  can have. If tube  $VT_2$  is a power tube,  $R_4$  must be no more than 100,000 ohms to 250,000 ohms in most cases.

If the tube is not a power tube, there is less gas trouble, and  $R_4$  can be larger in value. You may find voltage amplifiers using values from 250,000 ohms to as high as 10 megohms.

**Value of  $C_3$ .** The value of  $C_3$  needed for good low-frequency response depends on the value of  $R_4$ . If  $R_4$  can be a high resistance, then even a small capacity can be used without

excessive loss of the low frequencies. However, we have just shown that  $R_4$  cannot always be made so large; therefore,  $C_3$  may have to be fairly high in capacity.

► The low-frequency response of a radio receiver depends on the design of a number of things—the large amount of audio power needed for good low-frequency response requires an oversized loudspeaker, a large power supply, a high-power output stage, and a large console with a specially designed speaker compartment. All these items are costly, a fact that explains why the low-frequency response is sacrificed in the average receiver.

Once the limits have been set by the design of these components, there is no reason for making the voltage amplifier too perfect in its response; in fact, it is better *not* to have the low-frequency response too good, because if the amplifier can amplify very low frequencies, slight changes in the power supply voltages may cause it to oscillate. This oscillation prevents the passage of the signal, so all that can be heard is either a low-frequency howl or a “putt-putt-putt” sound called *motorboating*. Therefore,  $C_3$  and  $C_1$  of Fig. 1 are made no larger than necessary in practical amplifiers. Values found in the average radio are from .005 mfd. to perhaps .1 mfd., with the most common values being .01 or .05 mfd.

**Value of  $C_s$ .** The shunt capacity  $C_s$  is made up of the inter-electrode capacities within the tubes, and the capacities between wires and parts in the circuit.

The tube inter-electrode capacities are shown for a triode tube in Fig. 4. You will recall that a capacity exists between any two conductors separated by an insulator. Therefore, the metallic tube elements, separated by a vacuum,

form tiny condensers. As shown by Fig. 5, the capacity between the grid and the cathode (called  $C_{GK}$ ) is across the input circuit, and the capacity between the plate and the cathode (called  $C_{PK}$ ) is across the output circuit. Even worse is the capacity between the grid and the plate ( $C_{GP}$ ). Notice that this capacity acts as a coupling condenser between  $R_L$  and the input, so that part of any a.c. voltage across  $R_L$  will also be across the input. This means that part of the amplified signal voltage across the load will feed back to the input circuit through  $C_{GP}$ . Because of the phase relationships involved, the reaction of this feedback voltage on the input is such that the circuit acts as if the input capacity of the tube were far higher than the measured capacity between the grid and the cathode elements.

► Therefore, going back to Fig. 1, tube  $VT_2$  acts as if it had a high capacity between its grid and its cathode, in parallel with  $R_4$ . Also, tube  $VT_1$  has an output capacity between its plate and

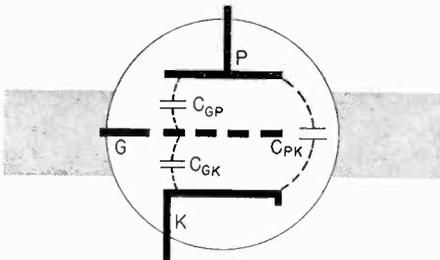


FIG. 4. The inter-electrode capacities within a triode tube.

its cathode (across  $R_3$ ), and there is stray capacity between the chassis and the wiring and parts. (Coupling condenser  $C_3$  is the most bulky part, so a considerable amount of stray capacity exists between it and the chassis. This is another reason for choosing  $C_3$  with a capacity that is no larger than necessary—if its physical size is kept down,

the stray capacity will likewise be minimized.)

Since the output capacity of  $VT_1$  and the input capacity of  $VT_2$  are in parallel, they can be lumped together and considered as a single condenser. Therefore, at high frequencies, the circuit is equivalent to that shown in Fig. 2C, where capacity  $C_s$  represents all the shunting capacities.

► Set designers make every effort to minimize the value of  $C_s$ . Once this

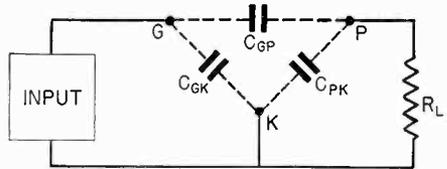


FIG. 5. The plate-grid inter-electrode capacity provides a coupling path so that the voltage across the load is transferred to the input circuit.

value is determined, it fixes the largest values of  $R_3$  and  $R_4$  that can be used and still have the required fidelity.

**Summary.** Tube  $VT_1$  is usually chosen to be a high- $\mu$  triode (or pentode in some cases). Tubes with high gain and low d.c. current values have been specially designed for use in R-C amplifiers. Even so, fidelity compromises mean that R-C amplifiers are used where gain is less important than fidelity, because it is possible to get good fidelity, despite what has just been shown, if the gain can be sacrificed. Hence, we can make the following rules for circuits like that in Fig. 1:

1. For maximum gain at the middle and low frequencies,  $R_3$  and  $R_4$  should be high in resistance.

2. For minimum loss of high frequencies,  $R_3$  should be low in resistance.

3. The larger  $C_3$  is in capacity, the better is the low-frequency response.

► As a serviceman, you will have to replace these parts whenever defects

occur in them. You should use replacements having values within 20% of the original values if you do not wish to change the circuit characteristics. If you do want to make a change, or if you do not know the original values, the rules just given will help you

choose the best replacements for the desired conditions. When you install replacements, take care to minimize the shunting capacity  $C_s$ . Don't add more wire than necessary, and don't disturb positions of parts more than is necessary.

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## More About Voltage Amplifiers

So far, we have considered the frequency response and frequency distortion of a resistance-coupled amplifier. Equally important is the amplitude distortion that can occur—in fact, even small amounts of amplitude distortion can be far more objectionable than fairly large amounts of frequency distortion. Let's continue with the resistance-coupled voltage amplifier as our example, and learn more about amplitude distortion.

### AMPLITUDE DISTORTION IN R-C AMPLIFIERS

You know that amplitude distortion is produced whenever the wave shape of a signal is changed. This distortion will occur whenever the tube's operating characteristic is curved. Therefore, to minimize the distortion, operation is limited to the straightest portion of a tube's characteristic.

► However, so far we have presented the darker side of this story because we have shown you only the characteristic curve of the tube *alone*. Such "static" characteristic curves show how the tube operates when there is *no plate load*. These curves are the kind the tube manufacturer can most easily furnish since he doesn't know just what load values may be chosen by the set designer. However, when the tube is operated with a load (the normal condition) its operation is over a "dy-

namic" characteristic that is considerably different in shape. Naturally, we are interested in dynamic curves because they show how a tube actually operates in a stage.

Dynamic curves can be found by actually setting up the circuit and measuring the response, or they can be plotted from the static curves. We won't go into their construction here, but the following facts will show why they differ so much from the static curves.

First, you know that the plate current of a tube with no load increases rapidly as the grid bias is made less negative. Curve *A* of Fig. 6 is a typical static curve, showing how the plate current and the grid voltage are related when the tube plate voltage is a fixed amount.

Next, you have learned that when a load resistor is used, some of the B supply voltage is dropped across it because the tube plate current flows through this resistor. Therefore, the actual plate-cathode voltage of a tube is the difference between the d.c. load drop and the supply voltage.

► Now, let's suppose we change the bias so that the grid is made less negative. Since the plate current tends to increase when the bias is less negative, there will be a greater drop across the load resistor, so the actual tube plate voltage drops. This reduction in plate

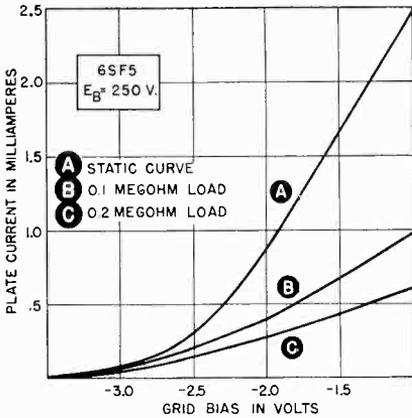


FIG. 6. Curves B and C show how increasing the load resistance tends to flatten the tube  $E_G$ - $I_P$  characteristic curve.

voltage counteracts some of the plate current change (the plate current depends on both the grid voltage and the plate voltage) so the actual change in current for a particular grid voltage change is not as large when a load is used.

Hence, when a load is used, the tube characteristic is "ironed out" so that it is much flatter and straighter than the static curve. The effect of this self-regulation is shown in Fig. 6, where curves B and C represent dynamic curves for two different load values for a triode tube.

You can see that there is far less curvature in these dynamic curves than there is in curve A. This means that there will be much less amplitude distortion when a load is used. That is, a change in grid voltage (such as is

caused by a signal) will cause much more nearly linear plate current changes.

► Furthermore, the higher the load resistance is, the flatter the curve. However, this brings back our compromise conditions; you know that high resistances cause excessive loss of the higher frequencies. Therefore, the designer must consider gain, frequency distortion, and amplitude distortion when he chooses values for the tube load.

## CASCADE AMPLIFIERS

These factors are further complicated when there is more than one stage of R-C amplification. For example, suppose we have two stages, as shown in Fig. 7. We'll consider tubes  $VT_1$  and  $VT_2$  to be the voltage amplifying tubes for these stages, with  $VT_3$  shown only to indicate the output coupling. Furthermore, let's assume that the characteristics of *corresponding* parts are alike: that is, the tubes are the same, condensers  $C_4$  and  $C_7$  have the same capacity, and resistors  $R_3$  and  $R_6$  have the same resistance, as do  $R_4$  and  $R_7$ .

The first thing we are interested in is the amount of amplification we get. Let's suppose that each of the stages is capable of giving a gain of 10 at the mid frequencies, and that a signal of 1 volt is applied to the input of  $VT_1$ . There will be a signal of 10 volts across  $R_3$ ; this voltage is applied to tube  $VT_2$ .

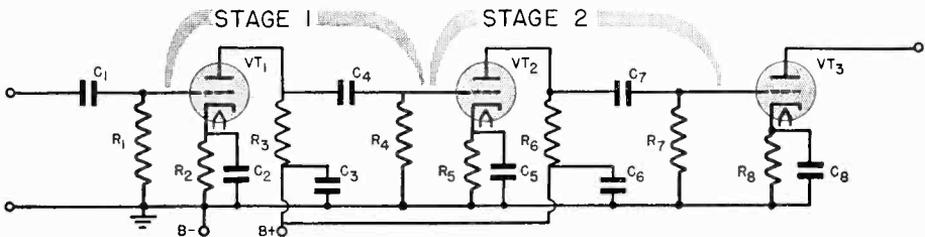


FIG. 7. A two-stage R-C amplifier.

This tube also has a gain of 10, so its 10-volt grid signal is raised to 100 volts\* across  $R_6$ . Thus, the gain of successive stages is the *product* of their respective gains; the gain of the  $VT_1$  stage is multiplied by the gain of the  $VT_2$  stage ( $10 \times 10 = 100$  and our 1-volt signal was raised to 100 volts).

When more than one stage is used, we say that the stages are "in cascade," because each one steps up the signal output of the previous ones by its own gain factor.

**Fidelity.** From this, we might at once assume that all we need to do to

with two stages it will be one-quarter (one-half times one-half) that of the mid frequencies.

► *This is one of the reasons why the number of low-frequency amplifier stages is kept at a minimum in radio receivers. It is far better to use a single high- $\mu$  triode or pentode tube than to use two or more stages of low-gain triode tubes.*

## TELEVISION AMPLIFIERS

Although careful design is necessary to have high fidelity in a resistance-

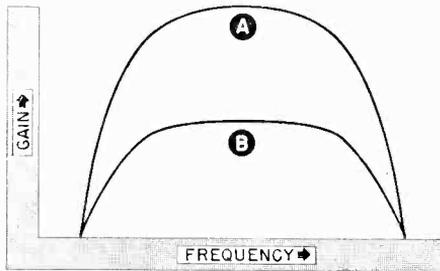


FIG. 8. Curve A shows that there is a higher mid-frequency gain when two stages are used, but there is a more rapid drop-off at the lower and higher frequencies.

obtain more gain is to add more stages. However, if we do, we find that the fidelity becomes much worse when more than a single stage is used.

For example, see Fig. 8. Curve B represents the fidelity of a single stage, and curve A represents that of two identical stages. Curve A shows that the gain is increased at the mid frequencies, but you can see that the addition of another stage exaggerates the frequency distortion already present at the low and high frequencies in one stage. Thus, if the gain at some particular frequency is only half that of the mid frequencies in a single stage,

coupled amplifier, this circuit still offers a greater range of frequency response than any other, so it is universally used (in a modified form) as the video intelligence signal amplifier in television receivers. We will leave complete details for later lessons, but some of the methods used to get a broad frequency pass band are of particular interest at this time.

For television, the frequency range is from about 10 cycles to as high as 3,000,000 or 4,000,000 cycles. This is many times greater than that found in any sound receiver. Getting this extended high-frequency response is quite a problem.

To reduce the effect of the shunting capacity (corresponding to  $C_s$  in Fig.

\*If the input was .01 volt, the output of  $VT_1$  would be .1 volt, and the output of  $VT_2$  would be 1.0 volt.

2C) and thus obtain better high-frequency response, a very small load resistor (1000 to 5000 ohms) is used. This means extremely low gain unless special high mutual-conductance pentode tubes are used.

Other steps are also taken to reduce the effects of the shunt capacity. As you learned, the shunt capacity is due

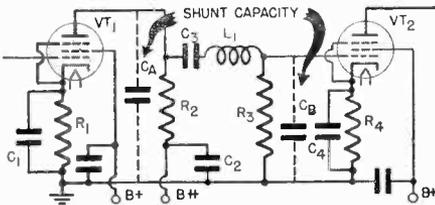


FIG. 9. The effects of the shunt capacities  $C_A$  and  $C_B$  are reduced by using  $L_1$ .

partly to the output capacity of one tube and partly to the input capacity of the other. In Fig. 9, these capacities are represented as  $C_A$  and  $C_B$ . One scheme used is to insert a coil ( $L_1$  in Fig. 9) between the plate circuit of  $VT_1$  and the grid circuit of  $VT_2$ . This serves to isolate  $C_B$  from  $R_2$ , so it reduces the effects of  $C_B$  on the load of  $VT_1$ . The isolation increases with frequency increases because the reactance of  $L_1$  increases with frequency. Hence, its increasing reactance counteracts the decreases of the  $C_B$  reactance insofar as  $VT_1$  is concerned.

Also it is possible to make  $L_1$  and  $C_B$  resonant at some rather high frequency. This forms a series resonant circuit so that a maximum voltage appears across  $C_B$  at the resonant frequency. This voltage is applied to  $VT_2$ , so the response at and near this resonant frequency is improved.

► Another scheme is shown in Fig. 10. Here, coil  $L_2$  is placed in series with  $R_2$ . As the frequency increases, so does the reactance of  $L_2$ . Therefore, as  $L_2$  is part of the load, the load impedance rises as the frequency increases. Al-

though the shunting effects of the stray capacity are increased, there is a better division of the  $\mu e_G$  voltage between the  $r_P$  of  $VT_1$  and the load, so the  $VT_1$  stage provides higher gain at the higher frequencies. This tends to minimize the drop-off in high-frequency response. Furthermore, by making  $L_2$  resonant with  $C_A$  (at a high frequency), a parallel resonant circuit is formed. This provides a rise in the load resistance at resonance, which in turn further boosts the gain at the high frequencies. Sometimes both schemes shown in Figs. 9 and 10 are used in the same amplifier.

► In addition to these methods of reducing shunt capacity effects, television engineers use sockets that have low capacity to chassis, and mount parts so that they are separated from the chassis as much as possible, yet try to keep the leads between stages short to avoid capacity between these leads and other wires. These precautions are necessary because of the very wide

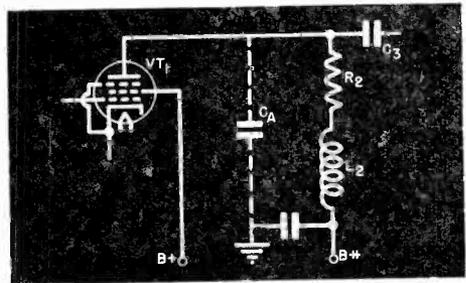


FIG. 10. Another compensation for shunt capacity is to use  $L_2$  in series with the load resistance.

band of frequencies passed by the television amplifier. Naturally, it is not necessary to go to such extremes in ordinary sound receivers.

## USING COILS FOR COUPLING

The resistance coupling you have just studied is the coupling method most commonly used today for voltage amplifiers. However, some amplifiers

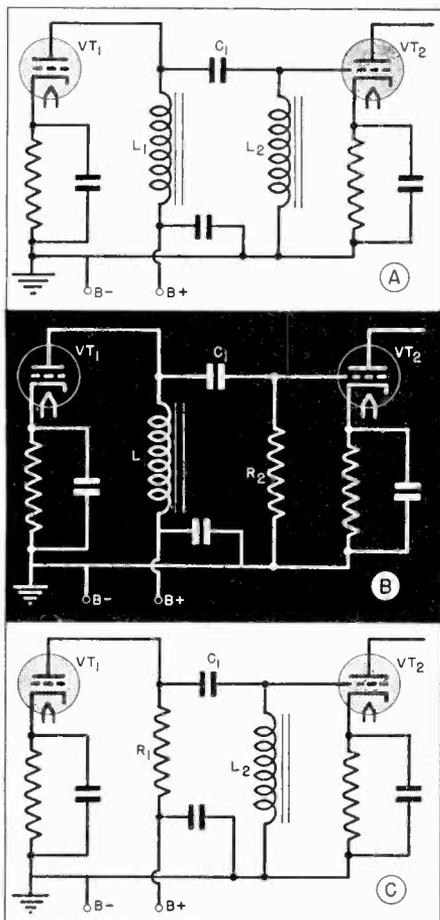


FIG. 11. Three forms of impedance coupling are used in low-frequency amplifiers.

use inductances (coils) for coupling. Some of these inductances are choke coils, others are transformers—we will study both.

**Impedance Coupling.** Essentially, an impedance-coupled amplifier is like a resistance-coupled amplifier except for the substitution of one or more iron-core choke coils for the resistors. Fig. 11A shows a full impedance-coupled amplifier. Here choke coil  $L_1$  is used as the plate coil, and choke coil  $L_2$  is used as a grid coil for tube  $VT_2$ . The operation is similar to that of re-

sistance coupling except for the effects of the coils on the frequency response, and the effect of  $L_1$  on the plate voltage.

Since  $L_1$  has a low d.c. resistance, there will be very little d.c. voltage drop across it. Practically the full supply voltage is applied to the plate of the tube. For this reason, this circuit is particularly useful where the B supply is very limited, as in battery-powered circuits.

Coil  $L_2$  also has a low d.c. resistance. Therefore, any gas current flowing through  $L_2$  will cause but a small voltage drop, so the use of a coil eliminates the bias shift encountered when a grid resistor is used with a gassy tube.

► Since both choke coils offer higher reactances as the frequency increases, we might expect the gain of the stage to go up as the frequency rises. However, good high-frequency response is not obtained, because the choke coils have rather high distributed capacities, which are effectively in parallel with the coils. The distributed capacity tends to cut off the high-frequency response even more rapidly than in a resistance-coupled amplifier, because the coil capacities are added to the normal shunting capacities in the circuit.

The low-frequency response is poor also, because the reactance values drop at low frequencies. Hence, unless very large values of inductance are used, the low-frequency response will be much poorer than that of the mid frequencies.

Fig. 11B shows a variation of this circuit in which coil  $L_2$  is replaced by resistor  $R_2$ . This eliminates the distributed capacity and inductance troubles of one of the choke coils, so this circuit is somewhat better in frequency response than that shown in Fig. 11A.

This is the impedance-coupled circuit most likely to be found in use, for it is considerably cheaper than the one in Fig. 11A.

Fig. 11C shows another possible variation, in which coil  $L_1$  is replaced by resistor  $R_1$ . (Choke  $L_2$  is used as the grid coil.) This circuit has little to recommend it unless tube  $VT_2$  is likely to be gassy and a high resistance is not wanted in the grid circuit.

**Transformer Coupling.** Fig. 12 shows an example of a transformer-coupled voltage-amplifier stage. As you learned in an earlier lesson, transformers have a special kind of amplification also; they can step up the signal voltage by their turns ratio.

Therefore, if tube  $VT_1$  in Fig. 12 has a gain of 10, and the transformer  $T_2$  a step-up ratio of 3, the first stage gain will be the product of the two, or 30. Hence, transformer-coupled amplifiers give much higher gain for the same type of tube than do either impedance-coupled or resistance-coupled amplifiers.

As you learned in an earlier lesson, the frequency response of a transform-

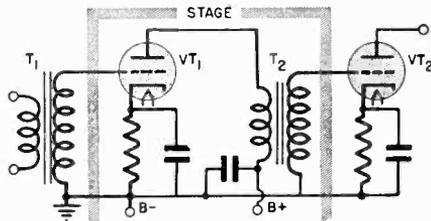


FIG. 12. A typical transformer-coupled, low-frequency, amplifier stage.

er is limited. The inductance of the primary winding must be very high to prevent an excessive loss of low frequencies because of the voltage division between the primary and the plate resistance of tube  $VT_1$ . In addition, the distributed capacity of the windings must be low to prevent an excessive loss of the high frequencies. Essentially, transformers give a frequency response similar to that obtained with impedance coupling.

► The resistance-coupled amplifier offers far better fidelity than either of the devices using iron-core coils. However, the high amplification obtainable from the transformer makes it a favorite where gain is more important than frequency response.

## Single-Ended Power Amplifiers

Once the voltage amplifier has made the signal sufficiently large, the voltage is fed into a power stage which supplies the power to drive the reproducer (a loudspeaker in a sound receiver). On paper, the power stage diagram is very similar to that of the other audio stages. It is not until we notice the plate current values and bias voltages that we see the difference.

Modern receiver loudspeakers require a considerable amount of a.f. power—anywhere from 1 watt to 12 watts on maximum output levels. Ord-

nary *voltage* amplifying tubes cannot furnish anywhere near this power. Most of them draw only from .25 watt to about 1 watt of d.c. power. This power is found by multiplying the B supply voltage by the d.c. plate current. For example, if the B supply voltage is 250 volts, and the d.c. plate current is 1 ma. (.001 amp.), then the d.c. power is  $250 \times .001$ , or .25 watt.

The total developed a.f. power is always far less than the d.c. power, and at least half the a.f. power is lost in the plate resistance of the tube. There-

fore, we must use an output tube capable of delivering a higher power than can a voltage amplifier.

This means a power tube must draw a higher d.c. power than a voltage amplifier in order to deliver the required a.c. power. Since it is desirable to operate all the stages at the same plate voltage to simplify power pack

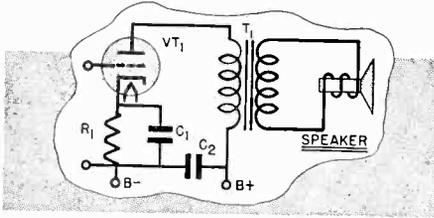


FIG. 13. The voice coil of the loudspeaker is matched to the plate impedance of the output tube by the output transformer.

design, power tubes are designed to draw more plate current. (Power is equal to voltage times current, so increasing the current increases the power.)

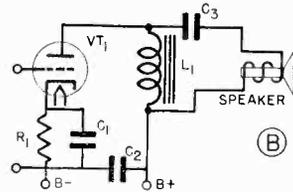
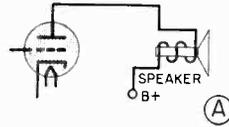
► To get the increased emission, the size (length and diameter) of the cathode is increased, and the filament current is increased to the value needed to heat the larger cathode. Then, the plate area is increased to correspond to the increased cathode size. This provides more electron paths (in parallel) from the electron cloud to the plate, and, as the same plate voltage acts across each, this has the effect of resistance in parallel—there is less plate-cathode resistance, so the plate current is higher for the same plate voltage.

For comparison, audio voltage amplifying tubes in radio receivers draw currents of from .5 ma. to perhaps 5 ma., but power output tubes draw currents of 30 to 50 ma. and more.

**Coupling to Speaker.** Fig. 13 shows the method of coupling from the plate of a power tube to the voice coil of a dynamic loudspeaker. Usually a

voice coil has an impedance of only 4 to 8 ohms, which is far lower than the plate resistance of the tube. Transformer  $T_1$  is used to match the two impedances.

**Load Matching.** As you know, maximum power output is obtained when the load impedance exactly matches the plate resistance of the tube. This does not assure minimum amplitude distortion. However, you will recall that increasing the load re-



Magnetic speakers were once very popular, and in radio servicing you may still encounter a receiver using one. An output transformer is not needed with them—they are made with many turns of wire, so it is possible for them to have impedances of 2000 to 6000 ohms, which is a close match to the required load for output tubes. In the direct coupling shown at A, the d.c. plate current flows through the speaker. This may be undesirable, for, if the wrong connections are made, the resulting d.c. flux will be opposite to that of the permanent magnet used in the speaker. This may ruin the speaker magnet. Also, the high current requires large wire sizes, which may make it difficult to wind the coil in the space provided. Hence, many sets used the choke-condenser coupling shown at B to avoid this; here the d.c. is blocked by C<sub>3</sub> and flows only through L<sub>1</sub>.

sistance value tends to flatten the triode tube characteristic curve. Therefore, it has been found that with *triode* tubes, a load equal to about twice the plate resistance value will give the maximum *undistorted* power output.

Notice the difference. The maximum power, with some distortion, is ob-

tained when the load and source values are matched. The maximum *undistorted* power (less than the maximum power but with less distortion) is obtained when the load is twice the tube resistance. (Further increasing the load resistance reduces the distortion some more, but the power output falls off because of the mismatched impedances.)

The load values and operating voltages listed in tube charts for triode power tubes are for the *maximum undistorted* power output, and the output transformer is designed for these values.

### PENTODE AND BEAM POWER-OUTPUT TUBES

The triode power-output tube offers very good fidelity, but it requires a high signal input to be driven to its full output. Also, it consumes a considerable amount of d.c. power. The latter is no problem in a large receiver with a heavy-duty pack, but is troublesome in a set using battery or a.c.-d.c. supplies, which are limited in their power. For these reasons, pentode and beam power tubes having higher efficiencies than triodes\* were developed.

The pentode and beam tube types require considerably less grid signal voltage than does a triode to produce the same output. For example, the type 2A3 triode tube requires a grid signal having a peak value of nearly 45 volts to give 3.5 watts output. On the other hand, the 6K6G pentode power tube requires a signal of only

\* For comparison, the a.c. power output (with permissible distortion levels in the case of the pentode) of the 6K6G pentode and the 2A3 triode are similar (about 3.5 watts). With the same plate voltage, the 2A3 draws 60 ma., but the 6K6G draws only 32 ma. The d.c. powers used, at 250 volts, are 15 watts for the 2A3, and 8 watts for the pentode.

about 18 volts peak to give 3.5 watts output.

This is important because it leads to a simpler voltage-amplifier design. In most receivers, one voltage-amplifier stage is plenty with pentode or beam output tubes, but triode output tubes frequently require two.

**The Pentode Tube.** Fig. 14 shows an output stage using a typical power pentode tube. As you will notice, this tube has three grids. From the cathode toward the plate, these grids are, respectively, the control grid, the screen grid, and the suppressor grid.

The control grid performs the same

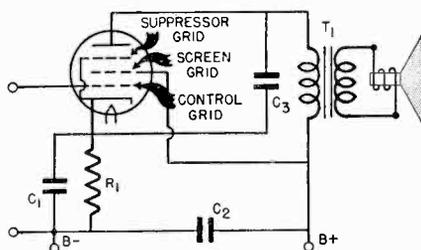


FIG. 14. The elements of a pentode power output tube are identified.

function as it does in a triode tube. ► The screen grid is a relatively coarse grid that is connected to a positive voltage—the screens of power types are usually connected directly to B+. Its high potential causes it to draw electrons from the electron cloud. The screen grid catches some of these, but because of its coarse structure (widely separated turns of wire) most of the electrons go through the mesh to the plate.

Since the screen grid is considerably closer to the cathode than the plate is, the plate current is much more dependent on the screen grid voltage than it is on the plate voltage. If the screen grid voltage is kept constant, the plate voltage can be varied over rather wide

limits without changing the plate current to any considerable extent.

Hence, in pentodes, the control grid has far more effect on the plate current than does the plate voltage, so the pentode has a high amplification factor. (As you know, the amplification factor or  $\mu$  of a tube is the ratio of the plate voltage change to the grid voltage change that produces the same plate current change.) This means that to obtain the same a.c. plate voltage ( $\mu e_G$ ) and hence the same a.c. power, the grid signal voltage required by a pentode is much smaller than that required by a triode power output tube. Thus, the *power sensitivity* of a pentode is much higher than that of a triode.

► The screen grid gives us the desired high sensitivity, but the presence of this second positive element causes a trouble that requires the presence of the suppressor grid.

The screen grid attracts electrons and speeds them on their way to the plate (in fact, it is sometimes called an accelerating grid). When these high-speed electrons strike the plate, other electrons are knocked out of the plate material. This emission of electrons due to bombardment is known as *secondary emission*.

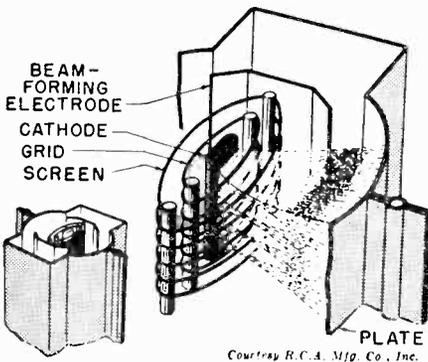


FIG. 15. The internal construction of the beam power output tube.

As long as the plate is more positive than the screen grid, these secondary emission electrons are attracted right back, so no harm results. However, the voltage developed across the plate load varies with the signal, and, of course, this drop will at times be so large that the plate voltage (supply minus load voltage) falls below the screen grid value. Whenever this occurs, the secondary emission electrons tend to move from the plate to the more positive screen grid. This reversal of electron flow will reduce the plate current, causing distortion.

Secondary emission is not prevented by the suppressor grid, but its bad effects are removed. This grid, tied to the cathode, is always at the cathode potential. It is therefore always negative with respect to the plate, and repels any electrons that try to move away from the plate, driving them back. Only the slower-moving secondary emission electrons are controlled by the suppressor grid; the high-speed electrons moving from the electron cloud fly past this grid so fast that its potential cannot block them.

**Beam Power Tubes.** The beam power tube is practically a pentode tube. However, instead of a suppressor grid, it uses a pair of beam-forming electrodes to get the same effect. Fig. 15 shows a cut-away view of such a tube.

The beam-forming electrodes are connected to the cathode, so they are negative with respect to the plate. They repel electrons, forcing the electrons to "bunch together" and pass through the openings between these electrodes to reach the plate. Thus, they force the electrons to form two streams or "beams." This concentrates the flow of electrons in a stream, actually producing a compact mass of them between the plate and the screen grid. There-

fore, any secondary emission electrons trying to move from the plate toward the screen grid are met at once by a large body of negative electrons (practically an electron cloud between the plate and the screen grid) that forces them back to the plate.

resistance of the 6K6G is about 50,000 ohms). Notice, too, that increasing the load resistance does not improve the shape of the characteristic. In fact, the opposite occurs; an increased load resistance increases the curvature at the right end of the dynamic curve.

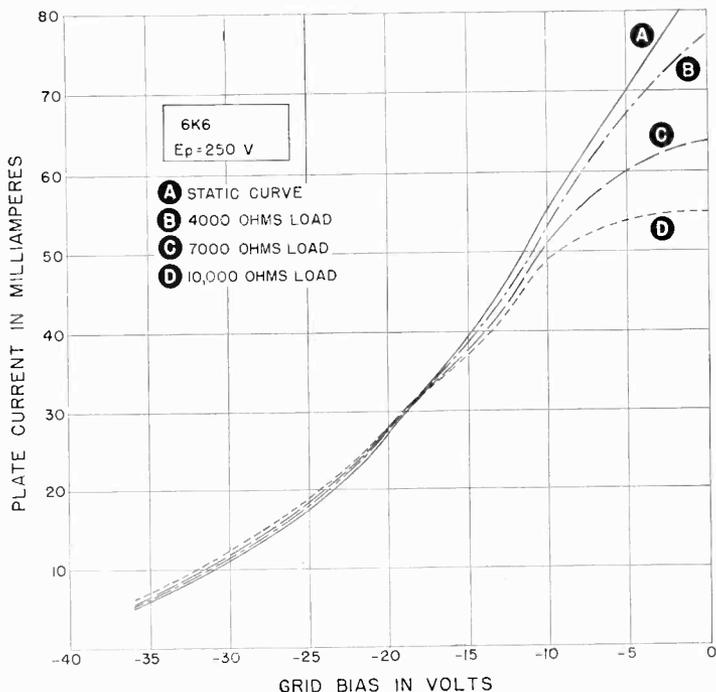


FIG. 16. These curves are typical of the static and dynamic curves of pentode power output tubes. Notice the upper bend curvature increases with load increases. Hence, the load must be kept low to keep down the third-harmonic distortion.

### PENTODE AND BEAM POWER DYNAMIC CHARACTERISTICS

The dynamic characteristics of pentode and beam power tubes are not as linear as are those of a triode. Fig. 16 shows the static curve for a typical pentode output power tube and also the dynamic curves for several load values.

Notice that the load values given in Fig. 16 are far below the plate resistance for the tube (the a.c. plate

This undesirable drop-off in the curve occurs because the plate voltage falls below a critical value. That is, the plate current of a pentode or beam power tube is relatively independent of the plate voltage until the load voltage drop causes the plate-cathode voltage to fall about 75 volts. Further reduction of the plate voltage causes the plate current to drop, too. Therefore, the plate current follows the grid voltage changes only up to this point where the plate voltage "takes over."



tion is used. Commonly used values range from .001 mfd. to .02 mfd.

Usually, the set is designed so that  $C_3$  and the output transformer resonate near the high-frequency end of the pass band and thus prevent some of the drop-off at these frequencies.

In some receivers,  $C_3$  may be connected directly across the primary of  $T_1$ , or else its lower end may go to the chassis instead of to the cathode. These changes do not affect the results as long as  $C_2$  and  $C_1$  are effective.

### PARALLEL OUTPUT TUBES

When more power output is needed than can be obtained from a single tube (and a more powerful tube is not available or is undesirable because of voltage requirements), more than one output tube must be used. One way in which two or more tubes can be used is to connect them in parallel, as shown in Fig. 17. The corresponding parts—plates, grids, and cathodes—of the two tubes are tied together.

Since the plate resistance of the two tubes in parallel is only half that of

one tube alone, the load used is likewise only half the value that would be used with a single tube. Both tubes furnish current, so the load current is doubled. The result is that the power output of this parallel arrangement is twice that of a single tube for the same applied signal voltage.\*

However, the parallel output stage is unstable (oscillations occur easily), and the fidelity is no better than that of a single output tube. When another arrangement, known as the push-pull connection, was developed, the parallel circuit was abandoned for receivers. The push-pull circuit provides even more power from two tubes than does a parallel connection and gives better fidelity. We'll study this circuit next. ► Incidentally, to distinguish between the push-pull and other connections, any power stage using a *single* tube or *parallel* tubes is known as a *single-ended* stage. The *push-pull* stage is called a *double-ended* stage.

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\* The power is  $I^2R$ , so doubling the current would give four times the power, but halving the load resistance gives us a net of twice the output.

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## Push-Pull Power Stages

The maximum power output that can be usefully obtained from a tube depends on the amount of distortion permissible. The push-pull circuit is particularly desirable because it eliminates all even harmonics that result from amplitude distortion in this stage. Thus, it has far better fidelity than does the single-ended output stage. This leaves only the odd harmonics so the power output can be increased until the odd-harmonic distortion level becomes objectionable. In fact, it is possible with the push-pull circuit to

obtain considerably more than twice the undistorted power that a single tube could deliver.

Essentially, a push-pull circuit is one in which the grids of two tubes are fed out of phase, and the resulting plate currents are recombined so that any *even* (second, fourth, etc.) harmonics *that are produced by distortion in the stage* are cancelled. (It is important to realize that any harmonics *that are part of the signal* will pass through the stage; it is only harmonics *added within the stage* that are can-

celled.) This circuit is used with triode, pentode, and beam power tubes; we'll take up each in order.

### TRIODE PUSH-PULL STAGES

The basic triode push-pull circuit is shown in Fig. 18. Tubes  $VT_1$  and  $VT_2$  are identical types. The plate supply is fed through the center tap of transformer  $T_2$ , so that the plate current of  $VT_1$  flows through one half of the transformer while the plate current of  $VT_2$  flows through the other half. Bias is obtained as a result of the flow of the combined plate currents through  $R_1$  from B— to the cathodes.

The grid circuits are completed through the secondary of transformer  $T_1$  to its center tap, where the return circuit goes through ground to  $R_1$ .

► Notice the direction of the electron flow in each plate circuit, as indicated by  $i_1$  and  $i_2$ . In each tube, electrons move from cathode to plate. Therefore, the two d.c. plate currents flow in *opposite directions* through halves of transformer  $T_2$ . Assuming we have identical tubes with the same bias and no signal, these plate currents are practically equal. Therefore, since the plate currents are equal in value, and flow in opposite directions through  $T_2$ , the fluxes produced by these *d.c. currents* cancel each other, and no resulting flux exists in the transformer core when there is no signal. (When current flows one way through a transformer, the flux lines in the core have a particular direction. When the current is reversed, the flux direction reverses. Therefore, if *equal* currents flow through equal windings in opposite directions, the two fluxes cancel each other.)

**Tracing The Signal.** Now let's assume that a sine-wave signal  $e$  is applied to the primary of transformer  $T_1$  in Fig. 18. A larger voltage will ex-

ist across the secondary, since this is a step-up transformer.

The induced voltage exists across the entire secondary winding of  $T_1$ . Therefore, when terminal 1 of the secondary of  $T_1$  is positive, terminal 3 must be negative, and vice versa.

► For the moment, let's assume that terminal 1 is positive. Then, the voltage  $e_1$  between terminals 1 and 2 is that applied to  $VT_1$ . This is half the total voltage, and at the moment, the grid end of the winding is positive. Therefore, at this instant, the positive signal voltage makes the grid of  $VT_1$  less negative (it subtracts from the bias voltage), so the plate current of  $VT_1$  increases.

At the same instant, terminal 3 is negative with respect to terminal 1. In fact, it is the most negative point on the winding, so the voltage  $e_2$  (between 2 and 3) is negative. This signal voltage adds to the bias, making the grid of  $VT_2$  more negative, so this tube's plate current decreases. Thus, you can see that the two grids are fed signals that are  $180^\circ$  out of phase, because when one is positive the other is negative.

Since on this half-cycle the current

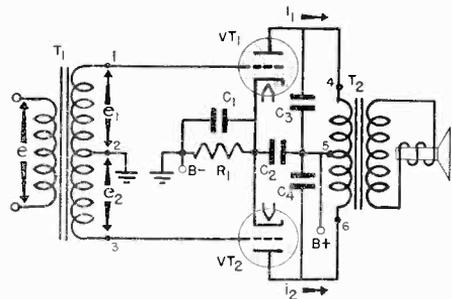


FIG. 18. A typical triode push-pull stage. In some cases you may find that by-pass condensers  $C_1$ ,  $C_3$ , and  $C_4$  may not be used in push-pull stages of this type. Condenser  $C_1$  is not required if the tubes are balanced (as you will see when you study inverse feedback), and  $C_3$  and  $C_4$  are needed more with beam power tubes than with triodes.

$i_1$  of tube  $VT_1$  is increased, and at the same time the current  $i_2$  of  $VT_2$  is decreased, we get an important action in the transformer  $T_2$ . This changing plate current has an a.c. component, so an a.c. voltage is produced across the primary of  $T_2$ . Let us say that terminal 4 is becoming more positive than 5, while 6 is becoming more negative than 5. Hence, terminals 4 and 6

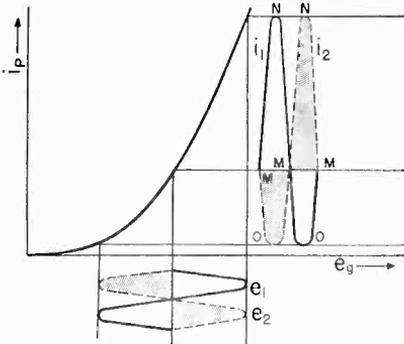


FIG. 19. When the operating point is on the lower bend of the tube's characteristic, the plate current pulses  $i_1$  and  $i_2$  will be distorted.

are opposite in polarity. *Insofar as the entire primary is concerned, therefore, the two voltages add together and act like a single a.c. voltage of twice the value of either component.* Also, the resulting a.c. current in the primary causes an a.c. flux in the core, which in turn induces a voltage in the secondary of  $T_2$ .

► In the next half-cycle the polarities of  $e_1$  and  $e_2$  are interchanged (but they stay  $180^\circ$  out of phase) so the two actions reverse: current  $i_1$  now decreases and  $i_2$  increases. This reverses the phase of the a.c. voltage across the primary of  $T_2$  and so reverses the direction of the flux in the core.

Since the a.c. voltage across the primary of  $T_2$  is produced by *two* tubes, it is twice the value produced by one tube; in other words, we have double the power output of a single tube.

However, it is possible to get more from this stage, because distortion is reduced to the point that a greater output is obtainable. Let's see how.

**Distortion Cancellation.** Let's again apply a sine-wave signal and operate near the lower bend of each tube's characteristic curve, as shown in Fig. 19. Considering first the grid signal  $e_1$  shown by the solid-line curve, the corresponding plate current curve is  $i_1$ —also shown by a solid curve. The plate-current wave shape is not equal in amplitude on the two sides of its zero axis—that is, the alternation  $M-N$  is greater than the alternation  $M-O$ —so we have added the second and other even harmonics to the original fundamental sine wave.

At the same moment the other tube is getting the grid voltage  $e_2$ . Therefore, as shown by the dotted lines in Fig. 19, its plate current  $i_2$  is distorted also on the  $M-O$  alternation. However, since  $e_1$  and  $e_2$  are  $180^\circ$  out of phase, the plate current alternation of  $i_1$  that is distorted occurs at the same moment as the portion of  $i_2$  that is *not* distorted (and vice versa).

► Now let's examine the action occurring in  $T_2$  when these plate currents flow in the primary. As shown in Figs. 19 and 20, the plate currents  $i_1$  and  $i_2$  are  $180^\circ$  out of phase. However, remember that they flow in *opposite directions* through the halves of the primary of  $T_2$ . Therefore, if  $i_1$  produces flux  $f_1$  as shown in Fig. 20, then  $i_2$  will produce the flux  $f_2$ . In effect, the reversal of plate current direction through the primary has resulted in  $f_2$  being "flipped over." And, as we explained earlier, the two fluxes add together as if they were produced by a single a.c. voltage across the entire primary winding. Hence, when  $f_1$  and  $f_2$  in Fig. 20 are added (thus,  $A-B$  plus  $C-D$  equals  $E-F$ ), the resultant flux

$f_c$  is NOT AS DISTORTED as are  $f_1$  and  $f_2$ . The second-harmonic distortion present in the plate currents has been eliminated by the action of the fluxes in  $T_2$ .

► Also, the same cancellation will occur on any other even harmonic, so no even-harmonic distortion of any consequence is produced in a push-pull stage. (IMPORTANT: Harmonics in

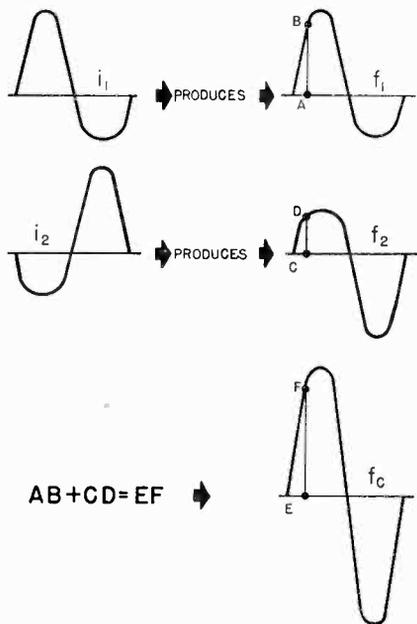


FIG. 20. The even-harmonic distortion cancellation occurs because of the manner in which the fluxes add in the output transformer.

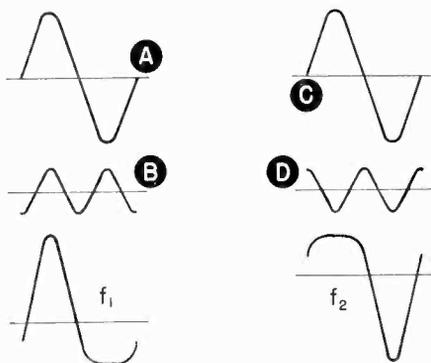
the original signal are not cancelled, because they act like any original signal applied to the push-pull stage. This action occurs only for harmonics added by the push-pull tubes.)

**Third Harmonics.** Third harmonics are not cancelled, because odd-harmonic distortion affects both halves of the flux cycle, and adding the two flux components would not remove the distortion. This is shown in Fig. 21. Flux  $f_1$  has a square-top shape (indicating odd harmonics have been added), and

$f_2$  has exactly the same shape. Adding the two together reinforces the odd harmonics instead of producing cancellation. The resultant  $f_c$  thus is just an enlarged version of its two components and contains just as many third harmonics (or other odd harmonics).

**Summary.** We can then say about push-pull circuits:

1. The d.c. plate currents are equal and opposite, so the d.c. fluxes in  $T_2$  cancel.
2. The a.c. plate currents are  $180^\circ$  out of phase, but since the currents flow in opposite directions through  $T_2$ , the fluxes add in the core of the transformer.
3. If even harmonics are added by amplitude distortion in the push-pull stage, they are wiped out when the fluxes are added in the output transformer.
4. If odd-harmonic distortion occurs in a push-pull stage, no cancellation exists—odd harmonics go through the



Perhaps the second-harmonic cancellation can be seen easier by considering the components of the fluxes. Flux  $f_1$  is the result of adding a fundamental A to a second harmonic B that is  $90^\circ$  out of phase with it. Similarly,  $f_2$  is the result of combining the fundamental C with the  $90^\circ$  out-of-phase second harmonic D. With  $f_1$  and  $f_2$  in the relationships shown, you can see that the fundamentals A and C are in phase and can be added directly. However, the second harmonics B and D are  $180^\circ$  out of phase with each other, so they cancel each other.

circuit in the same way as the fundamental wave.

► These facts are true whether the even and odd harmonics are the second and third or are the sixth and seventh—all *even* harmonics are eliminated, and *odd* harmonics are passed on.

Triode tubes have considerably more *even*-harmonic distortion than

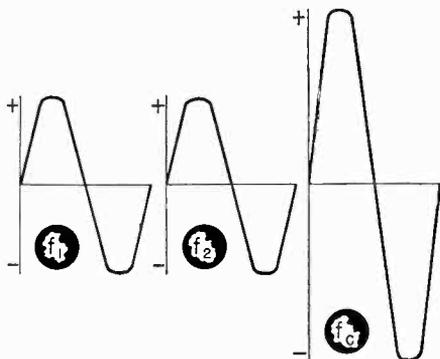


FIG. 21. Odd-harmonic distortion is not cancelled in push-pull circuits because both halves of the flux cycle are distorted, so when they are added, the combination must have the same distortion.

*odd*-harmonic distortion. Therefore, the push-pull circuit arrangement with triode tubes in a class A amplifier comes very close to the ideal in giving distortionless output.

**Increasing the Output.** The elimination of even-harmonic distortion in a push-pull stage permits us to get an undistorted output from the stage, that is more than twice as large as the undistorted output a single tube can give. We can get this greater output by doing two things:

1. We can make the load more nearly equal to the plate resistance.
2. We can adjust the bias so that the operating point is nearer the lower curved section of the characteristic, thus permitting a larger grid signal voltage, which in turn means a greater plate-current swing and more a.c. power.

Of course we could take both these steps with a single-tube amplifier, but doing so would increase the even-harmonic distortion in the amplifier. Such increased distortion also occurs in the push-pull stage, but the action of the stage promptly wipes it out; as a result, we get increased power without increased even-harmonic distortion. We can get about 2.5 times as much output power from a push-pull stage as from a single-ended stage before odd-harmonic distortion begins to become objectionable.

### PENTODE AND BEAM POWER PUSH-PULL STAGES

Pentode and beam power tubes are used in push-pull arrangements in the same way that the triode is. As shown in Fig. 22, the only difference is the substitution of a pentode (or a beam

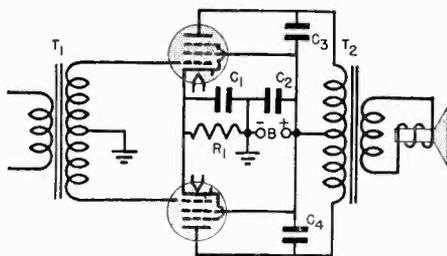


FIG. 22. A pentode push-pull power output stage.

power) tube in place of each triode. The operation of the stage is essentially the same.

However, pentode and beam power tubes have greater amounts of distortion, particularly third-harmonic distortion. Therefore, with push-pull stages using these tubes, it is common practice to use a low-resistance load.

As was shown for the 6K6G tube in Fig. 16, a small load causes less odd-harmonic distortion because it causes less of a bend in the dynamic curve

at the right-hand end. Such a load may cause more bend in the lower part of the characteristic, and thus cause more even-harmonic distortion—but we don't care; the even-harmonic distortion is eliminated by the push-pull arrangement.

In practice, a load resistance value is chosen to give all the power possible at the permissible value of odd-harmonic distortion. If, say, 5% third harmonic is permissible, the load resistance value is increased (thus increasing the power output) until the third-harmonic distortion reaches this value.

### CLASS B PUSH-PULL STAGES

The circuits we have just discussed have been adjusted for class A operation. The operating point is chosen so that the grid will never be swung positive, nor will the plate current ever be zero.

However, where high power and high efficiency are required, a *push-pull* circuit can be operated as a class B amplifier (that is, operated with a bias set at the plate current cut-off point) without creating excessive distortion. As Fig. 23 shows, operating the tubes with the bias at the cut-off point causes the plate currents to be half-wave pulses. Then (Fig. 24) the flux produced by the other will fill in the spaces. The combination produces the complete a.c. flux cycle  $f_c$ . In other words, one tube works on one half-cycle, and the other tube works on the other half-cycle in the class B push-pull stage.

The class B stage has an increased efficiency because each tube operates for only half of each cycle and passes practically no current the other half-cycle. Thus, there is no steady d.c. plate current to cause a constant power loss in the plate resistance of the tube.

This is important because the amount of power that can be delivered from a stage depends on the ability of the tube to dissipate the heat developed within it. Thus, if a tube can dissipate 5 watts safely, we can increase the output only to the point where this power is lost in the plate resistance. Therefore, if we reduce the loss, we can safely increase the grid signal voltage to where the same loss occurs.

This is exactly what happens in a class B stage. Reducing the average plate power by causing the plate current to exist for only half the time permits the grid signal level to be increased (the grid can even go slightly positive). In turn, this produces high plate current pulses, so a large amount of a.c. power is delivered to the load. Since operation requires a specially designed voltage amplifier, the entire system must be designed for it. Fur-

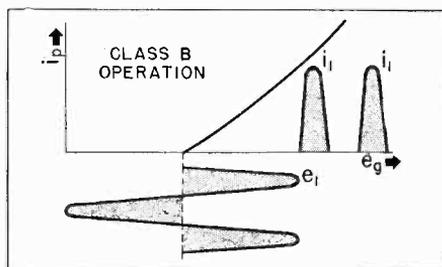


FIG. 23. Class B operation is at the plate current cut-off point, so the plate current pulses exist for half cycles.

thermore, class B operation in the low-frequency amplifier is possible ONLY in a push-pull circuit—otherwise half the signal would be destroyed.

► On paper, a class B stage looks much like a class A push-pull circuit. However, a fixed bias from the power supply is always used instead of self-bias, unless special zero-bias class B tubes are used. (Some class A stages use fixed bias also.) With a fixed bias

instead of  $R_1$ , Fig. 22 could be a class B stage, providing the following parts differences are considered:

Transformer  $T_1$  will be a step-down transformer, so its secondary will have low reactance. This is necessary because the grid is driven positive part of the time in class B operation, causing a grid current flow. If the trans-

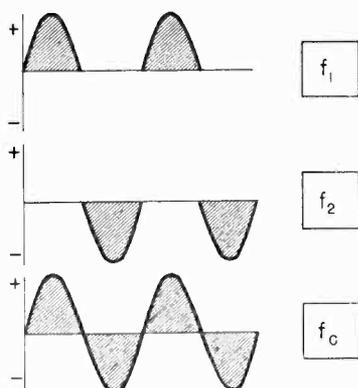


FIG. 24. As shown here, class B push-pull operation causes the tubes to supply current pulses alternately. The corresponding fluxes add to reproduce the entire a.c. cycle.

former had high reactance, the grid current flow through it would cause a voltage drop in the transformer, that would distort the peaks of the applied signal. Since the grids draw current, the grid circuit requires power—it is not just voltage-driven. Therefore, the

preceding stage must be capable of furnishing this power, and the transformer  $T_1$  must be an impedance-matching type. A low-power tube (in a class A circuit) is frequently used ahead of a class B amplifier. This tube is known as the “driver” tube, because it *drives* the class B stage to its normal operating point.

The class B amplifier delivers more power than is usually necessary in a home receiver. For this reason, true class B amplification is rarely found except in public address systems or in other applications requiring high power. However, class AB operation, in between class A and class B amplification, is found in many of the more powerful home receivers.

### AB OPERATION

Class AB operation uses a C bias somewhere in between the cut-off bias of class B operation and normal class A bias. Then, if the driving voltage is limited so that the grid never becomes positive, the operation is known as class  $AB_1$ . On the other hand, if the signal voltage applied to the grid is permitted to go slightly positive (approaching class B operation) the operation is known as class  $AB_2$ . Class  $AB_2$  operation delivers higher power than  $AB_1$  but less power than does class B.

## Inverse Feedback

The output of any amplifier using pentode or beam power tubes contains a considerable percentage of distortion. One effective method of reducing this distortion (and also reducing any excessive hum, oscillation, or noise) uses the principle of inverse feedback.

Feedback, as the name implies,

means feeding part of the output voltage of an amplifier back into its input circuit. If the feedback voltage arrives at the input exactly in phase with the original input voltage, we have positive feedback, often called *regeneration*. If positive feedback is carried far enough in an amplifier, the amplifier becomes

unstable and finally oscillates.

For *inverse feedback* or *degeneration*, the phase relation between input and feedback voltages is such that the feedback voltage is made to arrive  $180^\circ$  out of phase with the input signal. This means that whenever the input signal rises in a *positive* direction, the feedback voltage rises in a *negative* direction, and vice versa. The feedback voltage, therefore, always opposes the original input signal. The resultant input is the difference between the two voltages, so it is always smaller than the original signal.

Since inverse feedback *reduces* the effect of the original input voltage, it reduces the output also. This amounts to a reduction in the amplifier gain. (An important point to remember is that unlike positive feedback, which makes an amplifier unstable and sometimes causes oscillation, inverse feedback drops the gain and output, making the amplifier entirely stable and always under control.)

So far, the only result of inverse feedback we have seen is a reduction in gain. Let us see what benefits this sacrifice may offer.

## REDUCTION OF DISTORTION

Let us examine the circuit in Fig. 25. Except for the inverse feedback connections, it is a conventional single-ended power amplifier. Note that the lower end of input transformer  $T_1$  is not returned to ground, but is connected to voltage divider  $R_1$ - $R_2$  across the secondary of output transformer  $T_2$ . (Resistors  $R_1$  and  $R_2$  are large in value compared to the impedance of the loudspeaker voice coil, so they absorb very little power from the circuit.) With this arrangement, part of the output voltage is fed back into the grid circuit; the exact amount of voltage can be regulated by adjusting the

values of resistors  $R_1$  and  $R_2$ .

Now, what happens when an input signal is applied to the circuit? First, the input signal  $e_1$  produces in the secondary of  $T_1$  the grid voltage  $e_2$ , which after amplification appears in the output as  $e_0$ . For simplicity, let us assume that although  $e_2$  is a pure sine-wave input voltage as in Fig. 26A, the output  $e_0$  looks something like that in Fig. 26B. The "bump" on the output wave may represent noise, distortion, or hum (or all three) generated in the amplifier tube. The bump thus represents something that should be eliminated, because it was not present in the original signal.

Through our feedback circuit, a portion ( $e_3$ ) of the distorted output  $e_0$  is fed back to the grid in series with  $e_2$ . This  $e_3$  voltage is indicated by the dotted curves in Figs. 26B and 26C.

For our purpose, we can assume that all these things occur instantaneously;

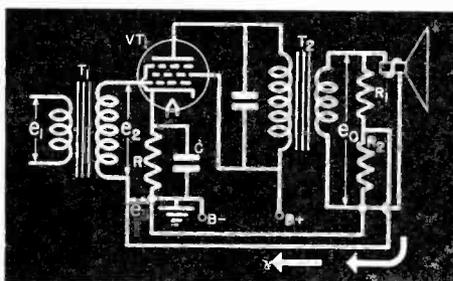


FIG. 25. A typical single-ended power output stage utilizing inverse feedback. The portion of the output voltage across  $R_2$  is fed back as  $e_3$  into the grid circuit.

that is, there is no delay between the time  $e_2$  is applied to the grid, and the instant that  $e_3$  is fed back into the grid circuit.

Suppose we have connected the windings of output transformer  $T_2$  so that feedback voltage  $e_3$  arrives exactly  $180^\circ$  out of phase with input voltage  $e_2$ , as shown in Fig. 26C. This

means we have inverse feedback. Since  $e_2$  and  $e_3$  are of opposite polarity, the resultant input is equal to their difference. This input is illustrated by the curve in Fig. 26D which is obtained by subtracting  $e_3$  from  $e_2$  in Fig. 26C.

The final output caused by the new reduced input will be something like that in Fig. 26E. It is apparent that

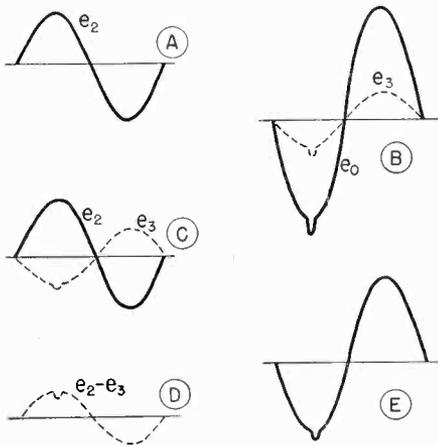


FIG. 26. These curves show how inverse feedback tends to correct amplitude distortion.

this is a much better reproduction of the original input than that obtained in Fig. 26B.

*Through the use of inverse feedback we have taken the amplitude distortion components of the output and put them back into the input in such a direction that they tend to cancel themselves.*

It is true that the output is reduced in amplitude in the process, because of the reduced effective input voltage (26D is smaller than 26A, so 26E is less than 26B,) but if the input signal voltage ( $e_1$  in Fig. 25) is increased, the output can be brought back to its former level while the distortion components remain substantially reduced.

The amount of reduction of all distortion, hum, and noise by inverse feedback depends upon the amount of output voltage fed back to the grid cir-

cuit and upon the gain of the amplifier without feedback. In general, the gain of an amplifier and the distortion in its output caused by the amplifier itself are reduced by an equal per cent. This means that if feedback voltage is increased until the gain has been reduced 50%, distortion will be reduced 50% also. If the input voltage is then doubled to return the output to its former value, the distortion in the stage will also double, but it will still be only 50% of what it would be without feedback.

► Suppose we have an amplifier that without feedback, has the frequency-gain characteristic curve 1 in Fig. 27. Let us say the poor response below 100 cycles is caused by coupling condensers that are too low in capacity, the peak at 3000 cycles by resonance effects in a transformer, and the loss of frequencies above 10,000 cycles by the shunting effect of tube capacities. Obviously, an amplifier with such a response would not give high-fidelity reproduction.

Now let us see what improvements inverse feedback might make. With a medium amount of output voltage fed back into the input in proper phase, we get response curve 2 in Fig. 27. This is a considerable improvement. Observe that although the over-all gain has been reduced, the low-frequency and high-frequency response is nearer the response of the intermediate range,

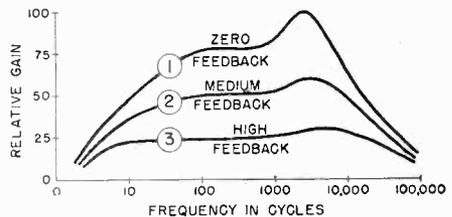


FIG. 27. Inverse feedback can also improve the frequency response, providing the gain can be sacrificed.

and the undesirable peak at 3000 cycles has been cut down.

How feedback brings this about may be visualized as follows: Below 100 cycles and above 10,000 cycles, where the amplifier's inherent gain is low, the output voltage also is low. This means the feedback voltage is small in magnitude, so the input is not reduced much. For the peak at 3000 cycles, however, the gain originally is high. The increased output voltage makes the feedback voltage quite large. When this is fed back, it cancels a large part of the original input and cuts the gain considerably. Therefore, the feedback corrects for frequency distortion as well as for amplitude distortion.

Even greater improvement in frequency response can be obtained if the value of the feedback voltage is increased, as in curve 3 of Fig. 27. Observe, however, that any increase in feedback is always accompanied by an additional reduction in gain. In-

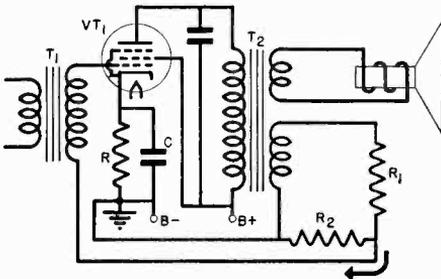


FIG. 28. Inverse feedback from a special winding on the output transformer.

verse feedback may be increased only to the point where it is impossible to get enough driving voltage (or where it becomes uneconomical to go further).

► The remarkable ability of inverse feedback to reduce distortion has resulted in its widespread use in modern receivers that use pentode and beam power tubes, particularly those using

single-ended stages. Let's study some of the typical circuits.

### TYPICAL INVERSE FEEDBACK CIRCUITS

In addition to the basic circuit you have just studied, there are a number of variations on the method of getting the required feedback. It is generally

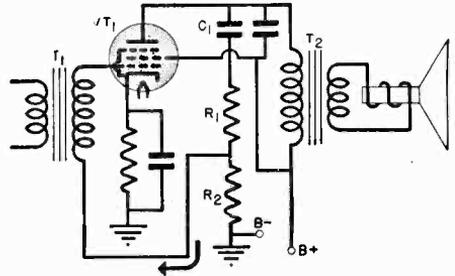


FIG. 29. Here the feedback is obtained from a voltage divider connected from plate to chassis.

unnecessary to feed back the entire output voltage to the grid circuit, and for this reason almost all feedback circuits use some form of voltage divider to reduce the feedback voltage. In the circuits shown in Figs. 28 to 32, the voltage divider resistors are marked  $R_1$  and  $R_2$ . The exact amount of feedback can be regulated by varying the resistance values.

► The output voltage developed across the average low-impedance loudspeaker voice coil (in a circuit like Fig. 25) does not always yield sufficient feedback voltage for satisfactory correction of distortion. However, an extra secondary winding having a greater number of turns can be added to the output transformer as in Fig. 28. This increases the feedback voltage with correspondingly better results.

► To get a high feedback voltage without an extra transformer winding, the voltage often is taken directly from

the plate of the tube (Fig. 29). Condenser  $C_1$  serves only as a blocking condenser to prevent short-circuiting the d.c. plate supply and prevent application of the plate voltage to the grid. The capacity of  $C_1$  should be high enough so that the reactance of the condenser is small compared to the resistance of  $R_1$  and  $R_2$ , otherwise the feedback will not be the same for all frequencies.

► The feedback voltage may be fed in *parallel* with the input instead of in series. This is usually done with resistance-coupled input circuits like that shown in Fig. 30. Note that  $R_2$  is not only part of the feedback voltage divider circuit, but it is also the grid resistor for the tube.  $C_1$ , as before, is only a blocking condenser.

► For feedback over two stages, the circuit in Fig. 31 is often used. Observe that the feedback voltage is inserted in the cathode circuit. The feedback could not be applied to the grid of the first tube because we have a phase

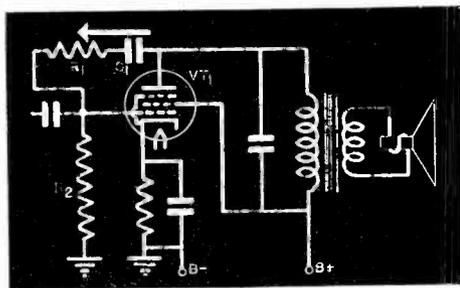


FIG. 30. The grid resistor  $R_2$  divides the feedback voltage with  $R_1$ .

shift of  $180^\circ$  in each tube, and with two tubes this makes a total shift between input and output of  $360^\circ$ . The voltage at the grid of the first tube, therefore, is exactly in phase with the output voltage, and coupling the feedback to the first grid would result in positive feedback.

► Feedback, however, can be applied to the grid of the first tube in a two-tube amplifier if an extra output transformer winding is used. Fig. 32 is an example of this. In this case, the connections to the extra transformer winding may be reversed, if necessary, for proper phasing.  $R_3$  serves only to load the transformer winding, and has no direct effect on the feedback voltage.

**Current Degeneration.** All the preceding systems are known as volt-

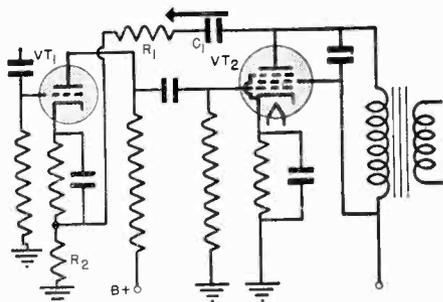


FIG. 31. Inverse feedback from the plate of the output tube to the cathode of a voltage amplifier.

age types, since it is the output *voltage* that is fed back. This system thus corrects the output voltage and hence reduces both frequency and amplitude distortion. There is another system which depends on the output tube *plate current*. This current type can correct for amplitude distortion but does not correct the frequency response.

A typical circuit is shown in Fig. 33. This is a conventional amplifier except that the bias resistor by-pass condenser has been omitted. With no condenser, the plate load consists of both the output transformer and the cathode resistor  $R_2$ , and a small part of the output voltage is developed as  $e_3$  across the cathode resistor. This voltage acts as a variable bias that opposes changes in the grid signal. When the grid voltage  $e_2$  goes positive, the increase in plate current through  $R_2$  increases the

bias ( $e_3$ ) and tends to make the grid more negative. When the signal swings negative, the bias  $e_3$  is reduced. Hence, it has the same effect as the inverse feedback circuits already described.

Since this circuit acts to minimize changes in *plate current*, it can correct amplitude changes but not the frequency response, because the same

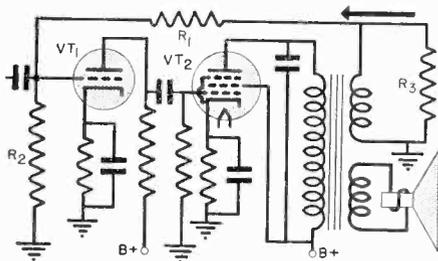


FIG. 32. Feedback from a special output transformer winding to the grid of a voltage amplifier.

plate current amplitude can cause differing output voltages if the load impedance changes with frequency.

### FEEDBACK LIMITATIONS

It has been assumed that the feedback voltage is always  $180^\circ$  out of phase with the input voltage in these feedback circuits. Unfortunately, this is not always true. In addition to the  $180^\circ$  phase shift in each tube, there is always some phase shift in each coupling condenser or audio transformer. These extra phase shifts are quite small at medium audio frequencies, but at extremely high and extremely low frequencies they can become quite large. When more than two stages are

included in the feedback path, it is possible that the net change in phase could result in positive instead of negative feedback, causing oscillation. Therefore, the design of these circuits must be carefully worked out. (It is necessary that any defective parts in these circuits be replaced by exact duplicates to avoid possible trouble of this kind.)

► All the inverse feedback systems shown here can be applied to push-pull stages as well. Incidentally, the cathode type of Fig. 33 introduces no degeneration in a push-pull stage (when the same resistor biases both tubes) unless the tubes become unbalanced. If the two tubes are the same, when one

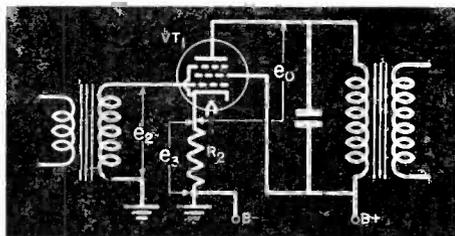


FIG. 33. Current degeneration is obtained when the cathode by-pass condenser is omitted. This will correct for amplitude distortion but not necessarily for frequency distortion.

plate current is increasing the other is decreasing, so the drop across the bias resistor is practically constant. Therefore, there will be no degeneration until the circuit is upset and needs it the most! For this reason, you will frequently find that the bias resistor for push-pull tubes is not by-passed.

# Phase Inverters

In our study of push-pull amplifiers, you undoubtedly noticed the use of two transformers. This was the standard circuit at one time. Modern practice, however, is to eliminate audio transformers whenever possible, since high-quality ones are bulky and expensive, and cheaper ones do not give faithful frequency response.

We can't, of course, get rid of the impedance-matching *output* transformer. (Nor, in class AB<sub>2</sub> and class B operation, can we avoid using the input transformer, for no other device can perform its two functions of offering low impedance and of matching impedances to the driver.)

However, in class A and class AB<sub>1</sub> operation, the *input* push-pull transformer is an ordinary voltage amplifying transformer whose only special function is providing a way to feed the two grids 180° out of phase. It can be eliminated if we can find some other way of doing this.

As you know, a vacuum tube stage reverses the phase of a signal 180° because the signal voltage across the load is opposite in polarity to the signal applied to the grid of a tube. Therefore, if we feed the grid of one of the push-pull tubes from a voltage amplifier, and then put in another tube (called a *phase inverter*) to reverse the phase to feed the other push-pull tube, we can have push-pull operation (class A) without using an input transformer.

A resistance-coupled stage is used as the phase inverter. Its use offers several advantages: higher fidelity characteristics, a reduction of the possibility of magnetic coupling and hum pickup, and a reduction in weight of the equipment. In general, it is cheaper to use a tube stage than the relatively expensive input push-pull transformer.

Of course, in addition to reversing the phase, the additional stage must not feed a greater signal to its push-pull tube than is fed to the other. Therefore, it is necessary to limit the gain so that the inverter tube will not have greater output than the voltage amplifier.

► Fig. 34 shows a simplified diagram of an inverter. *VT*<sub>1</sub> is a voltage amplifying stage. It amplifies the signal, and feeds tube *VT*<sub>2</sub> through the resistance coupling *R*<sub>3</sub>-*C*<sub>4</sub>-*R*<sub>4</sub>-*R*<sub>5</sub>. A voltage di-

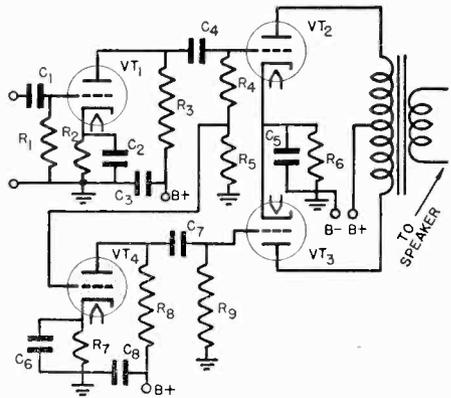


FIG. 34. A basic phase-inverter stage.

vider *R*<sub>4</sub>-*R*<sub>5</sub> is used to feed a portion of the output voltage from *VT*<sub>1</sub> to the phase inverter tube *VT*<sub>4</sub>.

The tube *VT*<sub>4</sub> amplifies in the normal manner. However, the voltage divider *R*<sub>4</sub>-*R*<sub>5</sub> is figured out so that if this stage has a gain of, say, 10, then only one-tenth of the voltage across the *R*<sub>4</sub>-*R*<sub>5</sub> combination is fed to the tube. Hence, there is effectively no gain in the *VT*<sub>4</sub> stage (10 times 1/10 is 1), and the voltage across *R*<sub>9</sub> is the same amount as the voltage across *R*<sub>4</sub>-*R*<sub>5</sub>, so tube *VT*<sub>3</sub> gets the same amount of signal voltage as does tube *VT*<sub>2</sub>. (When servicing this type of phase inverter, it is important to replace *R*<sub>4</sub> and *R*<sub>5</sub> with

exact duplicate parts when these resistors are found defective. This is necessary so that the proper voltage division will be obtained to counteract the gain of  $VT_4$ .)

What about the phase? When the signal in the grid circuit of  $VT_2$  is going positive, this same signal is fed to the grid of  $VT_4$ . The phase reversal of  $VT_4$  causes the grid of  $VT_3$  to be driven

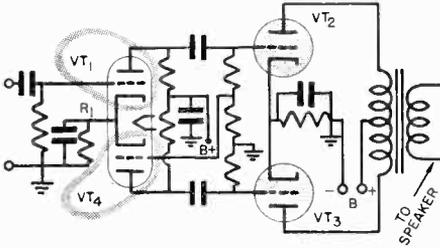


FIG. 35. The phase inverter and the driver are sections of the same dual tube.

negative at this same moment—just what we want to happen.

► Fig. 35 shows another arrangement using the circuit of Fig. 34. Tubes  $VT_1$  and  $VT_4$  are individual sections of a dual-triode tube. The same bias resistor ( $R_1$ ) is used for both sections, a practice that tends to equalize any slight irregularities in the characteristics of  $VT_1$  and  $VT_4$ .

### DEGENERATIVE PHASE INVERTERS

There is danger that aging of the tubes may upset the voltage balance between the outputs of  $VT_1$  and  $VT_4$  even when they are part of the same tube. If an unbalanced voltage is fed into the push-pull circuit, we will get a distorted output, as the even harmonics will not balance out.

One way around this is to use a single tube to feed both the push-pull tubes, as shown in Fig. 36.

First, let's examine the circuit of  $VT_1$ . Tracing the direction of electron

flow from cathode to plate, we find that the end 3 of  $R_4$  is negative with respect to terminal 4. Continuing around through the B supply, and through  $R_3$  back toward the cathode, we find that terminal 2 of  $R_3$  is negative with respect to terminal 1.

Now, let's consider terminal 4 of  $R_4$  to be grounded through  $C_4$  (insofar as a.c. signals are concerned), so that it connects effectively to terminal 2 of  $R_3$ . Comparing the voltages across these two resistors, we find that an increase in plate current makes terminal 3 of  $R_4$  more negative at the same instant that it makes terminal 1 of  $R_3$  more positive. Therefore, if we feed  $VT_2$  from terminal 3 of  $R_4$ , and feed tube  $VT_3$  from terminal 1 of  $R_3$ , the two tubes will be fed out of phase.  $R_3$  and  $R_4$  must be equal in size, of course, so that the voltages fed the push-pull tubes will be equal.

► Since  $R_3$  is part of the plate load, but is also in the input circuit in such

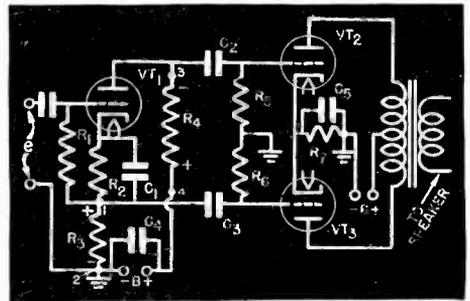


FIG. 36. Both push-pull tubes may be fed from a single tube in this unique phase inverter stage.

a way that the voltage developed across  $R_3$  opposes the input signal voltage, this circuit is degenerative. For example, if the stage gain is 50, then 1 volt across  $R_1$  (the grid resistor) will appear as 25 volts across  $R_4$  and  $R_3$  respectively. (25 plus 25 is 50, the a.c. voltage we would get from a gain of 50.) However, the 25 volts across  $R_3$

must be subtracted from the voltage  $e$  applied at the input terminals, so to get 1 volt across  $R_1$ , we must apply an  $e$  of 26 volts. Hence, the over-all gain of  $VT_1$  is only about 2 ( $50 \div 26$  is about 2). As only half the output voltage is supplied to each of the push-pull tubes, the effective gain is one, so the input voltage  $e$  we supply must be the same as the voltage necessary to drive the grid of one of the push-pull tubes.

As the tube  $VT_1$  ages, the voltage across  $R_3$  and  $R_4$  will tend to decrease, but this lets a slightly increased amount of voltage appear across grid resistor  $R_1$ , and conditions are restored approximately to normal. Furthermore, even if the input voltage increases to large amounts, the resulting increase in voltage across  $R_3$  prevents overload. Thus, all the advantages of degeneration are added to this phase inverter stage.

► The same idea of degeneration is also used in what is called a self-bal-

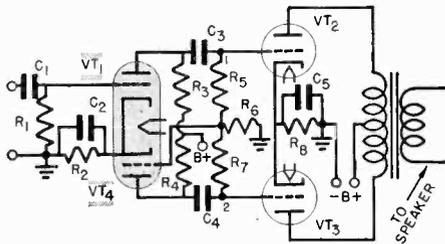


FIG. 37. By making a slight change in the grid feed circuit of the phase inverter stage, a balanced phase inverter, including degeneration, is obtained.

ancing inverter using two tubes. The circuit is shown in Fig. 37. At first glance, it appears to be the same as that of Fig. 35, but notice the position of resistor  $R_6$ . This resistor is placed so that degeneration occurs, and its value is increased to equal that of  $R_5$  (and  $R_7$ ).

Half the voltage between point 1 and ground (point 1 is the grid terminal of  $VT_2$ ) is applied to the grid

of  $VT_4$ . The amplified plate voltage of  $VT_4$  then appears between point 2 and ground; half of it is across  $R_6$ , and is fed back into the grid circuit of  $VT_4$ . Since (as in the circuit of Fig. 36) the output voltage is fed back to the grid of  $VT_4$ , the effective gain is only 2 in-

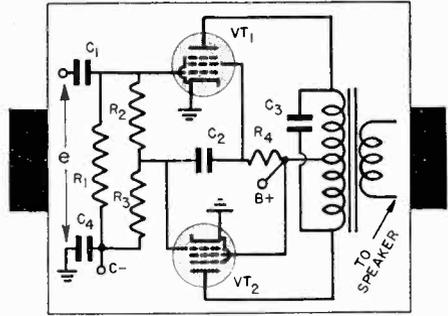


FIG. 38. A special phase inverter circuit used on certain Philco receivers. One of the output tubes is used as the source of the grid voltage for the other.

stead of being the natural gain of the stage (and, since  $VT_4$  gets only half the  $R_5$ - $R_6$  voltage, the gain from point 1 to point 2 is only one—the two voltages are equal). However, this gain is practically independent of the characteristics of tube  $VT_4$ , so the tube can age without upsetting the circuit.

► A somewhat unique phase inverter circuit is shown in Fig. 38. This is used in certain Philco receivers.

Here, the input signal  $e$  is fed directly to the grid of one of the push-pull tubes ( $VT_1$ ) through condenser  $C_1$ . The grid resistor for this tube is  $R_1$ . This tube amplifies the signal in the normal way.

Inserted in the screen grid circuit of tube  $VT_1$  is a small resistor  $R_4$ . This acts as a load resistor, in that a certain amount of the signal energy developed by  $VT_1$  appears across this resistor. In other words, the screen grid acts like a triode plate, and taps off a certain amount of the signal energy, with resistor  $R_4$  acting as its load. The

signal developed across  $R_4$  is fed to the grid of  $VT_2$  through  $C_2$ . Resistor  $R_3$  is the grid resistor for this tube.

Thus, the tube  $VT_1$  serves a dual purpose. It is one of the tubes in the push-pull amplifier, and also acts as the phase inverter for tube  $VT_2$ .

Considerable degeneration is necessary in this circuit to prevent distortion. This is furnished by resistor  $R_2$ , which is a resistance of higher value than either  $R_1$  or  $R_3$ . Through it, some of the energy fed back through  $C_2$  is applied to the grid of  $VT_1$ , opposing the original signal and causing degeneration in this tube. In addition, some

of the input signal  $e$  is fed through  $R_2$  to  $VT_2$  so that it is out of phase with the  $C_2$  voltage. Hence both tubes undergo a certain amount of degeneration. However, degeneration is not carried as far as in the phase inverters we previously described, because nearly normal performance is wanted from  $VT_1$ . Just enough is used to give reasonable fidelity.

► There are a number of other less commonly used phase inverter circuits. However, they all work essentially like the basic ones we have just described, so you will have little trouble in understanding them.

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## Facts About Low-Frequency Amplifiers

This lesson is an important milestone in your Course. Here, you gain a complete understanding of one of the major kinds of amplifiers. This lesson might also be considered a summary of your previous studies—because you use facts from all the earlier lessons. You put together your knowledge of how condensers and tubes work, facts you have learned about basic amplifiers, your knowledge of sound waves and their characteristics, and your knowledge of the kinds of distortion. So many important facts are in this lesson that you should come back to it from time to time and quickly run through it to refresh your mind on important circuit and stage actions.

► As a summary of the highlights of this lesson, the following facts will be helpful. If any are not clear in your mind, then it is advisable for you to go back over the lesson right now, because you will need these important facts for later lessons.

The low-frequency amplifier deals with the intelligence signal—audio or

video. In sound receivers, the audio signal must be increased in power to operate the reproducer properly, so a power output stage is used. However, power stages require considerable voltage to drive them, so voltage amplifiers are used to increase the output of the demodulator so that it can operate the power output stage properly.

The voltage amplifiers are named according to the devices used to couple the stages together (or to couple to the power output stage). Hence, voltage amplifiers are known as resistance-capacitance-coupled, impedance-coupled, and transformer-coupled stages.

Of these, the resistance-capacitance-coupled stage offers low gain but is capable of the best fidelity characteristics. It is possible to get a frequency range wide enough to make frequency distortion of minor importance. Amplitude distortion can be kept within reasonable limits also, and phase distortion is of importance only in television. (You will learn more about trouble with phase distortion as you study

television receivers in detail.)

When the voltage amplifier has built up the voltage sufficiently, this voltage is used to drive a single-ended or a push-pull power output stage.

► In the single-ended power stage, a single power tube (or tubes in parallel) will be found. The triode offers the best fidelity, but it requires a much higher signal voltage. For this reason, most modern receivers use pentode or beam power tubes. Both the pentode and the beam power tubes have higher power sensitivity than the triode (they require a much smaller grid signal voltage). For example, earlier radio receivers used two or sometimes three voltage-amplifier stages to drive a single triode power output stage. With modern receiver design, a somewhat greater output is obtainable from the demodulator, so it is possible for a single voltage-amplifier stage to give enough voltage to drive pentode or beam power stages.

You may wonder why both pentode and beam power tubes are used when they seem so similar. The beam power tube is somewhat more efficient than the pentode, and does not introduce quite as much third-harmonic distortion because its characteristic curve is shaped in a slightly different way. The pentode tube was introduced first, and is still used for low-power outputs. However, the more efficient beam power tube has replaced the pentode in the majority of receivers.

In power output stages, the high frequencies may be cut down somewhat by the necessity for a by-pass condenser across the plate load, and the low-frequency response depends upon the primary inductance of the output transformer. Even so, it is possible to make the frequency response of these stages fairly good.

However, the pentode and the beam

power tubes introduce considerable *amplitude* distortion. In the single-ended stage, much of the amplitude distortion can be corrected by inverse feedback, and the push-pull output stage has the important advantage of eliminating all even-harmonic distortion that originates in the output stage. Inverse feedback can be used for push-pull stages also, and will cut down the odd-harmonic distortion.

► In order for a push-pull stage to function properly, the grids of the two tubes must be fed  $180^\circ$  out of phase. This requires a special input transformer or a phase inverter tube stage. The input transformer has been practically eliminated from modern receivers because of its undesirable characteristics and rather high cost. Instead, a phase inverter is almost universally used to feed the grid of one of the push-pull tubes.

The two push-pull tubes must be fed with input voltages of the same amplitude. When a phase inverter stage is used, one of the push-pull tubes is fed directly from a voltage amplifier stage. A fraction of the output of this stage is also fed to the phase inverter stage. There this fractional voltage is amplified and inverted in phase; then it is fed to the other push-pull tube. The amplification received in the phase inverter stage is just enough to make the two push-pull grid voltages equal. (A single tube may feed both push-pull tubes if its load is split between its plate and cathode circuits so that phase inversion occurs.)

► Inverse feedback is a system in which part of the output energy is fed back to some point in an amplifier where it will be out of phase. Any distortion or other unwanted component that originates between these points (the output and the point where the feedback is returned) will be fed back

out of phase and thus will tend to cancel itself. Therefore, inverse feedback is particularly used with pentode and beam power tubes to correct their amplitude distortion. Of course, inverse feedback lowers the gain of the stage or stages within the feedback path, so a higher input voltage must be supplied at the input terminals. For this reason, inverse feedback is usually limited to the power output stage and perhaps to one preceding stage. Another voltage amplifying stage must be used ahead of the inverse-feedback circuit to deliver the required input voltage. Naturally, amplitude distortion must be controlled in this voltage amplifier.

The results of inverse feedback are such that a beam power output stage utilizing this circuit can deliver practically the same quality of output as the triode tube and still require somewhat less input voltage than would a triode tube giving the same power output.

► In modern radio receivers, you will recall that there are many kinds of power supplies. Some receivers operate from the power line and use a power transformer. It is almost universal to find a push-pull stage in such sets when they are intended to give good fidelity. On the other hand, if the receiver is an a.c.-d.c. type or is battery operated, there is a limit to the amount of voltage and current available. For this reason, such sets commonly use single-ended power output stages and smaller loudspeakers, which require less power.

In a television set, there may be from one to three voltage-amplifier stages. The television cathode ray tube—the tube on whose face the picture is reproduced—is the “output” stage in a television set. This tube is entirely voltage driven—very little power is needed to modulate the electron stream

within it. Therefore, in a television set, the low frequency (or video) amplifier consists primarily of voltage amplifying stages.

► As a radio serviceman, you are interested in all these stages because the parts are subject to natural breakdowns—condensers open, short-circuit, or change in value; resistors open or burn out; transformers short-circuit and open-circuit; and tubes change their characteristics with age. You need to know just how these amplifiers operate so that you can intelligently choose the proper replacement part.

Of course, when you have the manufacturer's instructions, or the part values are marked, you need only use parts like the originals to get the same results. However, there will be plenty of instances in which you will have to make a replacement and will not know the exact values of parts. Then you can be guided by your knowledge of what will happen if you use a resistor or a condenser that is larger or smaller than the original. Also, your knowledge of how standard circuits work will always be helpful when you meet non-standard circuits.

► You are now in a much better position to judge the fidelity that would be expected from different resistance-coupled amplifiers. However, always remember that some manufacturers make up for the deficiencies of one stage by purposely changing the response of another. Therefore, don't make the mistake of assuming that the response of the amplifier is necessarily poor because the design of one stage happens to sacrifice low frequencies or high frequencies. Another stage may have an excess response to these frequencies, so that the final result of the combination may be a relatively good output.

# Lesson Questions

Be sure to number your Answer Sheet 15FR-2.

Place your Student Number on every Answer Sheet.

Send in your set of answers for this lesson immediately after you finish them, as instructed in the Study Schedule. This will give you the greatest possible benefit from our speedy personal grading service.

1. Does the low-frequency amplifier obtain its input signal directly from: 1, the demodulator; 2, the i.f. amplifier; or 3, the power supply?
2. If the low-frequency response of an R-C coupled amplifier is to be increased, should the capacity of the coupling condenser ( $C_3$  in Fig. 1) be: 1, increased; or 2, decreased?
3. If a stage having a gain of 12 is cascaded with a stage having a gain of 10, what will be the total voltage amplification of the two?
4. Will an increase in the ohmic value of the load resistance for a triode tube make the dynamic  $E_G-I_P$  curve for the tube straighter (more linear)?
5. With a triode tube, is the maximum undistorted power obtained when the plate load: 1, is equal to; 2, is one half of; or 3, is twice the plate resistance?
6. What is the purpose of the plate-to-cathode condenser ( $C_3$  in Fig. 14) that is used in power output stages?
7. Does the push-pull stage eliminate: 1, even; or 2, odd harmonic distortion?
8. If there is no by-pass condenser across the bias resistor in the cathode circuit, is the inverse feedback: 1, the voltage type; or 2, the current type?
9. What is the purpose of the phase inverter stage?
10. Why is it important to replace resistors  $R_4$  and  $R_5$  in Fig. 34 with exact duplicate parts?

## HOW TO START STUDYING

For some people, starting to study is just as hard as getting up in the morning. An alarm clock will work in both cases, so try setting the alarm for a definite study-starting time each day. Start studying promptly and definitely, without sharpening pencils, trimming fingernails or wasting time in other ways.

*Beginning* is for many people the hardest part of any job they tackle. So formidable does each task appear before starting, that they waste the day in dilly-dallying, in day-dreaming, and in wishing they didn't have to do it. The next day, and the next after that are the same story. Indecision brings its own delays, making it harder and harder to buckle down to work.

Are you in earnest? Then seize this very minute; begin what you can do or dream you can. Boldness in starting a new lesson is a great moral aid to mastery of that lesson; only begin, and your mind grows alert, eager to keep on working. Begin, and surprisingly soon you will be finished.

J. E. SMITH

**RADIO FREQUENCY AMPLIFIERS  
FOR SOUND AND TELEVISION  
COMMUNICATION**

16FR-1

**NATIONAL RADIO INSTITUTE**

ESTABLISHED 1914

**WASHINGTON, D. C.**





# Radio Frequency Amplifiers for Sound and Television Communication

## WHAT R. F. AMPLIFIERS DO

**I**N sound or television receivers the modulated R.F. carrier current in the antenna circuit is usually not strong enough to be fed directly to the detector (where the sound or picture signals are separated from the carrier). Weak modulated carrier voltages must often be amplified as much as a million times before they can be satisfactorily demodulated by the detector; this can be done by an *R.F. amplifier* consisting of one or more stages.

This R.F. amplifier must not only *amplify* the desired incoming carrier signal, but must also be able to reject all undesired carrier signals which are present in the antenna circuit; the R.F. amplifier must have *selectivity*. Furthermore, the R.F. amplifier must be able to select, when the receiver controls are adjusted, any one of the many carriers which may be present in the antenna circuit; it must be possible to *tune* the R.F. amplifier to a desired carrier frequency.

*T.R.F. Receivers.* In one very common type of R.F. amplifier, all of the R.F. stages can be tuned to the frequency of the desired incoming carrier signal. A receiver using this type of R.F. amplifier is commonly known as a tuned radio frequency receiver, or simply as a *T.R.F. receiver*.

*Superheterodyne Receivers.* The most widely used R.F. amplifier is that in which only the first stage (or first two stages) is tuned to the frequency of the incoming carrier. The amplified carrier signal from the first stage then enters a mixer-first detector tube, where it is mixed with an R.F. signal from a local oscillator in such a way that the frequency value of the R.F. carrier is reduced to a lower R.F. value which is known as the intermediate frequency (I.F.). This intermediate radio frequency, still modulated with either the sound or picture signal, is then amplified by I.F. amplifier stages having fixed tuning and therefore giving maximum amplification at one particular frequency. This is the so-called superheterodyne (or super) principle of R.F. amplification, in which the tuning action takes place before frequency conversion and most of the amplification takes place after frequency conversion (in the I.F. amplifier stages, which are also known as low R.F. amplifier stages). Each R.F. and I.F. stage contributes something to the selectivity of the receiver, with the greatest amount of selectivity being secured in the fixed tuned stages (the I.F. amplifier).

*Side Bands.* Let us review briefly the characteristics of a modulated radio frequency carrier current. As you know, the process of modulating a carrier introduces side frequencies which are above and below the carrier frequency value. The greater the frequency range of a sound or picture signal being transmitted, the farther off from the carrier frequency will be the extreme side frequencies. The frequency range extending from the lowest to the highest side-band frequency is known as the *band width*;

this naturally varies with the nature of the intelligence being transmitted. In addition to the carrier frequency, then, the R.F. amplifier must therefore handle all of these side frequencies.

*Fidelity.* The R.F. amplifier in a receiver must be able to reject all signals except the desired R.F. carrier and its side bands, but at the same time it must not be so selective in its operation that it completely or partially rejects any of the side frequencies associated with the desired R.F. carrier frequency.

The fidelity (quality of reproduction) of an R.F. amplifier is impaired when the carrier frequency is amplified more than the side frequencies; distortion likewise occurs if the side frequencies higher than the carrier are amplified more than the side frequencies which are lower than the carrier.

The R.F. amplifier in a sound or television radio receiver must therefore be able to do these four things: 1, tune to the desired carrier frequency; 2, amplify the desired carrier frequency; 3, contribute selectivity by rejecting undesired carrier frequencies; 4, give uniform amplification for all side frequencies when high fidelity reproduction is required.

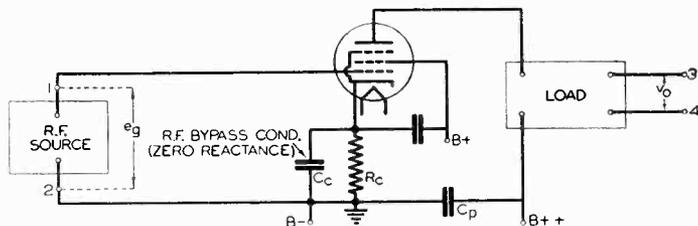


FIG. 1. Simplified schematic circuit diagram of a typical R. F. amplifier stage.

*R.F. Amplifiers in Transmitters.* In any radio transmitter, regardless of whether sound or picture signals are handled, the R.F. amplifier can be divided into two sections: 1, the section handling only the unmodulated R.F. carrier currents; 2, the section handling the modulated R.F. carrier current. The point of division is therefore the stage in which modulation takes place.

The first section of a transmitter creates, by means of an oscillator stage, an R.F. carrier which is amplified by succeeding R.F. stages until the R.F. carrier current is at the power level required for modulation purposes. Since only a single frequency passes through this first section, the R.F. stages may be very selective.

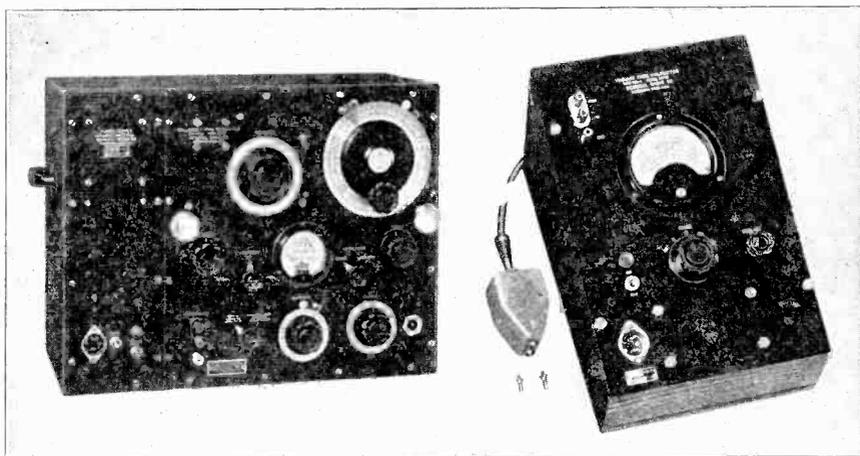
The process of modulation introduces side frequencies as well as harmonics. The second section of a transmitter must therefore pass a definite band width, amplifying all frequencies in this band equally well while raising the power of the modulated R.F. carrier to the desired value.

## RESONANCE CURVES

*What Is a Resonance Curve?* The manner in which the amplifying ability of a tuned R.F. amplifier stage varies for the different frequencies between the lowest and the highest side frequencies is of great importance;

with this information at hand, we can tell how much the stage amplifies, how well it rejects undesired signals, how it amplifies desired signals, and how it amplifies some frequencies more than others in the side bands associated with the desired carrier. All this information about a tuned R.F. amplifier can be presented with a *frequency response curve* which shows in graphical form *the manner in which the amplifier handles or amplifies the various frequencies in its operating range*. Since tuned R.F. amplifiers always contain at least one resonant circuit, this frequency response curve is often called a *resonance curve*.

*Typical R.F. Amplifier Stage.* The diagram in Fig. 1 presents in simplified form a typical tuned R.F. amplifier stage; after reviewing briefly the



Courtesy General Radio Co.

FIG. 2A (left). General Radio type 605-A signal generator, which consists fundamentally of an R.F. oscillator having a frequency range of 95 kc. to 30,000 kc., and means for varying and measuring the output voltage. There are also provisions for modulating the R.F. signal with an audio note of fixed frequency or with an external modulating signal when a modulated carrier is required; the meter on the panel indicates percentage modulation in this case.

FIG. 2B (right). General Radio type 726-A vacuum tube voltmeter, a multiple-range instrument designed to measure R.F. voltages from .1 to 150 volts with a high degree of accuracy. It is calibrated to read r.m.s. (effective) values of sine wave voltage.

operation of this stage, I will show you how a resonance curve for it can be obtained.

The R.F. source in Fig. 1 feeds to the grid-cathode of the tube (between points 1 and 2) an R.F. voltage which we will call  $e_g$ . As a result of the amplifying action of the tube and the effects of the load, there appears across the output terminals of the load an R.F. output voltage which we will designate as  $v_o$ . The net or true voltage amplification of this R.F. amplifier stage (expressed as  $A$ ) is the output voltage  $v_o$  divided by the input voltage  $e_g$ . The type of tube used, the operating voltages and the type of load applied determine to a great extent the behavior of this stage as an amplifier.

*How a Resonance Curve Is Secured.* Although an R.F. amplifier must handle many frequencies (the carrier frequency and the side-band frequencies) simultaneously in actual practice, it is obvious that we cannot study the performance of the amplifier while all these frequencies are pass-

ing through; the practical way is to send through only one frequency at a time, a simple sine wave R.F. signal, and measure how much it is amplified. The frequency of this test signal is then changed to other values, and the amplification is measured at each frequency.

Radio engineers use a signal generator like that shown in Fig. 2A to supply a sine wave R.F. signal whose frequency can be adjusted to the various values required when testing an R.F. amplifier. The large dial in the upper right corner of the panel controls the frequency of the output, the dial at its left changes the frequency range, and other dials change the strength and the modulation percentage of the R.F. output signal when a modulated signal is desired.

To measure the input and output voltages of an R.F. amplifier, engineers use a special voltmeter which employs a vacuum tube as a detector; a typical laboratory type vacuum tube voltmeter (abbreviated V.T.V.M.) is shown in Fig. 2B. The voltage being measured is applied to the wedge-shaped probe (at the left of the cabinet) in which is an acorn-type tube connected as a rectifier. A shielded cable connects the probe unit to the cabinet on which is mounted a multi-range voltmeter and the control switches. The amplification produced by an R.F. stage at any frequency can be determined simply by dividing the measured output voltage  $v_o$  by the measured input voltage  $e_g$ .

If the R.F. amplifier stage in Fig. 1 is set to amplify a 1,000 kc. signal, and we desire to obtain a resonance curve for the amplifier at this resonant frequency, we would connect the signal generator in place of the R.F. source and set it in turn at 980, 985, 990, 995, 1,000, 1,005, 1010, 1,015 and 1,020 kc., and for each setting connect the vacuum tube voltmeter across terminals 1 and 2 to measure  $e_g$  (if its value is not already known) and across terminals 3 and 4 to measure  $v_o$ . We would then compute the amplification  $A$  obtained at each frequency, and plot our information on graph paper in the manner shown in Fig. 3 to secure the resonance curve for the R.F. amplifier at 1,000 kc. Notice that as the frequency is increased from the lowest value, the amplification increases and reaches a maximum value at 1,000 kc., the resonant frequency of the stage. Further increases in frequency beyond 1,000 kc. give decreasing amplification, the amplification dropping to zero at about 20 kc. above the resonant frequency. A curve of this nature is called a *single-peaked resonance curve*, for it gives maximum or peak amplification at a single R.F. value.

When a 1,000 kc. R.F. carrier is modulated with a sound signal having a maximum frequency of 5 kc., you know that the side-band frequencies will extend from 995 to 1,005 kc. The amplifier represented by the response curve in Fig. 3 would amplify the carrier frequency about 65 times, but would amplify the lowest side frequency ( $f_1$  in Fig. 3) only 50 times and the highest side-band frequency ( $f_2$ ) 58 times.

An amplifier which has a single peaked resonance curve distorts a modulated carrier in two ways. If the corresponding upper and lower side frequencies are amplified equally well *but less than the carrier frequency*, the result after demodulation is comparable to frequency distortion (where the extremely high frequencies in the modulation signal are eliminated). If the corresponding upper and lower side frequencies are, in addition, *amplified unequally*, as is the case in Fig. 3, there is amplitude distortion as well.

*Defining Selectivity.* Since the U. S. A. radio stations in the broadcast band are only 10 kc. apart in frequency, it is essential that a receiver

tuned to one station does not also pick up the stations 10 kc. away on either side; insufficient selectivity in an R.F. amplifier therefore results in station interference. If the tuned R.F. stage whose performance is represented by Fig. 3 is the only one in a receiver, and it is tuned to a 1,000 kc. station, its carrier signal would be amplified about 65 times, while the carrier signals of adjacent stations (at 990 kc. and 1,010 kc.) would be amplified 15 times and 28 times respectively; this is clearly poor selectivity, for all three stations, if received with nearly equal signal intensity, will be heard at once. What, then, constitutes good selectivity?

Selectivity was formerly found according to the following procedure: The two frequencies at which amplification was 70% of the maximum amplification (the value at resonance) were found; the difference between these frequencies was determined, and the resonant frequency divided by this difference in frequency was said to be a measure of the selectivity of

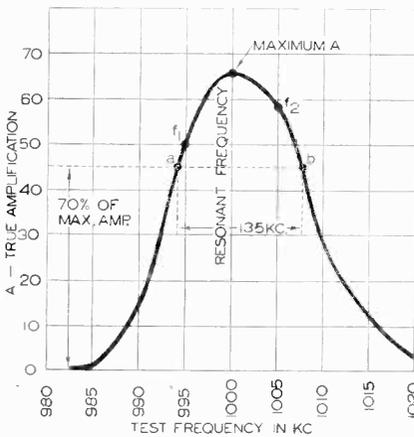


FIG. 3. Conventional resonance curve at 1,000 kc. for an R.F. amplifier stage.

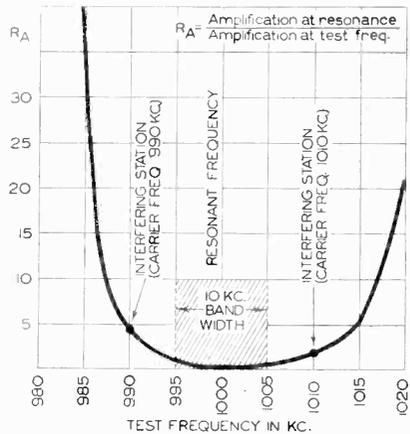


FIG. 4. Ratio resonance curve at 1,000 kc. for an R.F. amplifier stage.  $R_A$  is 1 at 1000 kc.

the amplifier. Points *a* and *b* in Fig. 3 correspond to the two frequencies at which *A* is 70% of its maximum value. The difference in frequency between these two points is 13.5 kc., hence the selectivity value for this particular amplifier stage would be 1,000 (the resonant frequency) divided by 13.5, or 74. Incidentally, this value is also the *Q* factor for the resonant circuit of this amplifier stage.\* This older definition for selectivity and *Q* factor is perfectly satisfactory when applied to simple or single resonant circuits, since all such circuits have essentially identical resonance curve shapes. The response curves of complicated or multi-resonant circuits have widely varying shapes, however, making this old definition of selectivity unsuited for present day use.

The modern definition for receiver selectivity is based upon the fact

\* In a series resonant circuit which is tuned to resonance, the ratio of the voltage across the coil or condenser to the applied circuit voltage is the *Q* factor of the circuit; *Q* factor is also the reactance of the coil divided by the circuit resistance. In a parallel resonant circuit the ratio of the coil or condenser current to the line supply current at resonance is the *Q* factor; *Q* factor is also the resonant resistance divided by the coil or condenser reactance at resonance.

that we are *primarily* interested in how much better the amplification is at resonance than at the nearest undesired frequencies. *For good selectivity, the desired signal frequency and its side frequencies must be amplified at least 1,000 times more than the nearest undesired signal and its side bands.* For fair selectivity this ratio (which we shall designate as  $R_A$ ) can be about 100, and for excellent selectivity the ratio must be higher than 10,000.

Fidelity is now defined in much the same way as selectivity. *For good fidelity the desired carrier frequency must not be amplified more than 1.25 times the highest side frequency desired with that signal;* in other words, amplification at the highest side frequency must be at least 80% of the amplification at resonance. These figures are by no means fixed, for they are based upon opinions rather than upon precise data, and in addition will vary with the purpose of and use to which an R.F. amplifier is put.

A better picture of the frequency response characteristic of an R.F. amplifier can be obtained by computing  $R_A$ , the ratio of amplification at resonance to amplification at the test frequency, for each value of test frequency and plotting these ratios against frequency in the manner shown in Fig. 4.\* A curve like this (called a *ratio resonance curve*) gives us both the fidelity and selectivity characteristics of an amplifier; for example, in Fig. 4 the highest side frequency (for a 5 kc. modulating signal) has a ratio of about 1.1 (at 1,005 kc.), which indicates excellent fidelity since it is considerably less than the maximum allowable value of 1.25. At 1,010 kc., however, the ratio is only about 2.3, indicating that the desired carrier frequency signal will be amplified only about 2.3 times more than the carrier signal of an undesired station at 1,010 kc. if both carrier signals have equal signal strength at the location of the receiver. Thus, fidelity is good but selectivity is very poor in the amplifier represented here.

*Amplifiers in Cascade.* If one R.F. amplifier stage raises the voltage at the resonant frequency 10 times, another similar stage connected in cascade will amplify the voltage 10 times further, giving a total amplification of 100; the addition of a third identical amplifier stage will boost the voltage 10 times further, giving a total amplification of 1,000. The amplification (gain) of an R.F. amplifier is therefore the product of the gain of each individual stage (the gains are multiplied together).

Resonance curves for two and three tuned R.F. amplifier stages in cascade are given in Fig. 5, with the curve for a single stage shown for comparison. The addition of stages makes the resonance curve sharper and higher in peak value. *Cascading of R.F. amplifier stages thus boosts the amplification and improves the selectivity of an R.F. amplifier.*

The resonance curve for perfect fidelity and ideal selectivity should be flat throughout the desired band width and should have straight vertical sides; adding stages makes the sides of the resonance curve more nearly vertical, giving better selectivity, but lowering fidelity (by making the resonance curve sharper in the band width).

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\* Laboratory engineers sometimes measure  $R_A$  directly. At each test frequency they increase the output of the signal generator until the measured amplifier output voltage  $v_o$  is the same value as it was at the resonant frequency; the signal generator output at the test frequency divided by the signal generator output at resonance gives the value of  $R_A$ .

Fidelity and selectivity characteristics of amplifiers in cascade are more clearly presented by plotting  $R_A$  against frequency, as was done in Fig. 4, this is done in Fig. 6. Let us analyze these three ratio resonance curves; for convenience I will set down again the practical definitions for selectivity and fidelity.

Good fidelity:  $R_A$  is lower than 1.25 at the *highest* side frequency.

Good selectivity:  $R_A$  is higher than 1,000 on both adjacent-channel carrier frequencies.

Checking fidelity first, we find that good fidelity is secured with one stage ( $R_A$  is less than 1.15 at 1,005 kc.) and with two stages ( $R_A=1.25$ ),

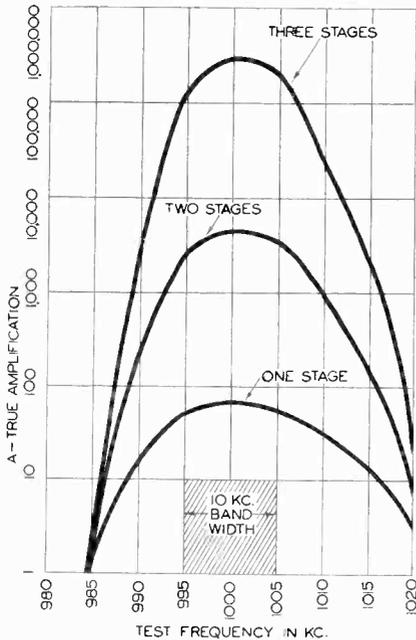


FIG. 5. Resonance curves at 1,000 kc. for one R.F. amplifier stage and for two or three similar stages connected in cascade. Amplification  $A$  is here plotted on a special condensed scale which corresponds to the response of the human ear. Resonance is at 1,000 kc.

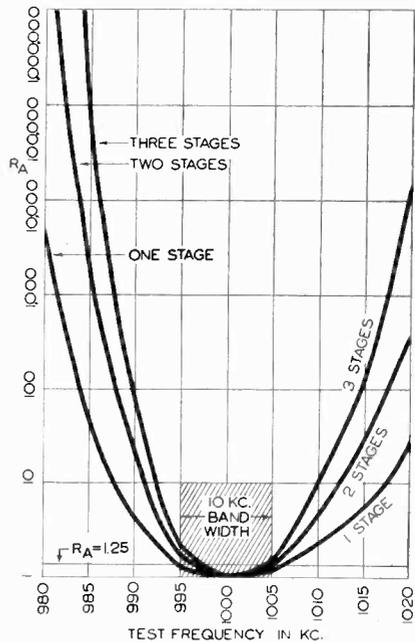


FIG. 6. These three ratio resonance curves give the fidelity and selectivity characteristics of one, two and three of the R.F. amplifier stages in Fig. 1. The scale at the left gives values of  $R_A$  (amplification at resonance = amplification at test frequency). Resonance is at 1,000 kc.

but three stages give poor fidelity ( $R_A$  is higher than 1.25). As to selectivity,  $R_A$  for three stages is about 100 at 990 kc. and is about 10 at 1,010 kc., these being the adjacent-channel carrier frequencies in the broadcast band. The lowest value governs, and hence selectivity is poor. For one and two stages, the values of  $R_A$  at these undesired carrier frequencies are lower, so selectivity is even poorer than for three stages.\*

\* When peaked R.F. amplified stages have symmetrical response curves, several stages may be connected in cascade to give good selectivity, and the attenuation of the higher modulation frequencies (the poor frequency response) can be compensated for by using an audio (or video) amplifier which will give greatest amplification at the higher frequencies. This fact makes it possible to equalize (adjust) a receiver for high fidelity even though individual stages in it do not have ideal characteristics.

You will notice that the vertical scales in Figs. 5 and 6 are arranged in a different manner from those in Figs. 3 and 4; this "condensing" was done in order to cover a wide range of values while still showing the nature of the curves clearly at the lowest values, around the resonant frequency. These condensed scales are known as logarithmic scales; they also tell us more clearly what the ear is going to hear, for our ear has this same logarithmic response.

Complete R.F. amplifiers, regardless of whether they are of the super-heterodyne or the T.R.F. type, are measured in exactly the same manner as the single stage shown in Fig. 1, and the results are plotted exactly as is done in Figs. 5 and 6. The resulting curves allow the radio engineer to study the amplification, selectivity and fidelity characteristics of the entire R.F. amplifier.

*Ideal and Practical Ratio Resonance Curves.* Ratio resonance curves for four different R.F. amplifiers, each of which contains several stages in cascade, are given in Figs. 7A and 7B. These curves give the value of the ratio  $R_A$  for different frequencies up to 20 kc. above and below resonance. You can consider  $R_A$  in two ways, as the ratio of the amplification at resonance to the amplification at various off-resonance frequencies, or as how much stronger the signal must be off resonance to produce the same output as at resonance. The higher the value of  $R_A$  for frequencies off resonance, the more selective is the amplifier.

Before you can study the ratio resonance curve for a particular R.F. amplifier, you must know at least two facts: 1, the frequency separation in kc. between the stations which are to be picked up by that amplifier; 2, the R.F. band width in kc. which is required to handle the entire range of frequencies for the audio or video signal being transmitted.

Stations in the broadcast band in the United States, radiating sound signals, are located 10 kc. apart and always have a frequency value ending in zero; thus you may receive stations at 570, 580, 590 kc., etc. For sound signals having a maximum range of 5,000 cycles, the side frequencies will extend 5 kc. above and below the carrier frequency. For good selectivity, then, the value of  $R_A$  should be greater than 1,000 at frequencies more than 5 kc. off the desired carrier frequency. To secure good amplification of the side frequencies, however,  $R_A$  should be as close to the value 1 as possible throughout the band width of 10 kc.

Curve 1 in Fig. 7A (the dash-dash curve) represents the ideal ratio resonance characteristics for an R.F. amplifier designed for use with broadcast band apparatus; all frequencies in the band width are amplified equally, while all other frequencies are completely cut off. Curves 2 and 3 in Fig. 7A and curves 4 and 5 in Fig. 7B progressively approach this ideal response, curve 5 being about the closest approach to the ideal which can be secured in practice. A radio expert would label curve 2 as "fair fidelity but very poor selectivity"; curve 3 would be designated as "poor fidelity and good selectivity" (this particular response curve might be satisfactory for radio-telegraphy work, where the side frequencies extend very little off resonance). In Fig. 7B, curve 4 represents an amplifier having good

selectivity and fair fidelity, while curve 5 has excellent fidelity and good selectivity.

Resonance curves very similar to those just studied are used in portraying the abilities of R.F. amplifiers designed for television purposes. Remember, however, that the resonant frequencies here will be in the range extending from 40 to 90 megacycles, the band width will be 5 to 10 megacycles wide and stations will be separated as much as 15 megacycles. The same resonance curves can be used to portray the behavior of R.F. amplifiers which follow the modulated stages in transmitters.

Now we are ready to consider R.F. amplifier circuits in detail; you will learn how their desired features are obtained. A knowledge of how their operation is affected by changes in circuit parts will be of great value when you have to make repairs on incorrectly operating receivers.

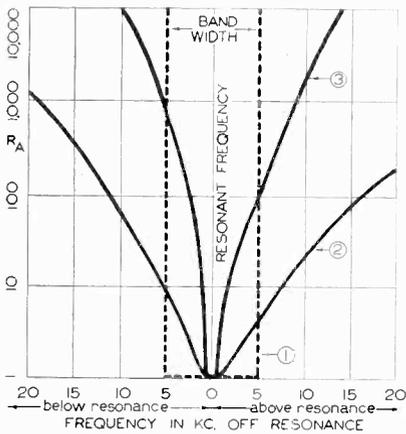


FIG. 7A. Ideal (1) and typical (2 and 3) ratio resonance curves for R.F. amplifiers.

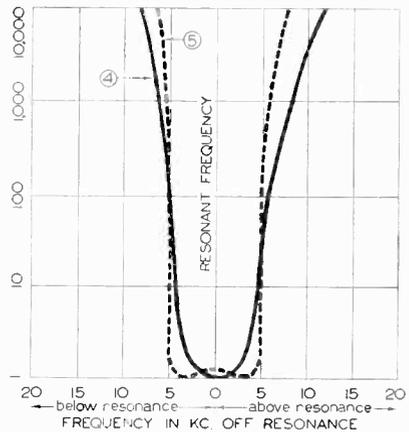


FIG. 7B. Ratio resonance curves of R.F. amplifiers designed to have 10-kc. selectivity. Vertical scale gives values of  $R_A$ .

## SIMPLE TUNED R. F. AMPLIFIERS

*Amplifier Loads.* Almost any circuit which uses a vacuum tube having a control grid can be used as an R.F. amplifier provided that the proper plate load is used. If a simple resistor or coil is made to serve as the plate load, we have an untuned R.F. amplifier circuit which will amplify R.F. signals over a *very wide band*; this poor-selectivity characteristic is highly desirable in those cases where other radio devices are used for signal selecting and tuning purposes. If a reasonable amount of voltage amplification is to be secured, the ohmic resistance (or the reactance in the case of a coil) of the plate load (which is untuned) should be greater than the A.C. plate resistance of the tube.

*Coil-Condenser Loads.* You are already familiar with the fact that a coil and condenser connected in parallel, as in Fig. 8, behave exactly like a resistor of high ohmic value at the resonant frequency (the frequency at which the coil reactance exactly balances the condenser reactance), but offer a lower reactance at other frequencies. Changing the value of either

the coil or the condenser changes the resonant frequency. Thus this parallel coil-condenser resonant circuit is a very desirable load for an R.F. amplifier, giving high amplification of the desired signal, giving low amplification of other frequencies and permitting tuning to a desired frequency.

If coil  $L$  in Fig. 8 has an inductance of 250 microhenrys and a resistance  $R$  of 20 ohms, and condenser  $C$  is adjusted to 100 micro-microfarads (mmfd.), this circuit will act like a 125,000 ohm resistor at its resonant frequency (this can be proved by actual test or by computation with a formula given later in this book). Thus, when this circuit is used as a plate load, a high value of A.C. voltage is produced across terminals 1 and 2 at the resonant frequency, but the circuit acts either as a condenser or coil of low reactance to other frequencies (off-resonance frequencies), with a resulting low A.C. voltage for these frequencies; the circuit therefore possesses selectivity characteristics. Decreasing the inductance of the coil or decreasing the capacity of the condenser raises the resonant frequency of the circuit. Another advantage of this coil-condenser tuned load circuit is that it is not seriously affected by capacity between the electrodes of the tube; a high plate-to-cathode capacity simply means  $C$  can be smaller.



FIG. 8. A coil-condenser load (a parallel resonant circuit) acts as a resistor of high ohmic value at the resonant frequency of the circuit.

*Coil-Condenser Load Connections.* There are a number of different ways of connecting a coil-condenser resonant circuit into the plate circuit of a tube; four of the most important of these are shown in Fig. 9. Remember that a small resistance (that of the coil) is always present in a resonant circuit (it is usually neglected when preparing diagrams, in order to simplify the presentation of resonant circuits).

In Fig. 9A, resonant circuit  $L-C$  (often called a tank circuit) is connected directly into the plate circuit, with D.C. plate supply current flowing through coil  $L$ . The A.C. voltage which is developed across the tank circuit terminals is impressed upon the grid of the next R.F. amplifier stage through coupling condenser  $C_K$ , which has negligible reactance to signal currents. This condenser also prevents D.C. plate current from flowing to the grid of  $VT_2$ , the next tube. Resistor  $R_g$  has an important function, that of connecting the grid to ground in order that the C bias voltage developed by cathode circuit resistor  $R_c$  can act upon the grid of the tube. Obviously the ohmic value of  $R_g$  must be much higher than the resonant resistance of the tank circuit if it is not to affect the plate circuit. (Since  $R_g$  is in parallel with the plate circuit, a low value of  $R_g$  would absorb power, lowering the resonant circuit resistance and thus reducing the net plate circuit resistance; the result would be a reduction in the amplification of the amplifier and a broadening of the resonance curve. This broadening improves fidelity, and is therefore desirable in some cases.)

In Fig. 9B, choke coil  $CH$  is used in place of grid resistor  $R_g$ . The reactance of this choke coil at any frequency to which the tank circuit may be tuned should be many times greater than the resonant resistance of the tank circuit, in order to secure a large A.C. voltage drop across the tank circuit.

In Fig. 9C the positions of the choke coil and the tank circuit are interchanged as compared with Fig. 9B; this arrangement, where the choke coil is in the plate circuit, is known as a shunt fed circuit, for the D.C. power is fed in parallel or in shunt with the  $L$ - $C$  circuit. When the resistance of  $R_g$  in Fig. 9A and the reactance of  $CH$  in Figs. 9B and 9C are many times greater than the resonant resistance of the tank circuit, the values of  $L$  and  $C$  in the tank circuit control the R.F. signal output; the resonant circuit thus becomes the plate load. The following important facts apply to all three circuits:

1. *The values of  $L$  and  $C$  determine the resonant frequency of the tank circuit.* For practical reasons it is customary to tune the condenser; decreasing the capacity of the condenser increases the resonant frequency of the tank circuit. Coil  $L$  is still shunted by a capacity, however, when condenser  $C$  is set at minimum capacity; this residual capacity consists of:  $a$ , the distributed capacity of coil  $L$ ;  $b$ , the minimum capacity of the condenser;  $c$ , capacities between the tube electrodes and between the leads in the circuit. In a well designed practical circuit the minimum or residual tank capacity is about one-ninth of the maximum tank capacity; this means that the maximum frequency to which the tank circuit can be tuned is about 3 times the lowest frequency. An example of this is the average broadcast band receiver, which tunes from about 500 to 1,500 kc. with a single set of coils.

2. *The effective resistance of the tank circuit at resonance is determined by the inductance of the coil, the capacity of the condenser and the inherent tank circuit resistance.\**

3. *If the tank circuit at resonance has a coil of high inductance and a condenser of low capacity, the resonant resistance will be high.* Radio men say that a tank circuit with large  $L$  and low  $C$  has a high  $L/C$  ratio. The lower the inherent resistance of the tank circuit, the greater will the resistance of a tank circuit be at resonance.

4. In a single-tube tuned R.F. amplifier stage which uses a parallel resonant circuit as a plate load, the maximum possible amplification is the  $\mu$  of the tube, and this value is obtained when the resonant resistance of the plate load is many times greater than the A.C. plate resistance of the tube. Triode tubes have relatively low A.C. plate resistances and hence this condition is quite easy to secure, but with tetrode and pentode tubes the load ordinarily cannot be made high enough in value to give maximum amplification.

5. When the tank circuit is tuned to higher frequencies in the usual manner by reducing the value of capacity  $C$ , the  $L/C$  ratio increases, raising the resonant resistance of the load.

*Tuned Transformer Circuit.* In Fig. 9D is perhaps the most widely used R.F. amplifier circuit, the so-called tuned transformer circuit. The power required for tank circuit  $L$ - $C$  must be supplied to the primary coil  $L_p$  by the tube. When the  $L$ - $C$  tank circuit acts like a resistor, as at resonance, the reflecting properties of the transformer make coil  $L_p$  act like a resistor (the

\* There are two formulas for securing the resonant resistance of a tank circuit like that in Fig. 8; the radio engineer uses whichever is more convenient. The

formulas are:  $R_R = \frac{L}{RC}$ , and  $R_R = \frac{\omega^2 L^2}{R}$ , where  $R_R$  is the resonant resistance of the tank circuit in ohms,  $R$  is the inherent tank circuit resistance in ohms,  $\omega$  is 6.28 times the frequency in cycles,  $L$  is the inductance of the coil in henrys and  $C$  is the capacity of the condenser in farads.

leakage inductance of primary  $L_p$  affects this condition, but this inductance is so small that it can be neglected). Facts 1, 2, 3 and 5 brought out in connection with the first three circuits in Fig. 9 hold true also for this circuit, and the following additional facts apply to this circuit only:

1. Increasing the number of turns on primary winding  $L_p$  or increasing the coupling between  $L_p$  and  $L$  raises the reflected resonant resistance appearing between terminals 1 and 2. We are actually increasing the mutual inductance  $M$  of the coils when we add turns or move the coils closer together; thus we can say that increasing the mutual inductance increases the effective resonant resistance of the plate load.

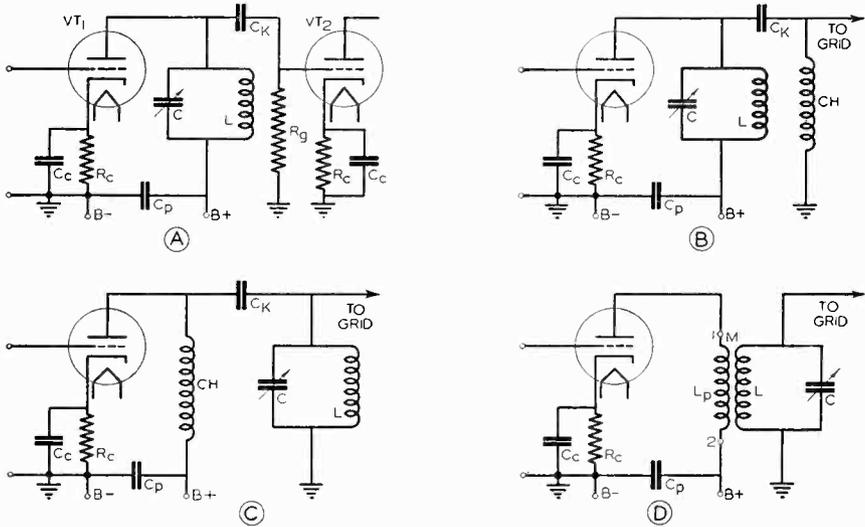


FIG. 9. Four ways of connecting the resonant  $L$ - $C$  load circuit into an R.F. amplifier stage so it will act as a plate load. These circuits apply also to screen grid and pentode tubes; triodes are shown here merely for simplicity.

2. When a tuned transformer load is used, maximum voltage amplification is obtained when the resonant resistance of the tuned load (as it appears in the plate circuit) is equal to the A.C. plate resistance of the tube, for the tank circuit then absorbs the greatest amount of power.

3. Circuit  $L$ - $C$  in Fig. 9D is a series resonant circuit (because the voltage induced in it acts in series with  $L$  and  $C$ ), and hence acts as a coil when tuned below resonance and as a condenser when tuned above resonance. The effects at terminals 1 and 2 are opposite, however, because of transformer action.

## HOW TUBE CAPACITIES AFFECT R. F. AMPLIFIER CIRCUITS

*Inter-Electrode Capacities.* When we study an ordinary triode tube in detail, we realize that the grid, plate and cathode are tiny condenser plates which introduce into the tube circuits the three capacities shown by dotted lines in Fig. 10. These inter-electrode capacities are: 1, the grid-to-cathode capacity,  $C_{gk}$ ; 2, the plate-to-cathode capacity,  $C_{pk}$ ; 3, the grid-to-plate capacity,  $C_{gp}$ . It is the grid-to-plate capacity,  $C_{gp}$ , which prevents the grid and plate circuits from being entirely independent of each other.

In the R.F. amplifier circuit in Fig. 10, the grid-to-cathode capacity  $C_{gk}$  acts simply as if it were an extra condenser connected in parallel with  $C_1$ ; the plate-to-cathode capacity  $C_{pk}$  acts as a condenser connected in parallel

with  $C_2$ . By-pass condensers  $C_c$  and  $C_p$  in Fig. 10 are of such high capacity that they act as low reactance paths for A.C. currents, and therefore need not be considered in this discussion.

In Fig. 11 the R.F. amplifier circuit of Fig. 10 has been modified to include only those parts which affect the tube as an amplifier;  $C_{gk}$  and  $C_{pk}$  have been placed in their effective positions. Capacity  $C_{gp}$  can, according to a complicated mathematical analysis which need not be taken up here, be considered as *equivalent* to an extra resistor  $R_R$  and an extra condenser  $C_R$ , both in parallel with the grid tank circuit  $C_1-L_1$ . Remember that  $R_R$  and  $C_R$  are present as *effects only*, not as actual devices; their exact equivalent values vary greatly with circuit conditions, being dependent upon the capacity of  $C_{gp}$ , upon the resonant frequency and nature (resistive, inductive or capacitive) of the plate circuit load, and upon the over-all amplification of the stage.

Coil  $L_2$  in Fig. 11 acts with  $C_2$  and  $C_{pk}$  to form a parallel resonant tank circuit which is the equivalent of a high resistance at resonance. When this parallel resonant circuit is tuned below resonance (to a frequency *lower*

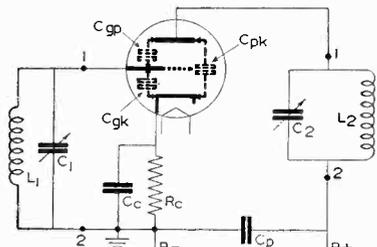


FIG. 10. The three inter-electrode capacities which enter into the operation of an R.F. amplifier are here indicated by dotted lines.

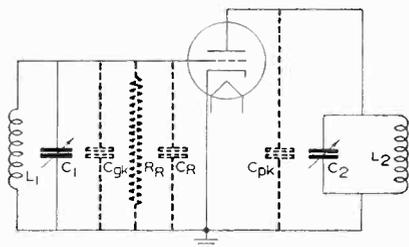


FIG. 11. The circuit of Fig. 10 is here modified to include only those parts which affect the tube as an amplifier. The effects of the grid-to-plate capacity  $C_{gp}$  are represented by  $R_R$  and  $C_R$ .

than that of the incoming signal), the tank circuit will act as if it were a condenser. (This is because decreasing the resonant frequency by increasing the capacity of  $C_2$  lowers the reactance of the condenser but does not affect the reactance of the coil; in a parallel circuit such as this it is the *lowest* reactance [the reactance of  $L_2$  or  $C_2$ ] which governs the nature of the tank circuit off resonance.) Likewise, when this parallel resonant circuit is tuned above resonance (to a frequency *higher than* that of the incoming signal), the tank circuit acts as a coil.\*

\* When a resonant load circuit is loosely coupled inductively to the plate circuit of a tube, as is done in Fig. 9D, the tuned secondary ( $L-C$ ) is purely resistive when tuned to resonance. Because of the leakage inductance of primary  $L_p$ , however, the effect of the load in the plate circuit (as measured between points 1 and 2) will be *inductive*. When the  $L-C$  circuit is tuned off resonance, the reactance which it reflects into primary  $L_p$  will always be opposite in nature to the reactance which the  $L-C$  circuit itself appears to have. Thus, when series resonant circuit  $L-C$  is tuned above resonance it acts as a capacity and is reflected as an inductance; when tuned below resonance it acts as an inductance and reflects as a capacitance. In order to make the effect of the load purely resistive across terminals 1 and 2, then, it is necessary to tune the load slightly below resonance, so it reflects a capacitance which will balance out the leakage inductance of the primary; in doing this we are really using the secondary  $L-C$  circuit to tune the primary  $L_p$  (which is in effect the load) to resonance.

The circuit in Fig. 11 possesses a number of interesting and important characteristics, some of which are portrayed by the curve in Fig. 12. These facts are listed here, for they mean a lot to any one who will work with R.F. amplifiers. The following facts apply equally well to the circuits in Figs. 9 and 10 (if points 1 and 2 in Fig. 9D are considered as the load terminals).

1. When the load is tuned exactly to the frequency of the incoming signal, the capacity of  $C_R$  is a maximum and is equal to the capacity of  $C_{rp}$  plus a value equal to the capacity of  $C_{rp}$  multiplied by the true amplification of the stage. When the load is tuned either above or below the signal frequency, the true amplification of the stage is greatly reduced and the capacity of  $C_R$  therefore decreases rapidly, as is shown in Fig. 12.

2. When the load is tuned exactly to the incoming signal, the equivalent ohmic value of  $R_R$  is so high that it can be neglected.

3. When the load acts as a condenser, the equivalent ohmic value of  $R_R$  is greatly reduced. The selectivity response curve of the amplifier is consequently broadened and the amplification of the stage is reduced.

4. When the load acts as a coil, the tube is actually feeding power back into the grid circuit, thus increasing the amplification of the stage and improving selectivity. For convenience this is often explained by saying that the ohmic value of  $R_R$  becomes *negative*; a negative resistance in shunt with a real resistance will actually increase the net resonant resistance of  $L_1-C_1$ , clearly indicating that the circuit is capable of producing a greater resonant voltage.

5. If the load acts as a coil of sufficiently high reactance, the tube will feed sufficient energy back to the grid to make the circuit oscillate at a frequency determined essentially by the values of  $L_1$ ,  $C_1$ ,  $C_{rp}$  and  $C_R$ .

*Practical Considerations.* The curve in Fig. 12 also contains the answers to some very practical questions concerning the operation of amplifiers in general, including audio or video amplifiers. Keep the following facts in mind when you work with amplifiers and you will have no difficulty in finding causes and cures for annoying squeals or for poor fidelity.

1. In an audio or picture frequency amplifier, a high plate load resistance or reactance will give high amplification and make  $C_R$  very high in capacity. The higher frequencies in the audio or picture signal will therefore be attenuated (by-passed to ground before they reach the grid). A low-impedance load gives less attenuation.

2. If high plate load impedance in an audio or picture frequency amplifier stage is accompanied by an inductive plate load condition, undesirable oscillation may take place.

3. In an R.F. amplifier using a triode tube, any stage having plate and grid circuits arranged as in Fig. 10 will go into oscillation whenever the plate tank circuit is made sufficiently *inductive* by tuning it above the signal frequency. Two R.F. signals, the incoming signal and the oscillating frequency, will then reach the grid and be amplified by the tube; these two frequencies will be mixed together by the detector stage, and if the difference in their frequencies is in the audio range, an annoying squeal will be heard in the loudspeaker.

4. Oscillations can take place even without an incoming signal if the resonant frequency of  $L_2-C_2$ , the plate load, is considerably higher than the resonant frequency of  $L_1-C_1$  in the grid circuit, for under this condition the plate tuned circuit will be *inductive* at the resonant frequency of the grid tuned circuit. These being R.F. oscillations, they will not ordinarily be heard in the loudspeaker. If, however, this condition for R.F. oscillation exists in two different stages, two R.F. signals may go through the system and beat with each other at the detector, producing an A.F. note which will be heard as a squeal. This explains why some receivers produce squeals even when not tuned to a station; the remedy simply involves adjusting one or more of the tuned circuits.

## GETTING RID OF FEED-BACK

When triode tubes are used in amplifier circuits, it is the grid-to-plate capacity which is of vital importance; the other two inter-electrode capacities,  $C_{gk}$  and  $C_{pk}$ , simply reduce the tuning range of the amplifier slightly at the highest frequencies.

**Grid Suppressors.** The oscillation or regeneration caused by the grid-to-plate tube capacity in a triode tube is an especially serious problem in single dial receivers, for these have no provisions for tuning out an annoying squeal at a particular setting of the tuning dial. Since only triode tubes were available in the early days of radio, engineers were compelled to develop a number of solutions for this regeneration or feed-back problem. One of these involved loading the grid tank circuit ( $L_1$ - $C_1$  in Fig. 10) with a resistor which made the resonant resistance of the grid tank circuit less than the ohmic value of the equivalent negative resistance  $R_R$ . (Engineers have proved that  $R_R$ , in addition to being negative must be lower in ohmic value than the resonant resistance of the grid tank circuit before oscillation can occur.) Another feed-back-killing scheme involved inserting a re-

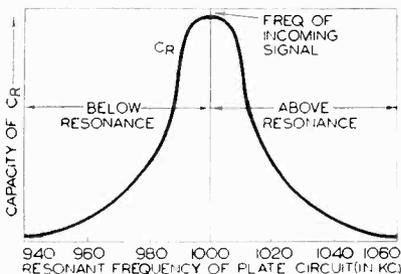


FIG. 12. This curve shows how the equivalent capacity  $C_R$  of the circuit in Fig. 11 varies when the plate tank circuit of an R.F. amplifier is tuned to, above and below the frequency of an incoming signal.

sistor in the grid lead, as at point 1 in Fig. 10; the resistor was in this case known as a *grid suppressor*. Lowering the amplification of the stage, such as by using a low  $L/C$  ratio in the plate tank circuit, was another way of preventing regeneration.

**Neutrodyne Circuits.** In the early days of radio, neutrodyne receivers were "all the rage"; these used an ingenious method of preventing regeneration while retaining all of the amplification of a stage. An A.C. voltage taken from the plate circuit of the R.F. amplifier was fed back into the grid circuit out of phase to balance out, buck out or *neutralize* the undesirable feed-back voltage caused by grid-to-plate capacity.

Figure 13A is an example of one of these early neutrodyne circuits; regeneration occurred here because the A.C. voltage across coil  $L_1$  fed back to the grid tank circuit through inter-electrode capacity  $C_{gp}$ . To offset this, coil  $L_2$  was coupled with and connected to coil  $L_1$  in such a way that the voltage which it fed through neutralizing condenser  $C_N$  to the grid was out of phase with the undesired feed-back voltage; the two voltages therefore bucked each other and, when  $C_N$  was adjusted to make the two voltages equal, they exactly cancelled each other and eliminated regeneration effects.

Figure 13B gives another neutrodyne circuit; no additional coil winding was needed here, for the required out-of-phase A.C. voltage was secured

by tapping the secondary winding of the R.F. transformer. There were many variations of these two neutrodyne circuits, but all were essentially the same in their operation. Neutrodyne circuits were not entirely practical, for they were easily thrown out of adjustment. Any changes in the circuit parts, or the installation of a new tube, made it necessary to readjust neutralizing condenser  $C_N$ .

Engineers realized that the grid-to-plate capacity in the tube was the real cause of all their regeneration troubles, and finally called in vacuum tube engineers to design a tube which had negligible grid-to-plate capacity. The screen grid or shielded grid tube, which you know as the tetrode, was the result; we will consider this tube next.

### THE SCREEN GRID TUBE

As you know, it is current flowing from the plate through inter-electrode capacity  $C_{gp}$  to the grid which results in regeneration in amplifier circuits. If an additional grid is placed between the control grid and plate of a triode, and this new grid, which we call the *screen grid*, is connected to

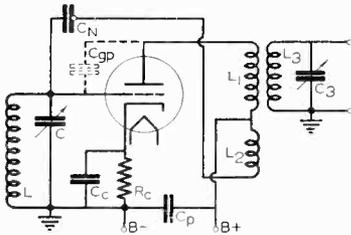


FIG. 13A. Early neutrodyne R.F. amplifier circuit, in which a special coil  $L_2$  provides the required neutralizing voltage.

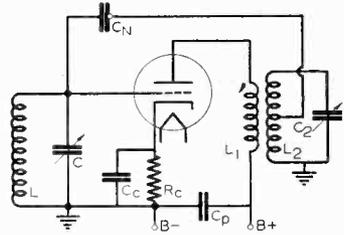


FIG. 13B. A tap on coil  $L_2$  in this neutrodyne circuit gives the required out-of-phase feed-back voltage.

ground, the feed-back current will flow to the screen grid and then to ground, and will not reach the control grid at all. Tube engineers also realized that this new grid could be made to aid the plate in pulling electrons out of the space cloud near the cathode. The resulting circuit arrangement, in which the screen grid is at zero potential for A.C. and at a high positive potential with respect to the cathode for D.C., can be seen in Fig. 14A. The screen grid is connected directly to one terminal of the power supply, and the plate is connected to a higher voltage terminal ( $B++$ ), while by-pass condenser  $C_2$  is placed between screen grid and ground to provide a path to ground for the A.C. feed-back currents. The capacity of  $C_2$  must be much higher than the capacity existing between screen grid and control grid in order to make the feed-back current take the desired path to ground.

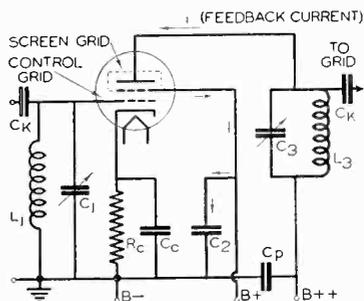
Now you can readily see that if the screen grid by-pass condenser in an R.F. amplifier (such as  $C_2$  in Fig. 14A) should open, as it often does in radio equipment, the feed-back current will flow past the screen grid to the control grid, and *oscillation will occur*, resulting in a squeal in the loud speaker output. Replacement of the screen grid by-pass condenser ( $C_2$ ) will eliminate the squeals.

The effect of the screen grid in reducing the grid-plate capacity of a tube can be determined from any tube chart. A comparison of the inter-electrode capacity values for a typical triode and a typical tetrode tube is given below:

Type 27 (triode) —  $C_{gp} = 3.3$  mmfd.;  $C_{pk} = 3.1$  mmfd.;  $C_{pk} = 2.3$  mmfd.  
 Type 24A (tetrode) —  $C_{gp} = .007$  mmfd.;  $C_{gk} = 5.3$  mmfd.;  $C_{pk} = 10.5$  mmfd.

Observe that in the tetrode the introduction of the screen grid has reduced the grid-to-plate capacity about 470 times, an amount which is more than sufficient to prevent regenerative feed-back.  $C_{gp}$  cannot be entirely eliminated, however, for there is always a little leakage current from plate to grid. Surrounding the plate completely with the screen grid eliminates a great deal of this leakage, at the same time increasing the plate-to-cathode capacity, but increasing  $C_{pk}$  does not appreciably affect the performance of a tuned R.F. amplifier; it simply means that less capacity will be needed in the plate circuit variable condenser, and a reduction in the highest frequency to which the T.R.F. amplifier can be tuned.

FIG. 14A. R.F. amplifier circuit using a screen grid tube. In such screen grid tubes as the 24A, the screen grid completely surrounds the plate, as indicated by the light dotted lines, but the schematic diagram for a screen grid tube generally shows only the heavy lines here indicated.



With the troublesome grid-to-plate capacity out of the way, radio tube engineers proceeded to build greater amplification into the screen grid tube by moving the control grid closer to the space cloud and moving the plate farther away from the cathode. While this did not materially increase the electron flow to the plate, it did increase the A.C. plate resistance. The screen grid tube therefore has a high amplification factor, a high A.C. plate resistance and an average value of mutual conductance; the following table gives a comparison of these values, together with rated plate current values for a triode and a tetrode:

Type 27 (triode):  $\mu = 9$ ;  $g_m = 1,000$  micromhos;  $r_p = 9,000$  ohms;  $I_p = 5.0$  ma.  
 Type 24A (tetrode):  $\mu = 400$ ;  $g_m = 1,000$  micromhos;  $r_p = 400,000$  ohms;  $I_p = 4.0$  ma.

Details of construction of a typical screen grid tube, the type 32, are shown in Fig. 14B; note how completely the plate is surrounded by the screen grid.

*The Performance of a Screen Grid Tube.* The behavior of a tetrode or screen grid tube as an amplifier is expressed by the family of  $E_p - I_p$  curves given in Fig. 15; these curves give average characteristics for the RCA Radiotron type 24A screen grid tube. Notice that plate current is referred to as  $I_b$  instead of  $I_p$ . In securing data for these curves, the tube was operated at a normal filament voltage of 2.5 volts (either A.C. or D.C.) and normal screen grid voltage of 90 volts D.C., and the plate voltage was

varied from zero to a maximum for each value of D.C. control grid voltage (for 0, -1.5, -3.0, -4.5 and -6.0 volts).

For any one value of control grid voltage, plate current rises quite rapidly at first, as the plate voltage is gradually increased from zero. The screen grid has such tremendous electron-pulling power, however, that electrons approach it at very high speeds, pass right through its widely spaced wires and hit the plate with terrific impact, knocking electrons out of the plate; this effect is known as *secondary emission*. Some of these secondary electrons return to the plate, but a great number of them are attracted to the screen grid, which also is at a high positive D.C. voltage.

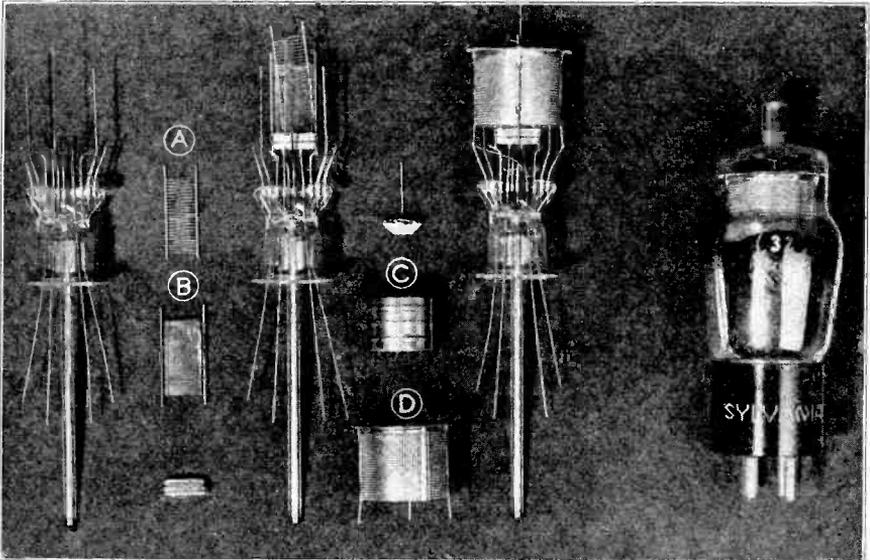


Fig. 14B. Construction of a typical screen grid tube, a type 32 tetrode. The control grid is at A. The inner part of the screen grid (mounted between the plate and the control grid) is at B, while the outer part of the screen grid (which surrounds plate C) is at D. At the right is the completed tube.

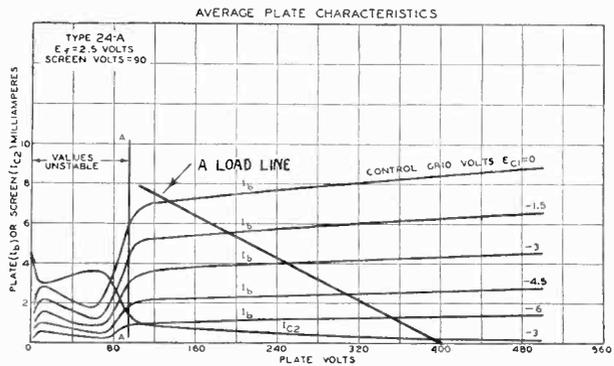
Curve  $I_{c2}$ , representing screen grid current, shows you that for low plate voltages the screen grid current is actually higher than the plate current, indicating that the screen grid collects a high proportion of the electrons which "bounce" off the plate and also attracts electrons directly (the screen grid is at a higher potential than the plate in the region to the left of line AA). It is for these reasons that in the region between zero and 90 volts (labeled "VALUES UNSTABLE" in Fig. 15), increases in plate voltage actually cause the plate current to go down. For plate voltages above 90 volts, plate current rises and screen grid current decreases gradually to a very low value.

When the plate tank circuit in an R.F. amplifier contributes its resonant-produced A.C. voltage to the D.C. plate voltage, it makes the plate-to-cathode voltage swing from high to low values. In order to show how plate current varies under this condition, we can use the load line in Fig. 15; this load line gives us plate current and plate voltage values directly, making it unnecessary to compute the voltage drop across the load.

The plate voltage of the screen grid tube in Fig. 15 cannot be allowed to swing below the screen grid voltage (90 volts in this case) if distortion is to be prevented. Incidentally, if the plate voltage in an R.F. amplifier circuit like Fig. 14A should swing down to about 40 volts, the circuit will act as an oscillator instead of an amplifier; under this condition the screen and plate possess a negative resistance characteristic and supply power to the plate tank circuit. A screen grid tube gives better results than a triode in an R.F. amplifier provided that we limit the plate voltage swing to the linear portion of the  $E_p$ - $I_p$  characteristic curve.

## THE SUPPRESSOR GRID TUBE

Tube designers quickly realized that some means of forcing secondary electrons to return to the plate in a screen grid tube instead of going to the screen grid would greatly improve the amplifying characteristics of the tube at low plate voltage. One scheme for accomplishing this, actually tried in the type 48 power tetrode tube, involved attaching thin metal fins



Courtesy RCA Mfg. Co., Inc.

FIG. 15. Family of  $E_p$ - $I_p$  characteristic curves for a typical screen grid (tetrode) tube, a type 24A tube. ( $I_p$  is here designated as  $I_b$ .)

around the inside of the plate electrode, as shown in Fig. 16A. These fins exert control over the secondary electrons for a considerable distance after they have bounced off the plate. The fins attract far more secondary electrons than does the screen grid, and these electrons then travel back along the fins to the plate. The chief objection to this method is that the fins are expensive to construct; electrical means for accomplishing the same results have been developed.

The introduction of an extra grid between the plate and the inner section of the screen grid proved to be the electrical answer to the problem of driving secondary electrons back to the plate. This extra grid is known as the suppressor grid, for it actually *suppresses* the effects of secondary emission.

Here is how a suppressor grid acts: First of all, it is connected to the cathode or to a terminal which is more negative than the cathode. When an electron bounces off the plate and tries to flow to the positively charged screen grid, it is repelled just enough by the suppressor grid to be forced back to the plate. Even when the suppressor grid is at zero or cathode potential, its action is sufficient to make secondary electrons prefer the plate to the screen grid.

Why is it that the suppressor grid has very little effect upon the electrons flowing from the cathode to the plate, but a great deal of control over those which reverse their direction at the plate? *Speed* is the answer; an electron moving from the cathode through the control grid and the screen grid to the plate has developed so much speed by the time it reaches the suppressor grid that it goes right through the coarse wire mesh without slowing up at all, but secondary electrons coming from the plate have very little speed. We might compare the suppressor grid to a thin sheet of steel, which is easily pierced by a bullet from a high powered gun but which is able to stop a slow speed bullet from a small rifle. This one-way action of the suppressor grid is so efficient that it is even possible to operate the screen grid at the same potential as the plate, improving the electron flow to the plate without affecting secondary emission.

Figure 16B shows a typical circuit for a suppressor grid or pentode tube. Notice that the suppressor grid is connected directly to the cathode. The screen grid is often connected to the same power supply terminal as the plate; thus terminals 1 and 2 may be connected together in some circuits, as they usually are when power pentode (suppressor grid) tubes are used.

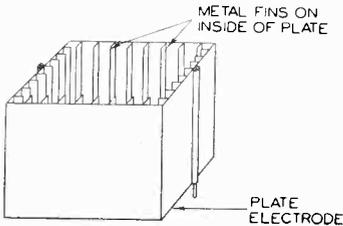


FIG. 16A. Sketch of plate electrode in a type 48 tube, showing metal fins used to limit screen grid current.

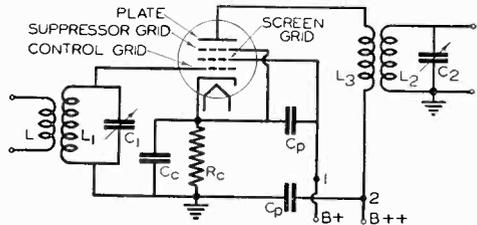


FIG. 16B. R.F. amplifier circuit using a suppressor grid tube. In most pentode tubes like this, the screen grid does not cover the outside of the plate.

The quality of a radio tube is judged entirely by its characteristics. A family of  $E_p-I_p$  characteristic curves for a pentode tube designed for R.F. amplification is given in Fig. 17; notice that the suppressor grid has eliminated the unstable region below the screen grid voltage value which was present in the screen grid tube characteristic curves in Fig. 15. The plate current of this pentode tube practically reaches its full value at low plate voltages, indicating that secondary emission is not depriving the plate of its current. A tube with characteristics like this can be made to swing over wide plate current and plate voltage values, giving maximum amplification and maximum power output for the tube.

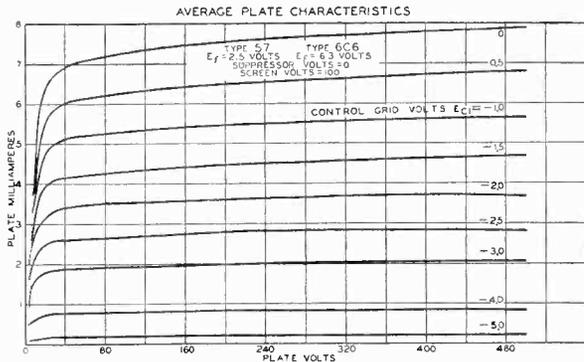
## VARIABLE-MU OR SUPER-CONTROL TUBES

You may already have heard of *variable-mu tubes*, also called *super-control tubes*, for they are quite commonly used in amplifier circuits where the gain (the amplification) is varied by varying the C bias voltage of the tube. These tubes solve the problem of controlling volume by providing a mutual conductance (and therefore an amplification factor  $\mu$ ) which can be varied by *changing the setting of the C bias voltage*. But let us first see what happens when we try to control volume in this way with ordinary tubes.

**Cross-Modulation.** When two stations are heard at the same time on a radio receiver, one being a weak station to which the receiver is tuned and the other being a powerful local station, and this is observed for a number of other desired weak stations as well, *cross-modulation* exists.

Courtesy RCA Mfg. Co., Inc.

FIG. 17. Family of  $E_p$ - $I_p$  characteristic curves for a typical suppressor-grid (pentode) tube, the type 57 and its equivalent in the 6-volt series, the 6C6.



Here is what happens: The weak signal is amplified in the normal manner by the first R.F. amplifier stage, but because of poor selectivity the strong signal also reaches the amplifier tube and is demodulated *because it swings the grid of the first tube beyond the linear region of the  $E_g$ - $I_p$  characteristic curve*. Plate current thus varies at the modulation frequencies of the strong station, just as in any detector, and these A.F. variations serve to modulate the desired R.F. carrier. This carrier, now having both the desired and the undesired modulation signals, goes through the receiver in

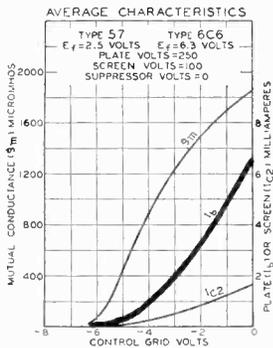


FIG. 18A. Characteristic curves for a type 57 or 6C6 pentode tube having both screen and suppressor grids.

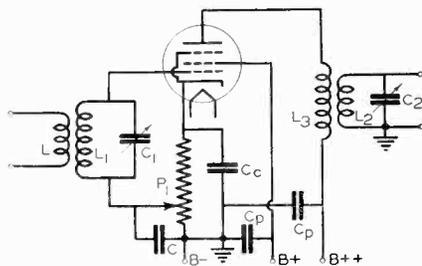


FIG. 18B. R.F. amplifier circuit using a pentode tube. Volume control  $P_1$  varies the automatic C bias value.

the normal manner and both modulation signals are demodulated together.

The characteristic curves for a typical R.F. amplifier tube are given in Fig. 18A; let us consider an actual example of cross-modulation, where the tube represented by these curves is placed in the R.F. amplifier circuit of Fig. 18B. A normal C bias of  $-3$  volts can be obtained for this tube by adjusting  $P_1$ . If the desired signal has a peak voltage of one-half volt, it will swing the grid over a linear portion of the grid voltage-plate current characteristic curve ( $I_b$ ) and the signal will be amplified in a normal man-

ner. If any undesired signal greater than 3 volts gets into this stage, however, it will swing the grid down beyond the bend in the  $I_b$  curve or even beyond the plate current cut-off point. The resulting rectification of the undesired signal causes cross-modulation.

*Modulation Distortion.* The  $E_g-I_p$  curve (curve  $I_b$  in Fig. 18A) for an ordinary screen grid tube has a very pronounced bend in the region of cut-off; the result is that signal voltages which swing the control grid *more positive* cause *large changes* in plate current, whereas signal voltages which swing the grid *more negative* give only *very small changes* in plate current. The harmonics produced under this condition can be partially filtered out by resonant circuits, but the more serious effect, distortion of the wave form of the modulation accompanying the carrier, is difficult to remedy; *modulation distortion* is the term applied to this operating defect of a screen grid tube.

In order to eliminate cross-modulation and modulation distortion effects yet still control volume by means of the C bias voltage, we need a tube which will reduce its mutual conductance as the C bias is made more negative, but which will not have a *sharp* bend in its  $E_g-I_p$  curve at low current values; we need a tube which requires a very high negative C bias to produce plate current cut-off.

An ordinary screen grid tetrode or pentode tube has a sharp plate current cut-off and, since the control grid wires are close together and near the cathode, only a small negative C bias is needed to drive the plate current to zero. This construction, as you know, gives a high amplification factor. If, now, we modify the construction so that some of the control grid wires are farther away from the cathode than others or so some of the control grid wires are closer together than others, we get a tube which has the desired characteristics. The variable spacing scheme is illustrated in Fig. 19A; this is the arrangement commonly used for variable- $\mu$  tubes, since it simplifies construction problems.

This variable spacing of control grid wires gives to the tube two distinct characteristics, one for each spacing, as you would expect. At small negative C bias voltages the action of the tube is essentially the same as if all grid wires were uniformly spaced at the maximum distance, but at high negative bias values the closely wound turns block electron flow; electrons continue to pass through the widely spaced grid turns to the plate until the highly negative cut-off bias is reached, and thus we obtain the desired remote cut-off characteristic.

The mutual conductance and plate current curves for a representative variable- $\mu$  tube are given in Fig. 19B, with the plate current curve ( $E_g-I_p$  curve) for an ordinary screen grid tube (the type 57) shown for comparison. If you compare the plate current cut-off points and the bends in the two  $I_p$  curves, you can verify the statements just made concerning variable- $\mu$  tubes.

Ordinarily screen grid and pentode tubes are quite satisfactory for those circuits where the C bias is such that the A.C. signal swings over a linear part of the  $E_g-I_p$  characteristic, but only a very limited variation in  $g_m$  is permissible if cross-modulation and modulation distortion are to be

eliminated. Variable- $\mu$  screen grid tubes and variable- $\mu$  pentodes (often called super-control pentodes) permit considerable variation in their values of  $g_m$ ; these tubes are well suited for those stages of a well designed amplifier where a C bias type of gain control is used, for they will reduce cross-modulation and modulation distortion.

## PRACTICAL R. F. PENTODE CIRCUITS

Amplifier circuits containing ordinary pentode tubes or super-control pentode (variable- $\mu$ ) tubes are basically the same as the triode amplifier circuits shown in Fig. 9. Remember, however, that pentode tubes have a very high A.C. plate resistance; this means that the conditions for maximum amplification (a load resistance many times higher than the A.C. plate resistance if an ordinary resonant circuit is used as load, or a load resistance equal to the A.C. plate resistance in the case of a tuned trans-

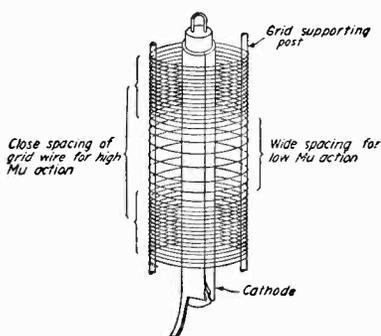


FIG. 19A. This sketch shows the variable spacing used for the control grid wires in a variable  $\mu$  tube.

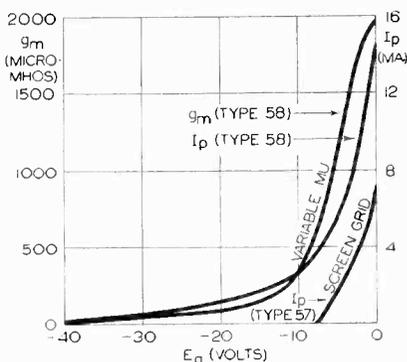


FIG. 19B. Characteristic curves for a variable  $\mu$  pentode tube (type 58), with plate current curve for ordinary pentode (type 57) shown for comparison.

former load for a single-tube stage) are seldom if ever secured. A tank circuit load which is designed for a triode tube is purposely given a low resonant resistance, so obviously a triode load will not be satisfactory in a pentode circuit.

A two-stage R.F. amplifier using pentode tubes and containing a number of different types of R.F. circuits is shown in Fig. 20. The first tube is an ordinary pentode; a 1,500 ohm cathode-to-ground resistor supplies it with an essentially constant automatic C bias. Condensers marked J (.1 mfd.) serve as low-reactance paths for A.C. currents. The second R.F. amplifier stage, containing a super-control pentode tube, is coupled to the first stage by condenser C, which is used both as a D.C. blocking condenser and as an A.C. coupling condenser. The capacity of C is quite low, being about 10 mmfd.; this capacity generally is secured simply by twisting two insulated wire leads together once or twice.\*

\* When the R.F. choke and the tank coil are constructed as a single unit (but with no inductive coupling between them), a stiff wire is sometimes connected to the upper end of choke RFC and looped partly around the grid end of the tank coil L<sub>2</sub> to provide the required low coupling capacity C.



response curve is the product of the responses of each stage. By careful adjustment, very nearly ideal response, with three more or less pronounced peaks, like those shown in Fig. 21, can be obtained. Considerable gain is of course sacrificed in order to secure this response; another disadvantage is that two tubes are required to provide the three tuned circuits.

It is possible, however, to obtain flat response with only a single R.F. stage; several such stages can be used to give the desired amount of amplification, for the gain of the stages is cumulative. Double and sometimes triple tuned circuits are used in place of single resonant circuits; a typical double tuned circuit which is widely used in the I.F. amplifiers of superheterodynes is shown in Fig. 22. Circuits  $L_1-C_1$  and  $L_2-C_2$  are usually identical circuits, coupled to each other by mutual inductance  $M$ . When both resonant circuits are tuned to give maximum output, a single peak response curve is obtained; if each tuned circuit has low circuit resistance, the frequency response curve will be pointed or peaked, whereas high circuit resistance gives a broad, rounded curve. When circuit  $L_1-C_1$  is tuned to a frequency *above* the incoming signal (by reducing the capacity of  $C_1$ ) and circuit  $L_2-C_2$  is tuned *below* the incoming signal (by increasing the capacity of  $C_2$ ), the resulting resonant curve will have a double hump. The

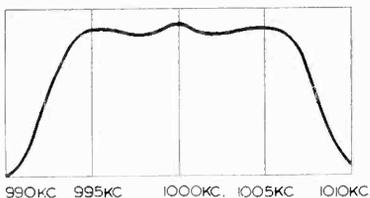


FIG. 21. When the three tuned circuits in the amplifier of Fig. 20 are properly adjusted for band-pass operation, the over-all response characteristic will be as shown here.

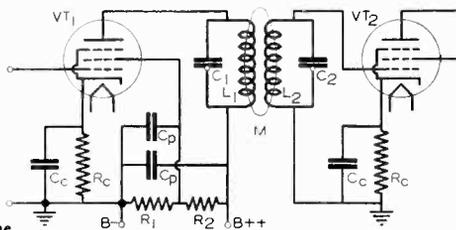


FIG. 22. A double-tuned band-pass radio frequency amplifier circuit, with inductive coupling between the two tuned circuits.

primary could just as well be tuned *below* resonance and the secondary *above* resonance for band-pass results.

The use of three resonant circuits, each coupled to the next and each adjusted to a different frequency, as the load of an R.F. amplifier gives a triple hump response curve. Of course, each extra coil wastes a certain amount of power, lowering the effective value of plate load resistance and consequently reducing the gain, but this loss in gain can be compensated for by using several such R.F. stages in cascade. A tuned amplifier having two or three tuned circuits obviously cannot give as much amplification, even when each circuit is tuned to resonance, as a single tuned circuit because of the losses in the extra coils. Single tuned circuits are therefore often referred to as high gain R.F. amplifiers.

Figure 23 shows another way of using two resonant circuits to give a double peak response curve; capacity  $C$  here couples the two circuits together, for the voltage developed across  $C$  by the  $L_1-C_1$  circuit becomes the source voltage for circuit  $L_2-C_2$ . Coil  $L_1$  is tuned by condensers  $C_1$  and  $C$  in series, while coil  $L_2$  is tuned by condensers  $C_2$  and  $C$  in series. Since condenser  $C$  is usually fifty to one-hundred times as large as either  $C_1$  or  $C_2$ , its tuning effect is negligible.



## POWER R. F. AMPLIFIERS

Maximum power amplification in any single-tube R.F. amplifier stage is obtained when the A.C. plate resistance of the tube is equal to the resonant resistance of the plate tank circuit. When maximum power is desired in the tank circuit, the tubes should have a high mutual conductance. In power R.F. amplifiers, especially those used in transmitters, the circuit must be designed to make maximum use of plate power; since it is ordinarily not feasible to design tank circuits which will have resonant resistances comparable to the A.C. plate resistances of ordinary tetrode and pentode tubes, special power R.F. amplifier tubes have been designed, these having a low A.C. plate resistance which is secured by reducing the amplification factor of the tube and by raising the cathode emission. The use of special tubes is the essential difference between power R.F. amplifiers used in transmitters and those used in receivers. Some power amplifier circuits are designed for peak response at a single frequency, while others (such as circuits handling modulated R.F. currents) are designed for band-pass operation.

*Efficiency.* In high power R.F. amplifiers such as are used in transmitters, the A.C. power in the tank circuits may be hundreds or even thousands of watts; in cases like these the efficiency of operation of the tube is of vital importance.\* When

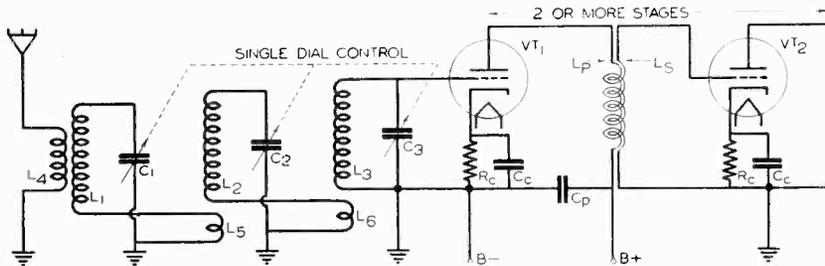


FIG. 24. A fixed R.F. amplifier circuit, with tuning being accomplished by the three resonant circuits which precede the first triode tube.

the C bias of an R. F. amplifier is set to make the input signal swing over a linear portion of the dynamic  $E_g-I_p$  curve, so the tube is operating as a *class A amplifier*, an efficiency of about 30 per cent is obtained even if the grid is driven positive for a part of each cycle.

The plate resistance of the tube is to blame for the greatest part of this wasted power, for all plate current must flow through this plate resistance. Greater efficiency can be obtained by cutting down the time during which plate current flows; by setting the C bias for plate current cut-off, no plate current will flow at normal plate voltage when there is no excitation, and plate current will flow only for one-half of each cycle when grid excitation is applied; this condition, where the tube is operating as a *class B amplifier*, is shown graphically in Fig. 25. Efficiencies of 50% are possible with this circuit. Observe that although the grid excitation voltage  $e_g$  is a perfect sine wave in form, the plate current  $i_p$  is a half sine wave; thus power is being supplied to the plate load circuit for only half of each cycle.

Engineers have proved, both by calculation and experiment, that the half sine wave plate current  $i_p$  in Fig. 25 is made up of a direct current component and an A.C. component having *even* (second, fourth, sixth, eighth, etc.) *harmonic components*; all these components are trying to flow through the  $L-C$  plate tank circuit. The D.C. component readily flows through coil  $L$  to the power supply, but the fundamental A.C. component encounters a high resonant resistance; part of this A.C. component is absorbed by the tank circuit, and the remainder by the tube.

\* The efficiency of an R.F. amplifier is the A.C. power output (in the tank circuit) divided by the D.C. power supplied to the plate circuit; multiply this ratio by 100 to get per cent efficiency. The power supplied to the plate circuit is, of course, the D.C. plate voltage multiplied by the D.C. plate current.

The harmonic components find condenser  $C$  to be a low reactance path; since only the tube is offering impedance to the harmonic currents, the entire harmonic power is wasted in the tube. A tank circuit having a high  $Q$  factor will have an unusually high fundamental A.C. component flowing through both  $L$  and  $C$ .

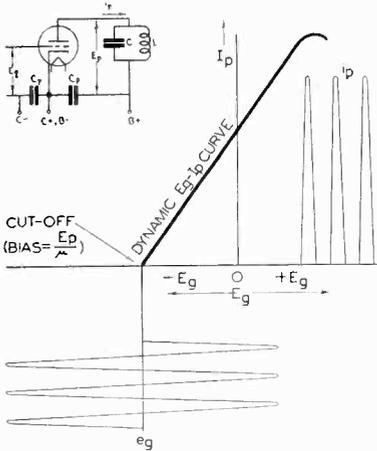


FIG. 25. Circuit diagram and characteristic curve for a typical triode tube operating as a class B amplifier.

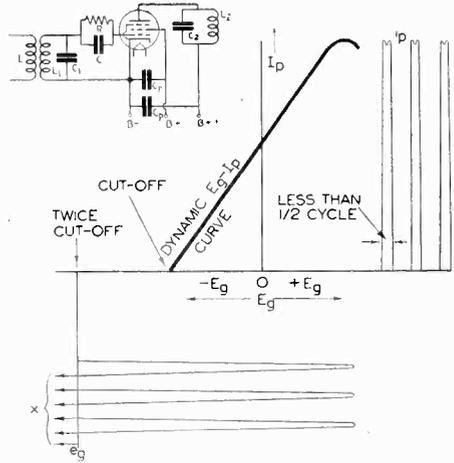


FIG. 26. Circuit diagram and characteristic curve for a typical pentode tube operating as a class C amplifier. The negative half-cycles at  $x$  have been omitted to conserve space on the drawing.

Even greater operating efficiencies (as high as 85%) can be obtained by making the C bias more negative than the cut-off value (usually about twice the cut-off value, as in Fig. 26) and increasing the grid excitation until the maximum possible plate current is obtained; the tube is here operating as a *class C amplifier*. Notice that the resulting plate current pulses in Fig. 26 are almost rectangular in shape, and flow for considerably less than a half cycle. Experiments have shown that

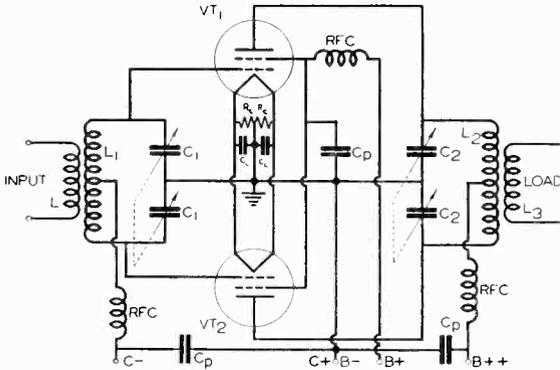


FIG. 27. Typical push-push power R.F. amplifier circuit.

with this wave form, more of the desired A.C. power is absorbed by the tank circuit.

Class C operation is widely used in power R.F. amplifiers where high efficiency and a large amount of power are desired. With both class B and class C operation, coupling the load to the tank circuit coil insures that the load will be fed with power at the fundamental frequency.

*Self-Biased R.F. Amplifier Circuits.* Although a separate C bias supply (like that in the circuit of Fig. 25) is generally used in power R.F. amplifiers, a self-

biased circuit like that in Fig. 26 is occasionally used. Here a pulsating rectified D.C. grid current flows through grid resistor  $R$  when the grid is driven positive by the excitation, thus making that resistor terminal negative which is closer to the grid. Condenser  $C$  in Fig. 26 provides a low reactance path for the A.C. excitation being applied to the grid and at the same time smooths out the pulsating direct current flowing through  $R$ , making the  $C$  bias voltage constant in value.

**Power Amplification.** Since the grid in the circuit of Fig. 26 is drawing current, power is being absorbed by the grid-cathode path in the tube; this power must be supplied by the grid excitation source. The power delivered to the plate tank circuit divided by the power fed to the grid gives what is known as the *power amplification* of a high-power R.F. amplifier stage. The power amplification value is usually about 10, which means that in order to get 1,000 watts out of a power R.F. amplifier, you must furnish about 100 watts of power to its grid.

**Push-Push Power R.F. Amplifiers.** In *audio frequency* amplifiers, a circuit containing two tubes arranged for push-pull or push-push operation is widely used to secure large power outputs, as you already know. These same two-tube circuits can, with slight modifications, be used to deliver high power output as R.F. amplifiers. A typical push-pull or push-push R.F. amplifier circuit is shown in Fig. 27. Split-stator condenser  $C_1$  divides the input voltage between the two tubes. A fixed  $C$  bias voltage is applied to the grids through an R.F. choke and coil  $L_1$ . Each tube feeds into the resonant load circuit made up of center-tapped coil  $L_2$  and split stator condenser  $C_2$ .

## TEST QUESTIONS

Be sure to number your Answer Sheet 16FR-1.

Place your Student Number on every Answer Sheet.

Never hold up one set of lesson answers until you have another ready to send in. Send each lesson in by itself before you start on the next lesson. In this way we will be able to work together much more closely, you'll get more out of your Course, and you will receive the best possible lesson service.

1. What four things must the R.F. amplifier do in a sound or television radio receiver?
2. What does a frequency response curve (or resonance curve) for a tuned R.F. amplifier show?
3. In the modern definition for good selectivity, how many times more should the desired signal frequencies be amplified than the nearest undesired signals?
4. How does the cascading of R.F. amplifier stages affect the amplification and selectivity of an R.F. amplifier?
5. When a tuned transformer load is used in a single-tube R.F. amplifier stage, under what condition is maximum voltage amplification obtained?
6. Of the four circuits shown in Fig. 9 (  $A$ ,  $B$ ,  $C$  and  $D$  ), which one uses a series resonant circuit?
7. Which inter-electrode capacity in a triode tube causes regeneration or oscillation?
8. In an R.F. amplifier like that shown in Fig. 14A, what happens when the screen grid by-pass condenser  $C_2$  opens?
9. What does the suppressor grid do in a vacuum tube?
10. Will variable- $\mu$  (super-control) pentode tubes reduce cross-modulation and modulation distortion?

## **FEAR LEADS TO FAILURE!**

No matter how hard a person may work for success, there is nothing which can help him if he is always doubting his own ability — if he is always thinking about failure.

To be ambitious for wealth yet always expecting to be poor is like trying to get past a vicious dog when afraid of the dog and uncertain of your ability to make friends with him—in each case, fear of failure is almost certain to result in failure. Success, on the other hand, is won most often *by those who believe in winning.*

Never doubt for a moment that you are going to succeed. Look forward to that success with just as much assurance as you look forward to the dawn of another day, *then work—with all that's in you — for success.*

**J. E. SMITH**

**HOW DETECTORS WORK  
IN RADIO AND  
TELEVISION RECEIVERS**

17FR-3

**NATIONAL RADIO INSTITUTE**

ESTABLISHED 1914

**WASHINGTON, D. C.**



# STUDY SCHEDULE NO. 17

For each study step, read the assigned pages first at your usual speed. Reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind, then answer the Lesson Questions for that step. Study each other step in this same way.

- 1. The Fundamentals of Amplitude Demodulation . . . . Pages 1-8  
After reviewing some of the fundamentals of amplitude demodulation, you learn that detection or demodulation consists of two steps: rectification and signal separation. Then you study the two basic methods of signal separation and learn some of the requirements that must be met by practical detectors. Answer Lesson Questions 1 and 2.
- 2. How Diode Detectors Work . . . . . Pages 8-14  
The diode detector is by far the most widely used type today. Here is the basic theory of this detector—the factors determining its fidelity and signal-handling ability. Answer Lesson Questions 3 and 4.
- 3. Practical Diode Circuits . . . . . Pages 14-17  
Here are circuits that you will meet. A brief introduction to a.v.c. is given, followed by details on multi-function tubes. There is also an introduction to a television detector. Answer Lesson Questions 5, 6 and 7.
- 4. The C-Bias or Plate Detector . . . . . Pages 17-20  
The differences between linear and square-law detection explained here will show you why square-law detectors are no longer used as demodulators.
- 5. Heterodyne or Beat Detectors . . . . . Pages 21-22  
The heterodyne or “first” detector in a superheterodyne is an important stage. Instead of being a demodulator, however, it serves to form the intermediate frequency carrier from the incoming and local signals.
- 6. Early Detector Types . . . . . Pages 22-25  
This short section reviews crystal, grid leak, regenerative and other early detectors. Answer Lesson Question 8.
- 7. Demodulators for F.M. Receivers . . . . . Pages 26-31  
Although f.m. demodulators will be described in detail in another lesson, an introduction is given here so that you can compare them with a.m. demodulators. As you will see, they are exactly the same except that the f.m. demodulator must be preceded by a “discriminator” circuit to convert the f.m. signal into an a.m. type. Answer Lesson Question 9.
- 8. Reviewing Methods of Providing C Bias . . . . . Pages 31-36  
Now that you have studied most of the basic radio stages, this review of battery, self, and fixed bias will help you understand some of the many circuit variations you will meet. Answer Lesson Question 10.
- 9. Mail Your Answers for this Lesson to N.R.I. for Grading.
- 10. Start Studying the Next Lesson.

# HOW DETECTORS WORK IN RADIO AND TELEVISION RECEIVERS

## The Fundamentals of Amplitude Demodulation

**T**HE radio signals that are picked up by the antenna of a sound or television radio receiver are modulated radio frequency signals, which are produced at the transmitter by varying an r.f. carrier in accordance with a sound or picture intelligence signal. This results in an r.f. signal that has either an amplitude or a frequency variation corresponding to the intelligence signal. Notice that the intelligence signal, as such, no longer exists. It has been converted into a *variation* in the r.f. carrier, and this variation must be reconverted at the receiver into a duplicate of the original intelligence signal. This reversion is called *demodulation* or *detection*, and is performed by a demodulator or detector stage in the receiver. Essentially, a modulated r.f. signal is fed into the demodulator, and a copy of the intelligence (audio or video) signal is obtained from it. (Superheterodyne receivers use an extra detector which serves another purpose; it is called a beat frequency or heterodyne detector. We will study first the demodulator, then this heterodyne detector.)

► As there are two systems of modulation (a.m. and f.m.) in wide use today, each requiring its own type of demodulator, we must divide detector circuits accordingly. In this lesson, we shall first describe the basic facts

underlying *all* detectors. Then, most of the remainder will be on detectors intended for amplitude-modulated sound waves, although we shall briefly cover several basic f.m. demodulators as well as a basic television detector. (Later in the Course, there will be complete lessons on f.m. and television receivers; in those lessons, these demodulators will be covered in greater detail.)

► To understand more clearly just what is necessary for detection, let us review a few facts about the process of amplitude modulation before we take up a detailed study of how detectors work.

### MODULATION PERCENTAGE

When the broadcast station to which a receiver is tuned becomes silent for a few seconds—say just after the station announcement—the r.f. carrier voltage that reaches the detector input will be unmodulated. The wave form of this unmodulated carrier is shown in Fig. 1A. The peak value  $N$  in this drawing is constant and is a measure of the *intensity* level of the carrier.

If a pure sine-wave sound is picked up by the microphone after the silent period, the carrier will be modulated with this sine-wave signal and, in the process of amplitude modulation, it will have the wave form in Fig. 1B. *M*

represents the peak value of the sine-wave modulation signal; the ratio of  $M$  to  $N$  is a measure of the amount of modulation. In fact,  $M$  divided by  $N$  and multiplied by 100 gives the *percentage of modulation*. For example, if the peak carrier voltage level  $N$  in Fig. 1B is 10 volts and the peak modulation voltage  $M$  is 4 volts, the percentage of modulation will be  $(4 \div 10) \times 100$ , or 40%.

A line drawn through the peaks of the r.f. carrier cycles (such as the dotted lines in Figs. 1B, 1C, and 1D) is called a *modulation envelope*. This envelope, which represents the manner in which the r.f. signal is being

level, the greater will be the demodulated output of the detector in a receiver. However, modulation percentages greater than 100% are prevented at the transmitter, because greater percentages cause interference with other channels and produce distorted signals.

► There are many different modulation envelope shapes since they depend on the modulation voltage and its variations. One possible envelope is shown in Fig. 2A. If a signal with this wave form is fed into an r.f. amplifier stage having a stage gain of 3, the signal that enters the detector will be like that in Fig. 2B, with three

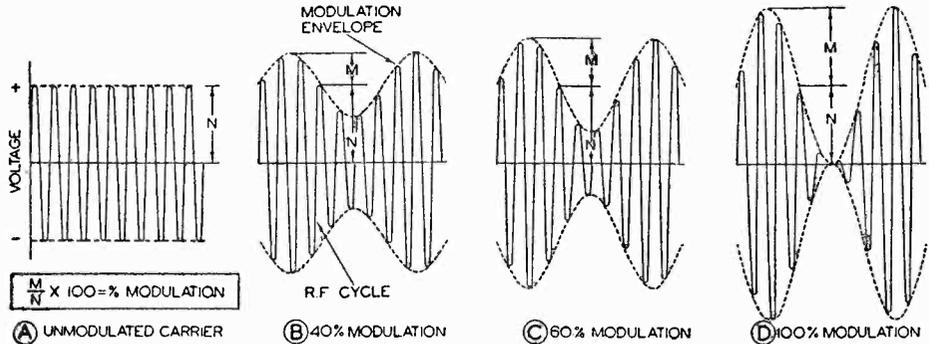


FIG. 1. The unmodulated r.f. carrier at A takes on the wave forms at B, C, and D when modulated with a sine-wave signal at various modulation percentages. These waves can represent currents as well as voltages.

varied, is an *imaginary* line representing a variation of exactly the same frequency as the modulating sound. It therefore *represents* the intelligence signal that we want to obtain from the carrier.

When the carrier level at the transmitter is held constant and the modulation signal level is increased, the percentage of modulation increases. When the two voltages,  $M$  and  $N$ , are exactly equal, there is 100% modulation, and the wave form will be like that in Fig. 1D. As we shall see, the greater the percentage of modulation of a transmitter for a given carrier

times the amplitude but exactly the same wave form as the input signal at A. Both carrier and modulation are amplified equally, so the percentage modulation remains unchanged after amplification.

In the study of detectors, just as in the study of other radio circuits, it is much simpler to consider only sine-wave modulation signals and sine-wave carrier signals. If the detector will handle a sine-wave modulated signal without distortion, then it will handle the corresponding component of a complex wave equally as well.

**Demodulation Steps.** Now, let's

see just what a detector is required to do. We know that the intelligence signal exists only as an r.f. variation. Our detector circuit, then, must respond to the *variations* in the carrier. In other words, when we feed a varying r.f. signal into our detector, the output voltage of the detector must be proportional to the variations in the signal (that is, this voltage must follow the modulation envelope). The output of the detector will then be the intelligence signal that we want.

As you see from Fig. 1, there are two modulation envelopes in the r.f. signal, one drawn through the positive r.f. peaks and the other drawn through the negative r.f. peaks. One is the exact opposite of the other and, as

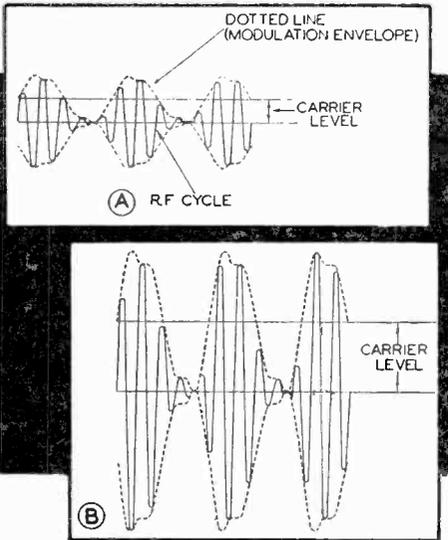


FIG. 2. One of the many possible wave forms of an r.f. carrier modulated with an actual sound signal (middle C as produced by a piano) is shown here before and after amplification.

shown in Fig. 2, an amplifier stage amplifies *both* of them. However, the demodulator must follow the variations of only one of the modulation envelopes, so we must get rid of one of them.

We do so by *rectifying* the r.f. signal. By cutting off half of each r.f. cycle, we will obtain a series of pulsations having peak values that follow one modulation envelope. Then, we use a *filter* to obtain the intelligence signal from these pulses.

► To sum up: To get exactly the signal we want from an amplitude

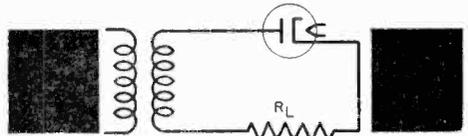


FIG. 3. A basic diode rectifier. This circuit performs one of the functions of a demodulator.

modulated r.f. carrier, we must do two distinct things to the r.f. signal. First, the incoming signal is rectified, so that half of each cycle is removed. The result is a series of r.f. current pulses that vary in accordance with the wave form of the intelligence signal. This rectified signal is then passed through a filter which, in effect, *separates* the desired intelligence from the undesired r.f. current. Keep these two steps (1, *rectification*; and 2, *signal separation*) in mind as you study the various types of detector circuits.

(Incidentally, some engineers call signal separation "restoration," since the intelligence signal is re-created from the r.f. pulses by the process.)

## RECTIFICATION

Suppose we see what happens when we feed r.f. signals into a simple half-wave rectifier circuit like that shown in Fig. 3. This circuit consists of a diode rectifier, a load resistance  $R_L$ , and a transformer through which we introduce the r.f. We will assume that the diode has a linear (perfectly straight) characteristic curve.

As you know from your previous

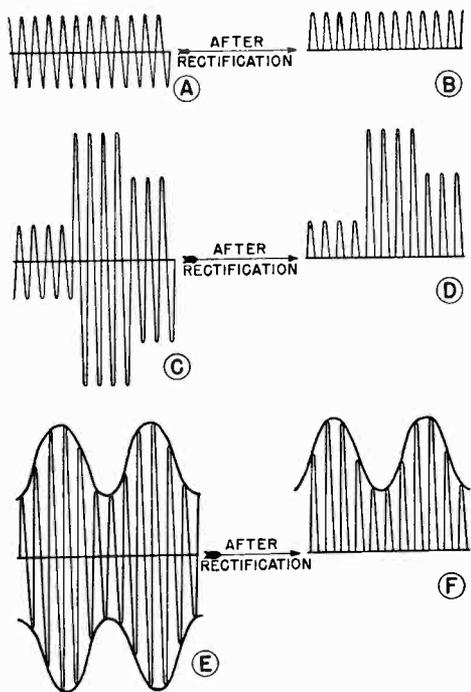


FIG. 4. The results of rectification of various signals are shown at B, D, and F.

study of half-wave rectifiers, if an unmodulated r.f. carrier (Fig. 4A) is fed in, the plate current will consist of a series of pulses (Fig. 4B).

Now, suppose we feed in a signal like that shown in Fig. 4C. The peak value of this signal is varied suddenly; first it is like that in Fig. 4A, then it is tripled in magnitude, and finally it is made double the peak value of the original. The rectified plate current (Fig. 4D) will again consist of pulses, this time of three different sizes that correspond to the three peak values of the unrectified signal voltage.

► Now, let us feed an r.f. carrier modulated by an audio-frequency sine-wave voltage (Fig. 4E) into our rectifier. This modulated carrier is much like the carrier in Fig. 4C, except that its peak values vary more smoothly, following the shape of the

modulating sine wave, instead of increasing or decreasing in sudden steps. When we rectify this sine-wave modulated signal, we get a series of pulses in the plate current (Fig. 4F), varying in peak value as the peak value of the modulated r.f. signal varies.

Thus, passing an r.f. voltage through a rectifier will give a series of half-wave r.f. pulses with an envelope or outline that follows the shape of the envelope of the modulated signal.

► This series of pulses is still not the intelligence signal that we want, but an analysis of these pulses would show that we have a complex wave containing the desired signal. As shown in Fig. 5, the pulses at A have at least three components: (1) a steady value of d.c. (Fig. 5B); (2) an a.c. component that has the same frequency as the modulating sine wave (Fig. 5C); and (3) an a.c. component that has the same frequency as the r.f. carrier

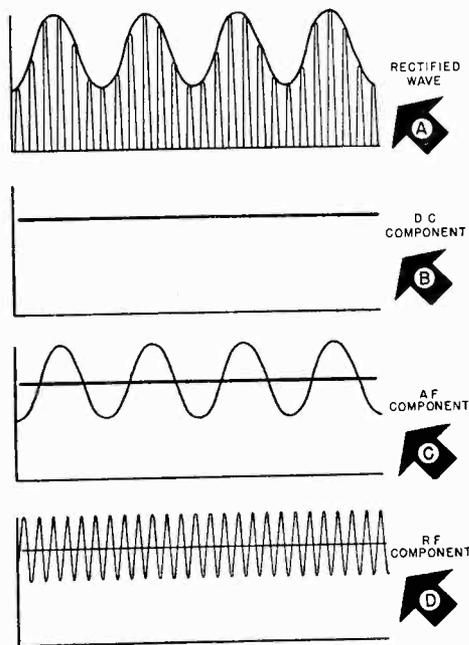


FIG. 5. A rectified wave has d.c., a.f., and r.f. components, as shown here.

(Fig. 5D). (In addition, there will be a number of harmonics of the r.f. carrier that we can ignore.) Therefore, all we need now is some way to separate the component we want from all the others.

► The currents and voltages in an actual detector circuit are never this simple, principally because, as you will learn a little later, we seldom get perfectly linear rectification. Non-linear rectification tends to distort

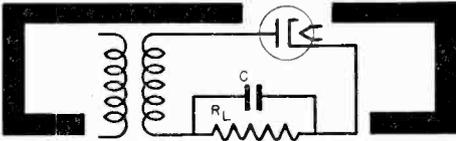


FIG. 6. This is a complete diode demodulator of a basic type. This circuit can rectify and separate the signal components so that the desired intelligence signal is available across  $R_L$ .

the wave form of the intelligence signal, adding harmonics to it. However, here at the beginning of your study of detectors, we will assume that we have perfect conditions and will leave the distortions produced in practical circuits for discussion in later sections of the lesson.

### SIGNAL SEPARATION

The rectifier circuit shown in Fig. 3 has accomplished one of the steps of demodulation. We must now separate the components of the pulses so as to obtain the desired intelligence signal. This is easily accomplished by using a filter which, in its simplest form, consists of a condenser placed across the load resistor as shown in Fig. 6.

This gives us the equivalent circuit shown in Fig. 7. The pulses can be considered as coming from the source  $E$ , and as being fed through the plate resistance  $r_p$  to the combination of  $R_L$  and  $C$ . Since the operation of this circuit depends to some extent on

whether the value of  $r_p$  is high or low, we'll study its workings under both conditions. First, let's see what happens when  $r_p$  is high.

**The R-C Filter Action.** If the value of  $r_p$  is high, and  $C$  is chosen to have a low reactance at the carrier (r.f.) frequency, then  $r_p$  and  $C$  will form an ordinary resistance-capacitance filter. Most of the r.f. component will be dropped across  $r_p$  and little will be left across  $C$  and  $R_L$ .

If  $C$  is properly chosen, it will have little effect on the intelligence frequencies, and none on the d.c. (assuming that there is no leakage), so these two components will appear across  $R_L$ . Of course,  $R_L$  and  $r_p$  form a voltage divider for these components, but if  $R_L$  is much higher in value than  $r_p$ , then most of these voltages will appear across  $R_L$ . (Since the condenser offers a lower impedance to the r.f. carrier than it does to audio frequencies, we may say that the condenser

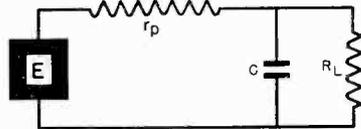


FIG. 7. The equivalent circuit for the signal separation function of a diode demodulator.

“by-passes” the r.f. pulsations around resistor  $R_L$ , leaving the d.c. and audio components.)

► The value chosen for  $C$  is important. The larger the capacity the better the r.f. components are filtered out, but  $C$  then filters out the higher intelligence frequencies as well. It is necessary to have  $C$  small enough so that the desired intelligence signal components will not be lost, even though this means less perfect r.f. filtering.

► When the filtering is accomplished as just described, by an R-C filter, the

current through  $R_L$  is the *average* of the diode pulsations, as shown by the heavy line in Fig. 8. The faithfulness with which this average value follows the peak value of the pulses determines the amount of distortion introduced by the detector, as we shall learn later.

### The Condenser - Input Filter.

When the value of  $r_p$  is very low, as it is in most diode tubes, we have to consider the action in Fig. 7 in a different manner. We now have to consider the tube as an electronic switch that completes the circuit through its low resistance whenever its plate is positive with respect to its cathode, but opens the circuit whenever the polarity is reversed.

Now, the condenser acts very much like the input condenser of a power-pack filter. It is charged up rapidly through the relatively low resistance of the tube when tube current flows, but, when the current is cut off,  $C$  must discharge through resistor  $R_L$ . The rate of discharge depends on the



FIG. 8. How the voltage across  $R_L$  follows the average of the rectified pulses when the plate resistance of the circuit in Fig. 7 is high.

time constant of  $R_L$  and  $C$  and (since  $R_L$  is a high resistance) it is far slower than the charging rate. This rapid charge — slower discharge — action means that the condenser voltage tends to follow the peaks of the r.f. pulses, as shown in Fig. 9. Hence, it almost exactly follows the peak value of the modulation envelope. Thus, it “smooths out” the r.f. pulses, leaving only the intelligence and d.c. components.

As the condenser voltage is also the

voltage across  $R_L$ , the load voltage also follows the intelligence signal instead of the r.f. signal—in other words, we have signal separation.

► It is usually better to have  $r_p$  low in value, rather than high. If  $r_p$  is low in value, the load voltage follows the peaks of the pulsations, and we ob-

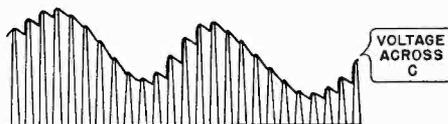


FIG. 9. A low value of plate resistance in the circuit in Fig. 7 permits the  $R_L$  voltage to follow the peaks of the rectified pulses.

tain a higher voltage across  $R_L$ . Further, as you will see a little later, we can get less distortion when  $r_p$  is low.

### THE DETECTOR OUTPUT

After signal separation — whether we follow the average or the peaks, or have a combination of the two actions — we will have across  $R_L$  an intelligence signal voltage, a d.c. voltage, and perhaps a small r.f. ripple. The signal across  $R_L$  is then fed to a low frequency amplifier—usually through a coupling condenser that blocks the d.c. component, passing only the desired intelligence signal (and a small r.f. ripple which we can ignore). As we shall see later, the d.c. component of the signal across the detector load is useful for control purposes.

► Incidentally, a certain amount of signal separation can occur in the low-frequency amplifier. However, if we apply high r.f. voltages to the low-frequency amplifier it may drive the first audio tube off the straight portion of its characteristic. This will introduce distortion, so we wipe out as much r.f. as possible before applying the signal to the low-frequency amplifier.

► This completes the story of the

*basic* steps of demodulation. However, there are many characteristics of detectors that we have not yet touched upon—distortion, for example. Suppose we now find out about some of the requirements placed on detectors, then go on to a study of typical circuits.

## DETECTOR REQUIREMENTS

There are three important performance characteristics on which detectors are rated: *sensitivity*, *fidelity*, and *voltage-handling ability* (sometimes called power-handling ability).

**Sensitivity.** Radio men are always interested in how much of the intelligence signal voltage they can obtain from a detector when it is fed an r.f. voltage at a definite carrier level and a given percentage of modulation. The sensitivity rating of a detector gives this information, for it is the *ratio of the intelligence voltage output to the modulated r.f. voltage input*. The more sensitive a detector is, the higher will be its output voltage.

With modern high-gain circuits, we are able to get more than enough amplification, so that we can afford to use detectors having no gain. However, in the early days of radio, before high-gain tetrode and pentode tubes were developed, high sensitivity was one of the most important requirements of a detector.

► We might take the time to cover the history of detectors briefly and see how this requirement has changed. The diode tube was used as a detector as soon as it was developed, but it was dropped after the invention of the triode, because triode detectors gave both amplification and detection. The triode was used first in a form of the C-bias circuit, but this was replaced soon by the grid leak-condenser square-law detector circuit which

gave a high output (along with severe distortion). When even more gain was sought, regeneration was added to this circuit. (These circuits will be studied later in this lesson.)

With the development of stable r.f. circuits and more powerful transmitters, there was less need for high gain, and the public demanded better fidelity. First regeneration was dropped, then the linear C-bias detector was developed. This type provided gain and improved fidelity.

Today, the diode has replaced both the grid leak-condenser and the C-bias types, as there is no longer any need for gain in the detector stage. These older types are of interest to us only because receivers using them are still being serviced. (The early crystal detectors are not found in broadcast receivers except in the simplest types intended for school-boys. However, a modified type of crystal detector is now coming into use as an ultra-high frequency detector.)

**Fidelity.** A transmitter is considered to have high fidelity when the envelope of the modulated-carrier current is a true reproduction of the intelligence signal. The receiver r.f. stages can alter the wave, particularly by side-band cutting, so it is only fair to consider detector fidelity in terms of the signal fed into it, rather than in terms of the radiated wave. Hence, we compare the wave form of the output of the detector with that of the envelope of the modulated signal fed into it; the more nearly these two wave forms correspond, the better is the *fidelity* of the detector. High fidelity—in other words, lack of distortion—is obviously desirable in a detector.

**Voltage Operating Range.** The range of input signal voltages over which a detector will operate with satisfactory fidelity is by no means un-

limited. Some detectors can handle only weak carrier signals without distortion, while others will distort on weak carriers yet work perfectly well on strong carriers. The percentage of modulation of the incoming signal also limits the operating range of a de-

tor; some circuits work well at high percentages of modulation, while others do not.

► Let us now study some of the practical detector circuits, and learn how they compare in their performance characteristics.

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## How Diode Detectors Work

We used a diode detector in the preceding section to explain the fundamentals of detection. However, in that preview, we made a number of assumptions and told only the basic story. We'll now study the details of diode detector operation.

### R.F. LOADING

Fig. 10 shows a diode detector circuit that is typical of those in modern superheterodyne receivers. The i.f. transformer feeding the detector is almost always tuned, so we can represent it by  $L_1-C_1$  and  $L_2-C_2$ . The intelligence signal developed across the load  $R_L$  is fed through coupling condenser  $C_4$  to the low-frequency amplifier.

The resonant circuit  $L_2-C_2$  acts as the source, and must supply power because the diode tube passes current whenever its plate is positive with respect to its cathode. This "loading" reduces the tuned circuit Q because the effect is that of connecting a resistance across the  $L_2-C_2$  terminals. As a reduction in Q reduces both the output and selectivity, we cannot permit too much loading.

As shown in Fig. 11, the diode plate resistance  $r_p$  and the diode internal plate-to-cathode capacity  $C_{PK}$  form a parallel group which is in series with the  $R_L-C_3$  group, with the entire

combination connected across the terminals of  $L_2-C_2$ .

We can assume that, at the carrier frequency, the reactance of  $C_3$  is very low so that  $R_L$  is effectively shorted for r.f. This leaves  $r_p$  and  $C_{PK}$  across the resonant circuit. We can dismiss the effect of  $C_{PK}$  because it serves only to detune the resonant circuit and we can adjust  $C_2$  to compensate for it. Hence, the actual loading depends on the value of  $r_p$ .

Fortunately, the effective value of  $r_p$  is increased and the loading is reduced by the action of  $C_3$  and  $R_L$ . You know that both the intelligence frequency component and the d.c. component are stored in  $C_3$ , with the polarity shown in Fig. 10. Hence, we have *two* voltages applied to the diode—the incoming signal and the voltage stored in  $C_3$ . Since the  $C_3$  voltage is only slightly less than the peak value of the incoming signal, and since its polarity tends to make the diode plate negative, the tube can conduct only during the relatively short time that the incoming signal is able to overcome the  $C_3$  voltage and make the plate positive. The plate current becomes a series of short, high pulses having a low average value.

Increasing the values of  $C_3$  or  $R_L$  will increase the time constant and will cause the  $C_3$  voltage to follow the peak value of the incoming signal even

closer. This reduces the plate current and the loading even further. However, as we will show, this increase in time constant lessens the ability of the circuit to pass the higher intelligence frequencies; this sets a limit to the values we can use. For this reason, we must accept some loading when we use a diode detector. This is not too serious since modern i.f. amplifiers usually provide enough selectivity and gain to make up for most of the effects of loading.

**Service Hint.** Notice the importance of condenser  $C_3$  to the r.f. circuit. If  $C_3$  opens, several things happen. The shunting effect of the internal tube capacity is reduced and, in addition, the loading on the tuned circuit is changed. More important is the fact that there is now a voltage divider (the tube and  $R_L$ ) for r.f., and a large portion of the signal voltage will be dropped across  $R_L$  instead of being applied to the tube. As a result of this reduced r.f. voltage across the

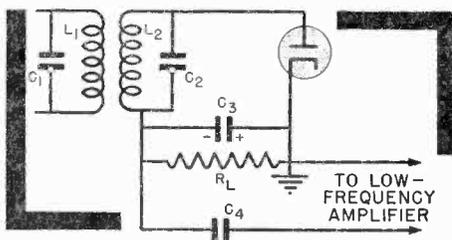


FIG. 10. The input and output connections of a diode demodulator.

tube, the plate current pulses will be far smaller, so the output will be reduced greatly. And, of course, the absence of  $C_3$  means that the detector output is now a series of r.f. pulses, and signal separation can occur only in the low-frequency amplifier.

► When you find that the output from a detector is far below normal, you should first suspect a defective

diode tube, then check the alignment. Keep  $C_3$  in mind, however, as it may be defective.

## FIDELITY CONSIDERATIONS

Naturally, we are always interested in how faithfully a detector reproduces the original audio signal — or, to put it another way, how much distortion is present in the output of a detector. There are two possible causes

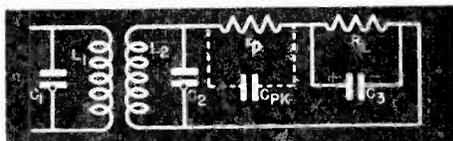


FIG. 11. How the diode circuit can act as a load on the input resonant circuit.

of distortion in the output of the diode detector we've just studied: (1) there will be distortion if the diode is not perfectly linear in its operation, and (2) there will be distortion if all the audio signal is not reproduced across the load. The first of these is primarily caused by the tube, while the second is primarily a function of the load and r.f. filter.

## NON-LINEAR TUBE CHARACTERISTIC

It's easy to see why a tube that is not linear in its response will cause distortion. A diode tube has linear response if its plate voltage-plate current characteristic is a straight line, because then equal changes in plate voltage will produce equal changes in plate current. The response of such a tube to a modulated r.f. carrier is shown in Fig. 12. As you can see from this figure, the envelope of the rectified pulses is exactly like the envelope of one half of the input signal. This means the a.f. current produced by the tube has the same wave shape as the

modulation envelope, and, if the voltage across the tube load has the same form as the tube current, there will be no distortion caused by the tube.

If the tube characteristic is not linear, but is instead curved like that in Fig. 13, then the envelope of the rectified pulses is no longer precisely like the carrier envelope; therefore, the a.f. voltage developed across the load resistor will not have the same form as the original audio. Distortion will have been introduced by the non-linear response of the tube. This distortion consists principally of the addition of *even* (the second, fourth, etc.) harmonics to the original signal.

Modern diodes have relatively straight characteristics, and the use of high load resistor values tends to straighten them out further. Even so, there is usually some curvature near the zero-current point of the curve. Let's see what this can cause.

### MODULATION PERCENTAGE AND SIGNAL INPUT LEVEL

The amount of signal and the percentage of modulation have much to do with how much distortion may occur in diode detectors.

As we have shown, most diodes have characteristics that are relatively straight except at the bottom, where

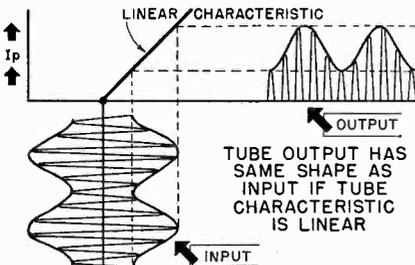


FIG. 12. The output follows the input exactly when the detector characteristic is perfectly linear.

they curve like the characteristic in Fig. 13. Such a tube will give undistorted response as long as the modulation envelope (the peaks of the r.f.) swings over the linear part of the characteristic, but will have a somewhat distorted response if the modulation envelope swings down over the curved part of the characteristic.

Now, the percentage of modulation

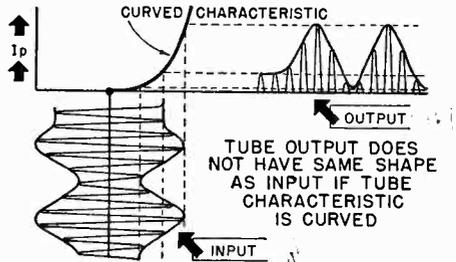


FIG. 13. A curved tube characteristic causes distortion.

partly determines whether or not the envelope will reach the curved part of the characteristic. Assuming we have only 40% modulation, like the wave in Fig. 1B, and the signal is fairly strong, the envelope does not reach the curved section at the bottom of the characteristic, and the detector output therefore will be undistorted.

If, however, the carrier is 100% modulated, as in Fig. 1D, the modulation envelope cannot help but swing down over the curved part of the characteristic. Part of the wave will then be distorted.

► The strength of the signal determines how bad the distortion will be. If the signal is weak, even the highest signal peaks will not go very far up on the characteristic. Consequently, the modulation envelope will swing mostly over the curved part of the characteristic. There may be severe distortion, even if the carrier is only 40% or 50% modulated. On the other

hand, if the signal is very strong, practically all the modulation envelope will swing over the linear part of the characteristic. Even if the modulation is 100%, only a very small part of the output will be distorted. Thus, a diode detector produces the least distortion when it is fed a strong signal.

This is a very important fact. Broadcast stations try to use a high percentage of modulation (up to 100%), which, as you just learned, will cause considerable distortion in a diode detector unless the signal is strong. Obviously, then, it is desirable to build up the signal as much as possible before applying it to the diode. This is one of the reasons that the diode detector was not widely used until high-gain tetrode and pentode tubes were developed for use in r.f. and i.f. stages preceding the detector.

**Power Detectors.** Today, the diode is commonly used as a power detector. This is simply a detector that can handle very large r.f. voltages (from about 5 to 50 volts) without appreciable distortion.

### THE R-C TIME CONSTANT

Distortion will be introduced also if the voltage developed across the load resistor does not have the same form as the envelope of the rectified current pulses. This distortion is caused primarily by the time constant of the r.f. filter condenser and the load resistance.

To see what will happen, let's assume that the time constant is too long to follow the modulation frequency shown by the dotted line in Fig. 14A. As you will notice, the condenser voltage follows the peaks satisfactorily between points X and Y of this figure, where either the carrier is

not changing or the modulation is swinging positive. When the modulation signal swings in the other direction, however, the condenser voltage decreases along the heavy line between points Y and Z instead of following the dotted line. This means that we get the output wave shown by the heavy line in Fig. 14B instead of the sine-wave signal shown by the dotted line. Higher modulating frequencies than that shown are followed even less well; they are not only distorted, but are also reduced in amplitude. Thus, too high a time constant causes two kinds of distortion: some frequencies are not reproduced as well as others (this is called frequency distortion); and the shape of the output

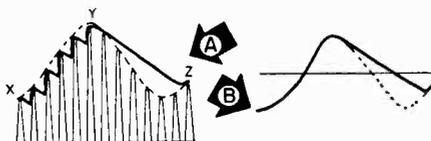


FIG. 14. If the R-C time constant is too high, a distortion known as "clipping" occurs, as illustrated here.

wave is changed, showing that harmonics have been introduced (amplitude distortion).

We are therefore definitely limited as to the time constant we can use. The values of  $R_L$  and  $C_3$  must be made sufficiently small to allow the condenser to discharge fast enough to follow the highest frequency that we want.

► In practice, design engineers usually choose the value of  $C$  first, using a value about 5 to 10 times the tube capacity  $C_{PK}$  to minimize the r.f. loss. Then, with this value fixed, they choose the highest value of  $R_L$  that will give the desired fidelity.

Fig. 15 gives examples of the attenuation introduced by different values of  $R_L$  when the capacity is fixed

at 250  $\mu\text{fd}$ . You can see that smaller values of  $R_L$  give less frequency distortion. (However, small values of  $R_L$  permit more loading on the resonant circuit and more amplitude distortion in the tube.)

► To have high-fidelity detection, we must reduce to negligible values both distortion produced by the tube and distortion produced by the load. Unfortunately, the measures we can take to correct the first tend to accentuate the second. That is, a diode is more nearly linear in its response when the load used with it has a high ohmic

(An r.f. choke coil can be used in place of  $R_L$ , but it is less expensive to use the resistor and the results are just as good.)

The improved filtering given by the R-C filter makes it possible for condensers  $C_2$  and  $C_3$  to be smaller. In fact, their sum can be less than the value needed for  $C_3$  in Fig. 10. This means that the higher intelligence frequencies are not cut off as much. This scheme allows higher fidelity, and permits the use of load values up to 500,000 ohms.

► It is possible to reduce the distortion in a diode detector stage to a very low value by selecting a tube that has a nearly linear characteristic and by choosing the values of the load with care. (The human ear will usually put up with more frequency distortion than amplitude distortion, so usually the load resistance is kept fairly high to reduce the amplitude distortion in the tube.) As a practical matter, this means that a diode detector can have virtually perfect fidelity, because even a highly trained ear cannot distinguish small amounts of distortion.

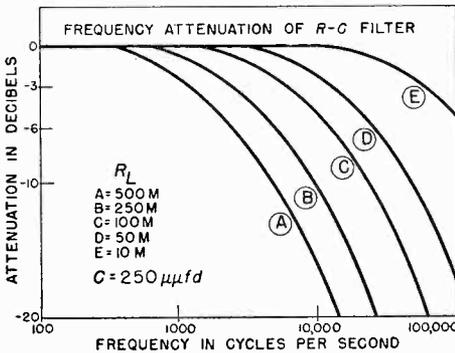


FIG. 15. How increasing the load resistance causes the high frequency response to fall off (a form of frequency distortion).

value, but this means that the time constant ( $R$  times  $C$ ) of our load circuit will be rather high; as you just learned, this tends to cut off the higher audio frequencies.

**R.F. Filters.** When the r.f. by-pass condenser ( $C_3$  in Fig. 10) is the only r.f. filter, its capacity must be kept high so that a minimum of r.f. voltage will be applied to the low-frequency amplifier. One way to keep this r.f. voltage low, and at the same time use a smaller capacity, is to use a complete r.f. filter as shown in Fig. 16. Here,  $C_2$  acts as an input filter condenser, while the  $R_1$ - $C_3$  combination forms a standard R-C filter section.

## ISOLATION OF VOLUME CONTROLS

Now, let's take up a variation in the connections that you will encounter—one that arises because many receivers use volume controls as the diode detector load. The control is used to vary the input voltage fed to the low-frequency amplifier.

A volume control has a movable slider which will eventually wear out the resistance element. The resulting poor contact will produce noise. These controls become noisy much more quickly when d.c. current flows through them. As you know, a d.c. current flows through the diode detector load resistance because the

filter circuit removes only the r.f. pulses and leaves the low-frequency and d.c. components of the signal. To increase the life of the volume control, some manufacturers isolate the control from this d.c. current.

Fig. 17 shows the usual connection with the volume control  $R_c$  used as a diode load. The coupling condenser  $C_2$  feeds the low-frequency component of the signal to the low-frequency amplifier, blocking out the d.c. from the amplifier input but not from the volume control.

By adding another blocking condenser and another resistor, as shown in Fig. 18, it is possible to keep d.c.

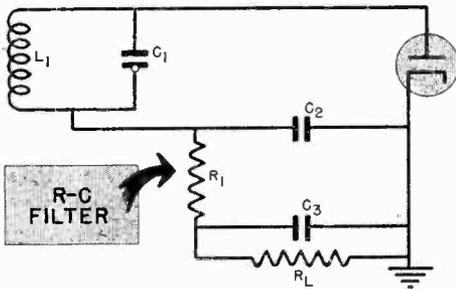


FIG. 16. The  $C_2$ - $R_1$ - $C_3$  filter shown here is found in most receivers, since it reduces the r.f. component without so much loss of the higher intelligence frequencies.

out of the volume control.\* Notice that the added resistor  $R_1$  is used as the diode load resistor, and the additional coupling condenser  $C_3$  passes only the low-frequency signal to the volume control  $R_c$ . From here, condenser  $C_2$  passes on the low-frequency signal to the low-frequency amplifier.

With this arrangement, only the low-frequency signal flows through  $R_c$ . The d.c. in the diode circuit flows

\*Of course, the circuit shown in Fig. 17 costs less than the set-up in Fig. 18. That is why the former is so widely used.

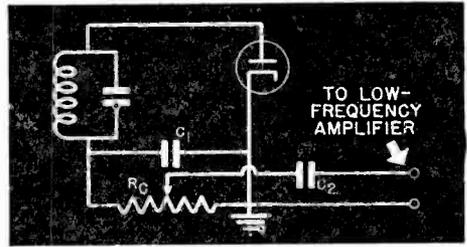


FIG. 17. Volume controls are frequently used as the detector load, connected in the manner shown here.

through resistor  $R_1$ , not through the volume control.

When the volume control is used as the detector load as in Fig. 17, its value is chosen just as that of a fixed load resistor is.

► When the control is isolated as in Fig. 18, however, it is desirable to use a higher resistance control than that used in Fig. 17. As you know, the reactances of  $C_3$  and  $C_2$  are negligible at the middle audio frequencies. This means that, as far as a.c. is concerned,  $R_c$  (and any resistance between the output terminals) will be effectively in parallel with  $R_1$ . Since two resistors in parallel have a total resistance that is lower than either of them has individually, there must be an increase in the values of  $R_c$  and  $R_1$  to make their total resistance higher, or we must accept a lower a.c. load resistance in the diode circuit.

A lower a.c. load would mean that

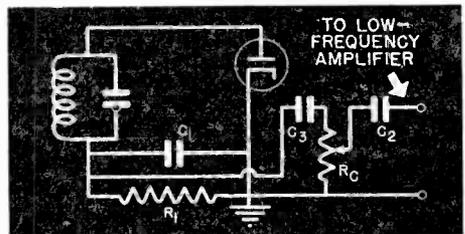


FIG. 18. There will be far less noise and longer control life when the volume control is isolated so that d.c. current does not flow through it.

less of the signal would be developed across the load, and also that a certain amount of amplitude distortion would occur.

► As a practical case, the volume control resistance in Fig. 17 is commonly 250,000 ohms. However, if the volume

control is isolated for longer life by using the circuit in Fig. 18, then resistor  $R_1$  may be about 250,000 ohms and the volume control  $R_c$  may be between 500,000 ohms and 1 megohm. (The additional coupling condenser  $C_3$  is usually from .05 mfd. to .1 mfd.)

## Practical Diode Circuits

In most practical diode detectors, the d.c. voltage developed across the load resistor is used for control purposes. A typical circuit is shown in Fig. 19.

In this figure, tube  $VT_2$  is the diode detector. Its circuit and action are precisely like the diode we just described.

The amplifier stage using  $VT_1$  is also standard, except that its grid return lead has  $R_2$  in it and is connected to the diode load resistor  $R_1$  instead of going directly to ground. Tracing from the amplifier cathode (point 1), you see that the bias voltage developed across  $R_g$  is applied to the grid through  $R_1$  and  $R_2$ . (Trace from point 2 through the chassis to point 3, then to points 4 and 5.)

The amplifier stage uses a variable-mu or super-control tube, so the am-

plifier gain will vary with the bias. The gain is maximum when the bias is minimum. In Fig. 19, the gain will be maximum when no signal is tuned in, for then the only bias is the self-bias developed across  $R_g$ .

However, when a signal is received, a d.c. voltage will appear across  $R_1$  with the polarity shown in Fig. 19. (Check the polarity by tracing the direction of electron flow in the diode circuit.) This voltage, whose value depends upon the strength of the carrier voltage applied to the diode, adds to the  $R_g$  bias voltage (trace from point 1 to point 4: the two voltages are in series, connected so that their voltages add). This increase in bias reduces the gain of the amplifier.

It is possible to choose part values for these circuits that will keep the

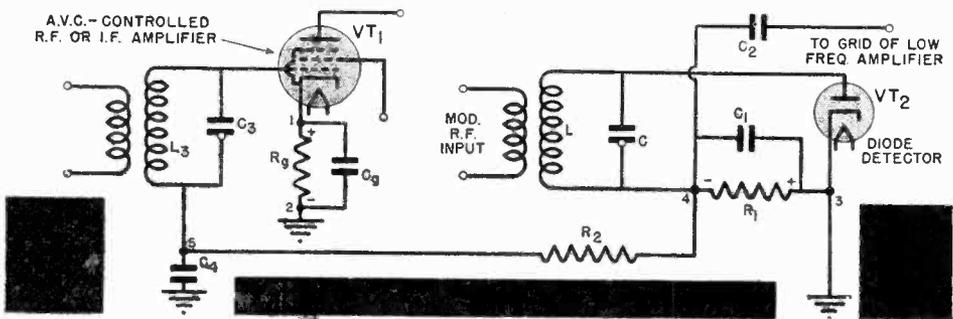


FIG. 19. A basic a.v.c. circuit.



making it desirable to use a separate diode for production of the a.v.c. voltage. Fig. 20 shows a circuit in which diode  $D_1$  is the detector, while diode  $D_2$  is used as a rectifier for the production of the a.v.c. voltage.

When a signal flows through  $L_1$ , a corresponding signal voltage is induced in  $L_2$ , which is tuned by condenser  $C_1$ . This applies an r.f. voltage directly to diode  $D_1$ . The pulses are fed through the r.f. filter  $C_2-R_1-C_3$  so that only the d.c. and a.f. components exist across the load resistance  $R_2$ . The low-frequency component is passed on through  $C_4$  to  $R_3$ , where it is applied to the grid of the triode section of the multi-function tube. The triode then acts as the first audio amplifier.

Condenser  $C_5$  is used to couple some of the r.f. potential into the circuit of diode  $D_2$ . As a result of rectification, a pulsating voltage exists across  $R_4$ , which is filtered of all except its d.c. component by the filter composed of resistor  $R_5$  and condenser  $C_6$ . This d.c. component is then applied to the a.v.c.-controlled tubes.

► In sets in which no additional biases are applied to the controlled stages, the same type of tube can be used with the extra diode connected in parallel with the detector diode, as shown in Fig. 21. The parallel diodes act as one diode with only half the usual plate resistance. Although this increases the loading on the tuned circuit, it provides a greater output.

The a.f. component developed across  $R_2$  in Fig. 21 is passed through  $C_4$  to the grid of the triode section. The d.c. component, fed through  $R_4$ , gives the a.v.c. action. The filter  $R_4-C_5$  removes the a.f. and r.f. components.

## TELEVISION DEMODULATORS

Fig. 22 shows a typical television detector. This must handle a very

wide frequency band, as the video signal includes frequencies from 10 to 4,000,000 cycles. To get the wide frequency range, very low load resistances—usually from 5000 to 25,000 ohms—are used. Two diodes in parallel are used to keep the output voltage as high as possible (these can be the two elements of a dual-diode tube).

The r.f. filter capacity must be kept small to handle the wide frequency

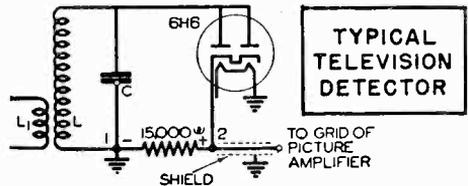


FIG. 22. A typical television demodulator.

range. As a matter of fact, a condenser may be unnecessary, for sufficient filtering capacity may be provided by the shielded lead going to the grid of the first picture amplifier tube. (Notice that there will be a capacity between the lead and its shield, which is effectively across the load resistance.)

► In the circuit shown in Fig. 22, terminal 1 of the load resistance is grounded instead of terminal 2. It is just as common to ground terminal 2 and to take the signal from terminal 1; the particular connection used depends upon the polarity of the signal wanted. (The television signal must be applied to the cathode ray tube with the proper phase, so that increases in brightness will cause brighter spots on the cathode-ray tube screen. As each amplifying stage reverses the phase by  $180^\circ$ , the number of video stages determines which detector connection will give the right polarity.) The subject of television detectors will be covered in another lesson, but you can see now that conventional circuits are used, the major

difference being the use of different part values.

### PUSH-PULL DETECTOR

The circuit in Fig. 23A is a push-pull, or full-wave, detector. When point 2 is positive with respect to point 1, the diode  $D_1$  passes current while  $D_2$  is cut off (because point 3 will then be negative with respect to point 1). On the next half-cycle, the polarities

of points 2 and 3 are interchanged, and the opposite action occurs. Hence, the diodes alternately pass pulses of current through the load resistor  $R_1$ . This is the same action as is obtained in full-wave rectification, which, as you know, doubles the frequency of the applied a.c. voltage. (A full-wave power-pack rectifier produces 120 cycles from a 60-cycle power line frequency.) In other words, if the r.f. carrier has a frequency of 400 kc., full-wave detection produces r.f. pulses with a frequency of 800 kc.

Doubling the frequency of the r.f. pulses across  $R_1$  does not greatly affect the envelope shape (see Fig. 23B). It does, however, permit a decrease in the size of condenser  $C_1$ , the r.f. filter, because at higher frequencies, condenser reactances are lower. Thus, a smaller condenser can do the filtering.

Since a reduced value for  $C_1$  gives greater fidelity, the push-pull detector has been tried in some high-fidelity receivers. It is not used very often today, however, except for an interesting variation of this circuit that is employed as the demodulator in f.m. receivers.

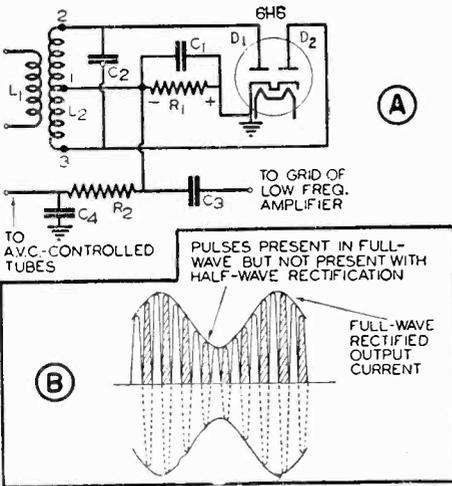


FIG. 23. A full-wave diode demodulator.

## The C-Bias or Plate Detector

In each of the diode circuits we have discussed so far, none of the tubes is powered by a B supply. The only power fed them (other than filament-heating power) is the modulated carrier voltage. Consequently, our circuits have given us no amplification, for, as you know from your earlier studies, a tube amplifies only if the input voltage is used to control the much larger voltages furnished by a B supply.

This lack of amplification is no problem in modern radios, for we can

produce all the gain we need by using amplifier stages before and after the detector. In earlier days, however, when high gain was not so easy to achieve, the diode detector was not in general use; instead, engineers developed circuits that permitted both detection and amplification in the same tube.

### LINEAR C-BIAS DETECTOR

The linear C-bias detector is really a vacuum tube amplifier stage supplied with a C bias almost large

enough to cut off plate current. It is operated with a plate voltage and a load resistance value which will make the dynamic  $E_g-I_p$  curve of the tube practically straight or linear. (The curve is similar to the linear type shown in Fig. 12.) Automatic C bias, supplied from a resistor in the cathode lead, is almost universally used.

**Triode Circuit.** An early linear C-bias detector circuit using a triode tube is shown in Fig. 24. Cathode resistor  $R_c$  provides an automatic C bias that is high enough to reduce plate current almost to zero when there is no r.f. input signal. When a modulated r.f. signal is applied to  $L_1$ , a voltage of similar wave form is induced in coil  $L$ . This voltage receives a resonant step-up from the L-C resonant circuit and is then applied to the grid of the tube.

Since the tube is biased almost to cut-off, plate current flows only when the r.f. input voltage is going through a positive half-cycle. Also, since the dynamic  $E_g-I_p$  characteristic of the tube is approximately linear, these plate current pulses have the same form and envelope as the positive half of the input r.f. voltage. Thus, the output of the triode is like that of the diode we have already studied, except that in this new circuit the pulses *have been amplified by the tube*. In other words, the tube amplifies in a normal manner, while detection occurs because the grid is biased so near cut-off

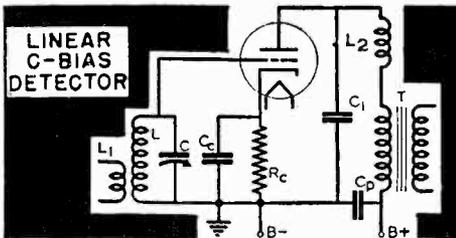


FIG. 24. The linear C-bias detector.

that only half of each signal pulse causes plate current to flow.

The primary of the audio transformer  $T$  is the load, in place of the resistor used in our previous diode detector examples. Condenser  $C_1$  is the r.f. by-pass condenser used for signal separation. In addition, the r.f. choke

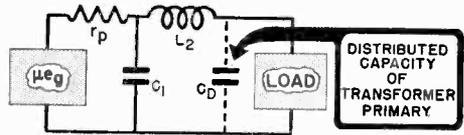


FIG. 25. The equivalent signal-separation circuit of Fig. 24.

$L_2$ , together with the distributed capacity in the transformer primary, forms a low-pass filter that assists in keeping r.f. from the load.

The equivalent r.f. filter circuit is shown in Fig. 25. We have a filter  $r_p-C_1$  and another filter  $L_2-C_D$ . This circuit follows the average of the plate current pulses rather than the peaks, so the load current varies in the manner shown in Fig. 8.

This circuit has the advantage of tube amplification and, as there is no grid current, it does not load the resonant circuit. However, it does not provide a d.c. voltage for automatic volume control. Also, there are limits to the input signal it can handle; if the signal exceeds the bias, so that the grid is driven positive, the plate current pulses will be distorted.

**Pentode Circuit.** Another linear C-bias detector circuit is shown in Fig. 26. This uses a pentode tube, and gives relatively high output voltages even for weak input signals. A similar circuit is found today in inexpensive t.r.f. receivers of the a.c.-d.c. type.

While this circuit looks more complex, its operation is much the same as that of the triode circuit in Fig. 24. The control grid is biased practically

to cut-off by plate current flowing through bias  $R_c$ , which is by-passed by  $C_1$ . R.F. filtering is obtained from  $C_2-L_2-C_3$ , which work with  $r_p$  to form the filter shown in Fig. 25. The a.f. voltage is developed across resistor  $R_L$ , and is passed on to the next stage through  $C_5$ . Condenser  $C_4$  keeps a.f. out of the power supply and also, in combination with resistor  $R_1$ , filters out any ripple from the power supply.

► In each of these C-bias detectors, the modulated signal is rectified because the grid of the tube is biased practically to cut-off—that is, to the point where the  $E_g-I_p$  characteristic makes a sharp bend. For this reason, a C-bias detector is often called a “plate-bend detector” or simply a “plate detector.”

### SQUARE-LAW DETECTION

All the detector circuits we have discussed so far have two common characteristics. In each, rectification of the modulated r.f. voltage has been

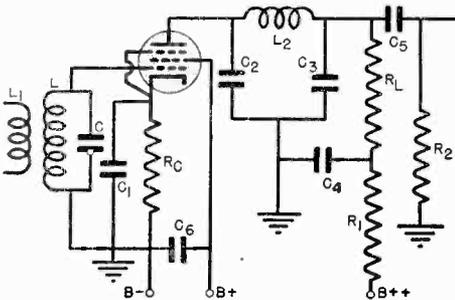


FIG. 26. A pentode C-bias demodulator.

complete (all negative half-cycles have been eliminated) and approximately linear (the envelope of the rectified pulses has been approximately the same shape as the envelope of the modulated carrier).

It is possible to have a detector in which rectification is incomplete and non-linear. Suppose, for example, we

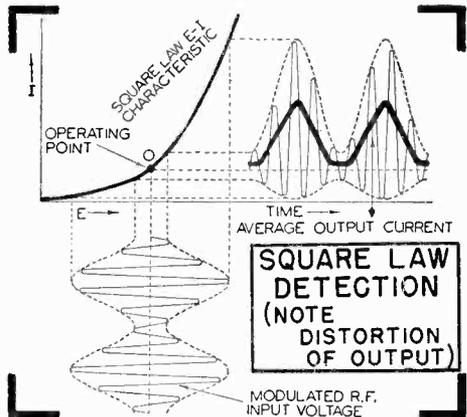


FIG. 27. This drawing shows how the square-law detector produces a distorted output signal.

have a triode which has a curved  $E_g-I_p$  characteristic like that shown in Fig. 27. Such a tube is said to have a “square law” characteristic, for its plate current is proportional to the *square* of the signal voltage applied to the grid. (That is, if the input signal is doubled, the output increases four times; if it is tripled, the output will be increased nine times, etc.)

Suppose, further, that we place this triode in a circuit like that in Fig. 24, and adjust the bias so that the tube will operate at point O in Fig. 27—the point where the  $E_g-I_p$  characteristic curves the most. If we now apply a modulated carrier to the input terminals of the stage, we will get the output shown in Fig. 27.

Here, each r.f. cycle has undergone incomplete rectification; each positive swing of the r.f. voltage has been increased, and each negative swing has been decreased, by the curved characteristic. Because of the incomplete rectification, the average value (a.f.) of the rectified current (shown by the heavy line in Fig. 27) does not have exactly the same wave form as the envelope of the positive current peaks.

This means that we will get a somewhat distorted reproduction of the

original a.f. voltage when we send the tube output through the signal-separating circuits of the tube load. (The amount of distortion produced depends on the percentage of modulation of the carrier; it is inappreciable for modulations less than 20%, but it goes up to 25% for modulations of 100%.) However, such a detector has the advantage of being very sensitive, for its output current (and therefore its output voltage) is proportional to the *square* of the carrier voltage. Thus, even very weak signals can be made to give an appreciable output.

► The detector we just described is known as a C-bias square-law detector. The tube used does not have to be a triode; many pentodes also have such a characteristic, and can be used as square-law detectors. In fact, the rectifying device does not even have to be a tube: a galena crystal has a square-law characteristic when used as a rectifier, and, as we shall see later in this lesson, such crystals were widely used as square-law detectors in the early days of radio.

Because they produce distortion, and because modern amplifier circuits make sensitive detectors unnecessary, square-law detectors have practically disappeared as demodulators. How-

ever, they are modified into superheterodyne first detectors, and they still are used in measuring instruments and in radio applications where sensitivity rather than freedom from distortion is desired.

**Combination Linear-Square Law Detectors.** Many tubes, if their operating voltages are properly adjusted, can be made to have an  $E_g-I_p$  characteristic which is square law in shape near the bottom and nearly linear farther up. If we use such a tube in a detector circuit, and adjust the grid bias so that we operate near the point of greatest curvature of the characteristic, we can get a combination of linear and square-law detection. Strong signals will swing over the linear part of the characteristic, and will give practically distortionless detection. Weak signals, however, will swing over only the curved part of the characteristic, giving us square-law detection. We then get sensitive detector action, which is desirable for weak signals—but we must, of course, pay the price of greater distortion for it.

These combination detectors were once very popular, but, like square-law detectors, they have almost entirely disappeared from home radios as demodulators.

# Heterodyne or Beat Detectors

The superheterodyne circuit requires two detectors; one is used as a demodulator to obtain the intelligence signal from its carrier (this is the second detector), and the other is used to produce the i.f. (this is called the first detector). This first or heterodyne detector is not used to demodulate the incoming signal; instead, it mixes the incoming and local oscillator signals to produce a beat frequency that is still modulated by the intelligence signal.

When two signals of different frequencies are mixed together and then fed into a detector, the resulting rectified current pulses will include the following components: the original frequencies; the sum of these frequencies; their difference; and various harmonics of all these. In the superheterodyne, we are interested in the difference frequency, which is called the intermediate frequency (abbreviated "i.f."). (This difference fre-

quency is also known as a *beat frequency* or a *heterodyne frequency*, and the method by which it is produced is called *heterodyne detection*.)

## HOW A HETERODYNE DETECTOR WORKS

The action of a heterodyne detector can be best understood by a study of the wave form curves in Fig. 28. When two r.f. signals that differ in frequency are mixed together, such as the signals in Figs. 28A and 28B, the result is the complex wave shown in Fig. 28C. (Here, the envelope of the wave represents the difference or beat frequency.) Notice that peak *a* is rounded, while the trough at *b* is sharp. Clearly, the envelope is not a sine wave.

If a voltage having the wave form shown in Fig. 28C is sent through a linear detector, the resulting demodu-

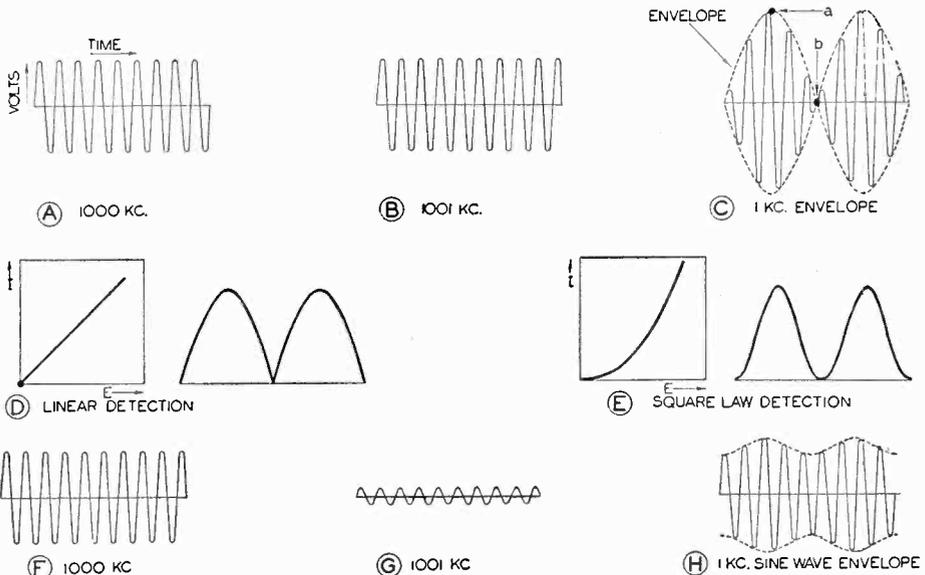


FIG. 28. A comparison between the linear and the square-law heterodyne detectors.

lated current shown in Fig. 28D is not sine wave in form. On the other hand, when the wave of Fig. 28C is sent through a square-law detector, the sharp troughs of the demodulated signals are distorted by the detector characteristic, giving the sinusoidal output shown in Fig. 28E. (Here we have a circuit introducing a distortion that compensates for another distortion!) Thus, when the amplitudes of the two signals being mixed are equal, the square-law detector gives a less distorted output than a linear detector.

► However, usually the signals mixed together are not equal in amplitude. Since the amount of heterodyne output depends on the amplitude of the two r.f. signals, the local oscillator output is generally made very strong so that the heterodyne output will be

strong even if the incoming signal is weak. Usually the output from the local oscillator is made about 10 times the strength of the strongest expected incoming signal.

When the two r.f. signals differ this much in amplitude, the envelope of the mixed signal is practically a pure sine wave. For example, if the local oscillator signal is represented by Fig. 28F and the incoming signal by Fig. 28G, we will obtain the complex wave shown at Fig. 28H when they are mixed. Either a linear or a square-law detector will give us a sine-wave output signal from this mixed signal, because it has a sine-wave envelope; therefore, under these conditions, we can use either kind of detector. You will study typical mixer-first detector stages in detail when you study the superheterodyne receiver.

## Early Detector Types

We will now take a brief look at some of the early detector circuits. You'll find these no longer used much, but you will meet them occasionally.

### CRYSTAL DETECTORS

Before the days of vacuum tubes, practically every radio receiver contained a crystal detector. Even today you can purchase "crystal sets" in many radio stores, for they give acceptable reception of local broadcasts with ample volume for headphone use.

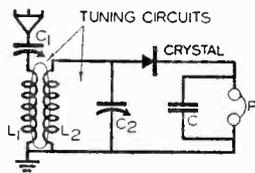


FIG. 29. A typical crystal detector.

Fig. 29 shows a crystal detector circuit that has been widely used. It is tuned by resonant circuits  $C_1-L_1$  and  $C_2-L_2$ , which are inductively coupled. R.F. filtering is provided by condenser  $C$  and the resistance of the crystal, which form an R-C filter. The crystal is a one-way device; it passes current much better in one direction than in the other. Hence, it provides approximate half-wave rectification of the applied r.f. signal. The audio component of the rectified pulses operates the headphones. Thus, rectification takes place at the crystal, and signal separation occurs at the load (the headphones). There is no amplification in a crystal detector.

The crystals are generally galena, iron pyrites, silicon, carborundum, or combinations of these minerals. A

fine phosphor-bronze wire, often called a "cat's whisker" because it was about as thin and springy as the whisker of a cat, was used originally to make contact with a sensitive spot on the crystal. This was a very delicate arrangement, for the slightest jar necessitated a resetting of the "cat's-whisker." More recently, fixed crystal detectors have been made by mounting a piece of carborundum permanently against a soft metal disc; these do not require adjustment, but are considerably less sensitive (give poorer rectification).

► In certain foreign countries where

was one of the first detector circuits that used a vacuum tube, and was for many years one of the most popular. Two forms of this circuit are shown in Fig. 30; both work in essentially the same manner.

We can consider the cathode and grid of the tube to be a diode in which the grid acts as an anode. Looking at the circuit from this viewpoint, you see at once that the cathode-grid circuit of the tube is really a simple diode detector.  $R_g$  and  $C_g$  form the signal separating load circuit, while the resonant circuit  $L_T-C_T$  is the means by which the modulated r.f. carrier

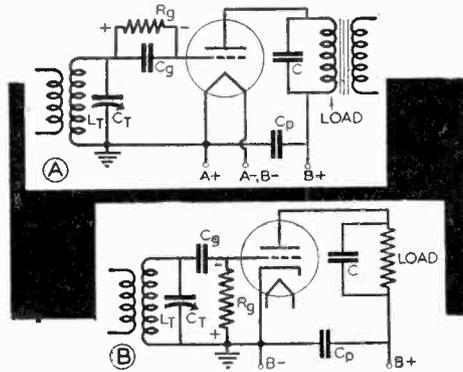


FIG. 30. Two forms of the grid-leak grid-condenser detector.

radio receivers are taxed according to the number of tubes they contain, crystals and special copper-oxide rectifiers (having elements about the size of a pinhead) are still used as detectors.

A recent development is the use of tiny fixed crystal units as detectors at micro-wave frequencies, where the effects of the transit time of electron flow through a tube make tube detectors unsatisfactory.

### GRID LEAK-CONDENSER DETECTORS

The grid leak-condenser detector

voltage is introduced into the detector circuit. When a signal is fed in, it undergoes detection in this "diode" circuit, and an a.f. voltage similar in form to the envelope of the carrier is developed across load resistor  $R_g$ . (The resistor  $R_g$  is called a "grid leak," while  $C_g$  is known as the "grid condenser;" from these come the name for this detector type.)

The a.f. voltage across  $R_g$  is applied between the cathode and grid of the tube, and thus produces a.f. variations in the tube plate current. The r.f. input signal is also applied between the grid and cathode, so the tube plate

current contains r.f. variations too. To keep the r.f. voltages out of the low-frequency amplifier,  $C$  is used in the plate circuit to form an R-C filter with the tube  $r_p$ . The a.f. component is amplified by the tube and appears across the plate load, ready to be fed to the a.f. amplifier.

Thus, a triode grid-leak detector acts like a diode detector to which one stage of audio amplification is added. Rectification and signal separation occur in the grid circuit; the a.f. produced is then amplified by the tube. Proper selection of the resistor  $R_g$  and condenser  $C_g$  will let the detector work as a square-law device for

tion envelope of the carrier.

One form of regenerative grid leak-condenser detector circuit is shown in Fig. 31. The grid circuit of the tube is exactly the same as that of the grid leak-condenser detector, but the plate circuit contains a coil  $L_2$  through which the r.f. plate current flows. This induces the desired amount of r.f. voltage in the grid tank circuit, through mutual inductance  $M$ . Variable resistor  $R$ , connected across the feedback coil, controls the r.f. current flow through the coil and so determines the amount of feedback. (Some receiver builders controlled the amount of feedback by varying the mutual in-

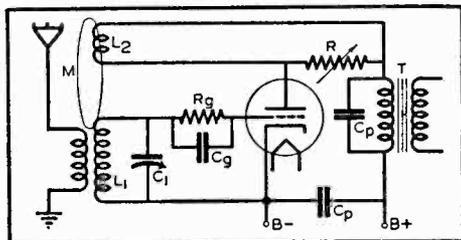


FIG. 31. A regenerative detector.

weak signals or as a linear detector for strong signals.

## REGENERATIVE DETECTORS

Even more gain than that found in grid leak-condenser detectors was sought in the early days of radio. Inventors found a way of using the undesired r.f. voltage in the plate circuit—they put this component to work in a regenerative detector, in which the r.f. current in the plate circuit of the detector was fed back into the grid circuit to reinforce the incoming modulated r.f. carrier voltage. Naturally a means of controlling the amount of feedback was necessary, for excessive feedback resulted in oscillation and complete destruction of the modula-

tion envelope of the carrier.)

The energy fed back from the plate circuit, in effect, reduces the losses in the grid circuit, so a far greater resonance step-up is obtained.

A regenerative detector has some good features, but also has several disadvantages. It is quite selective for the carrier frequency being tuned in, but attenuates side-band frequencies and produces severe frequency distortion. (For code reception and reception of ordinary speech, this frequency distortion is not at all objectionable; hence, regenerative detectors may still be found in certain communications receivers.) Furthermore, a regenerative circuit goes into oscilla-

tion quite easily, especially when condenser  $C_1$  is adjusted from a low to a high frequency. If no r.f. stage is used ahead of the detector, the resulting oscillations are radiated by the receiving antenna just as if the receiver were a small transmitter, and interference is created in neighboring receivers. In addition, an annoying squeal is usually heard from the receiver itself, for the frequency of oscillation generally differs only slightly from the frequency of the desired incoming signal, and the resulting beat frequency is an audio frequency.

**Super-Regenerative Detectors.** A regenerative detector is "supersensitive" (has its greatest sensitivity) when just on the verge of going into oscillation. However, when it is in this condition, small increases in carrier input will set the circuit into oscillation; once this occurs, decreases in carrier intensity will not stop the oscillation.

This difficulty was eliminated with the super-regenerative detector, a circuit based upon the principle that it takes a certain amount of time for a detector to go into oscillation. If the feedback can be reduced quickly enough during this interval of time, oscillation can be prevented. Feedback can be reduced temporarily either by increasing the negative C bias in the grid circuit or by decreasing the plate voltage. Either can be accomplished by introducing into the regenerative detector circuit a *varying voltage*, which may be sinusoidal in wave form, and which is known as the *quenching signal*. The quenching frequency must be somewhat higher

than the intelligence signal frequencies; otherwise, it would create interference. For code and speech reception, a quenching frequency of between 10 and 20 kc. is quite satisfactory.

A super-regenerative detector circuit, in which the tube acts both as

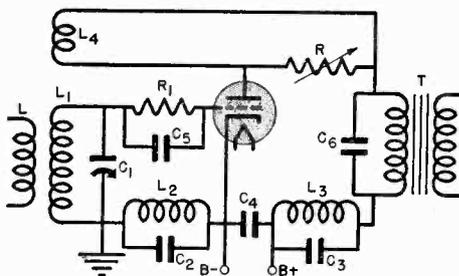


FIG. 32. One of the super-regenerative circuits.

a regenerative detector and as an oscillator for producing the quenching frequency, is shown in Fig. 32. Resonant circuits  $C_2-L_2$  in the grid circuit and  $C_3-L_3$  in the plate circuit act in conjunction with the tube as an oscillator, with feedback provided either by the grid-to-plate capacity of the tube or by mutual induction between coils  $L_2$  and  $L_3$ . The circuit can be tuned to maximum sensitivity without any danger that a carrier increase will send it into audible oscillation, for the inaudible quenching signal always reduces the plate voltage before the tube has time to break into oscillation at the incoming signal frequency.

Of course, a separate oscillator may be used to produce the quenching frequency; this is often done in order to secure independent control of the quenching signal.

# Demodulators for F.M. Receivers

So far, the detectors described have been intended to handle amplitude-modulated (a.m.) signals. To complete our study of detectors, we must also consider the fundamentals of detectors intended for frequency-modulated (f.m.) signals. Of course, we shall give more information on these later, when we discuss the complete f.m. receiver. For now, we shall present basic circuits so you can compare the detectors used in the two modulating systems. You may be surprised to see how little difference there is between *basic* circuits—a change in a part value here, or a different operating voltage there, may be all that is needed to make an a.m. stage into an f.m. stage. Again, you will see the value of understanding clearly how parts and circuits work.

► What is the difference between a.m. and f.m.? Essentially, amplitude modulation is a system in which the *amplitude* of an r.f. voltage is varied by an intelligence signal: the *amplitude* variation being proportional to the *amplitude* of the intelligence signal, and the *rate* of variation being proportional to the *frequency* of the intelligence signal.

In frequency modulation, the *amplitude* of the r.f. signal remains *constant*—the *frequency* is varied by the intelligence signal. Here, the *frequency* variation is proportional to the *amplitude* of the intelligence signal (louder sounds produce a wider frequency change), while the *rate* of frequency variation is proportional to the frequency of the intelligence signal.

Thus, in a.m., the *amplitude* changes of the intelligence signal produce *amplitude* changes in the carrier;

while in f.m. the *amplitude* changes of the intelligence signal produce *frequency* changes in the carrier.

► A tube is an amplitude-following device—the larger the input signal amplitude, the greater the resulting current. Hence, a.m. detectors produce an intelligence signal output that varies directly with the amplitude changes of the incoming r.f. signal.

On the other hand, the f.m. wave has a constant amplitude; there is no way in which a *tube* can produce an intelligence signal with a varying amplitude from such an r.f. signal. (The tube responds equally well to all r.f. signals over a wide range, as its output is relatively independent of r.f. frequency.) Therefore, it is necessary to add another step to the process of f.m. demodulation. *WE MUST FIRST CONVERT THE F.M. SIGNAL TO ONE WITH A VARYING AMPLITUDE BEFORE WE CAN DEMODULATE THE SIGNAL.* This is the fundamental difference between a.m. and f.m. demodulators. Let's see how the conversion is made.

## A BASIC F.M. DEMODULATOR

To convert an f.m. wave into one having a.m. characteristics, we must have a circuit or device that can *discriminate* between frequencies. That is, the circuit must give different output voltages for different frequencies. (As discrimination is a necessary step in f.m. detection, the f.m. detector stage is commonly called a *discriminator stage*.)

Frequency discrimination is obtained by the use of L-C resonant circuits. Let's look at several systems of discrimination, as differences in the methods of discrimination are the

major points of difference between f.m. demodulators.

► Let's suppose we have an L-C circuit with a resonance curve like that shown in Fig. 33. Notice that the L-C values of this circuit cause best response to the resonant frequency, which is 100 kc. It responds less well to frequencies of 90 kc. and 110 kc., and rather poorly to frequencies of 80 kc. and 120 kc.

► In an a.m. detector, we tune the

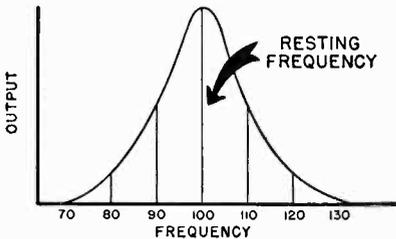


FIG. 33. A response curve for a resonant circuit.

L-C circuit to the frequency of the carrier in which we are interested; we then get maximum response from this frequency while nearby carriers are rejected. We depend on the modulation to give the varying amplitude signal needed for the proper operation of the detector.

The "carrier" in an f.m. system is called the "resting frequency." This is the frequency transmitted when there is no modulation. When modulation occurs, the frequency of the transmitted wave varies above and below the resting frequency. Now, if we feed an f.m. signal into this same tuned circuit, and the circuit is tuned exactly to the resting frequency, then we will not get the desired results. As the modulation swings the frequency up and down, we will get a variation in amplitude, but it will be always in the same direction—downward.

For example, if the resting frequency is 100 kc., and it is applied to an L-C circuit tuned to this frequency,

it will get the maximum resonance step-up. When the signal frequency swings up to 110 kc., we will get less output (see Fig. 33). On the other hand, when it swings down to 90 kc. we will also get less output. Therefore, we are not reconstructing the desired a.c. cycle; we want the output to go down when the frequency changes in one direction, and to go up when it swings in the other direction.

► However, we can use this same circuit if we tune it to resonance at a frequency higher than the resting frequency, as shown in Fig. 34. Under these conditions, the output for the 100-kc. resting frequency is not as great as the output for the 110-kc. signal. On the other hand, the output for 100 kc. is greater than the output for 90 kc. Therefore, variations in frequency above and below the resting frequency will give us variations in amplitude.

For example, when the incoming frequency increases toward 110 kc.,

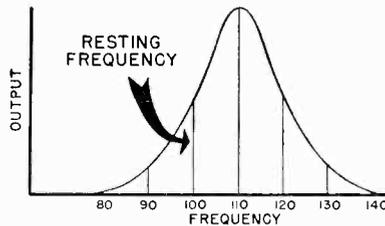


FIG. 34. The response curve of Fig. 33, with the circuit tuned to another frequency.

we will get a greater output from the resonant circuit and hence a greater voltage from the detector. On the next half-cycle, when the incoming signal swings downward in frequency towards 90 kc., we will have a smaller input to the detector, and thus obtain a reduced output. Thus, as the frequency changes above and below the resting frequency, we obtain a *varying amplitude* signal. Also, the greater

the frequency change from the resting value, the greater the amplitude variation and, hence, the greater the output. In other words, an ordinary resonant circuit that is *not* tuned to the resting frequency will convert an f.m. wave into an a.m. wave, which we can then demodulate in the usual manner.

► Actually, a simple f.m. demodulator could be made from *any* diode circuit just by changing the resonant point of the tuned circuit at its input. However, this is not the system used

the i.f. resting signal, but it is designed with a low Q so that it is sufficiently broad to respond well to frequencies on both sides of this value. This circuit induces voltages in both  $L_2$  and  $L_3$ ; these voltages cause electrons to flow through diodes  $VT_1$  and  $VT_2$ .

Since electrons move only from the cathode to the plate, electrons flowing through  $VT_1$  come “upward” through resistor  $R_1$ , producing a voltage drop across this resistor with the polarity shown. Similarly, electrons flowing through  $VT_2$  move “downward” through resistor  $R_2$ . The voltage drop across this resistor is, therefore, of opposite polarity to that across resistor  $R_1$ .

The resonant circuit  $M$  ( $L_2-C_2$ ) is tuned to a frequency somewhat lower than that of  $L_1-C_1$ , while  $N$  ( $L_3-C_3$ ) is tuned to a frequency somewhat higher. If the frequency of 4 mc. in Fig. 35B represents the  $L_1-C_1$  resonant frequency (the resting frequency), then 3.9 mc. may represent the resonant frequency of  $M$  and 4.1 mc. that of  $N$ . Hence, neither  $M$  nor  $N$  is tuned to favor the center frequency, although neither *excludes* signals of this frequency altogether.

The circuits are adjusted so that, at the resting frequency, the voltages applied to  $VT_1$  and  $VT_2$  are equal, and equal currents flow through  $R_1$  and  $R_2$ . As these are equal resistances, their voltage drops are equal under these conditions. Further, since their voltage drops are opposite in polarity (see Fig. 35A), there is no net voltage between terminals  $X$  and  $Y$  when only the resting frequency is fed into the stage.

► Now let us suppose the signal frequency is modulated so that it swings from 4 to 3.9, back through 4 to 4.1, then back toward 3.9, etc.

As the frequency approaches 3.9 (the resonant frequency of  $M$  — see

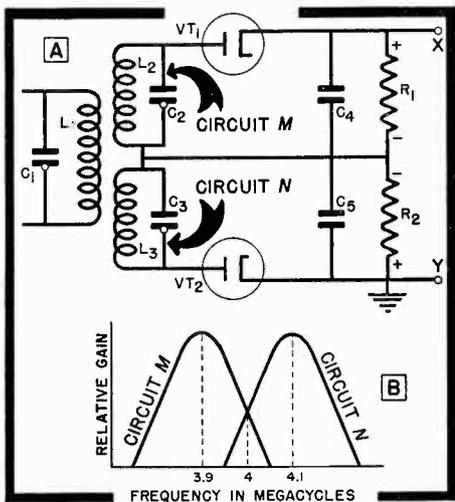


FIG. 35. An f.m. discriminator that used two resonant circuits.

in commercial receivers. There are alignment problems, and it is difficult to design a resonant circuit with a response curve that is straight enough to give freedom from distortion. Let's go on and look at commercial systems.

### DUAL L-C SYSTEM

Fig. 35A shows one commercial adaptation of the principles we have just described. The circuit works as follows:

Resonant circuit  $L_1-C_1$  is tuned to

Fig. 35), more voltage will be developed by  $M$ . This causes larger voltages to be applied to  $VT_1$ ; larger currents result and produce larger voltages across  $R_1$ . At the same time, the signal frequency is moving away from the resonant frequency of  $N$ , so less voltage is being applied to  $VT_2$ . The resulting smaller current produces less voltage across  $R_2$ . Hence, for this swing, the voltage across  $R_1$  increases while that across  $R_2$  decreases. The two voltages no longer cancel one another, so a net voltage (equal to the difference between them) appears between terminals  $X$  and  $Y$ .

For example, suppose the  $R_1$  and  $R_2$  voltages are initially 50 volts each,

**Summary.** A signal, shifting in frequency, is introduced into this stage. The frequency variations produce a varying voltage across terminals  $X$  and  $Y$ . Small frequency changes produce small voltages, as frequencies near 4 mc. do not "climb" as high on the  $M$  and  $N$  resonance curves. Larger changes, out to the limits set by the resonance points (3.9 and 4.1 in Fig. 35B) produce larger voltages. Thus, we have a circuit that produces an amplitude variation from the varying frequency signal.

This same circuit also gives us demodulation in the same manner as the diode detector we have discussed for amplitude-modulated signals. The

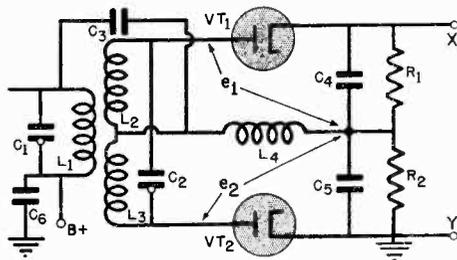


FIG. 36. This phase-operated f.m. discriminator is the type most commonly used.

and cancel exactly. On the swing just described, the  $R_1$  voltage may go up to 90 volts and the  $R_2$  voltage may go down to 10 volts. The difference is 90 minus 10 or 80 volts, which will appear between  $X$  and  $Y$ . (Terminal  $X$  will be positive with respect to  $Y$ , as the polarity will be that of the voltage across resistor  $R_1$ .)

When the swing reverses toward 4.1 mc., exactly the reverse action occurs. Now,  $N$  develops the larger voltage,  $VT_2$  passes the larger current, and the drop across  $R_2$  is larger than that across  $R_1$ . This reverses the polarity of the difference voltage existing across  $X$  and  $Y$ , since the polarity now is that of the voltage across  $R_2$ .

diodes  $VT_1$  and  $VT_2$  rectify the r.f. signal, while condensers  $C_4$  and  $C_5$  act as r.f. filters. Thus, the original intelligence signal appears across terminals  $X$  and  $Y$ .

### A PHASE-OPERATED DISCRIMINATOR

Fig. 36 shows the most widely used f.m. demodulator. The circuit is very similar to that of a push-pull a.m. detector described earlier in this lesson, except for the arrangement made to obtain frequency discrimination.

Complete details on the operation of this circuit will be given in a later lesson. Briefly, here's how it works:

$L_1$  is the primary, and  $L_2$  and  $L_3$  are

the two halves of the secondary, of a center-tapped i.f. transformer. ( $L_2$  and  $L_3$  are actually parts of the same secondary coil; we've given them separate names just for convenience in reference.) The circuit  $L_1-C_1$  is tuned to resonance at the resting frequency, as is also the circuit  $L_2-L_3-C_2$ . A choke coil  $L_4$  is connected to the center point of the transformer secondary, and also (through condenser  $C_3$ ) to the top of the primary  $L_1$ . This coil  $L_4$  is in the common plate return of both diodes.

If you trace the circuit from the top of  $L_1$  through  $C_3$ ,  $L_4$ , and  $C_5$  to ground, and from ground through  $C_6$  to the bottom end of  $L_1$ , you will see that  $L_4$  is effectively in parallel with the primary  $L_1$ . (We can ignore the condensers, because their reactances are negligible at f.m. frequencies.) Therefore, the primary voltage across  $L_1$  is also always across  $L_4$ , and we can call the voltage across  $L_4$  the "primary voltage" for convenience.

At any time, the voltage applied to the plate of diode  $VT_1$  is the sum of the voltage across  $L_2$  and the primary voltage across  $L_4$ . Similarly, the voltage applied to the plate of  $VT_2$  is the sum of the voltage across  $L_3$  and the primary voltage across  $L_4$ . These must be *vector* sums, since the voltages across these coils are not in phase.

This circuit is so arranged that when the resting frequency is fed in (that is, the frequency to which the tuned circuits are resonant) the voltage across  $L_2$  is  $90^\circ$  out of phase with the primary voltage. The voltage across  $L_3$  is then also  $90^\circ$  out of phase with the primary voltage, but in the opposite sense. Say we assume that the  $L_2$  voltage *leads* the primary voltage by  $90^\circ$ ; the  $L_3$  voltage will then *lag* the primary voltage by  $90^\circ$ . The *vector sum* of the  $L_2$  voltage and the

primary voltage will then be of the same magnitude as the vector sum of the  $L_3$  voltage and the primary voltage. Therefore, when the resting frequency is applied to this circuit, the total voltage applied to the plate of each diode is the same. That is, the voltage  $e_1$  for  $VT_1$  will equal voltage  $e_2$  for  $VT_2$ . Equal voltages cause the diodes to pass equal currents through  $R_1$  and  $R_2$ . Since the resistances are equal, their voltage drops are equal and opposite, so no net voltage appears between terminals  $X$  and  $Y$ .

► However, the two plate voltages do not remain the same when a modulated signal is fed in. As you know, a resonant circuit changes its characteristics when a signal other than the resonant frequency signal is applied to it. If an above-resonance frequency is applied, the circuit acts like a *coil*; if a below-resonance frequency is applied, the circuit acts like a *condenser*. In other words, *the phase of the circuit voltage changes when an off-resonance frequency is applied.*

When a modulated signal is applied to our discriminator circuit, phase changes occur both in the primary voltage and in the voltages across  $L_2$  and  $L_3$ . The amount of phase change will depend on how far from the resting frequency (how far off-resonance) the modulated signal swings. We might have, say, a total shift of  $45^\circ$ . The  $L_2$  voltage might then lead the primary voltage by  $135^\circ$  ( $90^\circ$  plus  $45^\circ$ ); the  $L_3$  voltage would then lag the primary voltage by only  $45^\circ$  ( $90^\circ$  minus  $45^\circ$ ).

Obviously, under these conditions the vector sum of the  $L_2$  voltage and the primary voltage is not equal to the vector sum of the  $L_3$  voltage and the primary voltage. Thus, the phase shift causes unequal plate voltages to be applied to the diodes. Therefore, the

$R_1$  and  $R_2$  currents are unequal and a net voltage appears between terminals  $X$  and  $Y$ . This voltage becomes larger the farther the f.m. signal swings from the resting frequency, and reverses in polarity as the signal swings above and below the resting frequency. It is, therefore, the amplitude-modulated signal we want. Detection of this a.m. signal occurs as it does in the circuit in Fig. 35; the diodes rectify the r.f., and condensers  $C_4$  and  $C_5$  act as r.f. filters, so the intelligence signal appears across terminals  $X$  and  $Y$ .

► Don't worry if all the details of these f.m. circuits aren't perfectly clear to you now—we will discuss f.m.

receivers thoroughly in later lessons. Even now, though, you can see that f.m. sets are not tremendously different from a.m. receivers. In fact, except for the addition of the frequency discriminator and the use of a circuit called a *limiter*, the f.m. receiver is designed very similarly to a wide-band high-fidelity a.m. receiver. And, as we have just shown, the f.m. signal must be converted into an a.m. signal at the detector before we can obtain it. The f.m. system of modulation has certain advantages; notably, high fidelity capabilities and, when a limiter stage is used, a remarkable freedom from noise.

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## Reviewing Methods of Providing C Bias

Now that we have studied nearly all the basic vacuum tube circuits, let us review the different ways of providing the negative operating C bias required in a.f. and r.f. amplifiers, detectors, oscillators—in fact, in any vacuum tube circuit. You are already familiar with most of these methods, but we will review them briefly here to give you a complete picture.

You will recall that a grid bias is a voltage applied to the grid to fix the operating point of the tube at the proper point on the characteristic curve. It is possible to get an entirely different operating action from a radio stage just by varying the bias voltage. For example, you know that a class A amplifier—the kind most commonly used in receiver circuits—has a bias voltage that makes it operate on the straight portion of its characteristic. The bias keeps the grid negative at all times (providing the signal level is not excessive) so it will not draw grid

current.

If we increase the bias enough so that the tube plate current is practically zero (in other words, if we bias the tube almost to cut-off), we have class B operation. We use class B stages in push-pull audio amplifiers, in transmitters, and as detectors.

Increasing the bias still further (well beyond cut-off) will give us class C operation. A class C amplifier is a highly efficient circuit used only for r.f. power amplification and for oscillatory circuits.

► Obviously, the bias voltage is important to the operation of the tube stage. We can secure bias in various ways: we can use a battery, or arrange a self-bias circuit in which plate, filament, or grid current provides the bias voltage;\* or we can use fixed

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\*A.V.C., an automatic C bias dependent on signal level, is not included in this review, but will be described in a later lesson.

bias circuits in which the bias voltage is obtained from the power supply. Any of these methods can be used with any tube which has a grid.

### BATTERY BIAS

Fig. 37 shows several methods of supplying a bias voltage from a battery. Exactly the same method is shown in *A* and *B*, except that *A* shows a tube with a filament type cathode, while *B* shows a tube with a separately heated cathode.

As you see, the *C* battery is connected in the grid return circuit.

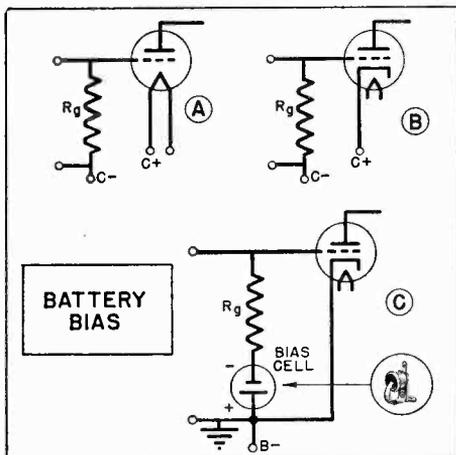


FIG. 37. Several methods of using batteries to furnish bias.

Hence, its voltage appears between the grid and cathode, and thus sets the operating characteristics of the tube.

► Since the grid is made negative by the bias voltage, no grid current flows. Therefore, the bias voltage causes no current flow through  $R_g$ , so this resistor does not enter into the biasing arrangement in these or any succeeding diagrams. The sole purpose of  $R_g$  is to provide a load across which signal voltage can be developed. (When signals are applied, there will

be a signal current through  $R_g$  that will cause a signal voltage drop across this resistor. This signal voltage will alternately add to and subtract from the bias voltage, swinging the grid voltage above and below the operating point.)

► In Fig. 37C, a bias cell is shown in use. A bias cell is actually a very small chemical-cell battery. Two types of bias cells are available; one type is rated at 1 volt while the other gives  $1\frac{1}{4}$  volts. Any reasonable bias voltage can be obtained by placing several in series. However, it is rare to find more than two used in a radio receiver.

The advantages of the bias cell are that it is small in size and that it can be wired into the circuit as a semi-permanent part of the circuit, since it does not require frequent replacement. Of course, the cell must be used properly to have reasonable life. Never try to measure the voltage of a bias cell with a voltmeter, because the current drawn by the meter may damage the cell. If you suspect that a bias cell is defective, remove it and substitute another, or a battery, in its place. You can determine whether the original is defective by observing the operation of the stage when the substitute is being used.

### SELF-BIAS METHODS

There are a number of biasing methods that depend upon plate, filament, or grid current flow. Let's take each up in turn.

**Plate Current Methods.** Fig. 38 shows several methods whereby the average d.c. plate current flow through a resistor produces a voltage that can be used for bias purposes.

In Fig. 38A, the plate current flows through resistor  $R_c$ . (Electrons flow from the cathode to the plate, through

$R_L$ , through the B supply, and through  $R_c$  back to the cathode.) As a result, a voltage drop that has the polarity shown in this figure exists across  $R_c$ . Since resistor  $R_c$  is between the grid and cathode, the voltage across it is also between the grid and cathode. This voltage makes the grid negative.

We can adjust this bias to the required amount by choosing the proper value of resistance for the plate current of the tube.

The condenser  $C_c$  by-passes the bias resistor. This prevents any of the a.c. signal voltage from being lost in  $R_c$ , and also prevents the plate circuit a.c. voltage from causing degeneration in the grid circuit.

which the plate current flow passes to the filament. In this case, remember that the current is doubled, since it comes from two tubes. As a result, the bias resistor should be half the size that is necessary for a single tube to produce the same bias voltage.

**Filament Current Methods.** Fig. 39 shows how the filament current can be used with battery type tubes to supply a bias. (We cannot use the filament current of a.c.-operated tubes as a bias source, because the filament current variations would introduce a 60-cycle variation in the plate current, producing hum. We cannot permit variations in the bias, so we can get bias only from a d.c. source.)

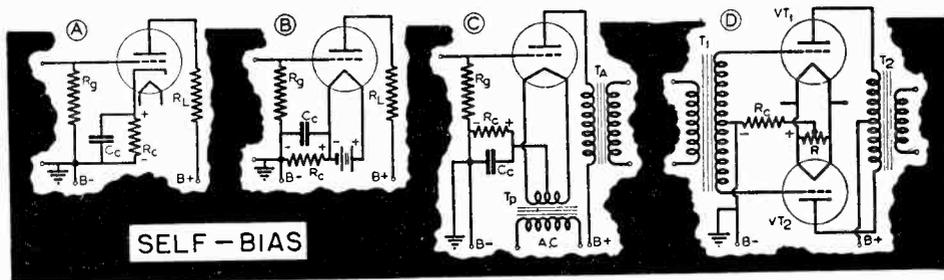


FIG. 38. Typical self-bias circuits.

► Fig. 38B shows the same kind of circuit for a tube using the filament as the cathode. It operates precisely the same way.

► Fig. 38C shows a similar arrangement for a tube with an a.c.-operated filament type cathode. In this case, there is a center tap on the filament winding. The plate electrons flow from the cathode to the plate, through the primary of transformer  $T_A$ , through the B supply, and through  $R_c$  to the center tap on the filament winding, then back to the filament.

► 38D shows the same arrangement for a push-pull stage. Here the plate currents of both tubes flow through  $R_c$  to the center-tapped resistor  $R$ , from

In Fig. 39A, electrons circulate from the battery through resistor  $R_F$ , through the filament, and back to the battery.

► The amount of bias depends on the reference point used. There are two systems of reference points used with these battery type tubes. As you know, there is a voltage drop along the filament wire, caused by its resistance, so that different points on the filament are at differing potentials with respect to the grid. Thus, the grid is far more negative with respect to one end of the filament than it is with respect to the other.

Formerly, the middle of the filament was taken as the bias reference

point. Following this system, the bias voltage in Fig. 39A will be half the filament voltage drop, plus the drop across the resistor  $R_F$ . This adds up to a bias of 3.5 volts. However, modern practice has been to assume that the negative side of the filament is the reference point, in which case the 1-volt drop across resistor  $R_F$  is the bias voltage.

► In Fig. 39B, we have a filament string so arranged that the drops

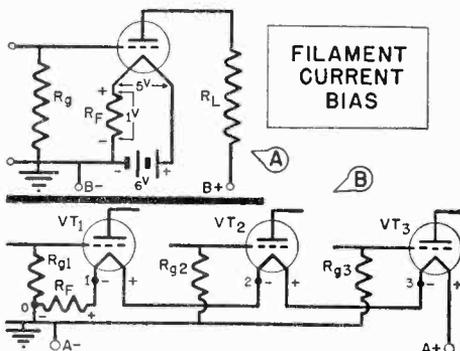


FIG. 39. How the filament current can be made to supply bias.

across the tube filaments become bias voltages. Let's see how this circuit works. We will consider the negative filament terminals to be the bias reference points.

The filament electrons move from A-, through the resistor  $R_F$ , through the filament of tube  $VT_1$  then through the filaments of  $VT_2$  and  $VT_3$ , to the A+ terminal. As far as tube  $VT_1$  is concerned, the only bias voltage between its negative terminal (terminal 1) and its grid is the drop across the resistor,  $R_F$ . This, then, is the bias for  $VT_1$ .

Now let's look at  $VT_2$ . The voltage between its grid circuit and point 2 on the filament string includes both the drop across  $R_F$  and that across the filament of tube  $VT_1$ . Therefore, this tube gets a higher bias, because its grid resistor  $R_{g2}$  returns to the point

0 on the filament string.

Similarly, tube  $VT_3$  has a bias equal to the drop across  $R_F$  plus the drops across the filament of  $VT_1$  and the filament of  $VT_2$ . In other words, the bias for this tube is equal to the sum of the drops between the return point of its grid resistor  $R_{g3}$  (point 0) and point 3 on the filament string.

Thus, by including more and more of the series filament voltage drops, the bias voltage on succeeding tubes can be increased. Or, we can adjust the bias by returning the grid resistor of each tube to the desired point on the string.

**Grid Current Methods.** You are already familiar with the fact that grid current flow through a grid leak resistor will provide a bias voltage. The method is shown in Fig. 40A. Whether the grid resistor is at  $R_{c1}$  or at  $R_{c2}$ , the same effect is obtained. In either case, the signal voltage swings the grid positive to a sufficient extent for grid current to flow. The

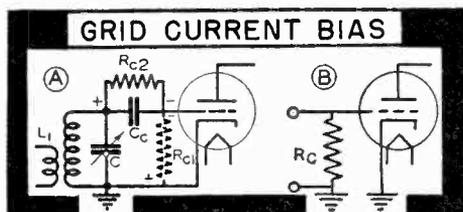


FIG. 40. The grid current can be used for bias purposes, as shown here.

grid current flow through the resistor then provides the bias voltage.

► A modern adaptation of this method is shown in Fig. 40B. Here, resistor  $R_c$  appears to be the usual grid resistor, used only as the load across which we develop a signal voltage. However, if we make this grid resistor very high in value, the tiny amount of convection current that normally flows through any tube will

set up a small bias voltage. (Convection current is the small grid current caused by some of the electrons speeding toward the plate happening to strike the grid wires. As these electrons are caught, they make the grid negative with respect to the cathode, so they tend to drift back to the more positive cathode through the grid circuit.) For some tubes, particularly certain first audio tubes, the bias developed in this manner is sufficient for normal operation.

Notice the difference between the circuits in Figs. 40A and 40B. In B, the bias is due to stray electrons,

but not always; in fact, in some circuits self-bias is undesirable for precisely this reason. There are other circuits in which it is impossible to use self-bias. For such circuits, we must use fixed bias sources—either batteries, which we have already discussed, or the set power supply. Fig. 41 shows several different ways in which a bias voltage can be obtained from the power supply.

In Fig. 41A, we have a combination of self and fixed bias. The plate current of the tube flows through  $R_3$  which provides a certain amount of self biasing. In addition, there is an

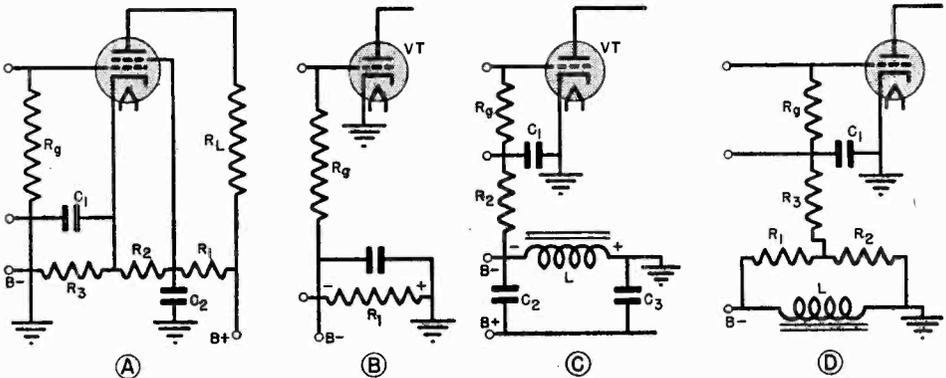


FIG. 41. When a fixed bias is needed, it can be obtained from the power supply by any of these methods.

which are captured by the grid and cause an extremely small grid current flow through a very high resistance. ( $R_c$  may be 10 to 20 megohms.) In this case, the tube has a bias whether or not a signal is applied. In Fig. 40A, however, bias exists only when a signal is applied.

### FIXED BIAS METHODS

Most of the self-bias methods are self-regulating—that is, the bias goes up when the tube current increases, and down when it decreases, and so tends to correct for variations in tube characteristics. This is usually an ad-

electron path from B— through  $R_3$ , bleeder resistor  $R_2$ , and the screen supply resistor  $R_1$  to B+. This extra bleeder current makes the current flow through  $R_3$  higher than the plate current value and increases the voltage drop across it. Thus, we can get the required bias with a smaller value of resistor  $R_3$ . Also, if we adjust the bleeder resistor  $R_2$  properly, we can make the bias voltage depend primarily on the bleeder current rather than on the tube plate current. The bias then becomes at least semi-fixed, as it does not depend so much on the tube characteristics.

► Fig. 41B shows another combination of self and fixed bias. The plate currents of all the tubes flow through  $R_1$ . The resulting voltage drop is used to bias  $VT$ .

► Fig. 41C is similar to Fig. 41B, except for the use of  $L$  instead of  $R_1$ . In this case, the filter choke (or the speaker field being used as a choke) is in the negative side of the power supply. The drop across its d.c. resistance becomes the bias for  $VT$ . As there is also a large a.c. drop across the inductance  $L$ , a filter  $R_2-C_1$  is used to prevent the a.c. components from reaching the grid.

► When the speaker field is used as a choke, the drop across it is usually larger than is required for bias purposes. In some cases a tap is placed on the field so that a portion of the drop may be used. The connections will then be similar to those shown

in Fig. 41C, except that  $R_1$  will go to the tap of  $L$  instead of its B—terminal.

Another arrangement uses a voltage divider like that shown in Fig. 41D. Resistors  $R_1$  and  $R_2$  divide the d.c. voltage drop across  $L$ , and the portion of the voltage across  $R_2$  becomes the bias. The amount of bias can be adjusted by properly choosing the values of  $R_1$  and  $R_2$ . These resistors have high values so they will not reduce the effectiveness of  $L$  as a filter. (High values are possible because no d.c. current flows in the grid circuit.)

► As you will notice, most of the fixed bias methods also are self bias systems to a certain extent. However, if the bleeder current, or the current from the other tubes, is a large portion of the current through the bias resistor (or coil), then the bias will be relatively fixed.

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THE N. R. I. COURSE PREPARES YOU TO BECOME A <b>RADIOTRICIAN &amp; TELETRICIAN</b> <small>(REGISTERED U.S. PATENT OFFICE)</small>
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# Lesson Questions

**Be sure to number your Answer Sheet 17FR-3.**

**Place your Student Number on every Answer Sheet.**

***Send in your set of answers for this lesson immediately after you finish them, as instructed in the Study Schedule. This will give you the greatest possible benefit from our speedy personal grading service.***

1. What are the two steps involved in the demodulation (detection) of an a.m. signal?
2. What are the three performance characteristics on which detectors are rated?
3. Does a diode give better fidelity on: 1, strong signals; or 2, weak signals?
4. What would happen to the higher audio frequencies if the value of the detector load resistance is increased (in a diode detector circuit)?
5. What use is made of the d.c. component of the load voltage developed in diode detector circuits?
6. Across which one of the following condensers in Fig. 21 is there neither r.f. nor a.f. voltage:  $C_2$ ;  $C_3$ ;  $C_5$ ?
7. In the television detector shown in Fig. 22, what furnishes the filtering capacity?
8. In a triode grid-leak detector, does rectification occur: 1, in the plate circuit; or 2, in the grid circuit?
9. What step, prior to rectification and signal separation, must be performed in an f.m. demodulator?
10. How does the tube in Fig. 40B get its C bias?

## ACHIEVEMENT BRINGS SATISFACTION

From my own experience, I know that the greatest satisfaction comes from doing a *difficult* job well—from mastering a tough new subject, locating a particularly elusive trouble in a receiver, or getting a broadcast transmitter back on the air a short time after a breakdown occurs.

The more difficult the achievement, the greater is the satisfaction we feel. In the words of the French writer Moliere, "*The greater the obstacle, the more glory we have in overcoming it.*"

Each achievement gives us the confidence to tackle still larger jobs and overcome greater obstacles. For instance, if this particular lesson is harder for you to understand than previous lessons, but you stick with it until you've mastered the important facts, you'll feel pretty good about completing it and you'll start the next lesson with real confidence in your own ability.

My greatest satisfaction today comes from teaching others how to achieve success, so your achievements mean a lot to me.

J. E. SMITH

**NOW PRACTICAL  
REQUIREMENTS CHANGED  
RECEIVER CIRCUITS**

18FR-2

**NATIONAL RADIO INSTITUTE**

ESTABLISHED 1914

**WASHINGTON, D. C.**



# STUDY SCHEDULE NO. 18

A bird's-eye view of the rapid growth of radio, from the crystal set of three decades ago to the superheterodyne receiver of today, is presented in this book. Stirring history-making events like the sending of the first wireless signals across the Atlantic are interwoven with achievements of men like Marconi, De Forest, Armstrong, Hertz and hundreds of others. You will learn how each new receiver evolved logically from the preceding, and how practical requirements of the listening public resulted in one improvement after another.

This book is in its present position in your Course because a knowledge of the technical problems which have been solved by the pioneers in radio will help you to understand modern circuits. Also, some of these older receivers are still in use; the information given will be of real practical value when they require servicing, and will tell you what to expect from these older sets in the way of performance.

- 1. Radio Wireless ..... Pages 1-5  
Radio circuits from the earliest types up to the introduction of the three-element tube and audio amplification. Answer Lesson Questions 1, 2 and 3.
- 2. Radio Broadcasting is Born ..... Pages 5-9  
The first broadcasts; development of the t.r.f. receiver. Answer Lesson Questions 4 and 5.
- 3. Receivers Grow Up ..... Pages 10-16  
Simplified tuning; battery eliminators; the a.c. receiver; dynamic loudspeakers. Answer Lesson Questions 6, 7, 8 and 9.
- 4. Radio Comes of Age ..... Pages 16-23  
New tubes; the rise of the superheterodyne; midget, auto and farm receivers.
- 5. Modern Receiver Types ..... Pages 23-28  
Receiver comparisons; all-wave and high-fidelity types; the f.m. receiver. Answer Lesson Question 10.
- 6. Mail Your Answers for this Lesson to N.R.I. for Grading.
- 7. Study One Circuit in Reference Text 17X.  
To get the most benefit from this valuable and practical reference text, study one circuit in it per lesson. One slow and careful reading will do, if you look at the diagram each time a part is mentioned. Do this for all the circuits, even though they may not be mentioned in the Study Schedules of other lessons.
- 8. Start Studying the Next Lesson.

# HOW PRACTICAL REQUIREMENTS CHANGED RECEIVER CIRCUITS

## Radio Wireless

**A**FTER the first novelty of owning a radio wore off, back in the days when the radio industry was just getting started, owners of sets began to want something better. Receiver designers learned of these requirements and undertook research with a view to improving the receivers. New discoveries resulted and were quickly adopted; innovation followed innovation as consumers kept demanding better receivers.

► The circuit chosen for any particular receiver depends upon seven important technical requirements demanded by the public for that type of receiver:

1. *selectivity;*
2. *sensitivity;*
3. *fidelity;*
4. *power output;*
5. *signal-to-noise ratio;*
6. *interference reduction;*
7. *ease of operation.*

Today the radio engineer can produce receivers which meet the requirements of the public, but it took a long time to reach this stage. Early radio receivers were of decidedly limited ability.

So that you can better appreciate the technical perfection of the present-day radio receiver, as well as realize why various receiver models today differ so greatly from each other, we shall trace the development of radio receivers from the days prior to the crystal detector to the all-wave, high-fidelity superheterodyne receivers of today.

### COHERERS TO DIODE DETECTORS

About 1864 James Clerk Maxwell, one of the great British scientists, predicted by means of complicated mathematics that an oscillating current would radiate electromagnetic waves which would be identical with light waves except for frequency. These predictions aroused considerable scientific argument. For twenty years Heinrich Hertz, a German scientist, attempted to prove that Maxwell's assumptions were wrong and that these electromagnetic waves could not exist. The results of these experiments quite surprised Hertz, finally converting him to the Maxwell theory. As a result, Hertz published in 1887 and 1889 the papers which brought him fame and gave to the world experimental proof that radio waves were entirely possible. Even today radio waves are designated by many people as *Hertzian waves*.

In 1890 Sir Oliver Lodge, another brilliant British scientist, actually set up an oscillating circuit (a charged condenser shunted by a heavy wire loop in series with a spark gap), and proved that another oscillatory circuit would pick up by induction the energy from the first circuit, if properly tuned.

**The Coherer.** The next important development before the actual sending of messages through the air was the coherer, a device which is based upon the fact that the enormous resistance offered to the passage of electric current by loose metal filings is greatly

reduced under the influence of alternating current.

In one very popular form, a coherer consisted of a two- or three-inch long glass tube, inside which were metal plugs spaced perhaps a quarter inch apart, with the space between the plugs filled with metallic filings. Various metals were used for the plugs or electrodes and for the filings. The passage of radio frequency current through the filings caused them to "cohere" or stick together, forming a conducting path for the direct current which actuated a telegraph sounder (an electromagnet which makes an audible click each time it operates). The great problem was to break up this path again after the r.f.

1896 to combine the results of all these scientists and produce one simple and workable radio system. His early apparatus was capable of transmitting messages for only short distances, but on December 12, 1901 he used the transmitter and receiver shown in Fig. 1 to transmit three dots (the letter S) over and over again from Poldhu, England to St. Johns, Newfoundland. To Marconi rightfully belongs the title "father of wireless," for he coordinated the work of others and put wireless on a practical working basis. (Before the advent of broadcasting, *radio* was known as *wireless*.)

► In order to secure long-distance transmission in those days, it was

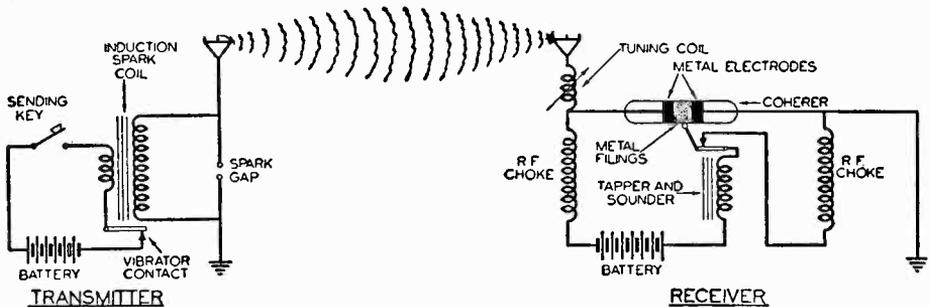


FIG. 1. Early Marconi wireless transmitting and receiving circuits. The vibrator at the transmitter operated as long as the key was pressed, interrupting the primary current of the induction coil and inducing in the secondary a high voltage. This high voltage caused an oscillatory spark discharge across the spark gap. The oscillations were radiated into space by the transmitting antenna, and induced r.f. currents in the receiving antenna. These currents made the coherer conductive, so that battery current passed through to operate the tapper and sounder. The buzzing sound of the tapper (an ordinary buzzer) was thus heard whenever the sending key was pressed.

current had ceased flowing; some experimenters mounted the coherer on a vibrating table, but the most popular arrangement was that used by Popoff of Russia and by Branly, in which a tapper attached to the telegraph sounder was used to jar the glass tube after the arrival of each signal.

► Scores of scientific men began experimenting with the newly-discovered Hertzian waves, each contributing additional data on the behavior of these waves, but it remained for Marconi in

necessary to use tremendously high-power oscillatory circuits, together with antennas much longer than those in common use today.

Having increased the power of transmitters to the practical limits of the time, scientists now turned their attention to improvements in receivers. The electrolytic detector, a fine platinum wire immersed in sulphuric acid, came into the picture around 1903. It was about this time also that the head-phone unit used in telephone systems

was brought into play to replace the telegraph sounder used for wireless. This one step increased the sensitivity of receivers considerably.

**Crystal Detectors.** At about this same time the crystal came into widespread use as a more convenient and

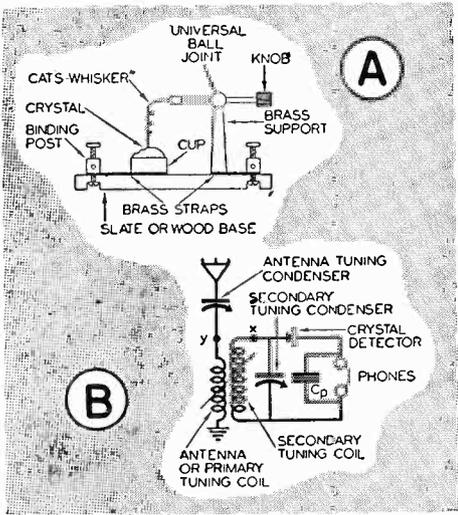


FIG. 2. A cross-section view of a typical crystal detector assembly is shown at A, and a typical crystal set diagram is shown at B.

more sensitive detector. Way back in 1874, Braun had discovered that two crystals making point-to-point contact with each other had a high resistance to the flow of current in one direction and a low resistance to current flow in the opposite direction. Single crystals in contact with a sharpened length of fine wire were found to give equally good rectifying action; materials such as galena, iron pyrite, fused silicon, carborundum, molybdenite, and other minerals came into use. A typical crystal detector is shown in Fig. 2A; this has the necessary mechanical adjustment which permits setting the cat's-whisker at a spot on the crystal which gives good rectification.

With the development of more sen-

sitive crystal detectors, together with sensitive headphones, long-distance reception became possible, and more and more transmitters went on the air. These were all code or telegraph transmitters, of course, and were operating on wavelengths longer than 600 meters (lower than 500 kc.). Congestion of the airways became severe, and steps were taken to make transmitters use narrower bands of frequencies and to make receivers more selective.

One of the earliest schemes for improving the selectivity (station-separating ability) of a receiver was the use of variable coupling between the antenna coil and the secondary coil, as illustrated in Fig. 2B. Additional coils, known as *loading coils*, were sometimes inserted in the tuning circuits at points *x* and *y* to increase the circuit inductance and thus give reception on the longer wavelengths (lower frequencies).

**The Diode Tube.** In 1904 Dr. Fleming invented the diode vacuum tube, then known as the *Fleming valve*.

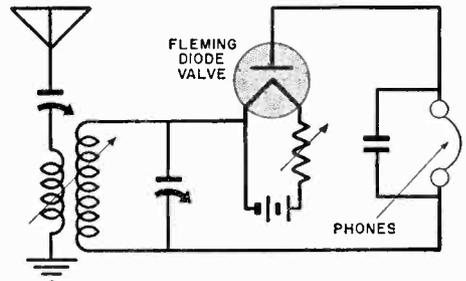


FIG. 3. An early diode tube detector circuit.

This contained a wire filament heated by direct current, and a metal plate or anode mounted a short distance away, both electrodes being enclosed in an evacuated glass envelope. A typical Fleming valve detector circuit is shown in Fig. 3. The same tuning controls are used here as in the crystal detector circuit of Fig. 2B to secure the required selectivity. Actually the two circuits

are identical except for the type of detector used.

## DE FOREST INTRODUCES TRIODE TUBES

The introduction of the grid in the Fleming diode tube by Lee De Forest in 1909 was an outstanding contribution to the radio art, and revolutionized the entire radio industry. The De Forest triode tube was first used only

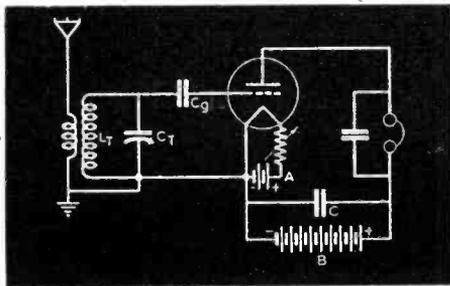


FIG. 4. One of the first triode tube detector circuits.

in detector circuits, where it provided a considerable amount of signal amplification in addition to demodulation.

A typical triode detector circuit is shown in Fig. 4. The output of the tuned circuit  $L_T$ - $C_T$  was fed to the grid and filament of the triode tube through series condenser  $C_g$ . The headphones, in series with a 22½- to 45-volt battery, were connected between the plate and filament of the tube. It was found that a leaky condenser gave best results for  $C_g$  (the reason for this was not at first apparent, but is now fully understood—the intelligence signal voltage was developed across the resistance of this leaky condenser and then amplified by the triode tube). Thus the filament and grid of the tube together acted as a diode rectifier, while the entire tube acted as a low-frequency amplifier. This circuit (known as a grid-leak and condenser detector) is still used to a certain extent.

Early triode tubes were rather poorly evacuated and therefore contained a certain amount of gas. This gas made possible a larger plate current than would exist in a pure vacuum. These gas tubes were commonly known as “soft” tubes and made very sensitive detectors. However, they were difficult to adjust properly as detectors, they were noisy, and their useful life was very short. Improving the vacuum in the tube (giving a so-called “hard” tube) gave better stability and longer life, but somewhat poorer sensitivity.

► The loss in sensitivity through the use of a hard tube was more than overcome by the invention of regeneration by De Forest in 1912 and by Armstrong in 1914. A simplified form of this regenerative circuit is shown in Fig. 5; here the modulated r.f. current

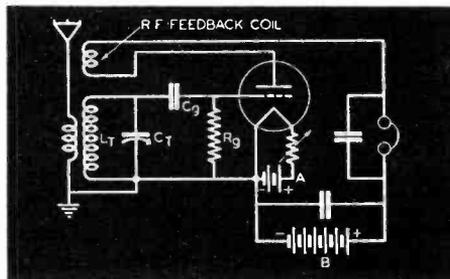


FIG. 5. The introduction of regeneration resulted in a tremendous increase in sensitivity. As condensers were improved, the grid leak  $R_g$  became necessary.

which exists in the plate circuit is fed back to the input circuit to reinforce the incoming signal. In the hands of an expert the regenerative detector gave amazing results, setting up new and more consistent long-distance receiving records.

## AUDIO AMPLIFICATION

Attempts to bring up the volume of very weak code signals by increasing the feedback of a regenerative detector were decidedly unsatisfactory, for the

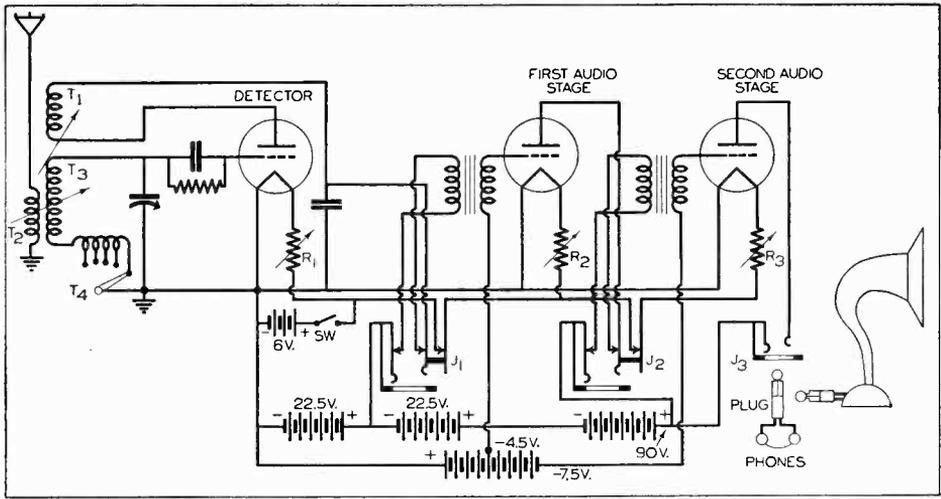


FIG. 6. A regenerative receiver of 1920-1924 vintage, having two a.f. stages. These receivers had an ON-OFF switch SW; separate filament controls  $R_1$ ,  $R_2$  and  $R_3$ ; a tuning condenser; a regeneration control varying the coupling between  $T_1$  and  $T_3$ ; variable coupling between  $T_2$  and  $T_3$  and sometimes a tapped loading coil arrangement like  $T_4$  to extend the tuning range. Thus, there were seven or eight controls on the panel of these receivers. When receiving strong signals, phones could be plugged into jack  $J_1$ , which disconnected the a.f. tube filaments, prolonging both tube and battery life. With the phones plugged into  $J_2$ , only the first a.f. stage was in use, while full amplification for weak signals was obtained with the phones in  $J_3$ . The first horn speakers could be plugged into  $J_3$  also.

circuit became unstable and often went into oscillation, producing very annoying squeals. Practical requirements demanded amplification of weak but clearly demodulated signals — audio amplification.

As soon as suitable tubes were available, one or more stages of audio amplification were added to the regenerative detector to increase the volume of weak code signals. Plug-in jacks were provided for each stage, as in the typical circuit shown in Fig. 6, so that

headphones could be inserted either in the detector stage, the first audio stage or the second audio stage. This was a crude but effective means of controlling volume. Although this was a great improvement over any other receiver available at the time, manufacturers of radio apparatus were not yet ready to scrap the then widely used crystal detectors; they simply manufactured a separate two-stage audio amplifier which could be connected to the output of any crystal detector.

## Radio Broadcasting is Born

With code wireless an established fact, engineers turned their attention to the transmission of voice and music through the air. The old spark coil transmitter, with its highly damped oscillations, was clearly inadequate for

the transmission of voice signals. In 1903 Poulsen had perfected a generator of high-frequency current which utilized an electric arc; he showed experimentally that telephonic communication through space over very short

distances was entirely possible with his transmitter if a microphone were used to modulate the radio waves. The Poulsen arc transmitter was used extensively for code transmission, but did not prove entirely satisfactory for voice signals.

► A high-frequency dynamo-electric generator was designed and constructed by Fessenden in 1906 for the production of continuous waves; with it he was able to transmit telephone messages for 15 miles.

► De Forest in 1907 was the first to use wireless for the broadcasting of phonograph records; he placed his musical program on the air without



FIG. 7. A photograph of an early receiver.

preliminary announcement, completely surprising the wireless code operators on duty at the time.

► The invention of the triode vacuum tube and its development as an oscillator for the generation of high-frequency current resulted in the world-famous demonstration on December 30, 1915, when the government radio station at Arlington, Virginia, succeeded in sending telephone messages through the air to Honolulu, Hawaii and Paris, France.

► During World War I (1914-1918) the large electrical manufacturers in this country had developed elaborate wireless research and manufacturing departments for the production of

radio apparatus for the government. The close of the war marked the collapse of the market for their products, and naturally these manufacturers tried to find new ways of stimulating wireless activities and creating new markets. The Westinghouse Electric and Manufacturing Company, believing that the future of radio was in the direction of voice and music broadcasting, delegated Frank Conrad to carry out research work along this line. Fortified with a vast amount of wireless experience, the latest scientific aids, and practically unlimited manufacturing facilities, Conrad developed the first broadcasting station. For months radio amateurs in the vicinity of Pittsburgh listened with interest to experiments in broadcasting, and then, on November 2, 1920, Westinghouse Radio Station KDKA astonished the world by broadcasting the results of the presidential election.

Wireless gradually came to be known as radio, and every one wanted to listen in on the broadcasts. Shortly after this epoch-making broadcast, Westinghouse opened Radio Station WJZ in New York City, the first *commercial* broadcasting station. Station after station sprang into existence—broadcasting swept the country, expanding into a vast industry and proving correct the early prophecies of Westinghouse.

► The nuisance of having to wear headphones whenever listening to radio broadcasts had long been recognized as a distinct drawback to the universal popularity of radio. In 1921 a megaphone was attached to an electromagnetic headphone unit, forming the first practical loudspeaker. The vibrating diaphragm of the phone unit, acting in conjunction with the outward flaring curve of the megaphone, served to excite the surrounding air waves so that all the persons in a room could



Courtesy Westinghouse

Broadcasting the election returns over KDKA on November 2, 1920. This historic picture shows the station engineer, announcer, two newsmen, and the transmitter.

listen to a single program. The immediate acceptance of this rather crude loudspeaker quickly resulted in the development of the balanced armature unit which gave greater diaphragm movement and consequently greater sound output.

During the period between 1922 and 1924, the receiver circuit shown in Fig. 5, used with batteries and a horn-type loudspeaker, was the accepted standard. At this stage in the development of radio, people wanted plenty of "gadgets" on their receivers; the more controls a receiver had, the better it was received by the public. A typi-

cal receiver is shown in Fig. 7.

Another form of regenerative receiver was developed about this time. This circuit, known as the super-regenerative, had a special means of preventing oscillation, so the set could operate at the point of maximum amplification. However, the circuit had extremely poor selectivity. It tuned so broadly it could not be used in crowded wavebands and thus lost favor.

## T.R.F. RECEIVERS

Regenerative receivers had one very serious fault; they would feed energy

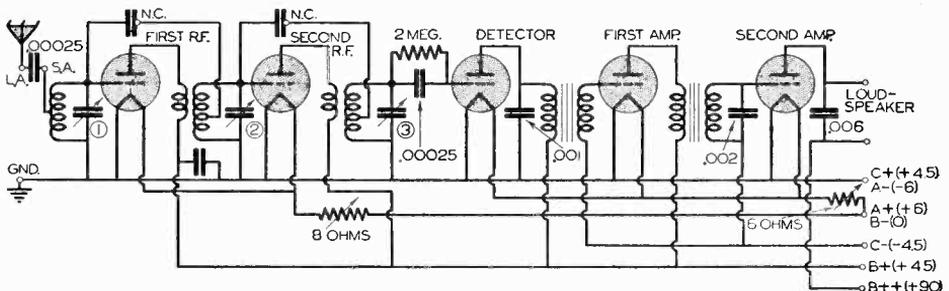


FIG. 8. Circuit of a typical five-tube neodydyne receiver, the Freed-Eisman model NR-5, which appeared in 1925. Feedback in the r.f. stages was cancelled out by adjusting the neutralizing condensers marked N.C.

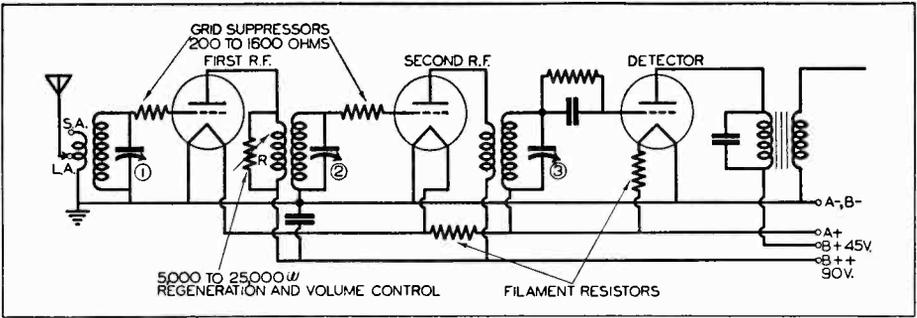


FIG. 9. The t.r.f. amplifier and detector section of a receiver which was popular about 1926. The grid leak and condenser detector was usually followed by two stages of a.f. amplification.

back into the receiving antenna, and this would be radiated, interfering with reception of signals by other receivers in the immediate neighborhood. Owners of radio receivers did not enjoy listening to ear-splitting squeals each time their neighbors decided to tune in a new station, and thus the public created a demand for the next improvement.

Around 1925 engineers seriously began to consider replacing regenerative circuits with circuits using r.f. amplification ahead of a non-regenerative detector. The big problem was making the then available vacuum tubes amplify r.f. signals without causing feedback which would result in shrill squeals accompanying the desired program. The neutrodyne circuit, as developed by Hazeltine, effectively prevented r.f. feedback and proved the solution to this problem.

A typical neutrodyne circuit, as used in receivers of this period, is shown in Fig. 8. There were three tuning dials, one for each resonant circuit tuning condenser in the r.f. tube stages and the detector. Other controls included a rheostat for varying the filament current to the detector and two a.f. amplifier tubes, and another rheostat to control the filament current to the r.f. amplifier tubes. Plug-in jacks (omitted for simplicity in the diagram) were inserted in each audio stage to allow the

listener to use headphones instead of the less-sensitive loudspeaker on weak distant stations. People still insisted upon being able to stay up into the wee hours of the morning, with headphones "glued" to their ears, trying for coast-to-coast reception. The neutrodyne receiver of 1925 was heralded as the marvel of its day.

This "squeal-less" radio receiver captivated the attention of the radio public. Many manufacturers turned over their entire facilities to the production of these receivers under the Hazeltine patent, in order to meet the rapidly growing demand. Other manufacturers developed their own r.f. feedback cancellation systems, all attempting to balance out the plate-to-grid circuit feedback voltage with an equal and opposite voltage.

Some manufacturers attempted to get around the neutrodyne patent by developing tuned radio frequency circuits which had enough loss in each r.f. stage to absorb the feedback energy and prevent squeals. This "losser" method proved fairly successful.

Grid suppressors, connected as shown in Fig. 9, also came into use as cures for feedback troubles in some of the earlier t.r.f. receivers.

► Other circuit improvements inaugurated at about this period included a variable resistor connected across the primary of the second tuned r.f. trans-

former for the purpose of controlling regeneration and volume. Fixed resistors and ballast resistors (such as the Amperite) in the filament circuits made it unnecessary to adjust filament current. Three tuning controls and a volume control now were all the adjustments used for a standard receiver.

The schematic circuit diagram of a tuned radio frequency amplifier using a grid-leak-condenser detector is shown in Fig. 9. Receivers of this type were designed to regenerate slightly when resistor  $R$  was set at its highest ohmic value, thus giving perfect control and high r.f. gain.

evident, and the popularity of home-made receivers began to fade.

Out of this period of home-made receivers there stands out one receiver which gave exceptionally good performance. This receiver contained a neutralized t.r.f. amplifier ahead of a regenerative detector, and was commonly known to the old-timers as the Browning-Drake receiver, after its inventors. The circuit diagram for this receiver is given in Fig. 10; one stage of audio application is shown here, but ordinarily a power stage was also used to provide loudspeaker operation.

► Home set-building died a natural

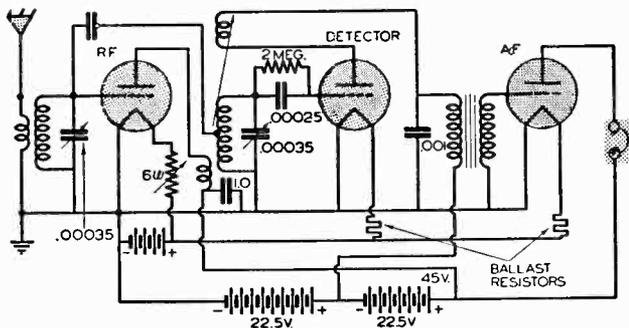


FIG. 10. One form of the famous Browning-Drake receiver circuit, the hit of the 1925-1927 period of home-made receivers.

## HOME-MADE RECEIVERS

From the very first days of wireless, the building of radio receivers in homes was a popular hobby. When radio broadcasting began, many more thousands joined the ranks of receiving set builders. Sets were first built from wiring diagrams, but soon complete kits of parts were made available. In those days just as many radio receivers were home-made as were produced by manufacturers. Gradually, however, as manufacturing techniques were placed on a production basis, the economy and superiority of commercial sets became

death soon after the introduction of the superheterodyne, primarily because of the increasing complexity of receiver circuits and the necessity of having special instruments for aligning the superheterodyne. Today, people who build their own receivers usually buy complete kits of parts, but the number of these experimenters is relatively small because little or no money can be saved. Modern radio factories, buying parts in very large quantities and using mass production techniques, can assemble complete receivers for less than the price which an experimenter would have to pay for the parts alone.

# Receivers Grow Up

The advent of t.r.f. receivers of the neutrodyne and grid suppressor types changed radio from a fad to an established industry, but prior to 1926 it was still looked upon as a fall-winter-spring affair. The radio industry struggled and worried through the summer months, wondering if business would return again in the fall. It did come back, bigger and better than ever each fall; with confidence established, business men began to invest great sums of money in radio. These investors could see no real reason for the summer

be made to operate exactly alike; referring to Figs. 8 and 9 where the tuning circuits are marked 1, 2 and 3, manufacturers designed circuits 2 and 3 to tune exactly alike, permitting control of the two tuning condensers with a single dial. Naturally, to increase ease of tuning, it was desirable to have the dial of circuit 1 (the input tuning circuit) have the same reading as the single dial used for circuits 2 and 3 when the set was tuned to a station. But the effects of the antenna on circuit 1 were quite marked—if the input circuit were designed for a short antenna (S.A.), then the connection of a long antenna (L.A.) would make the dial reading for tuning condensers 2 and 3 differ as much as ten dial divisions from that for tuning condenser 1. The difference in the characteristics of long and short antennas was overcome by providing two antenna posts, one for a long antenna and one for a short antenna; with the antenna connected to the proper post, and condensers 2 and 3 ganged together, a two-dial receiver was obtained. One form of two-dial control is illustrated in Fig. 11A; the third condenser was attached to the shaft of one of the two condensers shown.

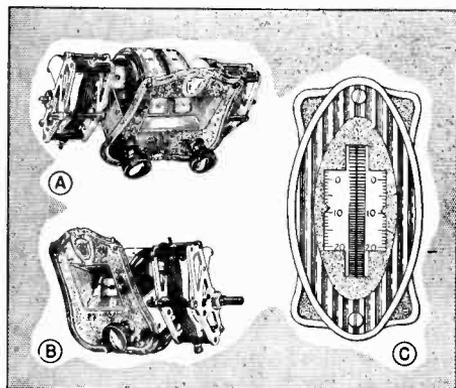


FIG. 11. Types of tuning dials in common use during the period when radio was passing from multi-dial to single-dial control.

slump, and demanded an organized effort to bridge the gap. Broadcasting stations boosted their power during the summer months, better and better programs were put on the air, and reasons for the summer slump in radio popularity were studied.

## SIMPLIFIED TUNING

It was found that many people objected to three-dial receivers on the basis that they had only two hands with which to tune. It was found that two of the three tuning circuits could

The next step in the simplification of tuning was the introduction of drum dials, which permitted adjusting the two tuning controls either separately or together. The face of one of these two-drum controls is shown in Fig. 11C; one drum controlled the condensers for tuning circuits 2 and 3, and the other controlled the single antenna tuning condenser. Some receivers used three separate drum dials, one for each condenser, while others had all three condensers controlled by a single dial, like that shown in Fig. 11B, for rough

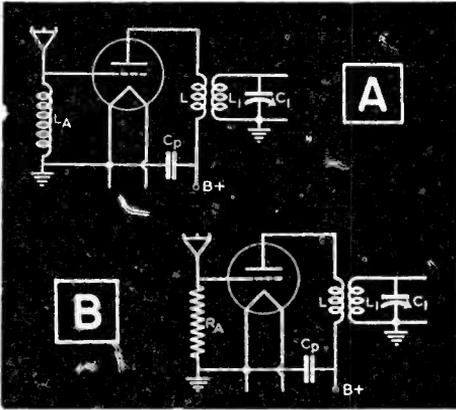


FIG. 12. Two forms of un-tuned antenna circuits used with r.f. receivers.

tuning to a station, and had some means for making fine (vernier) adjustments of each condenser when more accurate tuning was desired.

► The ideal single-dial receiver was still sought by radio designers and by the public, for with it even a child would be able to tune in radio broadcasts correctly and easily. It was fully realized that the antenna circuit was the trouble-maker. Un-tuned antenna circuits like those shown in Figs. 12A and 12B, followed by two stages of tuned r.f. amplification (three variable condenser circuits), were tried but found unsatisfactory. The trouble was that in the vicinity of powerful local

stations, the strong r.f. input signal reaching the first r.f. amplifier tube would make that tube work as a detector, and the strong incoming local signal would mix with or modulate whatever other incoming signal was being tuned in. This so-called *cross-modulation interference* effect was so objectionable that the idea of an un-tuned antenna system was dropped completely. Incidentally, cross-modulation is still a problem in radio receiver design.

► Engineers reasoned that if the r.f. section could be designed with plenty of gain, exact tuning in the antenna circuit would not be necessary and single-dial controls could be used. Tests proved this reasoning to be correct as far as local and near-distant stations were concerned, and the single-dial receiver became a reality.

The circuit of one of the first popular single-dial receivers is shown in Fig. 13. Notice that it contains a three-stage r.f. amplifier (four ganged variable condensers), a grid-leak-condenser detector and two transformer-coupled audio amplifier stages. The variable inductance, then known as a *variometer*, was coupled to the antenna exactly to resonance when a weak distant station was being received, but ordinarily was not

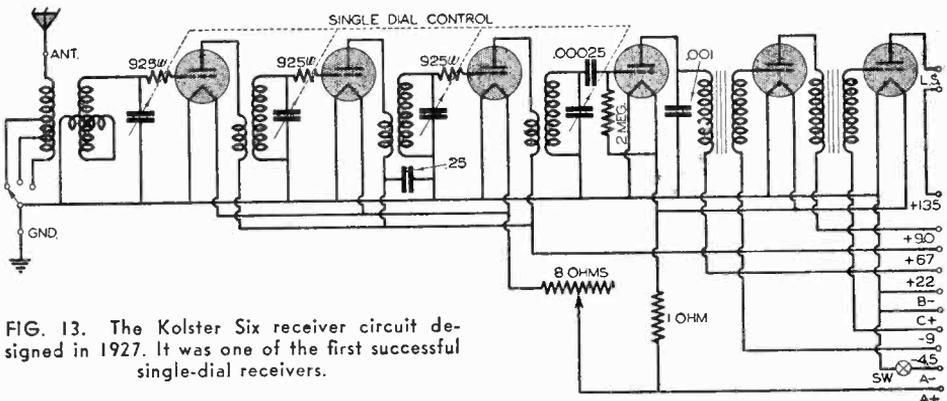


FIG. 13. The Kolster Six receiver circuit designed in 1927. It was one of the first successful single-dial receivers.

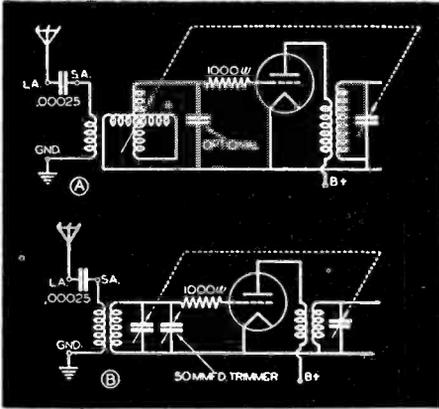


FIG. 14. Two methods of tuning the antenna input circuit to resonance with the other circuits, when tuning in weak distant stations.

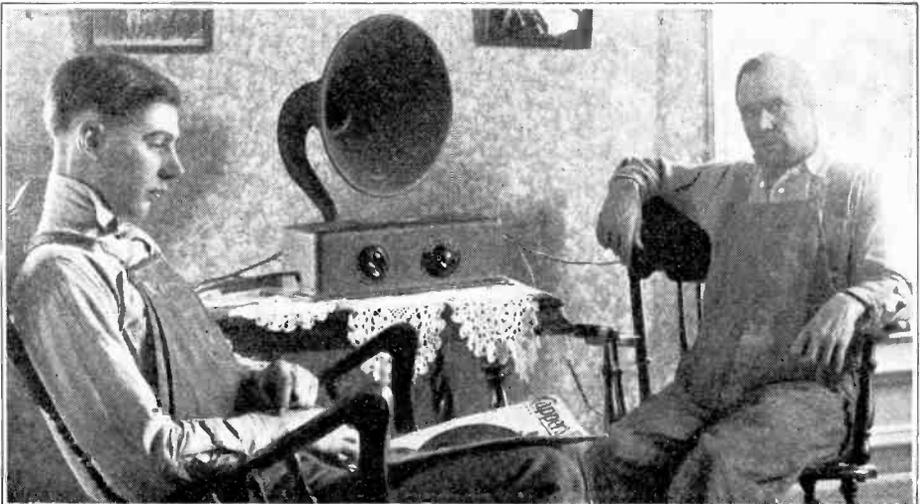
needed or used when tuning in local stations. There were several taps on the antenna coil for adjusting the circuit to various lengths of antennas; the tap switch was mounted on the front panel of the receiver. With this tap adjusted for loose coupling (only a few turns in the antenna coil), good reception of high-frequency stations was obtained. With tight coupling (many turns in the antenna coil), good

reception at low frequencies was secured.

► After the introduction of this first successful single-dial receiver, many others followed. Some omitted the first variable condenser and the variometer was ganged to the variable condensers in the manner shown in Fig. 14A. Some omitted the variometer entirely, using one section of the ganged variable condenser to tune the antenna circuit broadly, and using a trimmer or small variable condenser in parallel to permit accurate adjustment when receiving weak distant stations, as shown in Fig. 14B. At this stage in development, almost all receivers were tuned in exactly the same manner; strong stations were tuned in with the **single** tuning dial, while weak stations were first tuned in with this dial and then the antenna circuit was adjusted for maximum output by means of the antenna vernier trimmer.

#### A, B AND C BATTERY ELIMINATORS

All early radio receivers up to the single-dial control sets obtained power



An Atwater Kent single-dial battery set with horn loudspeaker. One of the very popular early models, widely sold in both urban and rural areas.

from batteries of one type or another. Dry cells built into convenient blocks of fifteen or thirty cells, with all cells connected in series, were used to supply d.c. voltages to the plates of the tubes. The storage batteries used for filament power had to be removed for recharging at intervals of one to three weeks; many set owners bought their own battery chargers for this purpose. Later, trickle (low rate) chargers were

surges of current produced by the "jerk" tube action from damaging the tube and also to reduce the resulting noise.

B and C battery eliminators met with instant popularity, and the public began calling for A battery eliminators as well. Engineers went to work again, and by using identical electrical principles succeeded in developing a low-voltage, high-current rectifier with a

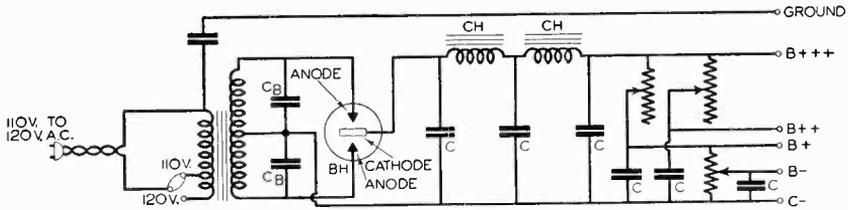


FIG. 15. An early B and C battery eliminator using the Raytheon BH tube as a full-wave rectifier. Paper filter condensers were used at C, while  $C_B$  were buffer condensers used to take out some of the noise interference developed in these tubes.

placed on the market for use in homes; these charged the battery at a low rate during periods when the radio receiver was not in use.

► The average radio receiver of the time required replacement of B batteries at intervals of two or six months, and replacement of C batteries every six to twelve months. This constant replacement of batteries was a decided nuisance, and the radio public in general began calling for the newly-developed battery eliminators.

In 1925 the Raytheon Company of Cambridge, Massachusetts, perfected the cold-cathode gaseous rectifier tube known as the Raytheon BH rectifier; this tube was quickly adopted by receiver manufacturers for use in B and C battery eliminators. A typical combination B and C battery eliminator using this tube is shown in schematic form in Fig. 15. You will note that it is simply a standard full-wave rectifier circuit except that buffer condensers  $C_B$  are connected across the two diode sections of the tube to prevent

ripple filter which satisfactorily eliminated the filament batteries. Various rectifying devices were used in A battery eliminators; many units were built with a special type A Raytheon cold-cathode rectifier tube or with the Tungar rectifier tubes originally developed for battery chargers, but the copper-oxide rectifier unit and the electrolytic rectifier unit were most widely used for A battery eliminators because of their lower cost and longer life. A typical A battery eliminator circuit

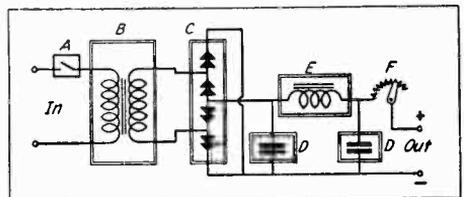


FIG. 16. Circuit of an A battery eliminator using a copper-oxide rectifier unit. B is a step-down transformer, C is a full-wave or bridge type copper-oxide rectifier unit, while D and E comprise a low-voltage, high-current filter. (Condensers D are 2000-mfd., 25-volt electrolytic condensers.) Rheostat F controls the output voltage, while A is the main power switch.

using a copper-oxide rectifier unit is shown in Fig. 16.

### THE A.C. RECEIVER

A person who went into a radio shop late in 1927 or early in 1928 to purchase a radio receiver had to choose five different and separate items (in addition to an antenna) if he desired a complete installation:

1. A radio receiver in its table model cabinet.

2. A loudspeaker.

3. A set of tubes.

4. An ABC power pack.

5. A suitable table on which to place the receiver and loudspeaker, with a shelf underneath for the ABC pack.

Each of these five units was available in a number of different brands, and oftentimes several different models were put out by one manufacturer.

The next step in the evolution of radio receivers was the combining of the ABC eliminator with the receiver itself to give a complete a.c.-operated receiver. At first, the ABC eliminator pack was simply placed inside an enlarged receiver cabinet, which made a somewhat bulky receiver. The attention of design engineers was then directed toward improvement of the power pack.

► As far back as 1925, RCA Manufacturing Company had been building into their receivers special innovations which the rest of the radio manufacturing industry was unable, either because of patent rights or because of high cost, to use. One of these special features was the hot-cathode diode tube used as a half-wave rectifier in the power pack; this tube was capable of furnishing as much as 60 ma. of rectified direct current, making it possible to connect the filaments of the type UV199 tubes in series and feed them with power obtained from the regular B and C battery eliminators. RCA was also the first to operate fila-

ment type power output tubes directly from a low-voltage a.c. source in commercially built receivers.

► Many schemes for operating receivers from a.c. power lines were being tried at about this time. In 1928 McCullough introduced the first a.c. tube. He reasoned that if the electron-emitting surface could be electrically separated from a source of heat, it would be possible to use alternating current for filament heating purposes without encountering trouble because of a.c. in the amplifying and detecting circuits. He designed the cathode as a metal sleeve coated with a good electron-emitting material, inside which was the heater filament, insulated from the sleeve by porcelain or some other insulating material. The McCullough tube, also known as the Kellogg tube (after its manufacturer) had its filament terminals at the top of the tube, with the other electrodes connected to standard tube base prongs.

The McCullough tube worked well as an a.f. amplifier, an r.f. amplifier or as a detector. Shortly afterwards McCullough introduced a power amplifier tube. (The principle of separating cathodes and filaments to permit a.c. operation of tubes is still in use today, although, of course, many refinements in tube construction have been introduced since these first a.c. tubes.) For B and C eliminators, most set manufacturers used the popular BII cold-cathode rectifier tubes.

► In 1928 RCA introduced their famous Radiola 17, an a.c.-operated receiver. At the same time they made available for general use three special a.c. tubes; the type 26 tube, having a filament which heated and cooled slowly and could, therefore, be operated from a low-voltage a.c. source when used in r.f. or a.f. amplifier stages; the type 27 heater type tube, having a separate cathode and designed primar-

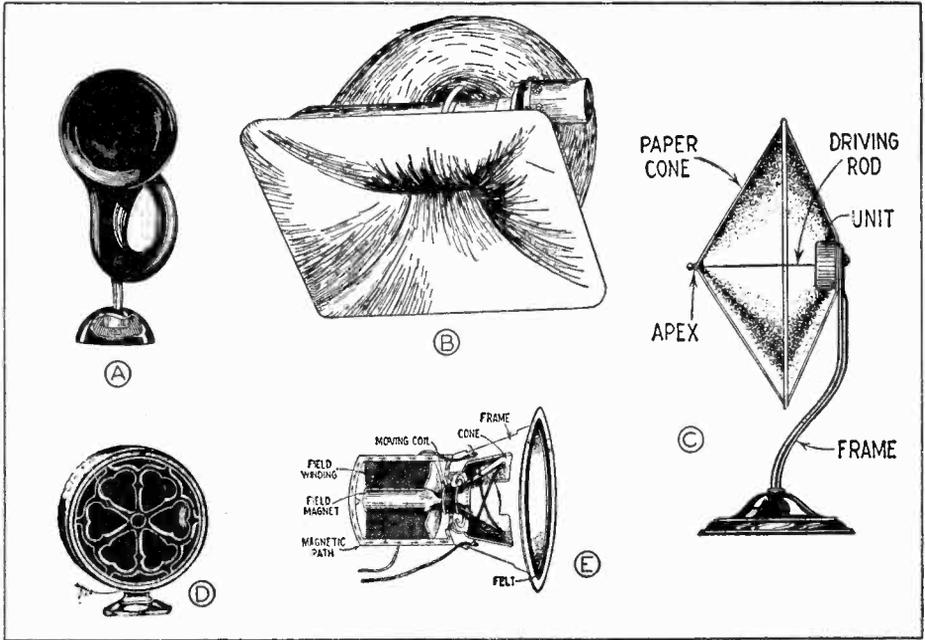


FIG. 17. Early forms of the horn, cone and dynamic loudspeakers.

ily for detector stages; the type 80 full-wave hot-cathode rectifier tube. The types 12 and 71 power tubes, originally designed for battery receivers, were found suitable for a.c. operation in the power audio stage. The year 1928 marked the general introduction of a.c.-operated receivers.

### DYNAMIC LOUDSPEAKERS

In the early days of radio, loudspeakers were made and sold independently of receivers, even after the introduction of a.c.-operated receivers. The early horn type loudspeaker, shown in Fig. 17A, used a single head-phone as a driving unit; greatly improved units, with balanced magnetic driving unit and long exponential (curved) horns like that in Fig. 17B, soon appeared and met with widespread acceptance. Next came the cone type loudspeaker, a paper cone driven by a balanced magnetic unit at its apex or peak. The first cone loud-

speaker was made up of two paper cones cemented together at their rims, with the driving unit inside, mounted at the apex of one cone and driving the apex of the other cone as shown in Fig. 17C.

Single-cone loudspeakers came next, with the edges of the cone attached to a fixed mounting ring by a soft leather washer. The driving unit was mounted inside, and was linked to the apex of the cone. Cone loudspeakers were made in various sizes ranging from 8 to 36 inches in diameter, with the large sizes naturally giving excellent bass response. Some were placed in metal housings similar to that shown in Fig. 17D.

► In 1925, Rice and Kellogg of the General Electric Company presented a technical paper describing their dynamic loudspeaker, a unit using a moving coil to drive a paper cone, the edges of which were clamped to a fixed rim; details of this loudspeaker appear in Fig. 17E. The edges of this cone were

made of soft, flexible leather, permitting the entire cone to move. These men also brought out the importance of the baffle in securing good low-frequency response.

In 1925, RCA Manufacturing Company used this first dynamic loudspeaker with one of their superheterodyne receivers, an a.c. set using the old UV199 tubes (with filaments in series, getting power from a diode rectifier) and two type 210 power tubes connected in push-pull for the output stage. Because of patent difficulties, however, other manufacturers were unable to take advantage of these new developments.

### THE RADIO RECEIVER PATENT SITUATION

To describe things mildly, the radio manufacturing industry prior to 1929 was decidedly "in a mess." Large-scale production of receivers was risky, for the important radio patents were in many different hands and there was always the possibility of lawsuits for infringement of patent rights. Rather than give away all their profits in the form of royalties to patent holders, many manufacturers took the risk of lawsuits, knowing full well that a patent showdown would come very soon.

The Radio Corporation of America, formed during World War I at the suggestion of the United States Govern-

ment in order to centralize radio facilities, had grown by leaps and bounds after the birth of radio broadcasting. This organization acquired the basic patents of the old American Marconi Company, together with American Telephone and Telegraph, Westinghouse and General Electric radio patents, the patents of other companies which it absorbed from time to time, patents obtained through outright purchase or through exchange with other radio manufacturers, and patents resulting from research in RCA laboratories. This organization was thus in a position to produce radio receivers and equipment without fear of serious litigation; it controlled the basic regeneration, t.r.f., superheterodyne, a.c. power pack, dynamic loudspeaker, vacuum tube, and vacuum tube circuit patents.

Realizing that order had to come from the existing chaos if the radio industry was to prosper, RCA proceeded to license reputable manufacturers. These then could, by combining their own patents with those of RCA and other patent-holding companies to whom they were paying license fees, manufacture radio receivers with reasonable freedom from lawsuits.

Manufacturers thus joined together, so today all reputable receiver manufacturers, both large and small, operate under a common patent agreement.

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## Radio Comes of Age

With patent problems at last out of the way, the radio industry definitely entered the "big business" class. Receiving circuits had been developed up to the point where the demands of the public were pretty nearly satisfied for a time, and manufacturers now turned

their attention to cabinets which would improve the appearance of their product.

Before 1929, a few radio manufacturers had recognized the desire of the public for a single cabinet which would house a receiver with all associated ap-

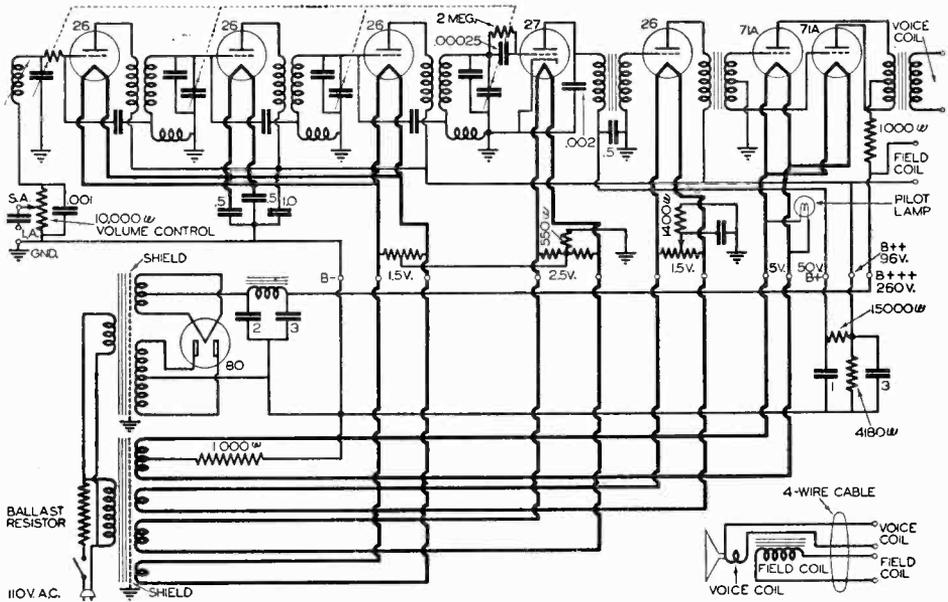


FIG. 18. Circuit of the Majestic Model 70B receiver, the hit of the 1929 radio season.

paratus. For some time, large furniture manufacturers had been producing adaptations of lowboy and highboy console cabinets with provisions for housing the radio chassis, the loudspeaker, and the ABC eliminator.

The production of complete radio receivers in several styles of console cabinets by the Grigsby-Grunow Company,\* incorporating all of the desirable technical features then available, was the outstanding event of the 1929 radio season. The Majestic line met with instant popularity, and many of these early console types of cabinet receivers are still in use today. The circuit of this Majestic receiver, shown in Fig. 18, is of special interest in connection with our studies of how practical requirements changed receiver circuits. You will observe that there are three t.r.f. amplifier stages (with four tuned circuits), a grid-leak-condenser detector, an a.f. voltage ampli-

fier stage, a push-pull audio output stage and a dynamic loudspeaker which received its field current from the power pack unit.

The Majestic receiver was entirely a.c.-operated, with type 26 tubes in each r.f. and a.f. voltage amplifier stage. It had a single tuning dial, with an additional control for tuning the antenna circuit exactly to resonance (by adjusting the inductance of the input coil) when receiving distant stations. A variable resistor which served as volume control coupled the antenna to the receiver input circuit. Each r.f. amplifier stage was neutralized. A very satisfactory sound output was obtained from the dynamic loudspeaker. An outstanding characteristic of this Majestic receiver was its excessive amplification of low notes; this false bass reproduction was pleasing to the public, and because of it, many radio listeners even today insist that the old Majestic was "king of them all."

\*Manufacturers of "Majestic" radio receivers.

► During the years of 1929, 1930, and 1931, the t.r.f. receiver with four or



increased power of broadcast transmitters, made the sensitivity of the square law grid-leak-condenser detector unnecessary. Designers turned to the linear C bias detector to secure less distortion in the detector stage and increase the gain and selectivity of the last t.r.f. stage. About 1930, tube manufacturers introduced the type 24 screen grid tube, primarily for r.f. amplification purposes; this gave far more amplification than a triode and made the old neutrodyne circuit obsolete.

With increased r.f. gain in receivers, it became possible to couple the antenna loosely and inductively to the first tuned circuit and secure true single-dial operation without any antenna trimmers. The well-designed receiver of 1930 consequently contained three stages of t.r.f. using screen grid tubes, a linear type 27 or type 24 detector tube, and a push-pull type 45 output stage preceded in some cases by an a.f. voltage amplifier stage. By this time the dynamic loudspeaker was considered an essential part of any receiver. The power pack used a full-wave type 80 rectifier tube. Volume control was commonly obtained by varying the C bias on the r.f. amplifier tubes.

## THE RISE OF THE SUPERHETERODYNE

The difficulties encountered in amplifying radio frequency signals above 200 kc. had long been recognized by radio engineers, particularly Armstrong, who had already contributed to the development of regenerative and super-regenerative receivers. The heterodyne method of reducing the frequency of the r.f. carrier practically suggested itself as a solution to this problem, and thus was born the superheterodyne principle of r.f. amplification, near the close of World War I. Armstrong obtained phenomenal gain

with intermediate frequencies of 30 to 45 kc. His superheterodyne patents were acquired by RCA, and about 1923 they produced the Radiola Super Heterodyne receiver, in which a 45-kc. i.f. section employing iron-core transformers and a sharp filter was used. RCA claimed the superheterodyne circuit for its own, and continued to develop it to a high state of efficiency. A number of radio parts manufacturers produced i.f. transformers and special preselector and oscillator coils for home experimenters, and home-made superheterodynes thus were the only other source of these receivers.

Late in 1930, the superheterodyne patents were released by RCA for use among their patent licensees, and t.r.f. circuits faded out of the picture, slowly at first and then with a sudden sweep as the superheterodyne circuit climbed to its throne as king of receiver circuits.

► In 1931, the RCA Manufacturing Company brought out the Radiola 80, a receiver which set a standard of performance that even today is difficult to exceed. The circuit diagram for this famous receiver is given in Fig. 19. Note that the two i.f. amplifier stages had screen grid tubes. The oscillator and second detector used triode tubes, with the latter serving as a linear power detector and feeding two type 45 tubes in a push-pull audio power output amplifier circuit. The first detector was preceded by one stage of tuned r.f. amplification, with a double resonant circuit serving as antenna coupler, to prevent interference troubles. All of the features necessary for good selectivity, high sensitivity, good fidelity, interference suppression, and high power output, are present in this circuit; its development over a period of years was an indication that receiver designers were giving attention to what the public wanted.

## MORE NEW TUBES

The inability of the C bias type of volume control to reduce the gain for strong signals without distorting them resulted (in 1932) in the introduction of the type 35 and type 51 variable mu tubes. These tubes had long  $E_g-I_p$  characteristic curves, so that very high negative C bias voltages were required to reduce the plate current to zero. Cross-modulation and modulation distortion effects were reduced by these tubes.

► The inability of screen grid tubes, even those of the variable mu type, to handle wide variations in plate voltage without secondary emission troubles practically forced the pentode r.f. amplifier tube into existence. In 1932 a new series of amplifying tubes, consisting of the type 56 triode, the type 57 pentode and the type 58 super-control pentode (which had a variable mu characteristic), appeared.

► The type 45 triode power amplifier tube did not have quite enough amplification to operate directly from a triode acting as linear C bias detector, unless special attention was given to the coupling transformer design. The type 47 power amplifier pentode came into existence to meet this deficiency; the 47 tube was used either singly or in push-pull circuits.

► When receivers were designed to be sensitive to weak signals, they invariably had too much gain for powerful local stations. For this reason the average broadcast listener had to tune with one hand on the tuning dial and the other on the volume control in order to prevent blasting when tuning from a weak to a strong signal.

To achieve more nearly ideal single-dial receiver control, automatic volume control was introduced around 1932. The first a.v.c.-controlled receivers contained an extra vacuum tube which

acted as a C bias detector and produced the variable negative C bias voltage required for control of the r.f. amplifier tubes. The result was automatic reduction in gain when strong signals were tuned in, and a certain amount of compensation for fading signals.

► The acceptance of the diode detector for signal demodulation purposes made unnecessary the use of a separate a.v.c. tube. The diode detector gave



*Courtesy Stewart Warner*

A large table model set, really a full-sized set in a table-type cabinet.

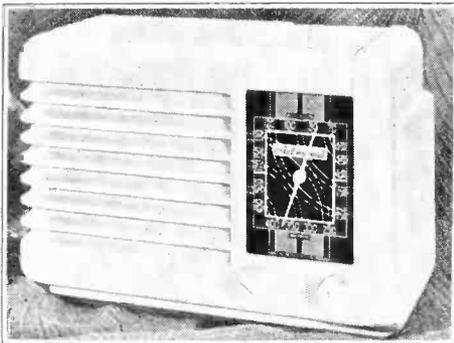
considerably less distortion than the linear C bias detector, was able to handle higher r.f. voltages, and at the same time developed a negative d.c. bias voltage for automatic volume control purposes. Diode detectors are now practically universal in the second detector stages of superheterodynes. The lower a.f. output of the diode detector required the use of an audio voltage amplifier. Rather than use an extra tube, two diodes and a triode were built into a single envelope, known as the type 55 duo-diode triode; here one di-

ode section served as second detector and the other supplied the a.v.c. voltage.

## MIDGET, AUTO AND FARM RADIO RECEIVERS

**Midget Receivers.** The year 1929 was the beginning of a period of business depression. Radio, being a new industry and a home substitute for more costly entertainment, kept right on going through the first years of the depression. By 1932, however, the greatly decreased income of the buying public began to make inroads on the volume of sales of high-priced radio receivers. Manufacturers decided that low-priced receivers were their only salvation.

The first midget receiver to attract widespread attention was the Jackson-Bell table model receiver, introduced about 1931. This first midget receiver was a t.r.f. set using screen grid r.f. amplifier tubes followed by a screen grid detector and a type 45 power am-



*Courtesy Admiral*

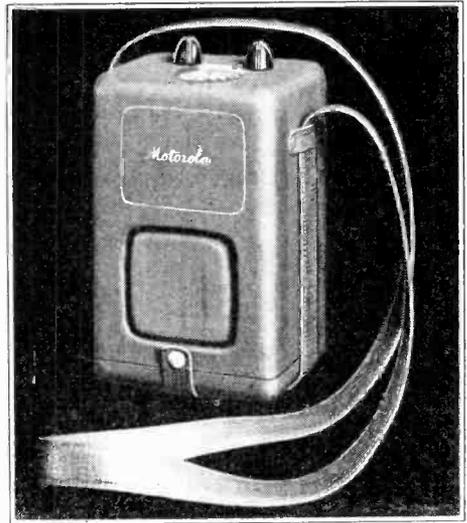
A modern type midget set.

plifier. The power pack used a type 80 full-wave rectifier tube.

Then came a flood of midget receivers, together with new tubes designed especially for them. In 1932, midget superheterodyne receivers were introduced. They invariably used the type 24 tube as a combination first oscil-

lator-mixer-detector, so a single tube performed the entire frequency converter function.

Large manufacturers of radio sets began turning out both t.r.f. and superheterodyne table model receivers in various sizes ranging from the "overcoat pocket" and "cigar box" sets to



*Courtesy Motorola*

A true portable, carried by the strap which goes over the shoulder. Special small-sized batteries and tiny radio parts make such receivers possible.

large-size table model receivers, all complete including loudspeaker, and some even having built-in antennas. Midget receivers, limited in both space and cost, resulted in two innovations—the use of multi-function tubes (particularly pentagrid converter tubes), and universal a.c.-d.c. operation (eliminating the need for a power transformer and permitting use of the receiver in either d.c. or a.c. localities). Midget receivers are available now in a wide variety of styles, sizes, and even colors.

**Auto Radios.** Mobile (movable) radio equipment was by no means new, for it was being used extensively by the War and Navy Departments, as well

as in airplanes. As far back as 1922, experimenters had been installing radio receivers in automobiles, but it was not until 1932 that radio manufacturers gave any considerable amount of attention to the auto radio receiver. Mass production technique was applied to the auto radio problem as another attempt to bridge the summer-time drop in radio sales.

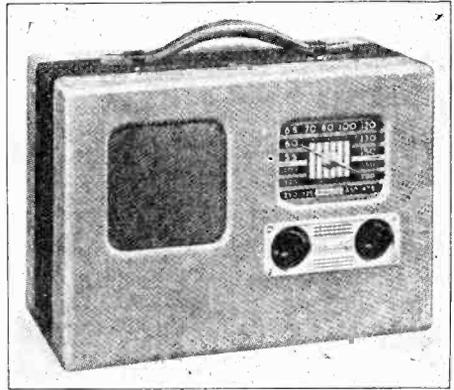
The first auto radios to appear on the general market included a radio chassis in a steel box, a remote tuning and volume control unit, either B batteries or a dynamotor for supplying the required high d.c. voltages, a separate loudspeaker, and suppressor resistors and condensers for the electrical system of the car to suppress ignition interference. The tube filaments (and the dynamotor) obtained power from the 6-volt auto storage battery.

It is interesting to note that auto radios went through the same development process as had home radios some years before. In each case the many separate units were gradually combined into one compact unit. The auto radio loudspeaker became an integral part of the radio chassis. The dynamotor was gradually replaced by the vibrator type of B supply unit, which also was built into the chassis. The superheterodyne became the standard auto radio circuit, and a.v.c. was introduced to compensate for fading. An entire auto radio set could now be bolted to the bulkhead or just behind the dash. Tuning controls which harmonized with the dash panel design of various car models were made available about 1935.

► Research work on auto radios naturally resulted in considerable benefit to radio receivers in general; perhaps the most outstanding contribution was the 6.3-volt line of vacuum tubes. The filaments in an auto radio receiver get their power from the 6-volt car stor-

age battery, the voltage of which may vary from 5 to 7.5 volts under ordinary driving conditions. This wide fluctuation in voltage made it impractical to use the existing 5-volt or 2.5-volt filament tubes, for these required accurate control of filament voltage. The new series of 6.3-volt heater type tubes could be connected directly to the storage battery. They were built to withstand the strain of voltage variations and yet each drew only about .3 ampere of filament current.

► Builders of universal a.c.-d.c. receivers tried using these 6.3-volt tubes, with the filaments connected in series,



*Courtesy Philco*

A modern three-way receiver, portable type, which will operate from a.c. or d.c. power lines or from self-contained batteries. Modern, highly efficient battery type tubes make these sets possible.

and obtained quite satisfactory results. A new group of tubes, drawing the same filament current as the 6.3-volt line but using higher filament voltages, now made its appearance; this group included the type 43 power output tube (with a 25-volt filament), together with the 12Z3 (12.5-volt filament) and 25Z5 (25-volt filament) rectifier tubes, designed specifically for use in midget receivers where the filaments of all tubes were connected in series.

**Farm Radios.** Throughout the phenomenal rise of radio receivers, the at-

tention of manufacturers had been focused on receivers designed primarily for either a.c. or d.c. power lines. Homes without electric power, particularly farm homes, were still struggling along with the old battery receivers like those shown in Figs. 8 and 9, purchasing new B batteries every few months and recharging storage batteries even oftener. To be sure, a few receivers were made with the low-drain type UV199 and UV120 tubes, but these tubes were entirely too fragile to meet with widespread acceptance.

► Late in 1930, the National Carbon Company introduced their air-cell battery, which could deliver slightly over 2 volts at a current drain of about .65

ampere for at least 1000 hours. When used as filament supply battery on a radio set operated an average of three hours a day, this battery had a useful life of about one year and required no recharging.

The air-cell battery was designed for use with the line of 2-volt tubes announced at about the same time by RCA. These tubes were rugged, small, and easily adapted to existing radio receiver circuits. It now became easy to design efficient battery receivers for farm homes, and the increased demand for these sets resulted in the production of a complete line of modern battery type tubes, far more efficient than early types.

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## Modern Receiver Types

After 1930, new tubes followed each other in rapid succession. In general these were produced either to satisfy a demand for tubes with special characteristics or to meet special power supply requirements.

With such a wide variety of tubes from which to choose, radio receiver designers now have little difficulty in designing a radio for any given power supply. It is perfectly possible to take a given signal circuit (r.f. amplifier, detector or audio amplifier) and adapt it to any power supply condition simply by rearranging the supply circuit and in some cases changing tubes. An example will show you how this is possible.\*

► Consider the superheterodyne circuit shown in Fig. 20, which is a bat-

tery-operated set. It uses a type 1A7 pentagrid converter tube which receives input power from the antenna through a transformer having a tuned secondary; this first tube acts as a combination oscillator and mixer-first detector. It is followed by an i.f. amplifier stage containing two i.f. transformers, after which comes a duplex-diode-triode tube functioning as a diode detector, a.v.c. tube and a.f. voltage amplifier. Then comes a pentode power audio amplifier stage having resistance - capacitance coupling. The supply connections for a battery-operated receiver are shown here, with all supply leads in heavier lines. The filaments of all tubes are connected together in parallel. All B+ leads can be traced to tube electrodes, then through the tubes to the filament circuit and to ground. All of the control grids in the r.f. section can be traced to the diode detector from which they receive a.v.c voltage. The power output tube grid can be traced to the

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\*Experienced servicemen seldom attempt to change a receiver over from one type of power supply to another, for many problems are encountered. The various circuits are presented here only to illustrate the principles involved.

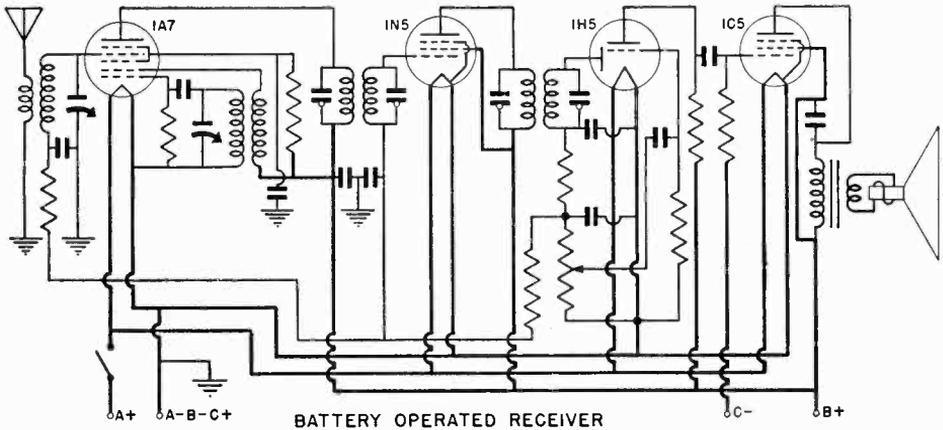


FIG. 20. A standard superheterodyne arranged to operate from batteries.

C— power supply lead. Either a magnetic or a permanent magnet type dynamic loudspeaker is used, so no d.c. power is required by the loudspeaker to produce a steady magnetic field.

► When this same signal circuit is adapted for a.c. operation, it will appear as shown in Fig. 21. Again the power supply circuit and all power supply leads are shown in heavier lines. Heater type tubes, designed to operate with a.c. filament voltages, have been

substituted for the low-drain battery type tubes. The power pack, which can be designed for any a.c. line voltage and frequency, uses a type 5Y3 full-wave rectifier tube. The filaments of all tubes are connected together in parallel and are completely isolated from the signal circuits. When electrode voltages lower than the power pack voltage are required, resistors are used in the power supply leads to drop the voltage.

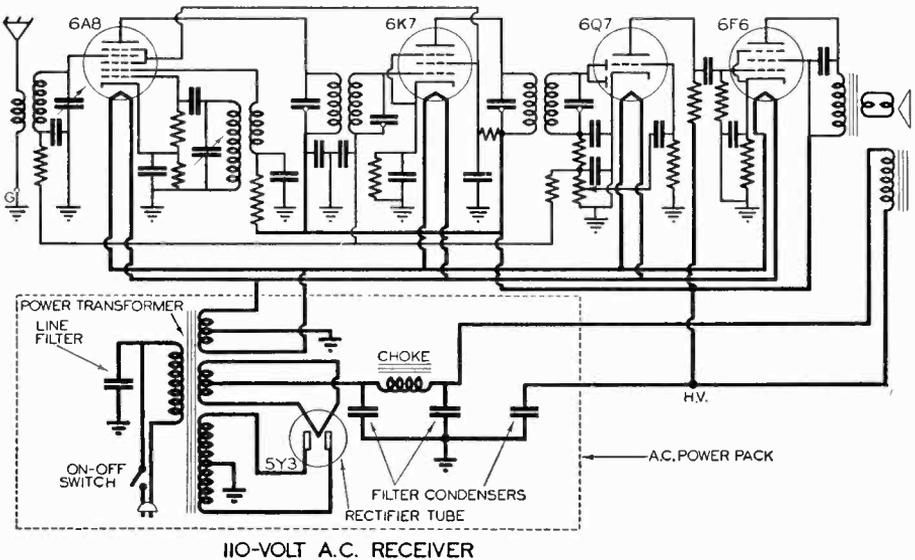


FIG. 21. The same basic circuit as in Fig. 20, arranged for operation from 110-volt a.c. power.

All of the *control grids* in this circuit can be traced to ground through resistors either directly or through the a.v.c. tube circuit. Other grid electrodes and all plate electrodes can be traced to the point marked H.V., which is the highest completely filtered d.c. voltage in the receiver. An electromagnetic type dynamic loudspeaker is used, with the field coil acting as a choke in the power supply filter.

► When this same signal circuit is used for an a.c.-d.c. universal receiver, the circuit takes on the form shown in

dentally grounded at a time when the chassis is connected to the "hot" side of the power line. No ground is used with this receiver. All other features of this universal receiver are similar to those for battery operation.

► The manner in which this basic receiver signal circuit is adapted for 6-volt operation, such as for automobile and farm radio receivers, is shown in Fig. 23. The filaments of all tubes are here connected in parallel directly to the 6-volt storage battery. A synchronous vibrator, also connected

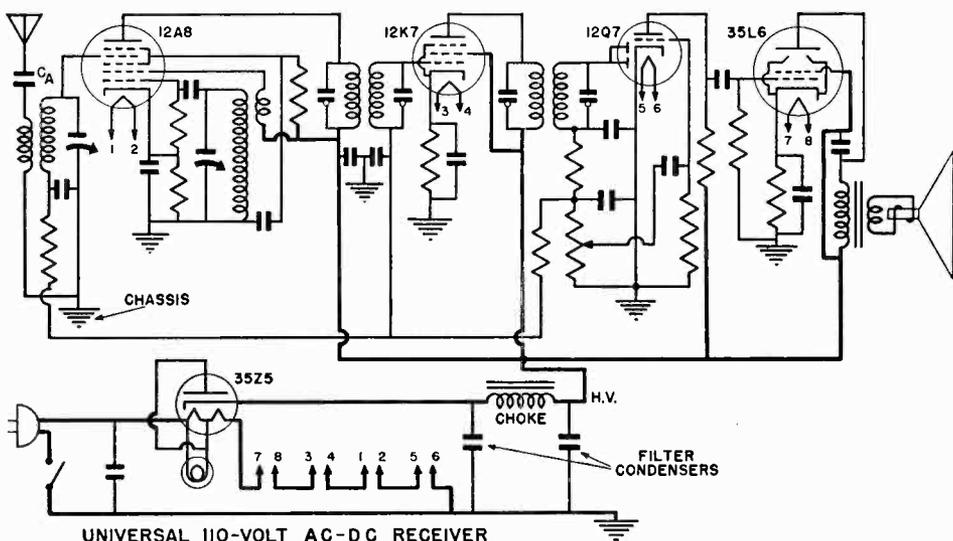


FIG. 22. The standard circuit of Fig. 20 arranged for universal operation from 110-volt a.c. or d.c. power lines.

Fig. 22. Here a diode rectifier tube is used, feeding into a filter which furnishes the d.c. electrode voltages. Supply leads are again shown in heavy lines; tube filaments are all connected together in series and are independently connected to the line voltage terminals.

Condenser  $C_A$ , in series with the antenna, allows r.f. current to flow but blocks d.c. and has a high reactance at power line frequencies. It prevents damaging the antenna coil primary in the event that the antenna is acci-

dentally grounded at a time when the chassis is connected to the "hot" side of the power line. No ground is used with this receiver. All other features of this universal receiver are similar to those for battery operation. The pulsating d.c. is filtered and then fed to the various tube electrodes except the control grids. All control grids can be traced to ground through the a.v.c. circuit. The chassis is the common return for all supply circuits in this set.

**Reading Circuit Diagrams.** The circuits in Figs. 20, 21, 22, and 23 show that with minor signal circuit changes, a standard radio receiver can

be adapted to any power supply. The analysis of these circuits has also made it clear that any circuit diagram can be traced or read in three distinct steps:

1. Trace the power pack and supply circuit leads through the signal circuits and tubes.

2. Trace the signal from the antenna to the loudspeaker (or image reconstructor in the case of a television receiver).

3. Trace any special control circuits, such as a.v.c. tone control, a.f.c., etc.

### ALL-WAVE AND HIGH-FIDELITY RECEIVERS

Along with the expansion of radio broadcasting, other radio services on land, at sea and in the air grew to unusually large proportions. Police, aviation, commercial and amateur radio transmitters began crowding the frequency channels above the broadcast band frequencies; furthermore, broadcast stations in many foreign countries were sending out programs on these high frequencies, primarily to reach far distant colonies.

Radio enthusiasts soon learned that many interesting messages and programs could be picked up on the high frequencies (short waves.) About 1930, receiver manufacturers took recognition of the growing public interest in short-wave reception and began the production of short-wave converters. These converters consisted of a mixer-first detector stage which, when connected to an ordinary t.r.f. receiver, used that receiver as the intermediate frequency amplifier, as second detector, a.f. amplifier and loudspeaker. A short-wave converter thus changed the broadcast band t.r.f. receiver into a short-wave superheterodyne receiver.

The addition of a separate unit to a receiver has always been followed by

a combination of that unit with the receiver; when the superheterodyne circuit became firmly established as the basic receiver circuit, this combination of broadcast and short-wave receiver units took place. With a superheterodyne, all that was required to make the change was a means of changing the preselector and oscillator coils for each frequency range desired. The i.f. value of all-wave receivers gradually rose from a low value of about 175 kc. to about 460 kc. in order to get away from the interfering signals which are inherent in a beat frequency system.



Courtesy General Electric

A modern phono-radio combination receiver.

► The radio listening public has always been divided into two distinct groups, one containing those enthusiasts who prefer DX (long-distance) reception which is reliable and free from interference, and the other group preferring local reception only, but of the highest possible fidelity of reproduction. To be sure, the first group also desired a certain amount of fidelity, and consequently manufacturers had to make some compromise. While design engineers strove to improve selectivity and sensitivity, they attempted at the same time to bring up the fidel-

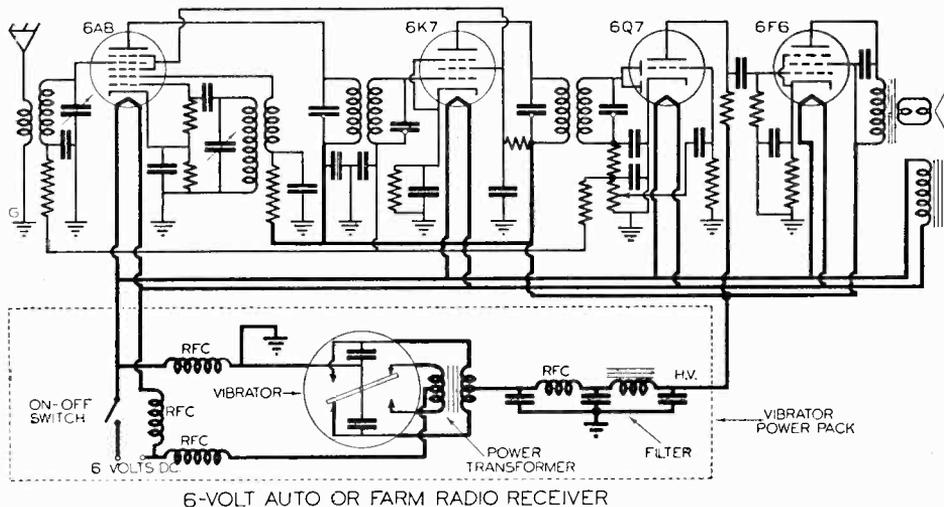


FIG. 23. The standard superheterodyne of Fig. 20 arranged to operate from a 6-volt storage battery.

ity as much as possible and eliminate interference troubles.

The demand for higher and higher fidelity continued, resulting in band-pass r.f. systems and flat-response a.f. amplifiers feeding into a high-fidelity loudspeaker with specially designed chambers or baffles. The problems of improving fidelity are by no means simple, for difficulties with voice, hum and volume range increase rapidly as fidelity is improved. Nevertheless, receivers are today on the market which will faithfully reproduce anything which can be put on the air by modern broadcast transmitters.

► Even though fidelity is all-important with certain listeners, they will not accept receivers which lack certain other desirable features. Therefore, the high-quality receiver of today incorporates a great many outstanding features, such as: high fidelity in the broadcast band; reception on all short-wave bands on which programs are regularly scheduled; compensated volume control; bass and treble control; automatic tuning of the push-

button or telephone dial type, and tuning indicators.

## TELEVISION

The radio industry, having gone through the code and sound signal development stage, now takes on the transmission of visual signals as well. The methods of modulating, broadcasting, and receiving visual signals do not differ in principle from the methods used for code and sound signals; only the ranges of modulation and carrier frequencies differ.

It is only natural that the well understood and thoroughly tested principles of the superheterodyne circuit should be extended for television service. Because of the high picture frequencies involved, ultra-high-frequency carriers are used. The preselector circuit in a television receiver must therefore tune to ultra-high-frequency carriers; the i.f. amplifier must handle a wide range of side frequencies; the detector circuit must have no frequency distortion, even at the highest pic-

ture frequencies, and the picture amplifier (if used) must likewise have no distortion. Special sweep circuits are needed to control the electron beam in the cathode ray image reconstructor tube.

Thus you can see how the amplitude-modulated radio receiver has passed through various stages of development, starting with the coherer, and moving in rapid succession to the basic circuits of crystal detectors, regenerative vacuum tube detectors, tuned radio frequency amplifiers, and finally to the superheterodyne circuit, which today is king of them all.

### F.M. RECEIVERS

Frequency modulation broadcasting is the culmination of years of research and development work by Major E. H. Armstrong and other radio engineers seeking to provide a broadcasting service which is free from interfering noise, free from station interference, and has true high fidelity. These three basic listener requirements are now met more completely than ever before. With f.m., radio reception acquires a breath-taking naturalness, and audio

frequencies as high as 15,000 cycles are reproduced faithfully.

F.M. receivers can be divided into three groups: 1, those designed only for f.m. reception; 2, those providing both a.m. and f.m. reception; 3, f.m. converters, which connect to the a.f. input terminals of an a.m. receiver.

Some sections of f.m. receivers employ entirely new circuits, while others simply use basic circuits which are suitably designed to handle all the frequencies encountered in an f.m. system. The a.f. sections and loudspeakers, however, can be the same in both f.m. and a.m. receivers because they handle the same types of signals. An f.m. receiver will usually be designed to give high fidelity, so as to take full advantage of the higher-quality signals put on the air by f.m. transmitters.

**Looking Ahead.** Modern receivers of the superheterodyne type, television and f.m. receivers are just mentioned here. You are going to study them in detail in later lessons. Also, receiver refinements, such as a.v.c., tone controls, a.f.c., push-button systems, etc., are all to be thoroughly covered in future lessons.

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THE N. R. I. COURSE PREPARES YOU TO BECOME A <b>RADIOTRICIAN &amp; TELETRICIAN</b> <small>(REGISTERED U. S. PATENT OFFICE)</small>	<small>(REGISTERED U. S. PATENT OFFICE)</small>
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# Lesson Questions

**Be sure to number your Answer Sheet 18FR-2.**

**Place your Student Number on every Answer Sheet.**

**Send in your set of answers for this lesson immediately after you finish them, as instructed in the Study Schedule. This will give you the greatest possible benefit from our speedy personal grading service.**

1. What seven important technical requirements, demanded by the public, control the circuit chosen for any particular receiver?
2. Who predicted mathematically that an oscillating current will radiate electromagnetic waves which are identical with light waves except for frequency?
3. Why were loading coils sometimes inserted in the tuning circuits of early receivers?
4. Why is the super-regenerative receiver unsatisfactory for use in congested bands?
5. Why were grid suppressors used in some of the earlier t.r.f. receivers?
6. What objectionable effect compelled set designers to drop the idea of un-tuned antenna systems in single-dial receivers?
7. In what radio apparatus was the Raytheon BH rectifier first used?
8. In addition to an antenna, what five separate items did a person have to select in the early part of 1928 in order to have a complete radio receiver installation?
9. In what year were a.c.-operated receivers generally introduced?
10. Name the three distinct steps involved in tracing or reading a circuit diagram.

## THE FABLE OF DISCOURAGEMENT

Once upon a time, so the story goes, the devil held a sale. To any one who would pay the price, he offered the tools of his trade. On a table, each with its price label, were *hatred, despair, sickness, jealousy, greed*, and all the other causes of unhappiness.

Off to one side, however, lay a harmless-looking wedge-shaped instrument marked *discouragement*. It was old and worn, but priced the highest of them all. When asked why the price was so high, the devil replied: "*Because this tool is one I can use so easily. Nobody knows that it belongs to me, so with it I can open doors that are immune to all other tools. And once inside, I can finish the job with almost any of the other tools!*"

Few people know how small is the margin between failure and success. Frequently, the separation is just the width of that one word—discouragement.

You can combat discouragement by cultivating confidence in yourself. Whatever you may desire of life—whatever your goal may be—you have only to work for it wholeheartedly, confidently, with that one goal always in mind, *and you will reach it.*

J. E. SMITH

**MANUAL AND AUTOMATIC  
VOLUME CONTROLS**

19FR-2

**NATIONAL RADIO INSTITUTE**

ESTABLISHED 1914

**WASHINGTON, D. C.**



# STUDY SCHEDULE No. 19

For each study step, read the assigned pages first at your usual speed. Reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind, then answer the Lesson Questions for that step. Study each other step in this same way.

1. Why Volume Controls Are Needed . . . . . Pages 1-6

The purpose of a volume control in a receiver, and the difference between automatic and manual volume controls, is explained here. Then, there is a discussion of R.F. and A.F. voltage control systems. Answer Lesson Questions 1, 2, and 3.

2. R.F. Gain Controls . . . . . Pages 6-11

When automatic volume control systems are not used, an R.F. control is necessary. There are many types of gain controls—some vary the load while others change operating voltages. Of these, the C bias and the antenna-C bias controls are the most popular. Answer Lesson Question 4.

3. Automatic Volume Controls . . . . . Pages 11-15

This is an explanation of the purpose and function of A.V.C. systems, how the A.V.C. voltage is filtered, and the importance of the time constant values of the filter. Answer Lesson Questions 5, 6, and 7.

4. Typical A.V.C. Circuits . . . . . Pages 16-24

Here are descriptions of typical A.V.C. circuits. There are many types; some use separate tubes, and some have tapped voltage dividers. Answer Lesson Question 8.

5. Construction of Manual Volume Controls . . . . . Pages 25-28

Few volume controls are made so that the resistance variation is linear. Most of them have either a left-hand or a right-hand taper. The proper taper is very important—sometimes more so than the resistance value itself. Study this practical section with great care. Answer Lesson Questions 9 and 10.

6. Mail Your Answers for this Lesson to N.R.I. for Grading.

7. Start Studying the Next Lesson.

# MANUAL AND AUTOMATIC VOLUME CONTROLS

## Why Volume Controls Are Needed

THE most important control in a radio receiver is unquestionably that which allows the listener to choose one desired station from the many which may be on the air at any one time. Next in importance is the *volume control*, which directly or indirectly controls the acoustical level or volume of the loudspeaker sound output. Volume controls are of two general types: 1, *automatic volume controls*; 2, *manual volume controls*. All radio receivers require manual volume controls, *even when equipped with automatic volume control*.

**Automatic Volume Control.** Automatic volume control (commonly abbreviated A.V.C.) keeps the output volume level of a radio receiver essentially constant despite variations in R.F. signal input voltage. These variations may occur while the radio receiver is tuned to a single station (fading) or while the receiver is being tuned from station to station.

The R.F. input signal voltages which modern receivers are called upon to handle may vary from 1 microvolt for low-power or distant stations to 1 volt for high-power local stations, and thus the strongest signal handled is a million times greater than the weakest signal. If the gain of the R.F. amplifier were *fixed* at a value sufficient for reproduction of the weakest signals, there would be *blasting* (excessive volume) when the set was tuned to a strong signal. A.V.C. counteracts this blasting by *raising the gain* of the R.F. amplifier for weak signals, and *lowering the gain* automatically and almost instantly for strong signals.

Blasting is particularly annoying

when the listener is tuning rapidly from station to station while searching for a desired program. Furthermore, blasting may be accompanied by *overloading* of the loudspeaker and of one or more stages in the receiver, with resulting distortion. A.V.C. prevents this overloading by reducing the gain of the R.F. amplifier for strong signals.

Programs received from distant and semi-distant stations, where the radio waves are totally or partially reflected from the sky before reaching the receiving antenna, are subject to *fading*. In other words, the strength of the signal at the receiving location varies considerably from minute to minute. A.V.C. can compensate for the effects of fading by raising and lowering the gain of the R.F. amplifier just enough to maintain the desired acoustical output, provided the signal does not become excessively weak. Automatic volume control is thus desirable in a radio receiver for these three important reasons: 1, *it prevents blasting*; 2, *it prevents overloading*; 3, *it minimizes fading*.

**Manual Volume Control.** A manual volume control is essential in a radio receiver for a number of reasons. Low volume may be desired when a musical program is used as atmospheric background while eating, reading, playing games, or carrying on conversation. Increased volume may be desired for news broadcasts, talks by important persons, radio plays, popular *all-star* broadcasts, symphonic music, or music for dancing; furthermore, some people can hear better than others, and the volume must be adjusted accordingly.

But what, exactly, is a volume con-

trol? To the average non-technical person, any control which permits him to change the volume of the sound coming out of the loudspeaker is a volume control. Radio men know,

rent swing over non-linear regions of the  $E_g-I_p$  characteristic of the tube. Because of this distortion, manual volume controls in receivers not having A.V.C. must be R.F. gain controls;

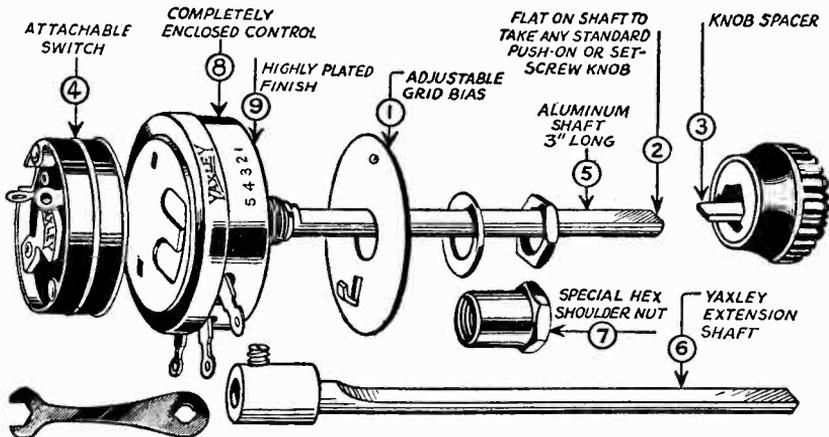


Fig. 1. Typical manual volume control with accessories (a unit made by the Yaxley Manufacturing Division of P. R. Mallory & Co., Inc.). A feature of this control is the adjustable grid bias plate (1) which permits adjusting the minimum resistance to 100, 200, 300, 400 or 500 ohms for fixed C bias purposes. Numbers on housing and indicating mark on plate tell which setting is used. With this plate removed, the control can be turned to zero resistance. The 3" shaft can be cut to any desired length with a file or hacksaw, or can be lengthened when necessary by using an extension shaft (6). A small metal spacer (3) is supplied with Yaxley controls for use with knobs designed for shafts having only 1/32" deep flats. The back cover plate of the control housing is easily removed to permit attachment of a power switch (4). The special nut (7) is used for mounting the controls on panels up to 3/4" thick. A small wrench like that shown is recommended for tightening this nut or the regular flat nut shown on the shaft, as pliers may wear off the corners of these nuts or scratch the panel.

however, that volume controls are either voltage controls or gain (amplification or sensitivity) controls. In receivers which have automatic volume control (commonly abbreviated as A.V.C.), the manual volume control is always an A.F. voltage control; in all other receivers the manual volume control is always an R.F. circuit control. (Regardless of whether the control is an R.F. or an A.F. type, the physical appearance will be almost the same. A typical control is shown in Fig. 1.)

There is a definite limit to the amount of signal voltage which can be handled by each stage in a radio receiver. Excessively high input voltage to a stage results in distortion, because high input signals make the plate cur-

they can then be adjusted to prevent overloading of *any* receiver stage.

## TYPES OF MANUAL VOLUME CONTROLS

The manual volume controls found in radio receivers are of three types, which may be used either singly or in combination: 1, *R.F. voltage controls*, which reduce the voltage of the R.F. input signal either before it reaches the first R.F. amplifier stage or before it enters an R.F. stage which may overload; 2, *A.F. voltage controls*, which reduce the A.F. voltage output of the demodulator stage or reduce the A.F. voltage at some point in the A.F. amplifier; 3, *R.F. gain controls*, which reduce the gain (amplification) of an

R. F. amplifier stage either by varying the load on that stage or by varying the mutual conductance of the tube in that stage. We will now consider the advantages and disadvantages of each of these forms of manual volume controls as applied to practical radio receiving circuits.

### R.F. VOLTAGE CONTROLS

R.F. voltage controls can either vary the R.F. voltage which is fed from the antenna to the input of the R.F. amplifier or can vary the R.F. voltage which is fed from one R.F.

movable contact arm connected to a third terminal (labeled *C*); the significance of these *R*, *L* and *C* (right, left and center) designations for these volume control potentiometers will be explained to you later. The value of the voltage fed to the following R.F. circuit depends upon the position of the movable arm of the potentiometer.

With the circuit of Fig. 2A, the fixed terminals of the potentiometer were connected to antenna and ground, and the variable tap on the potentiometer was connected directly to the grid of the first R.F. amplifier tube. The potentiometer thus con-

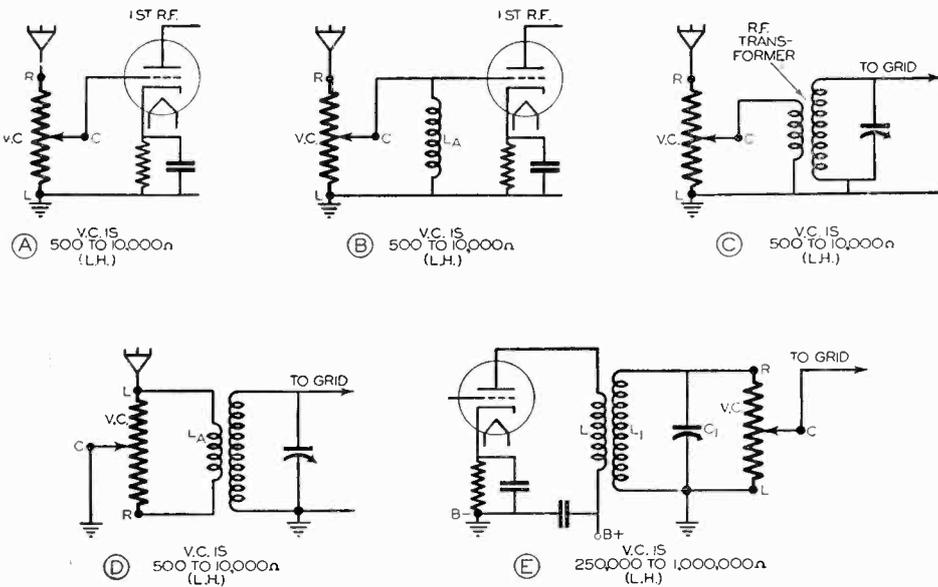


Fig. 2. Typical manual volume control arrangements which vary the amount of R.F. voltage transferred from one circuit to another. The resistance values indicated for V.C. are maximum values; the minimum resistance in each case is zero. The designations *R*, *L*, *C*, *R.H.* and *L.H.* are explained later in this text-book.

amplifier stage to another. These controls were widely used in older receiver circuits for the purpose of controlling volume, the antenna circuit control having been the more popular.

Five typical R.F. voltage control circuits are shown in Fig. 2. As you can see, the control device in each case was a potentiometer with two fixed terminals (labeled *R* and *L*), and a

controlled the R.F. voltage which was applied to the grid of this tube. This circuit had no selective or signal-boosting properties; therefore, only the characteristics of the antenna determined which signals would be favored. The antennas used at the time were generally of such a length that they were in resonance at the higher frequencies in the broadcast band,

and consequently these higher frequencies were favored.

To make the simple antenna circuit volume control of Fig. 2A respond equally well to low and high frequencies, inductance  $L_A$  was connected between grid and cathode of the first R.F. tube in the manner shown in Fig. 2B. The inductance and distributed capacity of coil  $L_A$  acted together to form a parallel resonant circuit whose resonant frequency was in the lower region of the broadcast band. This resonant circuit acted as a high impedance to low frequency signals, developing maximum R.F. voltage for transfer to the grid of the tube, and acted as a capacitive reactance path to ground (a shunting capacity) for higher frequency signals. By choosing the proper ratio of inductance to capacity for coil  $L_A$ , it was possible to make the antenna circuit respond uniformly to both low and high frequencies in the broadcast range.

The volume controls shown in Figs. 2A and 2B worked well in receivers located at a distance from powerful broadcast stations, but in the vicinity of a powerful station the capacity path between the leads and parts of the volume control was sufficient at radio frequencies to transfer the signal of the local station to the grid of the first tube at all volume control settings. When the volume control was turned up to receive a weak signal, the signal of the local station overloaded the first R.F. amplifier tube, making this tube act as a detector which modulated the weak signal with the program of the strong signal; cross-modulation resulted, with both the desired and the undesired programs being reproduced by the loudspeaker.

The next move of the radio receiver designer was to provide selectivity for the antenna circuit in order to coun-

teract cross-modulation effects. One of these selective circuits is shown in Fig. 2C. As you can see, the voltage output of the volume control was fed into the primary of an R.F. transformer, and the resulting R.F. current induced a stronger R.F. voltage in the tuned secondary of this transformer for transfer to the grid of the first tube. The chief drawback of this circuit was the fact that whatever resistance existed between terminals  $C$  and  $L$  of the potentiometer was reflected into the resonant circuit as a resistance; this made selectivity very poor at low volume control settings.

The modified potentiometer connection shown in Fig. 2D gave improved selectivity. Note that the variable tap of the potentiometer was grounded and that the entire potentiometer resistance was connected across the primary of the R.F. transformer. This resulted in a constant and high resistance being reflected into the resonant circuit, keeping the selectivity essentially constant. Antenna and ground connections were made to points  $L$  and  $C$  on the potentiometer. As the resistance between these two points was increased, more of the antenna current flowed through coil  $L_A$  and therefore a higher R.F. voltage was induced in the resonant circuit.

The usual form of R.F. voltage control used between two R.F. stages was that shown in Fig. 2E. Here the voltage developed across series resonant circuit  $L_1-C_1$  was fed to the two fixed terminals of the volume control potentiometer, and the variable tap of the potentiometer was connected directly to the grid of the following R.F. amplifier tube. The potentiometer resistance was generally between .25 megohm and 1 megohm, so the load placed upon the resonant circuit by the volume control was negligible. One disadvantage of this circuit was the fact that body capacity, such as

placing the hand near the volume control, tended to detune the circuit; this made it necessary to use a long shaft of insulating material between the potentiometer and its control knob. Furthermore, this volume control and the antenna circuit R.F. voltage con-

(acting as load).

When an audio transformer is used at the input, a common form of A.F. voltage control is that shown in Fig. 3A. Here audio transformer  $T$  couples the A.F. signal source to volume control potentiometer  $V.C.$  In any

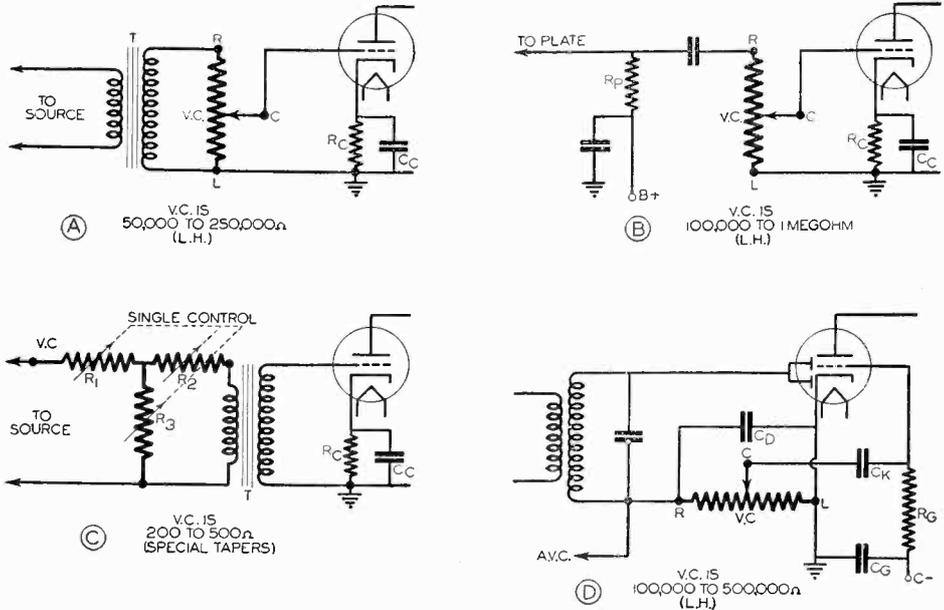


Fig. 3. Typical manual volume control arrangements which vary the amount of A.F. voltage transferred from a source of A.F. voltage to a following circuit.

controls were quite noisy, for they controlled low R.F. voltages in high-gain circuits.

## A.F. VOLTAGE CONTROLS

Several different ways of connecting manual volume controls into audio frequency amplifier circuits are shown in Fig. 3. Controls such as these are essential in the audio amplifiers of public address systems, in intercommunicating systems, in electric phonographs and in all radio receivers which employ A.V.C. in the R.F. amplifier. Basically, an A.F. voltage control varies the amount of audio frequency voltage which is fed from one circuit (acting as source) to another

circuit where the signal source has a high impedance, approaching that of the volume control, the transformer may be omitted and the source connected directly to terminals  $R$  and  $L$  of the potentiometer. Any desired portion of the voltage applied across the potentiometer is fed to the grid and cathode terminals of the following A.F. amplifier tube.

With resistance-capacitance coupled audio amplifiers, the arrangement of the A.F. voltage control is as shown in Fig. 3B.

When inserting a volume control in the input of a circuit which is acting as a load for an A.F. signal source, the volume control must not alter the impedance match between the source

and load. In cases like this, special constant-impedance volume controls or pads are used; these consist of two or more variable resistors which are adjusted simultaneously. The circuit in Fig. 3C is an example; here a so-called T-pad volume control having three variable resistors (this name results from the fact that the resistors are arranged to form the letter "T") is so connected that the resistance across the input terminals and the output terminals of the T-pad remains constant (at 200 ohms for a 200-ohm T-pad or at 500 ohms for a 500-ohm T-pad) regardless of the setting of the control knob. Only the A.F. voltage across the output terminals of the T-pad varies with the control knob setting. This subject is covered in greater detail in advanced lessons; for the present it is sufficient for you to know

what this type of A.F. voltage control is and why it is used.

A type of A.F. voltage control which you will constantly encounter in radio receivers using diode detectors and A.V.C. is shown in Fig. 3D. Here the volume control serves as load for the diode demodulator or second detector. The R.F. signal is by-passed around this volume control by condenser  $C_D$ ; this condenser also serves to smooth out the rectified R.F. voltage, producing the desired A.F. signal across the volume control. The A.F. voltage between points  $L$  and  $C$  is fed to the grid of the first A.F. amplifier tube (in the same envelope as the diode detector) through D.C. blocking condenser  $C_K$ . Other types of A.V.C. circuits in which manual volume controls are incorporated will be considered later in this book.

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## R. F. Gain Controls

**Volume Controls Which Vary the Load in an R.F. Circuit.** In the earliest battery-operated receivers, volume was controlled simply by varying the filament current of one or more tubes in the receiver. With the advent of A.C.-operated receivers, volume control became a more difficult problem, for varying the filament currents in heater type tubes did not give a satisfactory control over volume.

One of the first manual volume control schemes used in A.C. receivers was that shown in Fig. 4, which involved placing a variable resistor across the plate load resistance of an R.F. stage. Without this variable resistor the tuned secondary R.F. transformer made up of  $L_1$ ,  $L_2$  and  $C_2$  offered a definite load in the plate circuit of tube  $VT_1$ , and the stage consequently had a definite amplification or gain. Shunting a volume control across  $L_1$

permitted reduction of the plate load impedance and the gain to any desired lower value. Because of the fact that decreasing the load impedance increased the curvature of the  $E_g-I_p$  curve, making strong R.F. signals swing over non-linear portions of the curve, this scheme could be used only in the plate circuit of an R.F. stage which handled low R.F. voltages. This method of controlling volume also served as a control on regeneration, for it varied the A.C. plate voltage and thus varied the feed-back voltage.

**Volume Controls Which Vary the Screen Grid Voltage in an R.F. Stage.** The amplification of a vacuum tube amplifier can be controlled to a considerable extent by varying the D.C. voltages applied to the electrodes of the vacuum tube used in the stage. Any change in electrode operating voltages which affects the mutual con-

ductance of the tube will also affect the amplification provided by the stage. This is true because, as you will recall, mutual conductance is equal to A.C. plate current divided by A.C. grid voltage. Reducing the A.C. plate current therefore reduces the mutual

impulses created by a varying contact resistance at *C*.

**Volume Controls Which Vary the Plate Voltage of an R.F. Amplifier Tube.** A method used in early battery receivers for varying the plate voltage of a tube as a means of controlling volume was that shown in Fig. 5*B*, where a variable resistor was placed in series with the plate supply voltage and the R.F. plate load. This method was advantageous in that it did not place any unnecessary drain upon the B batteries, but had the disadvantage that a very high-resistance rheostat, which burned out easily, was required.

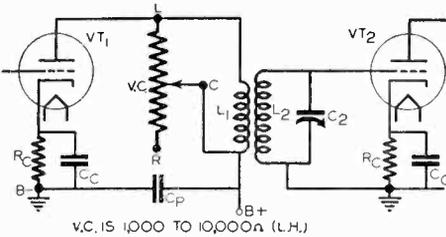


Fig. 4. This volume control varies the impedance of the load in an R.F. circuit and thus varies the gain of the R.F. amplifier.

conductance and, at the same time, reduces the A.C. voltage which is developed across the plate load for transfer to the following stage.

As a general rule, reducing the D.C. plate voltage, reducing the D.C. screen grid voltage or increasing the negative C bias voltage will reduce the mutual conductance of a tube. Making the suppressor grid negative with respect to the cathode and increasing its negative voltage will also decrease the mutual conductance. It is not customary, however, to vary the suppressor grid voltage for the reason that very large changes are required in voltage in order to secure any appreciable reduction in mutual conductance.

The usual method for controlling the screen grid voltage was that shown in Fig. 5*A*; the volume control potentiometer was made part of a voltage divider connected across the plate supply source, with resistor *R<sub>L</sub>* serving to determine the maximum screen grid voltage. Condenser *C<sub>s</sub>*, the screen bypass condenser, also helped to prevent noisy volume control action, for it shunted the *L-C* section of *V.C.* and thus by-passed to ground any current

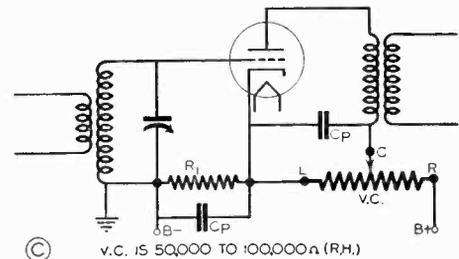
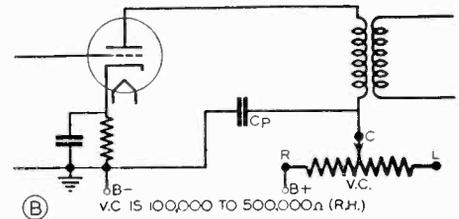
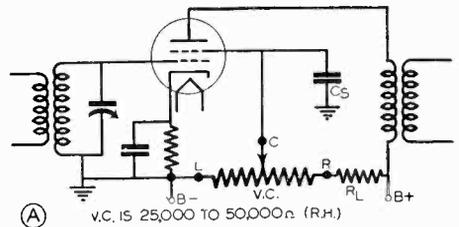


Fig. 5. Typical manual volume control arrangements which vary the mutual conductance of an R.F. amplifier tube by varying the screen grid voltage (A) or the plate voltage (B and C).

Plate voltage can also be controlled by using a potentiometer connected across the plate voltage supply in the manner shown in Fig. 5C. With this potentiometer across the B battery at all times, there was naturally a higher current drain than in the case of Fig. 5B, but the potentiometer was lower in resistance, making possible a more rugged design and consequently reducing the tendency to burn out.

Unfortunately, these methods of controlling volume by varying screen grid and plate voltage did not prove entirely satisfactory. Reducing these electrode voltages any reasonable amount affected the dynamic  $E_g-I_p$  characteristic of the tube enough to cause distortion at low volume control settings. These circuits were quickly replaced by the still popular C bias method of volume control, which will now be considered.

**Volume Controls Which Vary the C Bias Voltage of an R.F. Amplifier Tube.** The most effective way of varying the mutual conductance of a tube in order to control volume is by varying the control grid voltage (the C bias voltage). This method of controlling volume proved so satisfactory and came into such widespread use that tube designers developed special remote cut-off tubes, first the *variable mu screen grid tube* and later the *super-control pentode tube*, which gave even better results. These tubes allowed the A.C. grid voltage to swing over an essentially linear portion of the dynamic  $E_g-I_p$  characteristic curve at all volume control settings. A number of typical C bias volume control circuits are shown in Fig. 6; although triode tubes are indicated for simplicity, the circuits also apply to screen grid and pentode tubes.

A scheme widely used in early battery receivers for varying the C bias voltage was that shown in Fig. 6A, where the negative C bias voltage was

increased and the filament current was reduced simultaneously (by increasing the resistance of volume control V.C.) in order to reduce the mutual conductance of the tube. When the filaments of several tubes are in parallel and the grid return lead for each tube circuit is grounded, a volume control such as this will affect all such parallel-fed circuits.

Another C bias voltage control for battery receivers, used when filament voltage was fixed in value, is that shown in Fig. 6B. Here potentiometer V.C. and resistor  $R_1$  were placed in series across a C battery which was connected with the indicated polarity. The value of  $R_1$  must be low enough to develop the minimum C bias required for the circuit. (A high-resistance load on the C battery will make it last longer than the B batteries; with normal C bias and low B voltages, certain tubes will distort severely. For this reason, values for V.C. and  $R_1$  were chosen which were low enough to make the B and C batteries run down at approximately the same rate.) Switch SW opened both the A and C battery supply circuits in order to prevent current drain when the receiver was not in use.

The basic C bias voltage volume control for heater type tubes is that shown in Fig. 6C. The volume control is placed in series with the usual cathode bias resistor  $R_C$ , with condenser  $C_C$  connected between the cathode and the movable tap of the volume control. Increasing the resistance setting of the volume control increases the C bias voltage and therefore reduces volume; resistor  $R_C$  determines the minimum C bias voltage. The chief disadvantage of this circuit is that as the negative C bias voltage is increased, the plate current which flows through the cathode resistor combination is greatly reduced, counteracting the effect of the volume control. For example, if a cathode bias resistance

of 600 ohms gives a 3 volt negative bias, it may be necessary to increase this resistance to as much as 25,000 ohms in order to secure three times as much bias voltage ( $-9$  volts).

Where the movable tap of the volume control is grounded, as in Fig. 6C, receiver manufacturers will often use a potentiometer in which the movable tap or middle terminal *C* is grounded internally to the metal shaft of the

made essentially independent of the plate current. The volume control now has greater control over the *C* bias voltage and can be considerably lower in value than in the case of Fig. 6C.

In some receivers the volume control is actually a part of the power pack voltage divider system or is an additional voltage divider across a part of the main voltage supply in the manner shown in Fig. 6E. Here volume

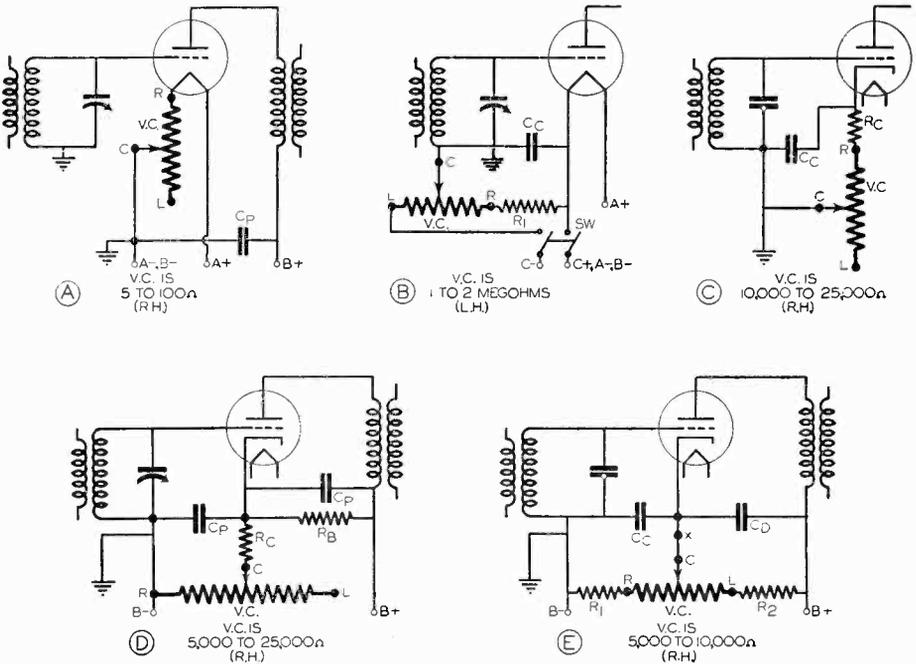


Fig. 6. Typical manual volume control arrangements which vary the mutual conductance of an R.F. amplifier tube (thereby varying the gain of the R.F. amplifier) by varying the *C* bias voltage applied to the control grids of one or more R.F. tubes.

control. This eliminates the need for making a soldered connection to the middle terminal, for mounting the potentiometer on the metal chassis automatically grounds the housing.

An improved *C* bias volume control arrangement is shown in Fig. 6D. By having between the plate voltage lead and cathode a bleeder resistor  $R_B$  which forces or "bleeds" a current through the *C* bias resistor  $R_C$  at all times, the volume control current is

control potentiometer *V.C.* varies the voltage between the cathode and the *B*- terminal, with resistor  $R_1$  determining the minimum *C* bias voltage and resistor  $R_2$  determining the maximum *C* bias voltage. The total resistance of  $R_1$ , *V.C.* and  $R_2$  in series across the power pack determines the value of bleeder current which flows at all times. Both the plate current of the tube and the bleeder current flow through  $R_1$  and the *R-C* section of the

volume control, producing a C bias voltage which is dependent upon the position of movable tap *C*.

**Volume Controls Which Vary Both the Antenna Input and C Bias Voltages.** We cannot pass over C bias

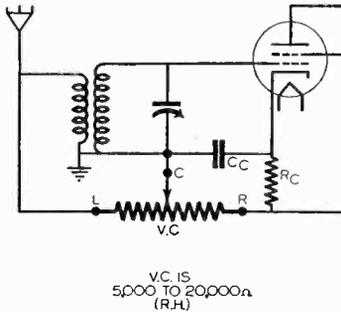


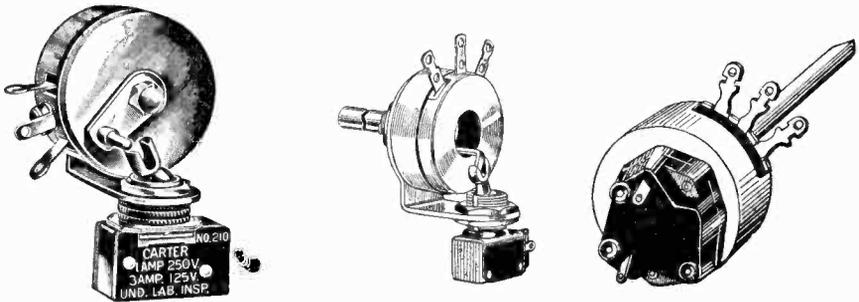
Fig. 7. A manual volume control arrangement which varies both the antenna input R.F. voltage and the C bias voltage of the first R.F. stage in order to change the R.F. amplifier gain.

voltage volume controls without pointing out that with the earlier triode and screen grid tubes, increasing the negative C bias beyond a certain point changed an R.F. amplifier stage into a

two types of volume control circuits in solving this problem.

One typical combination volume control circuit is shown in Fig. 7. Here fixed resistor  $R_C$  controls the minimum C bias voltage and hence the maximum amplification of the stage. That section of the volume control potentiometer between points *C* and *R* also serves as cathode bias resistor, while the remaining section shunts the input transformer primary winding. Moving contact *C* away from point *R* increases the C bias voltage and at the same time reduces the resistance across the primary winding, thus giving a double reduction in R.F. signal gain.

**Variations of Manual Volume Control Circuits.** You are now familiar with the important basic manual volume control circuits. Of course you will encounter many variations of these circuits in radio receivers, but as a rule these variations will be easily recognized as one of the types which you have studied.



Today it has become an almost universal practice among receiver designers to make the manual volume control or some other manual control serve also as an on-off switch. Earlier forms of dual potentiometer-switch arrangements appear at the left (a Carter unit) and in the center (a Yaxley unit); in both of these, rotation of the control knob to the extreme counter-clockwise position caused an arm on the control shaft to flip the toggle switch to its off position. At the right is a modern universal replacement volume control (a Philco unit) with a special power switch attached to its cover plate for this same purpose.

detector, with resulting modulation distortion and cross-modulation on strong signals. With variable  $\mu$  screen grid tubes and super-control pentode tubes this is naturally no longer a problem, but it will be of interest to see just how receiver manufacturers combined

Before beginning our study of automatic volume control circuits, we want to state definitely that with A.V.C.-controlled receivers, the manual volume control must be *in the audio system*, except in those cases where the manual control is of a type which

changes the effectiveness of the A.V.C. control. You can readily understand that any attempt to reduce the R.F. output of the R.F. amplifier manually

would simply cause the A.V.C. system to increase the amplification and counteract the change in the manual volume control setting.

## Automatic Volume Controls

A.V.C. action is basically simple. When the carrier level of the R.F. input signal is excessively high, A.V.C. lowers the gain of the R.F. amplifier by increasing the negative C bias voltage on one or more R.F. amplifier tubes. This is accomplished automatically by rectifying the R.F. carrier signal, then filtering out all but the resultant D.C. voltage, whose value is always proportional to the R.F. carrier level. This D.C. voltage is so applied to the grids of the R.F. amplifier tubes that *increases* in carrier level make the grids *more negative*. The R.F. amplifier gain is thus reduced enough to keep the R.F. amplifier output level essentially constant and prevent overloading of any tubes. Likewise, *reductions* in input carrier level result in *less negative* C bias voltages and greater amplification.

It is clearly impossible for A.V.C. to maintain the R.F. carrier level perfectly constant at the output of the R.F. amplifier, for it is the change in this level which produces the change in negative C bias voltage required for automatic volume control. With proper design, however, A.V.C. can keep the carrier level constant enough for all practical purposes. Exact control of carrier level is not required, for the output voltage of a receiver can be increased or decreased about 40 per cent before the change can even be detected by the human ear. When working with A.V.C.-controlled circuits, then, never depend upon your ears as a judge of performance; always use an output meter or some other type of indicator.

The action of a receiver having A.V.C. is best represented by overload curves like those in Fig. 8, which are obtained by plotting the R.F. input voltage of the receiver against the A.F. output voltage of the demodulator or second detector. Curve 1 in Fig. 8 represents the overload characteristics of a receiver not having A.V.C. Notice that overloading takes place at an R.F. input voltage of about 100 microvolts

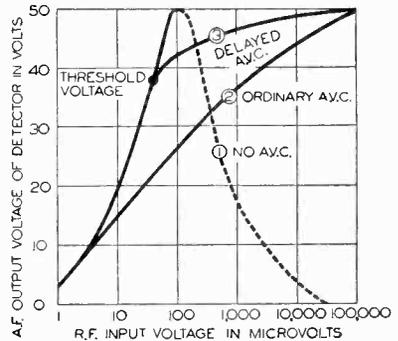


Fig. 8. Overload characteristic curves for three different superheterodyne radio receivers, obtained by feeding various R.F. input voltages into each receiver (the manual volume control being set for maximum volume) and measuring the A.F. output voltage of the second detector. Curve 1 is for a sensitive receiver which does not have A.V.C. Curve 2 is for the same receiver with ordinary A.V.C. added. Curve 3 is for the same receiver with delayed A.V.C. added. These curves illustrate receiver performance in general, and will naturally vary greatly with different receivers.

in this particular example. This is shown by the output falling off when the input is increased further. When this receiver is equipped with A.V.C., its overload characteristic is represented by curve 2. You can readily see

that with A.V.C., the receiver will handle all carrier signal levels below 100,000 microvolts (.1 volt) without overloading. On the other hand, however, this A.V.C. curve shows that the sensitivity of the receiver will be considerably lower with A.V.C. than without it for medium-strength signals. (R.F. input voltages in the region between about 10 and 100 microvolts do not deliver as much output.) Let us see how this drawback of A.V.C. is overcome in some receivers.

**Delayed A.V.C.** Since ordinary A.V.C. circuits prevent the maximum amplification of the receiver from being effective at low carrier levels, and since A.V.C. action is not particularly required at these low carrier levels since they cannot possibly cause overloading, the receiver designer simply arranges the A.V.C. system so it is inactive until the carrier level reaches a definite and fairly high value. This arrangement, which utilizes the full sensitivity of the receiver at low carrier levels, is known as *delayed A.V.C.* The *R.F. input voltage level at which A.V.C. action begins* is called the *threshold point* or the *threshold voltage*. The overload characteristic curve for a receiver having delayed A.V.C. is represented by curve 3 in Fig. 8. Notice that it follows curve 1, that for a receiver without A.V.C., up to the threshold voltage, after which it levels out and effectively prevents overloading.

Since the A.F. output voltage of the second detector in a receiver having delayed A.V.C. is quite high at the threshold point, further increases in output voltage should be prevented as much as possible. To do this, the negative C bias voltage produced by A.V.C. action should be applied to as many R.F. tubes as possible. If even this is insufficient, the A.V.C. voltage can be amplified by an extra vacuum tube stage, giving what is known as an

*amplified delayed A.V.C. circuit.*

**Definitions.** Those R.F. amplifier stages in an A.V.C. receiver which are to vary in gain as the incoming carrier level varies are called the *A.V.C.-controlled stages* or simply the *controlled stages*. The tubes in these controlled stages are called *A.V.C.-controlled tubes* or simply *controlled tubes*. The vacuum tube or tube section which converts the amplified modulated R.F. carrier into a D.C. voltage suitable for A.V.C. purposes is called the *A.V.C. tube* or *A.V.C. stage*, and the D.C. voltage is called the *A.V.C. voltage*. The A.V.C. voltage may be produced as a part of the action of demodulation, or may be produced independently by an extra stage in the receiver.

#### SIMPLE DIODE DETECTOR—A.V.C. CIRCUITS

A simple diode demodulator or detector circuit like that shown in Fig. 9A is not only capable of separating the modulation signal from the R.F. carrier, but can also produce the negative C bias voltage required for A.V.C. purposes. The modulated R.F. carrier signal at points 1 and 2 in the final I.F. amplifier stage passes through the final resonant circuits,  $L_p-C_p$  and  $L_s-C_s$ , and is applied directly to the plate and cathode of diode detector tube  $VT_2$ . Condenser  $C_D$  offers no opposition to this signal, for it has a low reactance at radio frequencies.

The modulated R.F. carrier is rectified by the diode tube, since this tube allows current to pass only in one direction; the wave form of the current passing through this tube is therefore like that shown in Fig. 9B. The charging and discharging action of condenser  $C_D$  on this pulsating current passing through  $R_D$  serves to filter out the R.F. variations, making the voltage across  $R_D$  have the wave form shown in Fig. 9C. Observe that this wave is made up of a D.C. component

which is proportional to carrier level and an A.C. component which is proportional to the percentage of modulation and the carrier level. If the values of  $C_D$  and  $R_D$  are properly chosen, this A.C. component will be an exact reproduction of the audio or video intelligence signal. This intelligence signal is fed into a low frequency amplifier (not shown in circuit) for further amplification by coupling the grid of the first low frequency amplifier stage to

the D.C. component of the voltage between terminals 3 and 4 is proportional to the level of the modulated R.F. carrier, these terminals may be used as a source for the desired A.V.C. voltage provided that the low frequency component is removed. Application of an A.F. signal to the grid of a controlled tube would place extra modulation on the carrier, a clearly undesirable condition. For this reason, it is necessary to filter the A.V.C. voltage in a radio

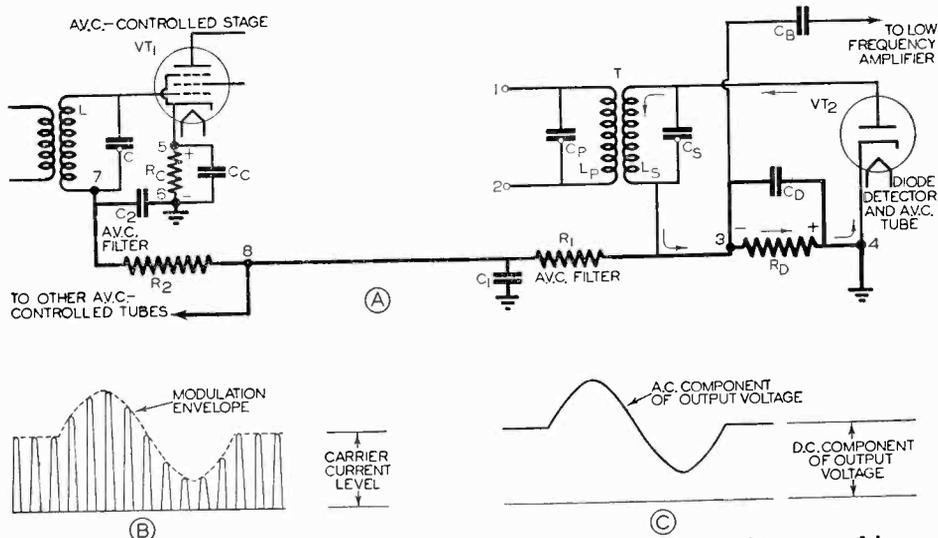


Fig. 9. These diagrams, together with the accompanying text, give the complete story of how an A.V.C. voltage can be produced by a simple diode detector, how this voltage can be filtered, and how the voltage is applied to an A.V.C.-controlled R.F. stage.

point 3 on resistor  $R_D$  through blocking condenser  $C_B$ , which blocks out the D.C. voltage component. The circuit is completed through the chassis.

Let us trace D.C. electron flow in the diode detector circuit of Fig. 9A. We start with the diode tube, for we know that electrons flow from the cathode to the plate. These electrons flow through coil  $L_s$  in the direction indicated by the arrows, then enter terminal 3 of resistor  $R_D$ , making this terminal negative with respect to the other resistor terminal (4); terminal 3 is therefore negative with respect to chassis or ground. Furthermore, since

receiver in order to keep the A.F. signal voltage out of the A.V.C.-controlled stages.

**Filtering the A.V.C. Voltage.** The fact that the control grids of R.F. amplifier tubes are negative, so that no D.C. grid current is drawn from the A.V.C. circuit, simplifies the problem of filtering the A.V.C. voltage. In Fig. 9A you will find two A.V.C. filters connected between point 3, at which both D.C. and A.F. components of voltage exist, and point 7 in the grid circuit of an A.V.C.-controlled stage, at which only the D.C. component of voltage is desired. These A.V.C. filters keep the

*A.F. signal voltage* out of the A.V.C.-controlled stages of the R.F. amplifier.

Let us consider first the A.V.C. filter made up of  $C_1$  and  $R_1$ . Notice that  $R_1$ - $C_1$  form a low-pass filter. The series element  $R_1$  offers a far greater opposition to alternating current than  $C_1$  does, so most of the A.F. is dropped across  $R_1$ . Resistor  $R_2$  and condenser  $C_2$  in the second A.V.C. filter provide additional filtering in the same way, making the voltage at point 7 a practically pure D.C. voltage. Condensers  $C_1$  and  $C_2$  naturally have no effect upon the D.C. voltage, and since no direct current flows through the filter circuit, resistors  $R_1$  and  $R_2$  likewise have no effect upon the value of D.C. voltage at point 7.

The flow of plate current through resistor  $R_C$  and condenser  $C_C$  in the A.V.C.-controlled stage containing tube  $VT_1$  produces across  $R_C$  a D.C. voltage which makes point 6 negative with respect to the cathode; this is ordinary automatic C bias action. The voltage drop across cathode resistor  $R_C$  is applied to the grid of tube  $VT_1$  through the chassis path between grounded points 6 and 4, then through  $R_D$ ,  $R_1$ ,  $R_2$  and coil  $L$  in turn, thus placing on the grid its normal C bias.

When an R.F. carrier signal is present in the receiver, the D.C. component of voltage produced across  $R_D$  acts in series with, and aids, the automatic C bias voltage. Thus the A.V.C. voltage and the automatic C bias voltage add together to make the grid of each controlled tube more negative than would be the case without A.V.C. An increase in carrier signal level boosts the D.C. component of voltage across  $R_D$ , driving the grid of each A.V.C. controlled tube more negative and thereby reducing the amplification of each tube sufficiently to keep the signal voltages in all stages of the receiver below the overload values. In an A.V.C. system a condition of equilibrium exists where

the carrier level at the detector is kept just enough above the desired constant value to provide the required A.V.C. voltage.

Condenser  $C_2$  in the circuit of Fig. 9A has another important task, that of providing a path to point 6 for the R.F. voltage developed across coil  $L$ . If this condenser were omitted, the R.F. current would have to flow through  $R_2$  and  $C_1$  to ground;  $R_2$  would naturally offer considerable opposition to the flow of R.F. current, and there would also be the possibility that R.F. current would stray into circuits where it could cause interference and undesirable feed-back. Since the reactance of  $C_2$  is less than the reactance of the  $R_2$ - $C_1$  path to ground, R.F. currents will take the  $C_2$  path to ground.

The A.V.C. filter system made up of  $C_2$  and  $R_2$  can be and often is omitted, leaving  $C_1$  and  $R_1$  to do the A.F. filtering and R.F. isolating, particularly when only one tube is being controlled by A.V.C. When several R.F. amplifier tubes are being controlled, it is customary to use an A.V.C. filter similar to  $R_2$  and  $C_2$  in each controlled stage, making connections from each controlled stage to point 8. This serves to isolate the tube circuits from each other, preventing undesirable feed-back.

#### TIME CONSTANT OF THE A.V.C. SYSTEM

An A.V.C. system must prevent blasting when a receiver is tuned suddenly from a weak to a strong signal, and must also compensate for more or less rapid fading effects. For this reason we are interested in knowing exactly how long it takes for the A.V.C. system to get into action when the R.F. carrier level is suddenly changed.

The D.C. component of voltage across  $R_D$  in Fig. 9A changes immediately after a change in carrier level,

but it takes a certain amount of time for condenser  $C_1$  in the first A.V.C. filter to charge or discharge to a new voltage value through  $R_1$ .

The amount of time delay introduced by A.V.C. filter  $R_1-C_1$  depends upon the ohmic value of resistor  $R_1$  and the capacity of condenser  $C_1$ ; this time, when expressed in seconds, is known as the *time constant* of the A.V.C. filter system, and can be computed quite easily. In the case of Fig. 9A, this can be done by multiplying the ohmic value of  $R_1$  in megohms by the capacity of  $C_1$  in microfarads. The result will be the time constant of the circuit in seconds, or the time required for the A.V.C. voltage to reach approximately 63% of its final new value after a change in carrier level. (It is standard practice among engineers to specify time constants for 63% of the total change, this having proved more convenient than a time constant based upon a total change.) The A.V.C. filter made up of  $R_2$  and  $C_2$  likewise introduces a time delay, which *increases* the time constant of the entire A.V.C. filter system. Remember—the time constant of an A.V.C. filter system is determined by *the values of the resistors and condensers in the A.V.C. filter system*.

A low time constant is naturally desirable in order to make the A.V.C. system respond as rapidly as possible to changes in carrier level; this can be secured by making the values of  $R_1$ ,  $R_2$ ,  $C_1$  and  $C_2$  low, but doing this impairs the filtering action which is so essential to the operation of an A.V.C. system. Receiver design engineers

therefore resort to a compromise which uses filter system parts large enough to provide satisfactory filtering and at the same time small enough to provide a sufficiently short time delay. A time constant of *one-fifth to one-tenth of a second* for the A.V.C. filter system is considered satisfactory by most engineers for the prevention of blasting and the reduction of fading.

The value for condensers  $C_1$  and  $C_2$  in an A.V.C. filter system have become essentially standard among receiver designers. A capacity of .1 mfd. for  $C_1$  and .05 mfd. for  $C_2$  are generally used, for these condensers are inexpensive and at the same time have a reactance of less than 20 ohms for any I.F. or R.F. signal which may be attempting to flow from resonant circuit  $L-C$  into the diode load. A .1 mfd. condenser, when used with a 1 megohm resistor, gives a time constant of one-tenth second. The filter action of these parts is such that they will reduce the strength of the lowest audio frequency signal which tries to get into the R.F. and I.F. amplifiers about 100 times. Two of these filter combinations would increase the time constant to one-fifth second and would increase the audio frequency filtering factor to 10,000 times. Ordinarily you will find that the values of resistors  $R_1$  and  $R_2$  range from .1 to 1 megohm, while condensers  $C_1$  and  $C_2$  range from .02 to .1 mfd. Do not be surprised, however, if you occasionally encounter quite different values than these; circuit conditions and the opinions of engineers vary widely. Changing the values of A.V.C. filter resistors or condensers affects the speed of A.V.C. action.

# Typical A.V.C. Circuits

Automatic volume controls are usually found only in superheterodyne receivers. As we will soon see, A.V.C. systems reduce the selectivity somewhat, and only superheterodynes have enough to spare. A.V.C. was tried on a few early high-gain T.R.F. receivers, but today, the only T.R.F. sets are inexpensive midgets which do not have this feature.

The sections of a superheterodyne receiver which are usually A.V.C.-controlled are as follows: 1, The *R.F. amplifier*, which amplifies the incoming modulated R.F. carrier signal; 2, the *mixer-first detector*, which mixes the incoming R.F. signal with the local os-

trolled. If there are three I.F. stages, the second is also A.V.C.-controlled. The final I.F. stage ordinarily has no A.V.C. control, or the control on this stage may be greatly reduced. This is because each succeeding stage handles greater and greater modulated carrier signal levels, and driving the C bias of a heavily loaded stage highly negative results in amplitude distortion and even partial or complete cut-off of the signal.

The more stages which are A.V.C.-controlled, the more uniform will be the receiver output and the less chance there will be for overloading. With these facts in mind, let us examine a

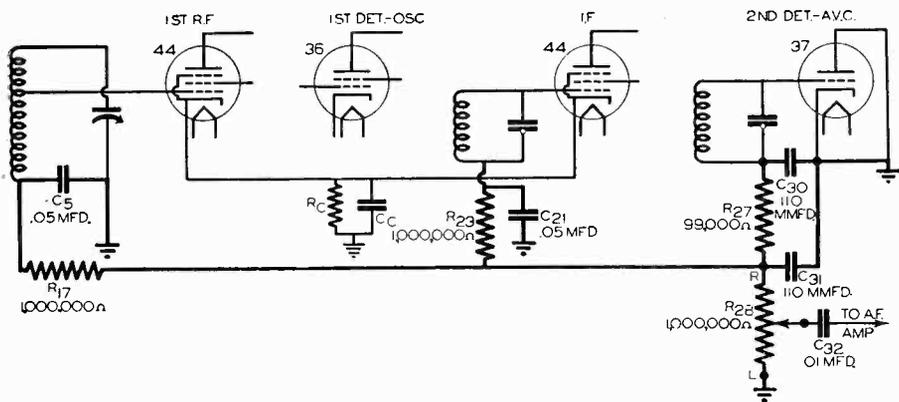


Fig. 10. Simplified diagram showing (in heavy lines) the A.V.C. system of the Philco Model 71 superheterodyne broadcast receiver. The values of the parts are the same as those used by the manufacturer.

cillator signal to give a modulated I.F. signal; 3, the *I.F. amplifier*.

It is common practice to apply the A.V.C. voltage to all preselector stages, in order to prevent overloading of the mixer-first detector. In those cases where a variable mu or super-control R.F. pentode tube is used for the mixer-first detector, this stage is sometimes A.V.C.-controlled. The first I.F. amplifier stage following the mixer-first detector is invariably A.V.C.-con-

rolled. Since we are interested only in the A.V.C. circuit and the A.V.C.-controlled stages, we will simplify the circuit diagrams by showing only these circuits.

A.V.C.-controlled tubes should always be variable mu screen grid tubes or super-control pentodes, for these tubes give essentially linear amplification over a wide range of C bias voltage values. If ordinary triode, screen



demands of receiver manufacturers who recognized the many advantages of a diode detector in A.V.C.-controlled receivers.

The diode detector load in Fig. 11 is made up of four resistors,  $R_7$ ,  $R_8$ ,  $R_9$  and  $R_{10}$ , connected in series between point 4 and ground. Condenser  $C_{17}$  provides a path to cathode for R.F. sig-

The mixer-first detector tube gets the lowest A.V.C. voltage, from point 1, while the I.F. amplifier tube gets a greater A.V.C. voltage from point 2. There are no filter resistors in the A.V.C. leads for these last two stages, fast A.V.C. action being desired in preference to complete protection from A.F. signal feed-back.

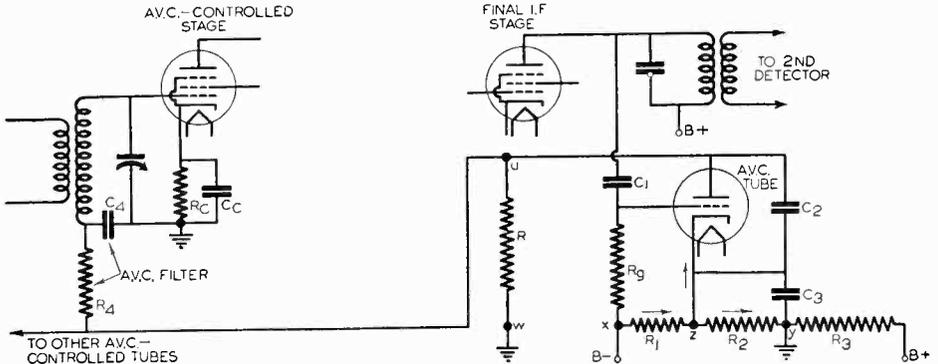


Fig. 12. Basic circuit for an A.V.C. system in which a separate triode tube is used to provide the A.V.C. voltage.

nals in the detector circuit. The A.C. and D.C. components of the detector output voltage appear across the resistor combination, with points 1, 2, 3 and 4 increasingly more negative with respect to the chassis or ground; when several different values of A.V.C. voltage are provided in this way, the voltages are said to be *staggered*. The A.F. voltage is fed to the control grid of the pentode section from the movable contact of potentiometer  $R_{11}$  which, with D.C. blocking condenser  $C_{16}$ , is connected across the diode load resistors.

As you can see, the grid of the first R.F. tube is connected to point 3, where it gets the highest A.V.C. voltage of any tube. The filter network for this tube, made up of  $C_4$  and  $R_1$ , has a time constant of .0075 second. This is the only tube in the receiver which has time delay, for fast A.V.C. action is desirable in an auto radio to compensate for changes in signal level when driving.

**Circuits with Separate A.V.C. Tubes.** Before the diode detector came into widespread use, a triode tube in a separate A.V.C. stage was commonly used to provide the required A.V.C. voltage. One basic circuit for this is shown in Fig. 12.

Let us consider the circuit first for the condition where no R.F. signals are being fed to the grid of the A.V.C. tube. Resistors  $R_1$ ,  $R_2$  and  $R_3$  form a voltage divider network which is connected across the power pack output terminals, hence electron flow is from B- to B+ through these resistors. This makes point  $x$  negative with respect to point  $z$ , and the grid of the A.V.C. tube (connected to  $x$  through grid resistor  $R_g$ ) is therefore *negative* with respect to its cathode.

How does the A.V.C. tube secure its plate voltage? Trace from point  $z$  through the cathode-plate path of the A.V.C. tube to point  $u$ , through A.V.C. load resistor  $R$  to point  $w$  and the chas-



this circuit appears in Fig. 13. First of all, notice that the R.F. amplifier tube, the mixer-first detector tube and the I.F. amplifier tube are A.V.C.-controlled. You can tell this because in each case the grid return lead from the grid coil does not go directly to ground or cathode but to points  $u$  and  $v$ , which are sources of A.V.C. voltage. The connection is in each case made through an A.V.C. filter consisting of a .05 mfd. condenser and either a 100,000 or 500,000 ohm resistor.

Resistors  $R_2$  and  $R_3$  in Fig. 13, serving as the load for the A.V.C. tube, divide the A.V.C. voltage into two values, the lower of which is fed to the mixer-first detector and the I.F. amplifier stage. Reduced A.V.C. voltage is necessary here because the mixer-first detector tube must handle a strong local oscillator signal in addition to the regular R.F. signal, and therefore cannot be driven as far negative as the amplifier tubes which get the full A.V.C. voltage.

Now let us see how the A.V.C. tube produces a negative bias voltage which increases with carrier level. We start at the B— terminal of the power pack, knowing that we should be able to trace all D.C. electron flow paths from this point to the B+ power pack terminal, and remembering that electrons flow into the minus terminal of a resistance and out of the plus terminal. One path is from B— to chassis through choke coil  $L_{18}$ , through the chassis to grounded points in the receiver stages, through a cathode bias resistor and through each tube from cathode to plate, and then back to B+ through the plate loads. This complete path through the first R.F. tube is indicated in Fig. 13, with the direction of electron flow shown by arrows and the chassis path indicated by a dotted line; the other paths between ground and the B+ terminal of the power pack,

one for each tube, are all in parallel with this one and can be traced in the same way. Another path is from B— through  $L_{18}$  to chassis and then through voltage divider resistors  $R_5$  and  $R_4$  directly to B+.

Now we are ready to trace electron flow through the A.V.C. tube (assume for the time being that no R.F. signals are present). Electrons flow from the B— terminal of the power pack through filter choke  $L_{17}$ , then through part of manual volume control potentiometer  $R_{16}$  to point  $z$ , where the electron flow divides. Some electrons go through the remainder of  $R_{16}$  and through resistor  $R_{17}$  to ground, from whence they take the various all-in-parallel paths through the tubes and voltage divider  $R_4$ - $R_5$  to the B+ power pack terminal, while other electrons go through the A.V.C. tube.

Since electrons enter  $R_{16}$  at  $x$  and travel to  $z$ ,  $x$  is more negative than  $z$  and the cathode of the A.V.C. tube. The grid of this tube, being connected to  $x$  through resistors  $R_{14}$  and  $R_{15}$ , thus gets a definite value of negative bias voltage which makes the tube pass a definite value of D.C. plate current. Electron flow is from  $z$  to cathode to plate, through R.F. choke  $L_9$  and A.V.C. tube load resistors  $R_2$  and  $R_3$  to ground, and then to the B+ terminal of the power pack through the chassis and the other tubes in the usual manner. The voltage drops produced across  $R_2$  and  $R_3$  by this current make points  $u$  and  $v$  negative with respect to ground (point  $w$ ); these negative voltages, applied to the grids of the controlled tubes through the A.V.C. filters, add to the automatic C bias voltages developed across the cathode resistors of the controlled tubes. There is thus a different normal C bias value for each setting of the manual volume control.

Potentiometer  $R_{16}$  serves as a manual volume control, for moving contact

arm  $z$  closer to  $x$  reduces the negative bias on the A.V.C. tube and at the same time increases the plate voltage of the tube slightly. This increases plate current flow, increases the voltage drops across  $R_2$  and  $R_3$ , increases the negative C bias voltage on the controlled tubes, reduces amplification and therefore reduces volume. Moving contact arm  $z$  away from  $x$  therefore increases volume; with  $z$  at  $t$ , we have the same arrangement as in the basic circuit of Fig. 12, where the A.V.C. tube has a high enough negative bias to cut off its plate current. Under this condition, it is obvious that no A.V.C. voltage is applied to the controlled tubes, and gain is a maximum.

The plate voltage on the A.V.C. tube is fairly low, for the maximum voltage it can obtain is that which is produced across choke  $L_{18}$  by the flow of power pack output current through the D.C. resistance of the choke. As you can see, the A.V.C. tube connects across this choke through  $R_3$ ,  $R_2$ ,  $L_9$ , the  $z$ - $x$  section of  $R_{16}$ , and  $L_{17}$ . The voltage across  $L_{18}$  is therefore divided among all these parts and the A.V.C. tube.

When an R.F. signal enters the A.V.C. tube through  $C_{18}$ , the tube acts as a C bias detector in rectifying the signal. R.F. components are filtered out by  $C_{20}$  and  $L_9$ , while the D.C. and A.F. components of the rectified R.F. signal appear across  $R_2$  and  $R_3$ . The A.F. component is filtered out in the usual manner by the A.V.C. filters. The D.C. component, proportional to carrier level, increases the normal D.C. voltages across  $R_2$  and  $R_3$  and thereby increases the bias voltages applied to the controlled tubes. An increase in R.F. carrier input to the A.V.C. tube produces an increased negative bias for the controlled tubes, reducing amplification just enough to keep the carrier input to the second detector essentially constant.

**How A.V.C. Affects Receiver Selectivity.** Undoubtedly you noticed in Fig. 13 that the R.F. input for the A.V.C. tube is taken from a point which is two tuned circuits ( $L_1$ - $C_1$  and  $L_2$ - $C_2$ ) ahead of the second detector. There is a definite reason for taking the R.F. input voltage for the A.V.C. tube ahead of these highly selective tuned circuits.

In ordinary A.V.C. circuits like those in Figs. 9, 10 and 11, where a diode tube serves both for A.V.C. purposes and for demodulation, we know that when all of the receiver tuning circuits are adjusted exactly to resonance the A.V.C. circuit will vary the gain of the R. F. amplifiers in accordance with the level of the desired carrier. But when the receiver is tuned slightly off the desired carrier frequency, as occurs when a station is tuned in, the carrier level at the A.V.C. input point is greatly reduced by the selectivity of the tuning circuits, and the A.V.C. circuit naturally increases the amplification to offset this. The broadcast thus comes in with about the same volume as before, but is slightly distorted due to incorrect tuning and has a strong noise signal in the background. In the older receivers this was quite objectionable, for listeners were accustomed to tune a receiver according to loudness and this A.V.C. action prevented them from using loudness as a guide. The radio expert would tune a receiver like this for minimum noise, but the average person did not know this little trick and blamed the receiver for the distortion and the apparently poor selectivity.

The solution to this tuning problem involved connecting the A.V.C. tube to a point farther ahead of the demodulator, where there was less selectivity and where the carrier level for the A.V.C. stage remained fairly constant as the receiver was tuned slightly off the incoming signal frequency. This kept the gain of the receiver fairly

constant, allowing the listener to tune for maximum output from the remaining I.F. stages. The R.C.A. circuit in Fig. 13 is an example of this arrangement; with their receiver, loudness could be used as a guide in tuning and good selectivity was obtained.

You will occasionally find the A.V.C. system connected considerably ahead of the second detector in modern receivers; although this practice is desirable it is not entirely necessary. Most receivers are now equipped with tuning aids to indicate when the receiver is correctly tuned.

### DELAYED A.V.C.

As you already know, A.V.C. action is undesirable on weak signals, for it reduces amplification a certain amount even on these weak signals. The "trick" which receiver designers use to delay the action of A.V.C. until a definite R.F. input signal (the threshold voltage in Fig. 8) comes through is quite simple, and involves merely the insertion of a fixed D. C. voltage in series with the load resistor of the diode A.V.C. tube.

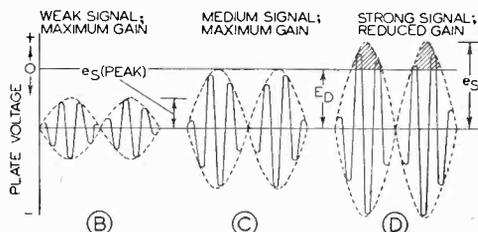
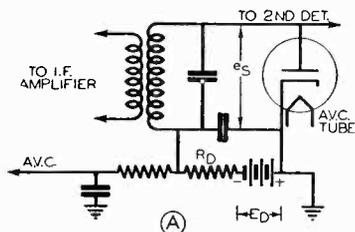


Fig. 14. Basic circuit for delayed A.V.C. action, and curves illustrating how delayed A.V.C. provides maximum amplification for weak signal.  $E_D$  represents the D.C. delay voltage, while  $e_s$  represents the peak value of the amplified R.F. signal.

The basic circuit arrangement for delayed A.V.C. is shown in Fig. 14A; a separate diode rectifier tube here is used for A.V.C. purposes. D.C. voltage  $E_D$  is placed in series with load resistor  $R_D$  with polarity as indicated. When there is no R.F. signal input, only this D.C. delay voltage is acting on the plate of the diode A.V.C. tube; it makes

the plate negative with respect to cathode, and thus no current can flow through  $R_D$ . Any R.F. input voltage  $e_s$  (secured from the I.F. amplifier) which has a peak value less than the delay voltage  $E_D$  will not make the diode plate positive, and hence there will be no A.V.C. voltage developed on weak signals. This is indicated in Fig. 14B, where the R.F. signal peak is considerably lower than the delay voltage  $E_D$  and cannot therefore make the plate positive. In Fig. 14C the peak input signal just equals the delay voltage; plate voltage is thus zero on peaks, but still is never positive. In Fig. 14D, however, the R.F. input signal peak is greater than the delay voltage, and the difference between these two voltages (the shaded area of each pulse) is effective in sending rectified current through  $R_D$  for the production of an A.V.C. voltage.

When the positive terminal of the delay voltage source is grounded, as in Fig. 14, this delay voltage places a negative bias on the grids of the controlled tubes at all times. By careful circuit design this can be made to serve

as normal C bias for the controlled tubes, eliminating the need for automatic C bias. The ground could just as well be at the negative terminal of  $E_D$ , however. This would put  $E_D$  in the cathode circuit of the diode, where it would have no direct effect upon the controlled tubes, and would serve only the function of delaying A.V.C. action.

A separate diode is a necessity when delayed A.V.C. action is employed to give maximum amplification of weak signals, for with a common detector-A.V.C. tube the delay voltage would prevent demodulation of weak signals and would cause distortion on medium-strength signals through cutting off of the negative peaks of the A.F. signal.

**Delayed A.V.C. Circuit Using Double-Diode Tube.** An excellent practical example of delayed A.V.C. is the Silvertone receiver circuit shown in Fig. 15. Section  $D_1$  of the double-diode 6H6G tube serves as second detector, section  $D_2$  serves as A.V.C. tube, and the voltage drop across resistors  $R_{10}$  and  $R_{11}$  in the power pack circuit provides the delay voltage  $E_D$ . Let us analyze the detector circuit action first.

The output R.F. signal  $e_s$  from the I.F. amplifier is fed directly to diode detector  $D_1$  through  $C_{15}$ , causing rectified current to flow from cathode to plate, through  $L$ ,  $R_4$  and  $R_5$  to the chassis, and then through the chassis to the cathode of  $D_1$ . The A.F. voltage developed across a part or all of  $R_5$  is fed through  $C_{22}$  to the grid of the 6F5G first A.F. amplifier tube. Incidentally, the grid of this A.F. tube gets its negative bias voltage from the voltage drop across resistor  $R_{11}$  (one of the delay voltage resistors).

Now let us analyze the delayed A.V.C. circuit. Condensers  $C_{14}$  and  $C_{15}$  feed the R.F. signal  $e_s$  into diode section  $D_2$ . The plate of this diode is made negative with respect to its cathode (which is grounded) by a connection through load resistor  $R_7$  to one end of resistor combination  $R_{10}$ - $R_{11}$ , through which the power pack output current flows to produce a delay voltage. This delay voltage also serves as minimum or normal C bias for the controlled tubes.

Weak signals undergo detection in the conventional manner in  $D_1$ , but

cannot overcome the delay voltage which makes the plate of  $D_2$  negative and hence no A.V.C. voltage is developed across  $R_7$ . Strong signals are likewise detected normally by  $D_1$ ; they also make the plate of  $D_2$  positive on the peaks of alternate half-cycles, and electron flow is from cathode to anode of  $D_2$ , then through A.V.C. load resistor  $R_7$  and through resistors  $R_{10}$  and

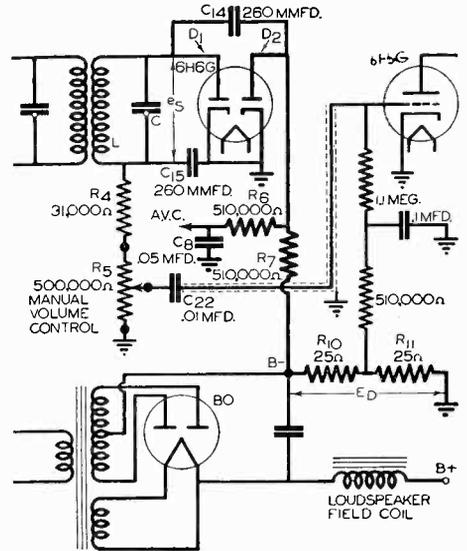


Fig. 15. Simplified diagram showing the delayed A.V.C. system of the Silvertone Model 1986-1987 superheterodyne broadcast receiver (chassis No. 100150). Section  $D_2$  of the double-diode tube produces across resistor  $R_7$  the desired A.V.C. voltage.

$R_{11}$  to ground and back to the cathode of  $D_2$ . The D.C. component of the voltage developed across  $R_7$  is the A.V.C. voltage; it adds to the normal negative bias voltage  $E_D$  and thus amplification is reduced when strong signals come through. The A.F. component of voltage across  $R_7$  is kept out of the controlled tubes by A.V.C. filter  $R_6$ - $C_R$ .

#### IDENTIFYING A.V.C.-CONTROLLED TUBES

Ordinarily you will have no trouble in locating the A.V.C. tube on a sche-

matic circuit diagram, for it is now general practice to identify tubes and their functions right on these diagrams. It is not customary, however, to indicate which tubes are A.V.C.-controlled. In order to determine this, you must know the usual methods of applying the A.V.C. voltage to a tube. Three common methods are shown in Fig. 16. You will observe that in each of

also as an R.F. by-pass from the tuning coil to ground.

In the I.F. amplifier stages of superheterodyne receivers the tuning condenser is of the trimmer type and is shunted directly across the tuning coil. The circuit arrangement in this case is as shown in Fig. 16B, the A.V.C. voltage still being fed through the tuning coil.

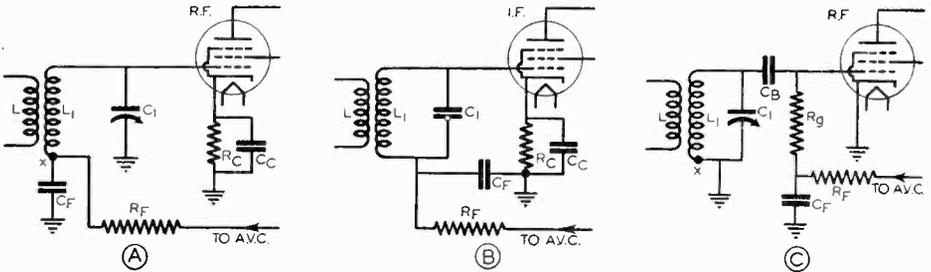


Fig. 16. Common methods of applying A.V.C. voltage to the grids of A.V.C.-controlled tubes.

these circuits the control grid does not trace directly through a conductive path to ground, chassis, B— or C—, but instead traces through an A.V.C. filter resistor to the A.V.C. tube load.

In the circuit of Fig. 16A the A.V.C. voltage is fed to the control grid through the coil ( $L_1$ ) of the tuned input circuit after being filtered by  $R_F$  and  $C_F$ ;  $R_C$  and  $C_C$  together furnish the minimum negative C bias for the tube, this being applied to the grid through the chassis and the conductive path of the A.V.C. system (this path is not shown here but can be traced in any of the A.V.C. circuits already studied). With this arrangement, tuning condenser  $C_1$  cannot be connected directly to point  $x$  on coil  $L_1$ , because of the fact that the rotor of this condenser is nearly always grounded directly to the chassis. Condenser  $C_F$  therefore serves

Occasionally the low R.F. potential end of the tuning coil (point  $x$  in Fig. 16C) is grounded, thus giving a direct connection between this coil and the grounded rotor of the tuning condenser. In this case the A.V.C. voltage must be applied through grid resistor  $R_g$ , with blocking condenser  $C_B$  used to prevent the A.V.C. voltage from being grounded by tuning coil  $L_1$ . Again the grid traces conductively to the A.V.C. system. To prevent loading of resonant circuit  $L_1-C_1$  and broadening of its response characteristic, resistor  $R_g$  must have a resistance of at least 500,000 ohms. Observe that the cathode is grounded directly; this indicates that the A.V.C. system furnishes the minimum negative C bias voltage in this arrangement. There are two A.V.C. filters,  $R_F$  and  $C_F$  being the first, and  $R_g$  with  $C_B$  making up the second.

# Construction Of Manual Volume Controls

Variable resistors or potentiometers which are used as manual volume controls are either of the wire-wound type or of the carbon type. Examples of each type are shown in Fig. 17; as you can see, the general appearance gives no clue toward identifying them.

The total resistance as measured with an ohmmeter between the left (*L*) and right (*R*) terminals of a volume control is a rough guide for identifying the type of construction used. Units which have resistances below 5,000 ohms are generally wire-wound; small-diameter resistance wire such as nichrome wire is wound on a thin, long and flexible rectangular strip of fibre and the strip is then curled into a semi-circle which fits into the cylindrical metal or bakelite housing of the unit. Insulating material separates the resistance unit from the housing. A movable contact arm, connected to terminal *C*, provides a means of contact to any point on the winding.



Fig. 17. Two representative manual volume control potentiometers. In each case two nuts are provided on the threaded tubular projection of the housing (through which the shaft runs) to permit clamping the control to a receiver chassis. Only a single mounting hole is needed.

Volume controls which have resistances above 10,000 ohms are ordinarily of the carbon type, although wire-wound resistances can be obtained with ohmic values up to 50,000 ohms for special purposes. It is safe to assume that all volume controls above 50,000 ohms are of the carbon type.

It is interesting to know how carbon type volume controls are made. In one

type of construction, a thin rectangular strip of insulating material is first coated with a kind of carbon paint, made by mixing highly pulverized carbon particles with water; one mixture of this nature is sold under the trade-name *Aquadag*. The strip is then curled as shown in Fig. 18A and mounted in the metal housing of the control.

The insulating strip is often cut in the shape of a round horseshoe like that shown in Fig. 18B, so the element can be mounted flat against the back of the

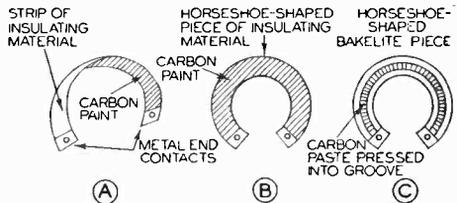


Fig. 18. Three types of construction used in carbon type volume control potentiometers.

housing without curling it. Another scheme involves the molding of a grooved circular horseshoe from bakelite, and pressing into the groove a carbon paste which serves as the resistance element; this type of construction is shown in Fig. 18C.

Each resistor manufacturer generally has his own method of applying the carbon solution or paste and treating it to give greatest dependability during use. Although sliding friction contact arms are used on wire-wound resistors, some form of roller contact which applies direct pressure to the resistance element is generally used with carbon type controls. This is necessary because the carbon elements do not stand up well under constant friction.

Wire-wound volume controls can be made with less than 5% variation in their resistance values (making the resistance of a 1,000 ohm unit anywhere between 950 ohms and 1,050 ohms;

engineers call this a 5% tolerance). Carbon resistors cannot ordinarily be made to such close tolerances; a tolerance of 20% appears to be customary with ordinary controls.

Exact values of resistance for volume controls are fortunately not required in ordinary circuits; only where

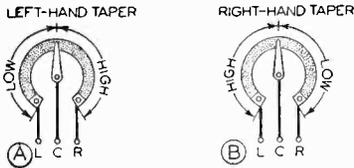


Fig. 19. Different resistances between the two halves of a volume control indicate a tapered control.

the volume control is a part of a power voltage-dividing system is it necessary to observe a close tolerance. Variations of from 20% to 40% in the total resistance of a volume control are perfectly satisfactory in ordinary receiver circuits. This is important for you to remember, as you can replace a defective 100,000 ohm volume control with either a 75,000-ohm or 140,000-ohm unit and still secure satisfactory results.

### TAPER

The manner in which the resistance of a volume control is distributed is of vital importance, even though the exact resistance of the unit can vary over wide ranges; this distribution of resistance is called the *taper* of the unit. If the resistance is uniformly distributed, so that varying the movable contact a definite amount causes a uniform change in resistance between points *R* and *C*, we say that the volume control has a *linear taper* and is therefore a *linear control*. Thus a linear control set at  $\frac{1}{4}$  of its total rotation would give  $\frac{1}{4}$  of the total resistance; similarly, at  $\frac{1}{2}$  of the total rotation there would be  $\frac{1}{2}$  of the total resistance.

Volume controls are usually expected to increase the receiver output volume

when the knob is turned in a clockwise direction (making the top of the knob move from left to right). The volume control is usually mounted on the radio chassis in such a way that the shaft, on which the knob is fastened, always points toward the person adjusting the control.

The diagrams in Fig. 19 show the essential elements of a volume control as they are when the shaft points toward you. If the resistance between terminals *C* and *L* increases uniformly as the contact arm is moved away from *L* and toward *R*, we have a linear control and curve 1 in Fig. 20 will repre-

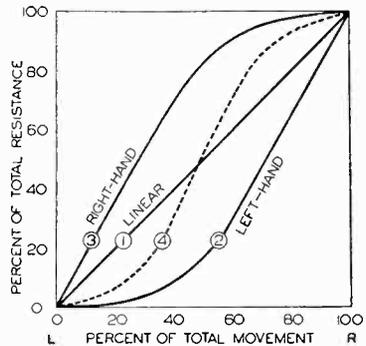


Fig. 20. Curves showing variation of resistance between terminals *L* and *C* with movement for three types of tapered volume controls and for a linear volume control.

sent its variation of resistance with control knob movement or rotation.

**Left-Hand Taper.** If, in moving contact arm *C* of a volume control unit from *L* to *R*, the resistance between *L* and *C* increases slowly at first and then more rapidly and uniformly after the half-way position is passed, we have what is known as a *left-hand taper* (curve 2 in Fig. 20). The taper or gradual change in resistance is here at the left-hand side of the control.

**Right-Hand Taper.** If the resistance between *L* and *C* in the above case increases more or less uniformly and

rapidly at first, and then increases less rapidly as the contact arm approaches terminal *R*, we have what is known as a *right-hand taper* (represented by curve 3 in Fig. 20). The taper or gradual change in resistance is here at the right-hand side of the control.

There is a simple way of telling whether a particular control has a right-hand or left-hand taper. With the shaft of the control pointed toward you, set the contact arm at its mid-position, as indicated in the diagrams in Fig. 19, and then measure the resistance first between terminals *C* and *L* and then between terminals *C* and *R* with an ohmmeter. If the lower resistance exists between *C* and *L*, you have a left-hand taper control, as in Fig. 19A; if the lower resistance exists between *C* and *R*, you have a right-hand taper control, as in Fig. 19B. If the resistances of the two halves are equal, you either have a linear taper or, in very rare cases, a combination right- and left-hand taper like that represented by curve 4 in Fig. 20.

### Why Tapered Volume Controls Are Needed.

The main purpose of a tapered manual volume control is to give changes in volume which sound uniform to the human ear for equal changes in volume control position. This means that the receiver designer must take into account the peculiar characteristics of the human ear, which prevent it from detecting changes of less than 3 db in the level of a complex sound. As examples, here are two circuits which use tapered volume controls.

A conventional diode detector circuit in which is incorporated A.V.C. and a manual volume control V.C. is shown in Fig. 21A. A study of this circuit will show you that as the movable contact on the volume control is moved from the cathode end to the diode plate end, the A.F. voltage fed to the triode

section of the tube increases. Hence, increased volume is secured when the contact arm is moved towards point *R*. If a conventional volume control unit is used, it should be so connected that terminal *L* on it connects to the cathode and terminal *R* to the tuned input circuit as indicated. This is done to make clockwise movement of the control knob give increased volume.

Supposing that a linear volume control were used for the circuit in Fig. 21A, let us see how it would act. We know that when movable contact *C* is at *R* the maximum available A.F. voltage will be applied to the grid of the triode section and full output volume will be obtained; likewise there will be zero A.F. voltage on the grid when *C*

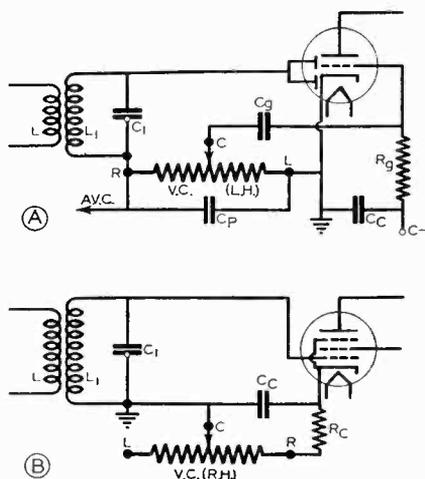


Fig. 21. Two examples of circuits which require tapered manual volume controls.

is at *L*, and volume will be zero. It is a known fact that a reduction of one-half in A.F. voltage corresponds to a 6 db change in sound level. Since the average human ear can just barely detect a 3 db change in the level of a complex sound, the 6 db change would be equivalent to two noticeable changes in sound.

With these facts in mind, let us reduce the volume from maximum to

zero by rotating contact arm  $C$  gradually from  $R$  to  $L$ . We listen carefully as we turn; at first we can detect no change in volume—our ear catches one change or reduction—we keep on turning until we can detect another reduction—now we check the volume control knob and find it is at the half-way position, so that half of the maximum A.F. voltage is being used.

We cut the voltage in half again by moving  $C$  from the  $\frac{1}{2}$  to the  $\frac{1}{4}$  position, and again get two noticeable changes in volume or a 6 db change in sound level. Moving  $C$  from the  $\frac{1}{4}$  to the  $\frac{1}{8}$  position again cuts the voltage in half, with two more detectable changes in sound. Thus as we move  $C$  from  $R$  to  $L$ , we get two noticeable changes in volume while moving the control knob first  $\frac{1}{2}$ , then  $\frac{1}{4}$ ,  $\frac{1}{8}$ ,  $1/16$ ,  $1/32$ , etc. of its total movement. Clearly a linear control gives extremely non-uniform changes in volume in this circuit, with the changes in volume being most apparent as the contact arm approaches  $L$ .

To overcome this non-uniform change in volume with control setting, the radio engineer uses a volume control which reduces the voltage rapidly as  $C$  is first moved away from  $R$ , then produces less change in voltage with change in setting, or allows the voltage to taper off slowly. This type of volume control would have a left-hand taper. *As a general rule, when the current through the entire volume control resistance is constant regardless of the setting of the movable contact, the left-hand taper is required.*

Another widely used manual volume control circuit is shown in Fig. 21B. Here resistor  $R_C$  sets the minimum C bias for the tube; increasing the resistance of manual volume control resistor  $V.C.$  increases this C bias and hence reduces the amplification and receiver volume. To make the volume increase

with clockwise rotation of the control, the right-hand terminal of the volume control must be connected to resistor  $R_C$  as indicated.

Maximum volume is, of course, obtained when contact arm  $C$  is at  $R$ . As  $C$  is moved away from  $R$ , the increase in C bias voltage is at first quite rapid, but increases less and less rapidly after that because advancing this volume control reduces the plate current of the tube. It therefore takes larger and larger series resistances in the cathode lead to get appreciable increases in C bias; what we need is a resistor which changes its resistance slowly at first, and then more rapidly. A right-hand tapered resistance, represented by curve 3 in Fig. 20, does this and is therefore used in a circuit like this.

*In general, when the current through the volume control resistance changes in value when the movable contact is adjusted, a right-hand tapered volume control is used.* In the C bias type of manual volume control such as this, the gain of the tube does not change uniformly with changes in C bias. To compensate for this deviation, special tapers are oftentimes used for the volume control. An ordinary right-hand taper control will generally suffice as a replacement in these cases, however.

You have undoubtedly noticed that the terminals of the manual volume controls in the circuits of Figs. 2, 3, 4, 5, 6 and 7 are marked  $L$ ,  $C$  and  $R$ ; these markings indicate how the volume control should be connected so as to get increased volume when the control knob is turned in the conventional clockwise direction. In addition, the ohmic values of the controls are given, and each volume control is marked either *L.H.* to indicate left-hand taper or *R.H.* to indicate right-hand taper. This extra information will be of practical help to any one who is called upon to service volume control circuits.

# Lesson Questions

Be sure to number your Answer Sheet 19FR-2.

Place your Student Number on *every* Answer Sheet.

*Send in your set of answers for this lesson immediately after you finish them, as instructed in the Study Schedule. This will give you the greatest possible benefit from our speedy personal grading service.*

1. Do receivers which employ automatic volume control also have manual volume controls?
2. Give three reasons why automatic volume control is desirable in a radio receiver.
3. Name three types of manual volume controls which are found in radio receivers.
4. What is the most effective way of varying the mutual conductance of a tube in order to control volume?
5. What is meant by the threshold voltage in a receiver having delayed A.V.C.?
6. What signal voltage (developed by the A.V.C. tube in a radio receiver) is kept out of the A.V.C.-controlled stages by A.V.C. filters?
7. What determines the time constant of the A.V.C. filter system?
8. What three sections of a superheterodyne receiver are usually A.V.C.-controlled?
9. Which type of taper (left-hand or right-hand) does a volume control unit have if, in moving contact arm C from L to R, the resistance between L and C increases slowly at first and then more rapidly after the half-way position is passed?
10. What is the general rule for using a volume control unit which has a right-hand taper?

## TAKE THE MIDDLE COURSE

Most of us realize the necessity for moderation in eating and drinking, but we often overlook the fact that moderation in all things is essential to happiness.

Consider, for example, the simple matter of opinions. If a man can see only his own opinions, and is unwilling to recognize that other people may also have good ideas, he is *opinionated*. A man with this fault is often unhappy, because he doesn't get along very well with other people. On the other hand, if a man yields his ideas to another's too readily, he is *weak-kneed*—and also unhappy.

If you can give and take—if you are open to reason—if you steer a *middle course*, you will be liked, people will be comfortable in your company, and you will be following one rule of happiness.

In your dress, be neat but not flashy. In your association with people, be courteous, but not affectedly polite. Be sympathetic, but not sentimental. Be self-confident, but don't be led into difficult situations by overconfidence. Don't believe everything you hear, but don't think everything you hear is false.

Let "moderation in all things" be one of the guiding principles of your life.

J. E. SMITH

**HOW SIGNAL CURRENTS ARE  
KEPT IN CORRECT PATHS**

20FR-3

**NATIONAL RADIO INSTITUTE**

ESTABLISHED 1914

**WASHINGTON, D. C.**



# STUDY SCHEDULE No. 20

For each study step, read the assigned pages first at your usual speed, then reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind, then answer the Lesson Questions specified for that step. Study each other step in this same way.

1. Introduction; By-pass Condensers . . . . . Pages 1-6  
How undesired signal currents get into a circuit; Methods of Keeping Signal Currents in Correct Paths; action of circuit having no by-pass condensers; Tracing Signal Currents; Cathode Circuit; Cathode C Bias Resistor By-Pass Condenser; Screen Grid Circuit; Screen Grid By-Pass Condenser; Plate Circuit; Plate Supply By-Pass Condenser; Practical Amplifier Circuit; By-Pass Condenser Rule. Answer Lesson Questions 1, 2, 3 and 4.
2. Simple Signal Current Filters; D.C. Blocking  
Condensers . . . . . Pages 6-10  
Simple Condenser Filter; Coil-Condenser and Resistor-Condenser Filters; Dual Condenser Filters; Pi or Low-Pass Filters; Practical Examples of Signal Current Filter Circuits; coupling problems; use of D.C. blocking condensers for coupling purposes. Answer Lesson Question 5.
3. Importance of Proper Connections . . . . . Pages 11-15  
Regeneration and degeneration effects due to stray coupling between leads, stray currents in the metal chassis or common coupling through the gang tuning condenser; Proper Connections in a Single R.F. Stage; Effects of Stray Chassis Currents; Stray currents in Ganged Variable Condensers. Answer Lesson Questions 6 and 7.
4. Shields for Electric Fields; Electromagnetic Shields;  
Positioning of Parts for Minimum Interference;  
Twisting of Power Supply Leads . . . . . Pages 15-23  
Electrostatic and electromagnetic induction; How Electric Fields Affect Grid Circuits; Sources of Electric Fields; Shielding of Coils; Practical Data on Electric Shields; electromagnetic shielding problems; Shields for Low-Frequency Magnetic Fields; Shields for High-Frequency Magnetic Fields; how positioning of parts can minimize interfering effects; Positioning of Wires; Positioning of Coils; magnetic coupling between parallel and twisted wires. Answer Lesson Questions 8 and 9.
5. Center-Tapped Filament Connections; Analysis of Signal  
Current Circuits in a Typical Radio Receiver . . . Pages 23-29  
Interference problems in filament-type tubes; A.C. hum interference; use of center-tapped filament resistor to eliminate hum; schematic circuit diagram of typical 5-tube superheterodyne receiver; tracing signal current paths; locating and identifying extra parts used to keep signal currents in correct paths; analysis of chassis layout to see how effects of stray electric and magnetic fields are minimized by proper placement of parts. Answer Lesson Question 10.
6. Mail Your Answers for this Lesson to N.R.I. for Grading.
7. Start Studying the Next Lesson.

# How Signal Currents Are Kept in Correct Paths

## Introduction

UNDESIRED signal currents can enter a circuit in three different ways: 1, through a *conductive path* such as a wire or a metal chassis; 2, by *electrostatic induction*, where the electric field set up by one circuit repels and attracts electrons in another circuit; 3, by *electromagnetic induction*, where the magnetic field set up by one circuit induces interfering currents in another circuit.

Undesired signal currents are kept out of circuits and desired signal currents are kept in their correct paths by many different methods; although these are generally applied to radio apparatus by design engineers, a knowledge of how each method works is needed by the radio operator or serviceman who is called upon to locate and remedy a defect in some circuit.

The most important of these methods for controlling signal current paths are

listed below; many of these have been covered in previous lessons, but this entire subject is of such great practical importance that a more detailed study is entirely justified.

## By-Pass Condensers

The importance of by-pass condensers can best be understood by considering the behavior of an amplifier stage which has no by-pass condensers. Such a stage is shown in Fig. 1A, where the input device  $R_s$  (across which the input signal voltage  $e_s$  appears) and the plate load  $R_L$  are shown as resistors. Although tuned circuits and transformers could be used at these locations in practical circuits, under ideal conditions they have the effect of resistors and will therefore be considered as such here.

When the input signal voltage  $e_s$  in Fig. 1A is zero, the control grid of the tube is at a fixed potential, the value of

### METHODS OF KEEPING SIGNAL CURRENTS IN CORRECT PATHS

1. Using by-pass condensers.
2. Using R.F., I.F. and A.F. signal current filters made up of resistors, coils and condensers.
3. Using D.C. blocking condensers.
4. Making proper circuit connections, in order to prevent stray coupling between leads and parts and to eliminate stray currents.
5. Shielding signal circuit parts against the effects of stray electric fields.
6. Shielding circuit parts against the effects of stray magnetic fields.
7. Positioning of power pack and signal circuit parts in such a way that magnetic and electric fields cannot cause interference.
8. Twisting filament leads and power supply leads which carry A.C., or running such leads parallel to each other and close together.
9. Using center-tapped filament connections for grid and plate return leads.
10. Using neutralizing circuits to counteract the effects of R.F. and I.F. feed-back currents. (This last method is covered elsewhere in the Course and will not be repeated here.)

which determines how much D.C. plate current and D.C. screen grid current will flow. The directions of electron flow in various parts of the amplifier under this condition are indicated by arrows directly on the circuit lines in Fig. 1A; you can easily verify these directions by remembering that electrons leave the minus terminal of the B battery (the only D.C. source in the circuit) and travel from cathode to plate or from cathode to screen grid in the tube. The D.C. screen grid and plate currents both flow through resistor  $R_C$ , developing across it a D.C. voltage drop which serves as the normal C bias for the tube and which therefore serves to determine the values of these D.C. electrode currents.

When an input A.C. signal voltage  $e_s$  is applied across  $R_s$ , it makes the control grid alternately more and less negative than the normal negative C bias value; when this input signal swings the control grid less negative (more positive), D.C. plate and screen grid currents both increase a certain amount, and when the input signal swings the control grid more negative, the D.C. plate and screen grid currents decrease. The application of an input signal on the control grid is thus causing the D.C. plate and screen grid currents to vary continually above and below their normal no-signal values, and we actually have pulsating D.C. currents in these two circuits.

We know definitely that the vacuum tube, when acted upon by the input signal, causes these variations in the electron current which is sent through the screen grid and plate circuits by the B battery; it is quite permissible, therefore, to think of the vacuum tube as simply a variable resistance acting in the plate circuit (and also in the screen grid circuit) and thereby producing these variations or pulsations in

plate and screen grid currents. If we say that this tube resistance is varying at exactly the same rate and in the same manner as the grid input A.C. signal, we can neglect the grid circuit entirely and concentrate our attention upon the plate and screen grid circuits, through which the B battery is forcing the pulsating currents.

In an amplifier circuit we are primarily interested in the pulsations or variations in plate current, for these develop across load resistor  $R_L$  the desired amplified signal voltage. We could consider only this A.C. or signal component of the plate current, and think of it as acting in a circuit made up of the B battery and a varying cathode-to-plate resistance in the tube, but experience has shown that an analysis or study of the circuit under this condition becomes quite complicated.

There is a much easier way of dealing with A.C. or signal currents in the plate and screen grid circuits of an amplifier—a method which is almost universally used by engineers because of its simplicity. This method involves neglecting the D.C. electrode currents entirely, and considering the *vacuum tube* as our source for A.C. or signal currents. (The battery is neglected as a voltage source and considered simply as a resistor equal to the internal battery resistance  $R_B$  in Fig. 1A). In other words, we replace the entire grid circuit and the cathode-plate path of the tube with an A.C. generator having a given internal resistance (the A.C. plate resistance) when we deal with the plate circuit, and we likewise consider an A.C. generator in place of the cathode-screen grid path of the tube when dealing with the screen grid circuit. (You will recall that this is exactly what was done in the equivalent tube circuits studied in a previous lesson.)

The tracing of signal currents (A.C. components of electrode currents) in Fig. 1A is quite simple now that we can consider the tube as the A.C. source. A.C. plate current  $i_p$  flows through the cathode-plate path of the tube (its source) to the plate, flows through load  $R_L$ , flows to point 2 (the chassis) either through the B battery or through voltage divider  $R_1$ - $R_2$ , and then returns to the cathode through  $R_C$ ; arrows labeled  $i_p$  indicate these paths. A.C. screen grid

control grid along with the input signal voltage  $e_s$ , and *degeneration* occurs. What actually happens is this: When the input signal  $e_s$  is increasing in a positive direction, making point 3 more positive with respect to point 2, the plate and screen grid signal currents through  $R_C$  will be increasing. Since electron flow for these currents is from point 2 to point 1, point 2 will be made increasingly more negative with respect to point 1. We thus have acting

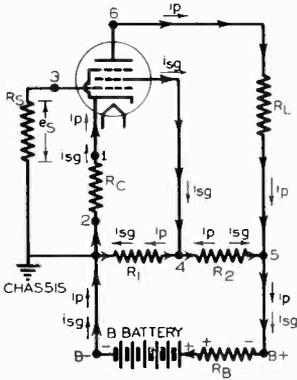


FIG. 1A. Simplified pentode amplifier circuit, with all by-pass condensers omitted. Arrows on circuit lines indicate D.C. electron flow, while other arrows indicate paths taken by signal currents.

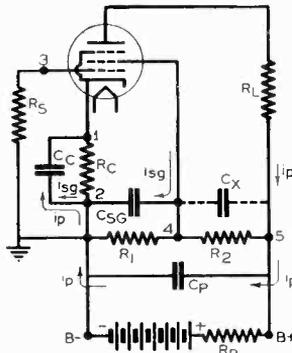


FIG. 1B. Three by-pass condensers inserted in the amplifier circuit of Fig. 1A, as shown above, effectively serve to keep signal currents in their proper paths, eliminating interfering effects.

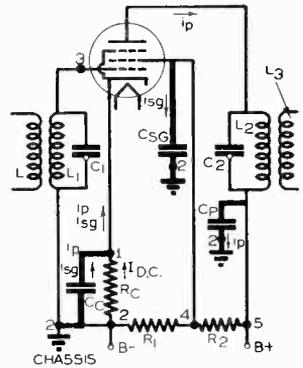


FIG. 1C. Pentode R.F. amplifier circuit of Fig. 1A with suitable by-pass condensers to provide low-reactance paths for signal currents. Actual tuned input and output circuits are shown instead of resistors.

current flows through the cathode-screen grid path of the tube (its source) to the screen grid electrode, flows to point 2 (the chassis) either through  $R_1$ , or through  $R_2$  and the B battery, and then returns to the cathode through  $R_C$ ; arrows labeled  $i_{sg}$  in Fig. 1A indicate these paths.

**Cathode Circuit.** Both the plate and screen grid signal currents flow through C bias resistor  $R_C$ , producing A.C. voltage drops across it. Since the control grid voltage is developed across  $R_C$ , these A.C. voltage drops act on the con-

on the control grid of the tube (applied to the grid and cathode terminals of the tube) the positively increasing input signal voltage and the negatively increasing A.C. voltage developed across the C bias resistor. These A.C. voltages oppose each other (are out of phase with each other) and hence the net A.C. voltage on the grid is reduced, with *degeneration* or reduction of signal current as a result.

**Cathode C Bias Resistor By-Pass Condenser.** The method commonly used to keep signal currents out of the

C bias resistor and thus prevent degeneration from occurring in the cathode circuit involves placing a by-pass condenser across the C bias resistor, making the reactance of this condenser low enough at the lowest signal frequencies so that the signal currents will take the path through the condenser rather than through the C bias resistor. This is done in Fig. 1B, where  $C_C$  is the cathode by-pass condenser which is connected across C bias resistor  $R_C$ . The A.C. voltage drop across this condenser will be negligibly low because the by-pass condenser has practically no reactance at signal current frequencies, and hence the only A.C. voltage acting on the grid will be the input signal voltage. The D.C. components of screen grid and plate current will continue to flow through  $R_C$ , producing across it the desired C bias voltage.

*Screen Grid Circuit.* Now let us see what undesirable effects are produced by A.C. screen grid current in the amplifier circuit of Fig. 1A. We note that the screen grid gets its D.C. voltage from a tap at point 4 on the voltage divider network made up of  $R_1$  and  $R_2$ . The voltage of this point with respect to point 2 (B-) depends upon the ohmic values of  $R_1$  and  $R_2$  and upon the current which flows through each resistor to produce across it a voltage drop. From Kirchhoff's voltage law we know that the sum of the voltage drops across  $R_1$  and  $R_2$  must equal the B battery voltage; stated in another way, the screen grid supply voltage (the drop across  $R_1$ ) will be equal to the B battery voltage minus the voltage drop across  $R_2$ .

Consider the A.C. screen grid current path through  $R_2$ . When a positively increasing control grid voltage causes screen grid current through  $R_2$

to increase, the voltage drop across  $R_2$  will increase and consequently the voltage drop across  $R_1$  (the screen grid voltage) will decrease. A decreasing screen grid voltage means a decreasing A.C. plate current, and consequently we have *degeneration*, a reduction in signal output, as an undesirable effect of A.C. screen grid current in this circuit.

*Screen Grid By-Pass Condenser.* A.C. screen grid current could be kept out of  $R_2$  by placing across it a by-pass condenser ( $C_X$  in Fig. 1B) which provided a low-reactance path around this resistor for this current, but a more practical solution is one which places a by-pass condenser between the screen grid lead (point 4) and the cathode lead (point 2); condenser  $C_{SG}$  in Fig. 1B is connected in this way. Now A.C. screen grid currents have a direct low-reactance path from point 4 to point 2 and then through  $C_C$  to the cathode, and consequently there will be no degeneration. This practical solution keeps A.C. grid current  $i_{sg}$  out of the source as well as out of  $R_2$ .

*Plate Circuit.* We still have the effects of A.C. plate current to consider in the circuit in Fig. 1A. This current ( $i_p$ ) flows through the B battery along with the A.C. screen grid current  $i_{sg}$ . Any voltage source has a certain amount of internal resistance; this resistance can be considered as acting in series with the source in Fig. 1A, and resistor  $R_B$  therefore represents the battery resistance through which these signal currents must flow.

We have seen how screen grid signal current can be made to take a proper path, but we still have plate signal current flowing through this battery resistance and producing a voltage drop across it. By considering the directions of electron flow (indicated by arrows

in Fig. 1A for the case where the grid signal is positively increasing) you can see that this voltage drop causes the potential of point 5 (the supply voltage) to decrease with respect to point 2 when the grid input voltage increases. The flow of signal current through the battery thus reduces the plate supply voltage and the plate current, and again we have *degeneration*. Furthermore, any A.C. plate currents which take the  $R_1$ - $R_2$  path from point 5 to point 2 instead of going through the battery may also cause undesirable effects.

Radio apparatus today generally contains a number of vacuum tube stages all connected to a common supply source. Any other tubes connected to the battery in Fig. 1A would consequently be affected by the signal voltage drops across the battery resistance and would undergo either regeneration or degeneration, depending upon whether the undesired voltage drops aided or opposed changes in plate current in a particular stage. Clearly it is undesirable to allow signal currents to flow through the plate supply and the voltage divider.

*Plate Supply By-Pass Condenser.* A by-pass condenser connected directly across the plate supply, like  $C_P$  in Fig. 1B, will offer a low-reactance path for signal currents to point 2 (the chassis), thereby keeping them out of the higher-resistance paths through the plate supply and voltage divider and eliminating undesirable degeneration or regeneration effects.

The plate by-pass condenser also lowers the impedance of the A.C. plate current path, allowing more A.C. plate current to flow through load resistor  $R_L$  and thereby developing across it a higher output signal voltage. No useful A.C. plate voltage is wasted now in the

internal battery resistance  $R_B$ .

*Practical Amplifier Circuit.* One example of a practical circuit which requires by-pass condensers is that shown in Fig. 1C, representing a pentode I.F. amplifier stage of a superheterodyne receiver. The connections to the three by-pass condensers are shown by heavy lines to make them easy for you to locate. In this circuit, bypass condensers  $C_P$  and  $C_{SG}$  are both connected to point 2 on the chassis, to which  $R_C$  is also connected, and low-reactance by-pass condenser  $C_C$  is relied upon to provide a path from the chassis to the cathode for the signal currents. Oftentimes, however, these by-pass condensers are connected directly from the plate and screen grid circuits to the *cathode*. Design engineers choose whichever method gives the shortest leads, for long leads result in electric and magnetic fields which produce undesirable coupling between circuits.

#### BY-PASS CONDENSER RULE

*In a vacuum tube circuit, all signal currents are by-passed through condensers to the cathode after they leave the electrodes or circuit parts through which they must flow to give the desired circuit action. Signal currents will take the path through a by-pass condenser in preference to other possible paths between two points in a receiver circuit because the by-pass condenser has practically no reactance at signal current frequencies; in fact, a by-pass condenser is the equivalent of a direct wire connection for these currents.*

Occasionally a design engineer omits a by-pass condenser specifically to secure regeneration or degeneration, or uses a single C bias resistor and by-pass condenser for two or more tubes.

A few typical examples will fix in your mind the general rule for apply-

ing by-pass condensers. A simple triode vacuum tube circuit like that shown in Fig. 2A requires two by-pass condensers. A simple tetrode circuit like that in Fig. 2B requires three by-pass condensers. A circuit containing a pentode tube in which the suppressor grid is tied directly to the cathode will likewise need three by-pass condensers. In a pentode tube circuit where the suppressor grid is not at cathode potential, as in Fig. 2C, four by-pass condensers are needed to keep signal cur-

Parts in parallel all have the same voltage, and here the resistor and condenser are essentially in parallel with the A.C. generator because the resistance of the connecting wires and the internal resistances of the generators are negligibly small. The condenser has simply increased the load on the A.C. generator and caused more current to be drawn from it, without appreciably lowering the A.C. voltage across the load. Of course, if either generator had a high internal resist-

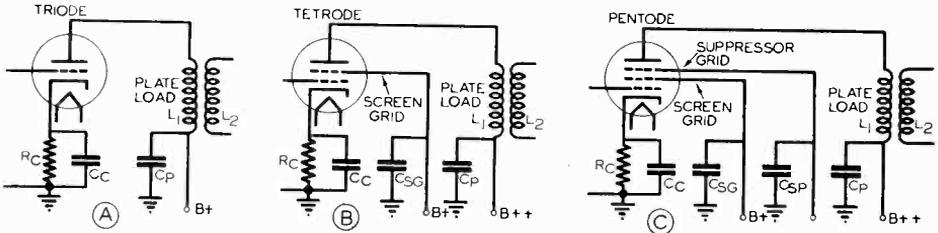


FIG. 2. Here are three applications of the general by-pass condenser rule. Observe that signal currents in the plate circuit are by-passed to the chassis after they leave the plate load, while screen and suppressor grid signal currents are by-passed to chassis after leaving their respective electrodes. In each case the cathode by-pass condenser provides a low-reactance path from chassis to cathode for signal currents.

rents in their correct paths. Occasionally a vacuum tube is used in a circuit where the control grid is at cathode potential (zero C bias); in cases like this the cathode resistor and the cathode by-pass condenser in the circuits of Fig. 2 would be omitted and the cathode would be grounded.

### Simple Signal Current Filters

*Simple Condenser Filter.* To understand why a simple condenser filter will not always give the desired results, consider Fig. 3A where condenser  $C_F$  is connected across load resistor  $R_L$ . These two parts are fed from a source which provides both A.C. and D.C. voltage, but we wish to apply only D.C. voltage to the load resistor. The reactance of condenser  $C_F$  is low for A.C., but this does not solve the problem, because we still have the full A.C. generated voltage across the load.

ance, the increase in alternating current would cause most of the A.C. voltage to be dropped inside the generator, and little would appear across  $C_F$  and  $R_L$ .

*Coil-Condenser and Resistor-Condenser Filters.* Let us consider the filter circuit arrangement of Fig. 3B first with only  $R_F$  connected ( $CH$  removed). We see that the A.C. voltage drop across  $C_F$  is applied across  $R_F$  and  $R_L$  in series. If  $R_F$  is large in ohmic value with respect to  $R_L$ , most of the A.C. voltage will appear across  $R_F$ , and very little across  $R_L$ . However, the D.C. voltage also divides between  $R_F$  and  $R_L$  in the same ratio as the A.C. voltage, and  $R_F$  contributes no filtering action.

To get filtering action and maximum D.C. voltage across load  $R_L$ , we replace resistor  $R_F$  with choke  $CH$ . The choke has a much higher reactance to

A.C. than  $R_L$ , so nearly all of the A.C. voltage is dropped across the choke. For D.C., the choke resistance is low compared to that of  $R_L$ , so most of the D.C. voltage appears across  $R_L$ .

Just as in Fig. 3A, condenser  $C_F$  in Fig. 3B is ineffective as a filter except when the generators have high internal impedance. We can make this condenser contribute to filtering action, however, by inserting a resistor at point  $x$  in Fig. 3B. The alternating current flows through this series resistor and the low combined impedance of  $C_F$  and  $R_L$ , and most of the A.C. volt-

age is dropped across the choke. Any A.C. voltage across  $C_{F1}$  is further reduced by the voltage-dividing action of high-impedance  $CH$  (or  $R_F$ ) and low-impedance  $C_{F2}$ , so there is very little A.C. voltage across the load.

*Pi or Low-Pass Filters.* When the choke coil is used in the circuit of Fig. 3C, we have a *pi filter* (so-called because on a circuit diagram it resembles the Greek letter  $\pi$ , which is pronounced "pie") or *low-pass filter* which can separate low frequency signals from high frequency signals. This filter has a resonant action at a definite fre-

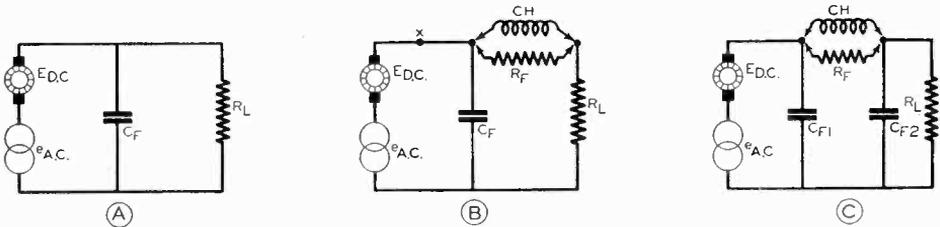


FIG. 3. The actions of the signal current filters commonly used in radio circuits are readily understood by studying these basic filter arrangements. Note the two-circle symbol for an A.C. generator; this symbol is widely used by radio and electrical men.

age drop now appears across the series resistor. The D.C. voltage divides between  $R_L$  and the series resistor, however, so we may not get enough D.C. voltage across load  $R_L$  if we use too large a series resistor for filtering purposes.

If high D.C. load voltage and adequate filtering are both required, a choke must be used at  $x$  in Fig. 3B instead of a series resistor. This gives two choke coils and one condenser in the filter network, and is a highly effective arrangement because the chokes offer high impedance to A.C. but very little opposition to D.C.

*Dual-Condenser Filters.* In practical circuits the source resistance is usually appreciable, and here the filter arrangement shown in Fig. 3C is highly effective. It has only one choke coil, and will work fairly well even with re-

quency known as the *cut-off frequency*; signals below this cut-off frequency pass through the filter readily and enter the load, for the choke coil then has very little reactance and the condensers have such high reactances that there is little shunting effect on signals. At frequencies slightly above the cut-off value, resonant action prevents transfer of signals through the filter. Signals considerably above the cut-off frequency are blocked by the choke coil and passed by the condensers, and hence cannot reach the load.

The cut-off frequency value depends upon the inductance of the choke and the capacities of the condensers (which are usually of equal size). The cut-off frequency is easily found by considering the filter as a resonant circuit made up of an inductance equal to one-half the inductance of the choke coil and a

capacity equal to that of one of the condensers in the filter, then determining the resonant frequency by the usual procedure for resonant circuits.

*Practical Examples of Signal Current Filter Circuits.* The tuned radio frequency amplifier circuit in Fig. 4A contains two examples of the resistor-condenser filter shown in Fig. 3B. Condenser  $C_{F1}$  provides a direct path to cathode for grid circuit R.F. currents, while the  $C_{F1}$ - $R_{F1}$  filter combination prevents any A.C. voltages which may exist across  $R_C$  from affecting the grid of the tube (there may be small

power supply and prevents any alternating current in the power supply from getting into the plate circuit. Filter resistor  $R_{F2}$  naturally lowers the D.C. plate voltage a certain amount; if this reduction in voltage is undesirable, this resistor may be replaced by an R.F. choke coil. Unfortunately, this substitution of a choke coil only serves to prevent plate circuit signal currents from getting into the power supply; the choke coil is not effective in keeping low frequency power pack ripple currents out of the plate circuit unless an additional iron-core choke

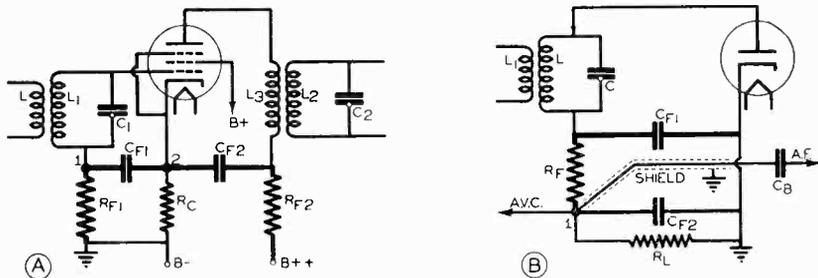


FIG. 4. The signal current filter is shown by heavy lines in each of these radio circuits.

amounts of A.C. plate current and power pack ripple current flowing through  $R_C$  and developing A.C. components of voltage across it). What happens is this:  $C_{F1}$ , having a very low reactance, is in series with  $R_{F1}$ , a very high resistance, across  $R_C$ , so what little alternating current gets through  $R_{F1}$  produces only a negligible A.C. voltage drop between points 1 and 2 as it flows through  $C_{F1}$ . Only the desired steady D.C. voltage across  $R_C$  can act on the grid through  $R_{F1}$ . Remember this important fact: A signal current filter will act both ways, preventing signal currents which are in one circuit from getting out and preventing other alternating currents from getting in.

In the same manner, signal current filter  $C_{F2}$ - $R_{F2}$  prevents plate circuit signal current from getting into the

designed for this particular purpose is used in series with the R.F. coil. For effective two-way filter action,  $R_{F2}$  should be replaced by R.F. and A.F. choke coils in series. Because of the relatively high cost of choke coils, it is more economical to use a filter resistor and either increase the D.C. supply voltage or adjust the circuit to operate satisfactorily at a reduced value of plate voltage.

The diode detector circuit shown in Fig. 4B contains a practical example of the dual-condenser filter represented by Fig. 3C. The path taken by the rectified signal current is through the diode tube, input coil  $L$ , filter resistor  $R_F$ , and diode load resistor  $R_L$ . R.F. current is kept out of the diode load by the filter combination made up of  $C_{F1}$ ,  $R_F$  and  $C_{F2}$ . In some circuits you will



vides the screen grid with the correct portion of the supply voltage, but  $R_2$  also serves with  $C_2$  as a filter which prevents screen grid signal currents from flowing through the supply lead and the power pack to the cathode.  $C_2$  provides a direct path from screen grid to cathode for these currents.

### D.C. Blocking Condensers

When the plate circuit of one stage is to be coupled to the grid circuit of a following stage and both stages have common supply terminals, the coupling device must be of such a nature that it keeps D.C. supply current out of the grid circuit of the following stage. (This discussion does not apply to direct-coupled amplifiers.) If this precaution were not observed, the grid would become positively charged with respect to its cathode, and correct operation would not be secured. When radio or audio frequency transformers are used as plate loads, the signal is transferred from one stage to another by induction and the D.C. component of plate current can flow only in the primary of the transformer, effectively solving this problem. When resistors or choke coils are used as plate loads, however, special means must be used to prevent D.C. supply current from entering the following grid circuit.

A universally used solution to this coupling problem is that shown in Fig. 6A, where coupling condenser  $C_B$  provides a low-reactance path for signal currents between the two stages and effectively blocks any flow of direct current. The currents flowing through each part in this circuit are indicated. You can readily see that if there were a D.C. conductive path for electrons from point 2 to point 1, electrons starting from point 3 (which is at ground or B— potential) would pass through

grid resistor  $R_{g2}$  to point 2, and then to point 1 and through resistors  $R_P$  and  $R_F$  to the B+ terminal of the power pack, making point 2 positive with respect to ground (point 3) and thereby placing a high positive bias on the grid of the following tube.

Quite often an external D.C. voltage is to be applied to a tube electrode which is already connected to ground by a D.C. conductive path. The R.F. amplifier circuit in Fig. 6B is an example; oftentimes an A.V.C. voltage must be applied to the grid of the tube, yet because the rotor of the tuning condenser is permanently grounded to the chassis and at the same time connected to one side of low-resistance coil  $L_1$ , an A.V.C. voltage applied directly to the grid would be shorted to ground through  $L_1$ .

One solution to this problem appears in Fig. 6C. The A.V.C. voltage is applied to the grid of the tube through an A.V.C. filter in the usual manner, with resistor  $R_g$  inserted as shown to prevent the A.V.C. circuit from shunting to ground (through A.V.C. filter condenser  $C_F$ ) the R.F. signal voltages produced across tuning circuit  $L_1-C_1$ . The value of  $R_g$  may be several hundred-thousand ohms; since the grid of the tube is negative at all times, it draws no current, and any current-limiting effects of resistors in the grid circuit are unimportant. In addition, D.C. blocking condenser  $C_B$  is required to prevent shorting out of the A.V.C. voltages by coil  $L_1$ . This is known as the *shunt feed* method of applying an A.V.C. voltage, for the A.V.C. voltage acts in parallel or in shunt with the signal input voltage.

Another solution to this same problem, used when the coil can be disconnected from the chassis, is the *series feed* method of applying the A.V.C.



power supply terminal are to be avoided, for they can give trouble in the form of capacitive or inductive coupling between each other.

Quite often a receiver manufacturer will allow a certain amount of regeneration or degeneration to exist because it does not appear to be objectionable at the time, but will later be compelled to modify certain sections of the receiver because the undesired effects become too objectionable after the receiver has been in use for some time. This is why you will occasionally find receivers of the same model, but made at different times, with minor changes in connections and parts values. Furthermore, since Radiotricians are called upon to make these corrections on earlier receivers which have developed trouble, the practical importance of this problem of making proper connections is quite evident.

*Proper Connections in a Single R.F. Stage.* As a practical example, let us consider how connections should be made in a typical pentode R.F. amplifier stage, such as that represented by the schematic circuit diagram in Fig. 7A. This diagram tells us that A.V.C. filter  $R_1-C_3$  is used to keep A.F. signals out of the grid circuit, to delay the A.V.C. action, and to provide a low-reactance path from the grid circuit to the chassis for R.F. signals. By-pass condenser  $C_4$  provides a low-reactance path for signal currents around C bias resistor  $R_C$ , while screen grid filter  $R_2-C_5$  keeps screen grid signal current out of the power supply; plate supply filter  $R_3-C_6$  does the same thing for plate circuit signal currents. This diagram does not show, however, how these signal currents actually travel between points 1, 2 and 3 through the chassis to point 4. The paths taken by these currents

are quite important, for degeneration or regeneration may occur if the signals from the screen grid, plate and control grid get a chance to produce appreciable voltages and mix before they reach the cathode. (Although the signals mix in passing through  $C_4$ , the reactance of this condenser is so low that the undesirable voltages produce negligible regeneration or degeneration effects.)

The production design engineer, who must decide beforehand the best location for each connecting lead and for each part, might redraw the schematic circuit diagram of Fig. 7A in the manner shown in Fig. 7B in order to indicate that points 1, 2 and 3 be connected directly to point 4. Doing this prevents the screen grid, control grid and plate signal currents from wandering through the chassis and mixing together before reaching point 4. All by-pass condensers in a single R.F. stage should be connected to a common point in the stage in order to prevent undesirable direct coupling between the different circuits in the stage.

The arrangement of connections shown in Fig. 7B may insure freedom from troubles due to improper direct connections, but we can still have trouble due to capacitive or inductive coupling between leads. Grid and plate leads in an R.F. stage must be kept as short and as far apart as possible in order to prevent capacitive and inductive feedback of signals from the plate circuit to the grid circuit. Suppose that, through faulty chassis layout, plate lead  $z$  is placed close to control grid lead  $x$ ; this would allow signal feedback from the plate circuit to the grid circuit, causing either regeneration or degeneration. Keeping the input and output leads in their proper places, as far away from each other as possible,

is a highly important duty of the radio apparatus manufacturer; since the exact positions of connecting wires are seldom shown in radio receiver service manuals, it is essential that you be familiar with approved methods of making actual connections.

Actual chassis connections for the pentode R.F. amplifier stage of Fig. 7A

point; in fact, one is above the chassis and the other is below.

*Effects of Stray Chassis Currents.* Suppose that instead of making connections as shown in Fig. 7C, points 1, 3 and 4 were simply connected to the nearest convenient points on the chassis, as indicated in Fig. 7D. Now we would expect the grid circuit signal

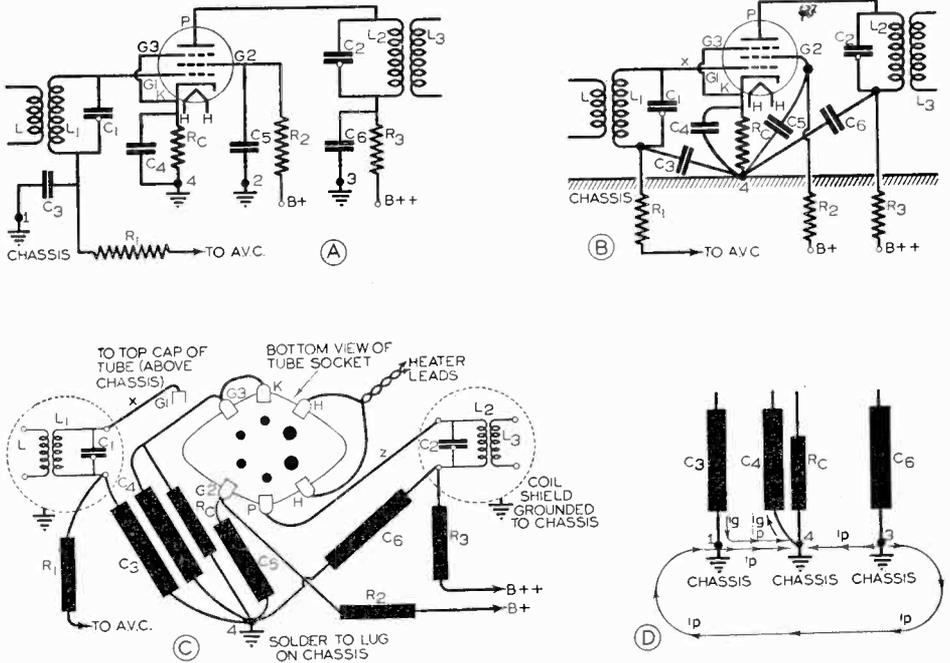


FIG. 7. An ordinary pentode I.F. amplifier circuit appears in its simplest form in the schematic diagram at A. Diagram B shows actual connections of by-pass condensers to a common point, but still in schematic form. Diagram C shows actual connections of parts on the chassis, while D shows how improper grounding of by-pass condensers to the chassis can result in interfering currents.

are shown in Fig. 7C, as they would appear when looking at the bottom of the chassis. The input and output coils with their trimmer condensers are, of course, on the top of the chassis, as also is the connection from the input tuned circuit to the top cap of the tube. Take particular notice of the fact that the control grid lead  $x$  and plate lead  $z$  do not run close to each other at any

current  $i_g$  to flow from point 1 through the chassis to point 4, and A.C. plate current  $i_p$  to flow from point 3 through the chassis to point 4. This is what actually takes place, but the A.C. plate current, which is many times greater than the grid current, spreads out over a wide path through the chassis in traveling from point 3 to point 4. One possible curved path is indicated in

Fig. 7D; notice that now some of the A.C. plate current  $i_p$  is flowing over the same path as the grid circuit signal current  $i_g$ . The A.C. plate current which takes the curved path causes an additional A.C. voltage drop between points 1 and 4 in the grid circuit, and this may cause undesirable regeneration or degeneration, depending upon the phase relationship between the signal currents involved.

If the grid by-pass condenser in a following stage is connected by mistake to a convenient point on the chassis, this signal current can likewise wander through the chassis, with one of its many possible paths being from point 1 to point 4 in Fig. 7D. This current can likewise cause undesirable regeneration or degeneration, depending upon the phase relationships of the currents.



Courtesy Sola Inv. Corp.

Although you ordinarily think of power pack filter circuits whenever electrolytic condensers are mentioned, here is an electrolytic condenser which is widely used to keep audio signal currents in correct paths. It is a low-voltage, high-capacity tubular dry electrolytic, and is used to by-pass a.f. currents around cathode resistors in the a.f. amplifier stages of radio receivers.

If it is desirable from a production viewpoint to connect points 1 and 3 directly to the chassis, as indicated in Fig. 7D, a length of heavy copper wire running from point 1 to point 4 and then to point 3 will provide a single path for all these A.C. currents, keeping them from straying through the chassis. Since this heavy wire will have low resistance, the current will take this path in preference to the high-resistance path through the metal

chassis. You will occasionally find this practice followed in radio receivers; a single heavy bus wire (copper wire having a square cross-section) is run from point 1 to point 4 to point 3 of one tube to point 1 to point 4 to point 3 of the following tube and in turn to each following tube. The connection must be made in this logical order if interference between currents is to be prevented.

*Stray Currents in Ganged Variable Condensers.* The ganged variable condenser in a sensitive modern all-wave receiver will generally have three tuning sections, with the rotors grounded (theoretically, at least) to the chassis through the frame of the condenser, as shown schematically in Fig. 8A. There will be a connection from each stator section to the grid of a tube and to a tuning coil.

With the stator sections so close to each other on the condenser unit, you can readily see how signals in one section might affect an adjacent stator section, causing feed-back effects. To eliminate this possibility, shielding plates are usually placed between the stator sections, these plates being grounded to the frame of the condenser and therefore to the chassis. (Electrostatic shielding of this type will be taken up later in this lesson.)

The average technician fails to realize that the current in each resonant circuit must flow through the rotor shaft in order to get to the condenser frame and then to the chassis. The construction of the rotor plate assembly is such that this path has appreciable resistance, and a certain amount of regeneration or degeneration occurs when different signal currents flow through the same section of the rotor.

In the circuit of Fig. 8B, for example, signal currents through condenser section  $C_2$  can either flow through the apparent rotor resistance  $R_2$  of this condenser and then through rotor resistance  $R_3$  of condenser  $C_3$  to ground, or can simply flow through the rotor resistance  $R_1$  of condenser  $C_1$  to ground; in either case there will be interference between signal currents. In order to eliminate this trouble, it is necessary to provide low-resistance paths to ground for the signal currents of each

making the stray currents appreciable in relation to desired signal currents. Rather elaborate precautions must often be taken to eliminate these stray currents; a study of the gang tuning condensers in a few modern all-wave receivers will reveal the exact manner in which low-resistance paths to ground are provided for each rotor section.

### Shields for Electric Fields

Having seen how interfering currents can get into circuits by actually

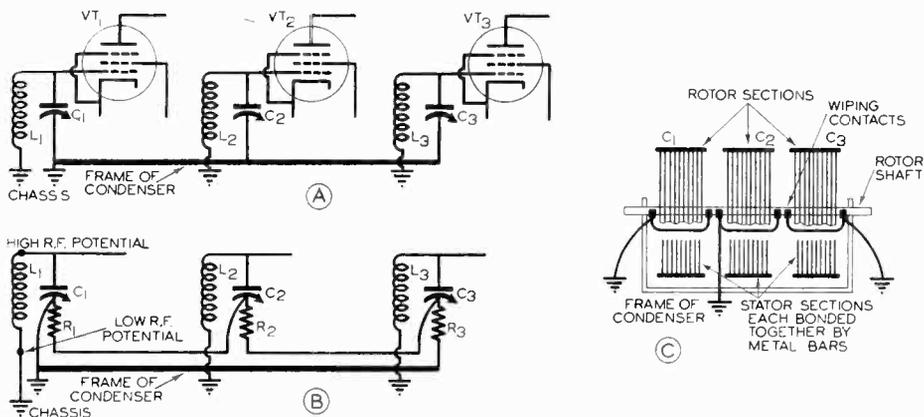


FIG. 8. Connections for a typical three-section ganged variable condenser appear at A. The effective locations of rotor shaft resistances are shown at B (curved arrows represent rotors), while one method used to offset the effects of rotor resistance and eliminate stray currents is shown at C.  $R_1$ ,  $R_2$  and  $R_3$  are not actual resistors, but represent resistances existing in each rotor section because of its mechanical construction.

rotor section. Thus you will often find wiping contacts on each side of a rotor section, with each pair of contacts connected together and also connected directly to the chassis, as shown in Fig. 8C. A better connection is from the rotor wiping contacts directly to the low or grounded R.F. terminal of the tuning coil associated with a rotor section.

Since the currents flowing in the rotor sections are resonant stepped-up currents, they may often be quite large,

flowing over undesired conductive paths, we are ready to consider how interfering currents can invade a circuit even when there are no direct wire connections. When the medium by which the interfering signals enter is an electric field, we call the action *electrostatic induction*, and when a magnetic field is to blame, we call it *electromagnetic induction*. We will consider electric fields first.

*How Electric Fields Affect Grid Circuits.* The grid or grid lead of a vacuum

tube is particularly susceptible to stray electric fields, for extremely small induced A.C. grid voltages can produce strong A.C. plate currents. The action is as follows: When the grid lead of a vacuum tube is located in the electric field which surrounds a highly charged object, variations in the electric charge (potential) of the object will cause the electric field to vary correspondingly, and this varying electric field will influence the free electrons in the grid circuit. When the object is negatively charged, it will repel the grid electrons; when the object is positively charged, it will attract the grid electrons (like

tion or in connecting leads. There will also be an interaction of electric fields when plate leads run near grid leads. These are just a few typical examples of how interfering currents are produced by stray electric fields. These interfering currents can result in regeneration, in degeneration or even in undesirable modulation of an R.F. signal, depending upon circuit conditions. Excessive regeneration, which results in a hissing noise in the loudspeaker or even in squeals due to oscillation, is easily recognized as being due to undesired coupling between circuits. Likewise, a 60 or 120 cycle hum modula-

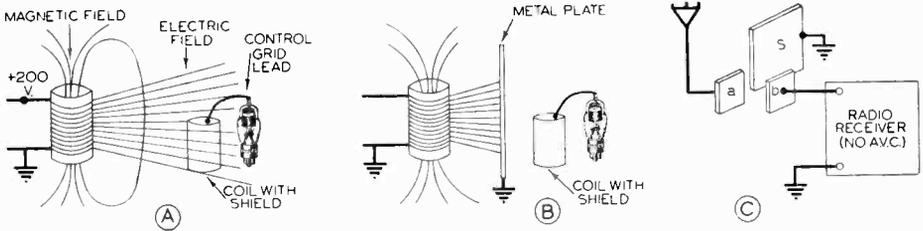


FIG. 9. These diagrams show how an electric field can affect electrons in the grid circuit of a tube, and how a metal plate serves as a shield for electric fields or electric lines of force.

charges repel, and unlike charges attract). If the object is being charged by an A.C. potential, the free electrons in the grid circuit will move back and forth to produce an alternating current of the same frequency.

*Sources of Electric Fields.* The plate of a vacuum tube is one object which can have a varying potential and produce an electric field which affects other circuits. The electric field associated with a power transformer or audio transformer can also induce currents in nearby tubes and in conductors. The stator plates of one section of a variable condenser can produce varying electric fields which induce interfering currents in another stator sec-

tion can be recognized and often traced to undesirable coupling between a power supply part and some signal circuit.

*Shielding of Coils.* The electric field around a coil which is carrying a current can be represented as shown in Fig. 9A. When this electric field passes near the control grid lead of a vacuum tube, the electrons in this lead will move back and forth in accordance with variations in the current through the coil, and consequently an interfering current will be induced in the control grid lead.

If a metal plate is inserted between the coil and the exposed grid lead of the tube, as indicated in Fig. 9B, this

metal plate will serve as a shield which prevents the electric fields (lines of force) from going any farther. Only materials which are good conductors of electricity will serve as shields for electric fields; insulating materials have little or no blocking effect upon an electric field.

The effectiveness of a shield in suppressing an electric field is easily demonstrated if you have access to a radio receiver which does not have A.V.C. Couple the receiver to the antenna system in the manner shown in Fig. 9C, so that the only connection to the antenna is by capacity coupling be-

*electric shield in radio apparatus.*

*Practical Data on Electric Shields.* The use of metal plates as shields between interfering parts is an old idea in radio, and this practice is followed even today in isolating the sections of many ganged variable condensers. Metal plates are entirely impractical, however, for shielding other parts in a receiver, as an electric field radiates in all directions and can affect many parts simultaneously.

The solution to the problem of suppressing electric fields lies in surrounding with a metal case the coil or other part which produces the field in order

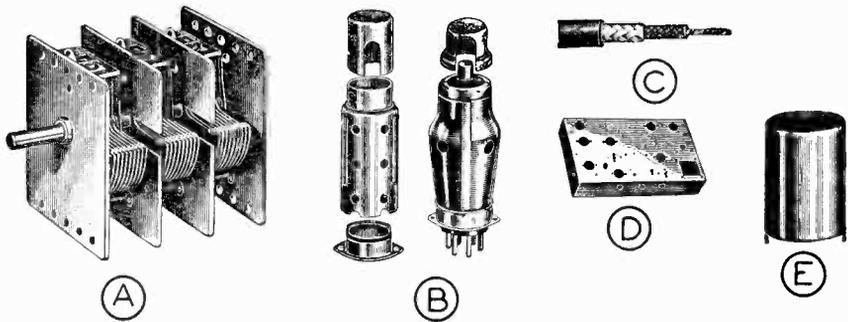


FIG. 10. Examples of electric shields used for radio apparatus.

tween metal plates *a* and *b*. With plate *a* removed, tune the receiver to a station which can just barely be heard. Now as you bring plate *a* near plate *b*, the volume will increase. Inserting a third metal plate *S*, larger in size, between plates *a* and *b* without touching them will have no effect on volume, for plate *S* merely relays the electric charges from plate *a* to plate *b*. When plate *S* is grounded, however, the electric field set up by plate *a* is grounded through *S*, and the volume will drop to the point where it is just barely audible. *This little demonstration shows clearly the im-*

to prevent spreading of the field. Parts which may be affected by electric fields are surrounded with metal shields in much the same manner. Thus, grid leads are surrounded by flexible metal sleeves (with insulation between the sleeve and the grid wire to prevent a short circuit), metal shields are used over glass tubes, or tubes which have all-metal envelopes are used to overcome the effects of electric fields.

Typical metallic shields which have been found effective in preventing interference by electric fields are shown in Fig. 10. The metal plates used between sections of a ganged variable

condenser are illustrated at *A*. Two types of metal shields for glass tubes appear at *B*. At *C* is shown a shielded wire; an ordinary insulated wire is covered by a flexible metallic braid or metallic loom, and the metal is in turn covered by a layer of insulation. Sometimes this outer layer of insulation is omitted, and sometimes there may be two or more insulated wires inside the metallic loom.

In addition to supporting the various parts, the chassis of a radio receiver also serves as a shield which is effective in preventing interference between parts above the chassis and those below the chassis; a typical chassis made of heavy gauge sheet metal is shown at *D*.

An example of an aluminum shield used for R.F. coils and R.F. transformers appears at *E*. In all cases it is essential that the shield be grounded to the metal chassis of the receiver at several points.

Shielding is so important to the operation of a high-gain, high-quality radio receiver that totally exposed parts (except resistors) are becoming quite rare. Paper condensers are automatically shielded if that lead marked "outer foil," "grounded end," or "ground" is connected to the chassis; a great many types of electrolytic condensers are made with metal shields, these being automatically grounded by bolting the unit to the chassis. The envelopes of all-metal tubes are likewise grounded automatically by inserting the tube in its socket, for one prong of the tube is always connected to the metal envelope, and the socket terminal for this prong is usually grounded.

### Electromagnetic Shields

Stray magnetic fields, which are produced by coils or wires carrying varying electric currents, can induce inter-

fering voltages in other coils or wires. These induced voltages are especially troublesome in grid circuits.

Two distinct electromagnetic shielding problems arise in the construction of radio equipment: 1, the elimination of interference due to parts which carry low-frequency power supply currents or A.F. signal currents; 2, the elimination of interference produced by parts which carry R.F. or I.F. currents.

*Shields for Low-Frequency Magnetic Fields.* In the case of magnetic fields which vary at a low frequency, the solution is simple. The part in question, usually an A.F. or power line frequency choke or transformer, is placed in an iron or steel housing which is so designed that leakage magnetic fields will flow through this housing and return to their correct paths instead of straying through the receiver and making trouble. Surrounding a coil with iron or steel lessens the temptation for lines of force to take high-opposition paths through air; iron or steel housings or shields are therefore widely used for low frequency coils and transformers.

*Shields for High-Frequency Magnetic Fields.* At frequencies above about 50 kc., it is not necessary to use an iron or steel housing for shielding purposes; in fact, *materials with good conductivity*, such as aluminum and copper will give better shielding effects than poor-conductivity iron and steel.

You can perform a simple experiment which will show how R.F. magnetic fields are affected by shields. Two coils,  $L_1$  and  $L_2$ , are connected as shown in Fig. 11. Any air-core coils will do; the numbers of turns are not important; Coil  $L_2$  is connected to the antenna terminals of a radio

receiver which does not have A.V.C. and which is tuned to a strong local station. Coil  $L_1$  is connected to the antenna and ground, as indicated. Antenna current flowing through  $L_1$  sets up a varying R.F. magnetic field which induces an R.F. voltage in coil  $L_2$ , and signals are therefore transferred to the receiver input.

Now insert between coils  $L_1$  and  $L_2$  a reasonably large copper, brass or aluminum plate, or place an ordinary aluminum cooking pot or frying pan between the two coils. You will find that this prevents transfer of the mag-

nets produce magnetic fields to oppose the original magnetic fields; this is simply Lenz' Law, with which you are already familiar. The opposing magnetic flux repels the original flux, sending it back and thus preventing the original flux from penetrating the metal shield. The circulating currents produced in the metal shields are called *eddy currents*. Of course, a certain amount of power is wasted when eddy currents flow in a conductor, but if the shield is not placed too close to the strongest part of the magnetic field, the loss will be negligible.

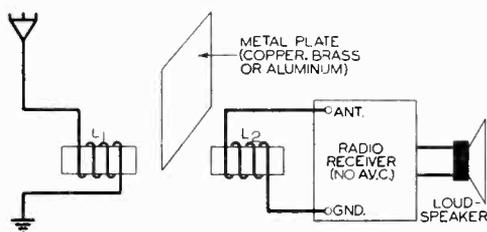


FIG. 11. In this experiment, a conductive metal plate inserted between the two coils prevents R.F. signals from reaching the receiver.

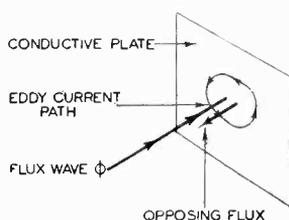


FIG. 12. How eddy currents in a metal plate can set up an opposing flux.

netic field from one coil to the other, and therefore prevents the local station from being heard. If this metal plate is grounded, it will also act as a shield for electric fields.

The action of the metal shield in Fig. 11 can be explained as follows: In attempting to pass through the conductive metal plate, the varying magnetic flux sets up very small circulating currents in the metal plate. You can consider this plate as being made up of many small circular metal rings, each of which is a complete closed circuit for electrons. The varying magnetic flux induces a voltage in each of these closed circuits in such a manner that the currents flowing through the cir-

High-grade R.F. and I.F. coils are usually placed in aluminum or copper cans, but copper-plated steel housings are also used as shields for coils. Aluminum and copper are good conductors of electricity, and therefore allow larger eddy currents to flow; these currents in turn produce stronger opposing magnetic fields to keep the original flux from passing out of the can. By placing the shield a reasonable distance away from the coil (at least one-half the diameter of the coil), little loss of signal power is incurred. Since the shield in effect acts as a larger number of short-circuited secondary windings on the coil, the main flux of the coil is reduced and the

inductance of a shielded coil is therefore lower than that of an unshielded coil. The relationship of the original flux, the eddy current rings and the opposing flux are indicated in Fig. 12. Typical shielded coils are illustrated in Fig. 13.

### Positioning of Parts for Minimum Interference

When shielding of parts is impractical or is not completely effective, inter-

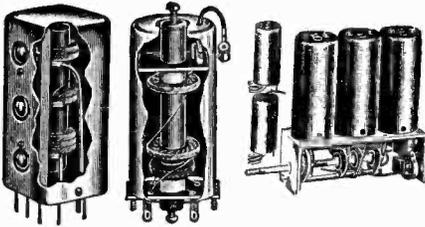


FIG. 13. Examples of shielded R. F. coils.

fering effects can be kept at a minimum by proper positioning of those parts and wires which are most affected. A

the vicinity of a coil is indicated in Fig. 14. For purposes of simplicity, an iron-core choke coil having open core ends was chosen. When a wire which is a part of a closed electrical circuit is placed in this magnetic field in such a way that the field cuts across the wire (or loops through the closed circuit formed by the wire), as is the case for wire *a* in Fig. 14, a voltage will be induced in the wire. When the wire is placed in position *b* (Fig. 14), the magnetic field will be parallel to the wire instead of cutting it or linking through the closed circuit, and no voltage will be induced. Thus you can see that for minimum interference, a wire should be parallel to the magnetic lines of force in its vicinity.

When a coil is placed in the magnetic field produced by the iron-core choke in Fig. 14, the relative positions of the two coils will determine whether or not a voltage is induced. For example, coil *A* in Fig. 14 is placed with its axis parallel to that of the choke coil, and hence the magnetic field links through coil *A*,

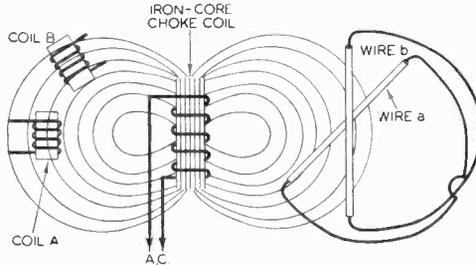


FIG. 14. The position of a coil or wire in a magnetic field determines whether or not a voltage will be induced.

review of the nature of magnetic fields and their voltage-inducing characteristics will reveal the best positions for various parts and wires.

The nature of the magnetic field in

producing an induced voltage. Observe that the magnetic lines of force which pass through coil *A* are parallel to the axis of the coil. The axis of coil *B*, however, is at right angles to the mag-

netic lines of force; in this case no flux flows through the turns of the coil and consequently there is no voltage induced.

*Positioning of Wires.* When it is necessary to place two wires close together, and one of the wires carries a varying current which can produce a varying magnetic field and in turn induce an interfering current in the other wire, the two wires should be placed at right angles to each other in the manner shown in Fig. 15A. Doing this makes the magnetic lines of force produced by wire  $x$  cut across wire  $y$  twice in opposite directions, with the result that the

paths are at right angles to each other, as indicated in Figs. 15B and 15C, and should be as far apart as possible from circuits which are most affected. This rule is, of course, only approximate, for the fields produced by these devices may be considerably distorted. Final positions for minimum interaction can be determined by experimentation; in the case of audio and power transformers, this can be done by feeding an A.F. or power line voltage to one coil and connecting a pair of headphones either directly across the other coil or through a separate audio amplifier which boosts the voltage and gives

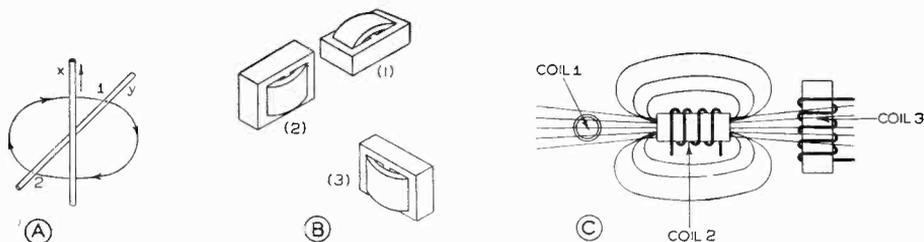


FIG. 15. Interfering currents due to electromagnetic induction are a minimum when wires, iron-core devices and coils are positioned at right angles to each other, as shown here.

two voltages induced in  $y$  cancel each other and have no effect upon the circuit of which wire  $y$  is a part. For example, one line of force produced by wire  $x$  may cut wire  $y$  at point 1 at a certain instant, producing in  $y$  an induced voltage having a certain polarity, but at the same instant this line of force will be cutting wire  $y$  in the opposite direction at point 2, producing a voltage of opposite polarity at 2. Placing wires at right angles to each other in this way also reduces interaction of electric fields.

*Positioning of Coils.* Coils and transformers should be positioned in such a way that their main magnetic field

more positive results. Adjust the position of either transformer until minimum hum is heard in the headphones. In actual radio receivers you can, of course, adjust the position of the transformer until minimum hum is heard.

### Twisting of Power Supply Leads

Having seen how the low-frequency magnetic fields which are produced by power transformers and power pack filter chokes can be prevented from straying, we are ready to consider how the connecting leads to these parts can be prevented from setting up interfering magnetic fields. The power line

leads which pass through the OFF-ON power switch to the power transformer are one possible source of trouble, and the low voltage, high current A.C. leads to the filaments of all tubes are another common cause of trouble. A study of the magnetic fields produced when two wires are close together will show how these magnetic fields can be suppressed.

Figure 16A shows two parallel wires,  $W_1$  and  $W_2$ . Alternating current flows out over one wire and back to its source over the other, but at any instant of

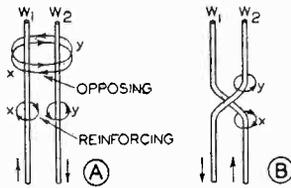


FIG. 16. Magnetic fields produced by two wires which carry the same current in opposite directions.

time the electron flow is in opposite directions as indicated. The magnetic lines of force produced by these currents have circular loops as indicated, with loops  $x$  produced by  $W_1$  and loops  $y$  produced by  $W_2$ .

Some of the magnetic loops will enclose both wires; notice that when this occurs, the magnetic lines of force will be in opposite directions and will cancel. Some of the magnetic loops will surround only their own wires, and in these cases the magnetic lines of force produced between the wires will all be in the same direction, and will reinforce each other.

Separating the wires allows more of these magnetic lines of force to pass between the wires and reenforce each other, and consequently the external magnetic field in the vicinity of the two

wires becomes stronger. Placing the wires closer together, on the other hand, forces more of the magnetic lines of force to encircle both wires and cancel each other, thus reducing the strength of the external field. Twin conducting leads should therefore be very close together if interfering magnetic fields are to be reduced to a minimum.

Even greater reduction in the external magnetic field produced by two current-carrying wires is possible when the wires are twisted together. The wires must, of course, be insulated in this case in order to prevent shorting them together. Figure 16B shows how twisting or transposing leads  $W_1$ , and  $W_2$  makes the individual flux paths  $x$  and  $y$  take opposite directions. In the vicinity of the wires these lines of force do not entirely interlock and cancel, but at a distance from the leads they actually do cancel and greatly reduce the external magnetic fields.

The more twists or transpositions there are in a given distance for two wires, the more reduction there is in external or stray magnetic fields. Incidentally, this reduction in the external magnetic field actually reduces the inductance of the conductors (any two wires which are close to each other and in the same circuit have a certain amount of inductance due to the fact that they form a single-turn coil).

Now you can understand why filament leads in radio apparatus are often twisted together or run very close together, and you can likewise understand why the power line leads to the power transformer primary are also twisted or placed close together. In some receivers, particularly those having low A.C. filament currents, manufacturers have found it unnecessary to twist filament wires or run them close together; elaborate precautions are

needed only when trouble is actually encountered.

You will usually find that the power transformer is located at a point on the chassis very close to the place where the power line cord enters. The power switch, however, is usually located at the front of the chassis; the two leads running to it are usually quite long, and hence should be twisted together for minimum magnetic effects.

There is still a certain amount of leakage magnetic flux even when leads are twisted, and therefore filament and power supply leads should be kept as

providing conductive paths for signal currents.

With the heater-type tubes which are so widely used in radio receivers at the present time, there is no direct connection between the filament and the cathode; if there is no leakage between these two elements, there will be no chance for signal currents to get to the filament circuit and for power line A.C. to get out of the filament circuit into signal circuits.

In the audio stages of radio receivers (particularly the power stages) and in the R.F. stages of radio transmitters.

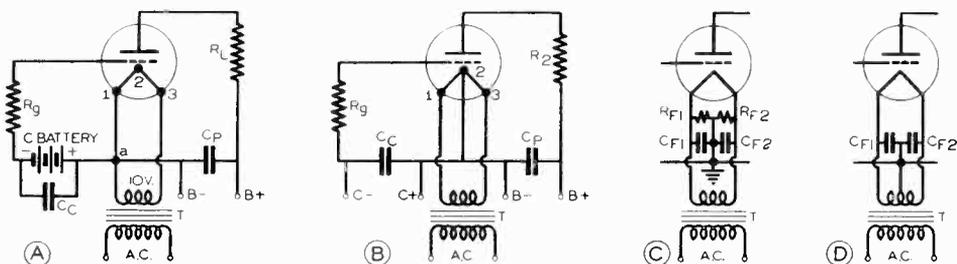


FIG. 17. When a filament type tube is being supplied with an A.C. filament voltage, as at A, the method shown at B, C or D is used to prevent the A.C. filament voltage from affecting the grid or plate circuits.

far as possible from the grid leads in the first stages of a receiver or amplifier. Even extremely weak interfering signals entering these stages may become appreciably strong after amplification by the entire system.

### Center-Tapped Filament Connections

The importance of preventing alternating current in filament leads from affecting signal circuits by magnetic induction has already been pointed out; it is just as important to prevent alternating current in the filament itself from affecting the signal circuits and to prevent filament leads from

filament type tubes are still widely used; here the filament serves also as cathode, and hence preventive measures must be taken.

Let us first see what happens when we connect a filament-type tube as indicated in Fig. 17A. Consider a simple circuit like that in Fig. 17A, where a C bias battery provides the normal negative bias voltage for the grid of the tube and the filament is fed with an A.C. voltage of, let us say, 10 volts. The grid and plate return leads for signal currents go to point a on one filament lead. Point 2, at the electrical center of the tube filament, can be considered as the cathode of the tube, even though all points on the filament are

emitting electrons. (For each point on the filament which is more negative than point 2 at any instant, there will be another point which is an equal amount more positive than point 2, regardless of the direction of current flow; as a result, the average of the voltage values between the grid and each point on the filament will equal the voltage between the grid and point 2, and likewise the average of all filament-to-plate voltages will equal the voltage between the plate and point 2.) The plate current will, therefore, be determined by the potential between point 2 and the grid (the grid bias voltage) and by the potential between point 2 and the plate (the D.C. plate voltage).

We can readily see, now, that the effective grid voltage will be the C battery voltage plus the voltage drop between points 1 and 2 on the filament. With 10 volts A.C. applied between points 1 and 3, there will be a 5 volt A.C. drop between points 1 and 2. This voltage drop will act in series with the C battery voltage insofar as the grid is concerned, making the grid bias voltage alternately 5 volts greater and 5 volts less than the battery voltage. Naturally this A.C. voltage of 5 volts applied to the grid will cause the plate current to vary correspondingly. The A.C. voltage drop will likewise act in series with the B battery voltage in determining the effective plate voltage, but the variations in voltage will obviously be more serious in the grid circuit because of the high amplification of the tube. Clearly the arrangement in Fig. 17A will result in excessive A.C. hum interference.

A theoretical solution to this problem is indicated in Fig. 17B, where the A.C. filament voltage is eliminated from the signal circuits simply by con-

necting the plate and grid return leads directly to point 2 by means of a tap on the filament. Since few if any tube filaments are made with center-tap connections, this solution cannot be utilized in practice.

The practical solutions to this problem, indicated in Figs. 17C and 17D, involve the use of external electrical mid-taps between the two filament leads. These mid-taps can be provided by a center-tapped resistor, as in Fig. 17C, or by a center-tapped filament winding, as in Fig. 17D. Often the mid-tap on the resistor is made variable, so it can be adjusted to compensate for unbalanced filaments and other conditions. When filament-type tubes are used in radio frequency amplifiers, bypass condensers are generally connected between the mid-tap and each filament lead to provide low-reactance paths to either filament terminal of the tube for signal currents; in the case of Fig. 17D, these condensers also serve to prevent signal currents from taking the path through the filament winding of the power transformer.

### **Analysis of Signal Current Circuits in a Typical Radio Receiver**

The complete schematic circuit diagram of a typical 5-tube superheterodyne receiver is shown in Fig. 18 just as you might find it in the service manual for the set. We will study this diagram in detail, first tracing the paths of signal currents, then locating and identifying the various extra parts used by the designer to keep signal currents in their correct paths. Having done this, we will analyze the chassis layout of the receiver to see how the effects of stray electric and magnetic fields are minimized by proper placement of parts. This analysis will not only serve to summarize for you the

many practical facts covered in this lesson, but will also give you additional experience in reading circuit diagrams.

A practical radio man is able to read a circuit diagram like that in Fig. 18 almost at a glance simply because he has learned to consider only the important fundamental signal circuit parts, neglecting the various resistors and condensers which keep signal cur-

rent from left to right. The R.F. signals picked up by the antenna are transferred to a grid of the first tube by  $T_1$ , which we recognize as a tuned-secondary antenna transformer. The first tube has five grids and is therefore a pentagrid converter serving as mixer, first detector and oscillator. The first two grids of the tube act with parts  $C_3$ ,  $L_3$  and  $L_4$  as an oscillating circuit

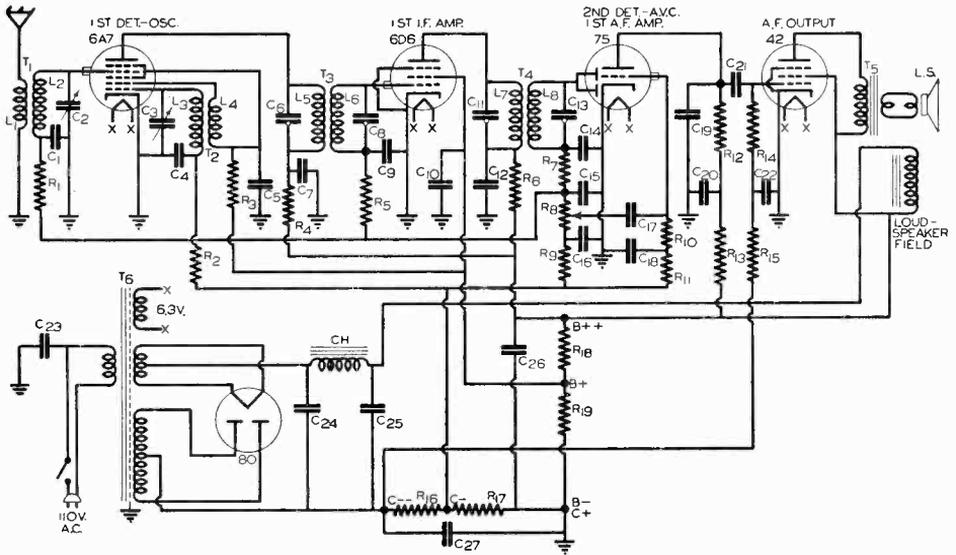


FIG. 18. Complete schematic circuit diagram of a representative 5-tube superheterodyne receiver. This circuit is presented here only to show circuit functions and illustrate the filtering and by-passing methods described in this book; it is not intended for constructional purposes, and does not represent any one particular manufactured receiver.

rents in their correct paths, and neglecting the power supply leads. In Fig. 19 are shown only those parts which serve to identify the various stages of this receiver circuit; we will study this at first, in preference to the complete diagram in Fig. 18, for you are not as yet expected to be able to eliminate the less-important parts mentally.

We start at the upper left in Fig. 19, since the upper part of a radio circuit diagram is invariably arranged to read

which produces the desired local R.F. carrier to mix with the incoming R.F. signal and produce in the plate circuit of this 6A7 tube the desired I.F. signal.

Adjustable condensers  $C_3$  and  $C_8$  identify  $T_3$  as an I.F. transformer which passes on this I. F. signal to the 6D6 first I.F. tube for amplification, while I.F. transformer  $T_4$  transfers the amplified I.F. signal to the diode section of the type 75 tube. Here you will recognize the conventional second detector-A. V. C. arrangement, with diode

current developing across the detector load the desired A.F. component and a D.C. component of voltage for A.V.C. purposes. The first detector and the I.F. amplifier are A.V.C.-controlled for their control grid leads do not return directly to their respective cathodes.

The A.F. signal is fed through coupling condenser  $C_{17}$  to the grid of the

through output transformer  $T_5$ .

D.C. voltages for the various tube electrodes are supplied by the power pack, made up of power transformer  $T_6$ , a type 80 full-wave rectifier tube, a filter system containing  $C_{24}$ , filter choke  $CH$ ,  $C_{25}$ , the loudspeaker field (which serves also as a filter choke) and  $C_{26}$ . Voltage divider resistors  $R_{16}$ ,

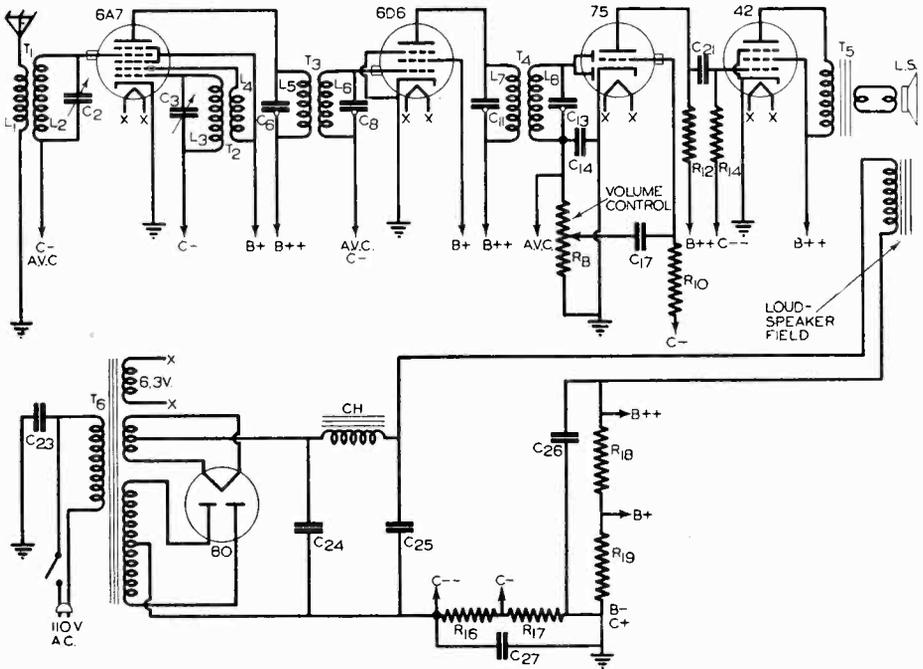


FIG. 19. Simplified version of the 5-tube superheterodyne receiver circuit shown in Fig. 18.

triode section of the 75 tube, which consequently acts as an A.F. amplifier stage. The value of the A.F. voltage which is fed to this first A.F. stage is controlled by the setting of manual volume control potentiometer  $R_8$ . The amplified A.F. signal is transferred to the type 42 output tube through resistance-capacitance coupling, and the A.F. output of this tube is fed to the voice coil of the dynamic loudspeaker

$R_{17}$ ,  $R_{18}$  and  $R_{19}$  then provide the required values of B and C voltages, with  $C_{27}$  providing additional filtering for the C voltages.

Now that we are familiar with the general function of each stage in our receiver circuit, we can return to Fig. 18 and try our skill at identifying the various by-pass condensers and signal current filters which must be added to the basic circuit of Fig. 19 in order to

secure satisfactory performance. We see immediately that  $C_1$  and  $R_1$  act as the A.V.C. filter for the first detector, for  $R_1$  connects to a point in the load circuit of the diode second detector.  $C_1$  also serves to provide a path to ground and cathode for grid circuit R.F. signals, and to prevent shorting to ground of the D.C. A.V.C. voltage.

Next we come to the  $R_2$ - $C_4$  combination. Since  $R_2$  connects to the C— point of the power pack voltage divider, we know this is not an A.V.C. filter; it is therefore a simple resistor-condenser signal current filter, which keeps oscillator R.F. signal out of the power pack.  $C_4$  also permits application of the C bias voltage to the grid without interrupting the continuity of the oscillator tank circuit  $C_3$ - $L_3$  for R.F. signals.

$R_3$ - $C_5$  is likewise a signal current filter, serving to keep screen grid (grids 3 and 5) and oscillator plate (grid 2) R.F. currents out of the power pack and at the same time by-pass these signal currents to the cathode of the 6A7 tube.

$R_4$ - $C_7$  is a signal current filter which by-passes plate signal currents of the 6A7 tube to its cathode and at the same time keeps these currents out of the power pack.

$R_5$ - $C_9$  are readily recognized as another A.V.C. filter once you trace connections from  $R_5$  to the second detector.

$C_{10}$  is clearly the screen grid by-pass condenser for the first I.F. amplifier.

$R_6$ - $C_{12}$  form a signal current filter which by-passes plate signal currents of the first I.F. tube to cathode and keeps them out of the power pack.

Dual-condenser filter combination  $C_{14}$ - $R_7$ - $C_{15}$  serves to keep R.F. current out of diode detector load  $R_8$  by providing low-reactance paths to cathode for this current.  $C_{16}$  and  $R_9$  together

serve as a signal current filter which permits application of normal C bias (developed across  $R_{17}$ ) to the 6A7 and 6D6 A.V.C.-controlled tubes while preventing power pack hum currents from reaching these tubes and at the same time keeping signal currents out of the power pack.

$C_{17}$  is simply an A.F. coupling condenser and D.C. blocking condenser.  $R_{11}$  and  $C_{18}$  act as a signal current filter in providing a return path to cathode for A.F. grid currents of the first A.F. amplifier, keeping these currents out of the C bias supply circuit, and also serve to keep power pack ripple currents out of signal circuits.  $R_{10}$  provides a conductive path for the D.C. bias voltage, since it is high in ohmic value, the entire A.F. voltage fed through  $C_{17}$  by potentiometer  $R_8$  is developed across it.

$C_{19}$  is an R.F. by-pass condenser which shunts to ground any R.F. currents which may reach the plate of the type 75 tube.  $R_{12}$  is the load resistor for this tube, with signal current filter  $R_{13}$ - $C_{20}$  providing a path to ground and the cathode of the 75 tube for A.F. currents after they leave the load, thereby keeping these currents out of the power pack;  $R_{13}$  and  $C_{20}$  also prevent power pack ripple currents from entering the plate circuit.

$C_{21}$  is a D.C. blocking and A.F. coupling condenser, while  $R_{15}$  and  $C_{22}$  act as a signal current filter for the grid of the type 42 output tube.  $R_{14}$  is a grid resistor which permits application of the bias voltage to the grid.

Filament current for the four signal circuit tubes is provided by a separate winding (marked XX) on the power transformer. Since heater-type tubes are used, there is no electrical connection between filament and signal circuits, and no filter or by-pass conden-

sers are required. The filament leads will, of course, be twisted or run close together under the chassis to lessen their magnetic effects on adjacent wires and parts. Notice the dotted line in the core of power transformer  $T_6$ ; this represents an electrostatic shield made of a layer of metal foil actually placed between the primary and secondary windings to eliminate electrostatic coupling which might, under certain conditions, transfer undesir-

that just studied is given in Fig. 20, which represents the top view of the chassis. In a receiver of this type, only the parts shown in Fig. 20 are mounted above the chassis; all the other parts are located underneath, with the chassis serving as an effective shield. Notice how parts which are connected together are placed close together; for example, the oscillator tuning condenser is alongside the oscillator coil, and the antenna transformer and tuning

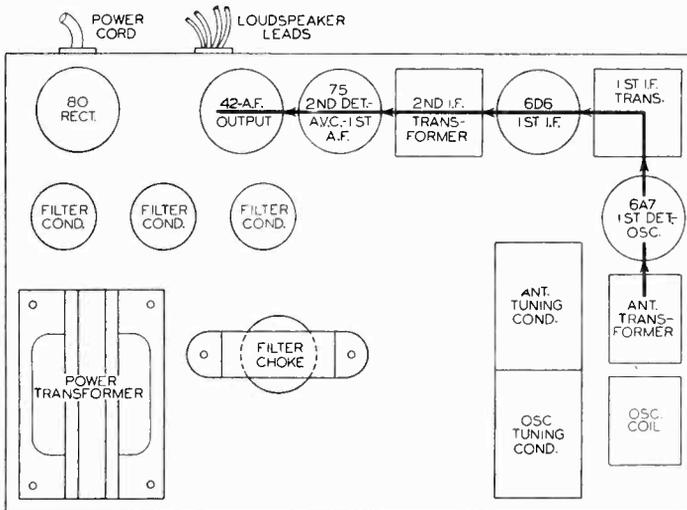


Fig. 20. Logical chassis layout (top view) for the 5-tube superheterodyne receiver circuit of Fig. 18.

able power line disturbances to the secondary winding and the receiver circuits.

We have thus made a thorough analysis of the receiver circuit. Very seldom does a practical radio man have to study a circuit as thoroughly as this, but since he may have to analyze a particular section of the circuit in which trouble has developed, it is well for you to go over complete circuits like this occasionally during your training period.

A common and widely-used arrangement of parts for a 5-tube receiver like

condenser are likewise near each other. The signal circuit tubes and transformers are arranged in the same order as on the circuit diagram, as indicated by the arrow line in Fig. 20. In any radio receiver you will find the parts so arranged on the chassis that the path taken by signal currents from tube to tube *does not cross itself anywhere, with the end of the path (the output circuit) as far as possible from the starting point (the antenna circuit); this lessens the chances for feed-back and simplifies the shielding problem.*

Notice also that the power pack com-

ponents—the power transformer, filter choke, filter condensers and the type 80 rectifier tube—are all grouped at one end of the chassis, as far as possible from the antenna input circuit, with the power cord entering the chassis just behind the rectifier tube. This lessens the chances for A.C. power line hum to get into the input circuit and be amplified by the entire receiver along with signal currents. You will find that the power transformer is completely shielded by its steel housing. As a further precaution against magnetic interaction, the filter choke is located at right angles to the power transformer.

All signal circuit coils and transformers above the chassis are in cop-

per or aluminum shields, which prevent interference effects of stray electric and magnetic fields. Since the tubes all have glass envelopes, they are likewise covered with metal shields.

Wiring and leads for all components under the chassis of this receiver will in general be as short as possible, with all chassis connections for any one stage being made to the same point to eliminate the effects of stray signal current in the chassis. If you encounter leads in a manufactured receiver which seem unnecessarily long, do not change them unless you know definitely that they are causing trouble; oftentimes it is necessary to use long leads in order to get around certain parts and eliminate interfering effects.

## TEST QUESTIONS

Be sure to number your Answer Sheet 20FR-3.

Place your Student Number on every Answer Sheet.

Never hold up one set of lesson answers until you have another ready to send in. Send each lesson in by itself before you start on the next lesson. In this way we will be able to work together much more closely, you'll get more out of your Course, and you will receive the best possible lesson service.

1. In what three different ways can undesired signal currents enter a circuit?
2. Why is a by-pass condenser commonly placed across the C bias resistor in the cathode circuit?
3. How can signal currents be kept out of the high-resistance path through the plate supply of a radio circuit?
4. Why do signal currents take the path through a by-pass condenser in preference to other possible paths between two points in a receiver circuit?

5. Can a signal current filter act both ways, preventing signal currents from getting out of a signal circuit and preventing A.C. power supply currents from getting into the signal circuit?
6. Why should all by-pass condensers in a single R.F. stage be connected to a common point in the stage?
7. Why must grid and plate leads in an R.F. stage be kept as short and as far apart as possible?
8. Is it important to ground an electric shield?
9. Name two materials which serve best as magnetic shields at high frequencies (above about 50 kc.)?
10. Why are chassis parts so arranged that the signal current path from tube to tube does not cross itself at any point, and its end (the output tube) is as far as possible from the starting point (the antenna circuit)?

## YOUR HEALTH

There is a definite relationship between a man's mental ability and his physical condition; for example, overeating is generally followed by a lazy feeling and a desire to sleep. The mind becomes less active. A headache may develop, along with a gloomy, crabby, or disgusted-with-life-in-general feeling. Certainly a man cannot do his best work when feeling this way.

Blue Mondays are quite real, and are caused by too much food and too little mental and physical exercise on Sunday, combined with a troubled sleep or too little sleep Sunday night. It takes several days for the human system to get back to normal after a week-end of excesses, so it may not be until Wednesday that you work with a clear mind. Then you find it easy to concentrate, and work becomes a pleasure. You say to yourself: "How much happier I would be if every day were like this!"

But every day *can be like this*—if you take proper care of yourself, with physical exercise each day in the open air, and a good sound sleep each night. See a doctor if your sleep is not entirely restful.

Give your health the attention it deserves, and you will be rewarded many times by increased happiness and increased success in your work.

J. E. SMITH

**THE VACUUM TUBE  
AS AN A.C. GENERATOR IN  
RADIO-TELEVISION CIRCUITS**

21FR-2

**NATIONAL RADIO INSTITUTE**

ESTABLISHED 1914

**WASHINGTON, D. C.**



# STUDY SCHEDULE No. 21

For each study step, read the assigned pages first at your usual speed. Reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind, then answer the Lesson Questions for that step. Study each other step in this same way.

- 1. Practical Importance; Coil and Condenser as Oscillator;  
How a Vacuum Tube Oscillator Circuit Works . . . Pages 1-8  
This first section is extremely important, because it gives you the basic principles upon which practically all radio oscillator circuits depend. If you understand these principles, you will not have to bother remembering the details of the many variations in oscillator circuits which are encountered in radio. Answer Lesson Questions 1, 2, 3, 4 and 5.
  
- 2. Effects of Circuit Changes; Type of Self-Excited  
Oscillators . . . . . Pages 8-14  
Here, in one brief section, are a lot of answers to that common question, "What would happen if I did this or that?" You get a lot of interesting answers, then consider briefly some of the common types of oscillators encountered in radio receivers and transmitters. Answer Lesson Question 6, 7, 8 and 9.
  
- 3. Applying Oscillator Loads; Oscillator Power Supply  
Connections; Frequency Stability; Crystal Oscillators . . . . . Pages 14-21  
If you study this section with the idea that someday you will actually be working with oscillator circuits and dealing with the various problems taken up, you will find this section intensely interesting. Try to locate the important idea in each paragraph, as an aid to understanding important facts. Answer Lesson Question 10.
  
- 4. Ultra High Frequency Oscillators; Dynatron Oscillators; Relaxation Oscillators; Audio and Beat Frequency Oscillators; A Serviceman's Modulated Test Oscillator . . . . . Pages 21-29  
Now we come to special types of oscillator circuits which, though interesting, need not be fully mastered at this time. If you get the general idea of how each of these circuits works, so that you can recognize the circuit when you encounter it, you can always return to this section later when you actually are working on these types of oscillators. Nevertheless, this section as well as every other section in this lesson will require careful study, because the subject of oscillators is so important to a radio man.
  
- 5. Mail your Answers for this Lesson to N.R.I. for Grading.
  
- 6. Start Studying the Next Lesson.

# The Vacuum Tube as an A. C. Generator in Radio-Television Circuits

## PRACTICAL IMPORTANCE OF TUBE OSCILLATORS

**R**ADIO, as a means of transmitting intelligence through space, depends upon the *production of high frequency A.C. currents*, and television requires the *production of ultra high frequency currents*. The operation, efficiency and stability of the super-heterodyne type of receiver depends upon the oscillator, a *generator of A.C. currents*. Some of the most successful electronic control systems are possible only because of the tube oscillator. Maintenance, testing and servicing of radio equipment can be satisfactorily carried out only with special oscillators called signal generators. Thus the very existence of sound and television broadcasting depends upon tube oscillators.

## THE COIL AND CONDENSER AS AN OSCILLATOR

Long before radio was even considered as a public servant, scientists knew that an A.C. current would flow through a coil when it was connected to a charged condenser. The frequency of this current, as you already

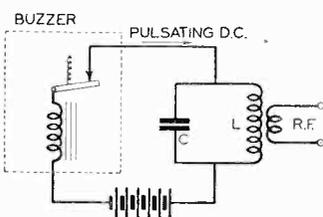


FIG. 1A. An early oscillator circuit; each closing of the buzzer contact recharges or primes condenser *C* in the coil-condenser circuit, and oscillations continue in the *L-C* tank circuit when the buzzer contact opens.

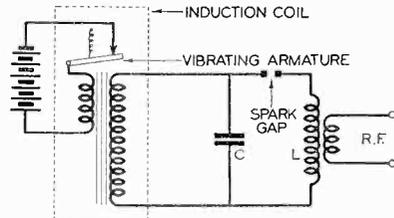


FIG. 1B. Early spark transmitter circuit. The induction coil charges condenser *C* with a high potential, the condenser voltage causes a spark to jump across the gap, and an oscillatory current flows through coil *L* and the spark gap as long as the condenser voltage is high enough to maintain the spark. Each closing of the induction coil contact recharges the condenser for another cycle of operation.

know, essentially depends upon the inductance and capacitance values. The duration of the current is fixed by the resistance in the circuit; as the energy travels back and forth between coil and condenser the losses in these two parts reduce the energy a little on each "trip," until finally the oscillations cease. Clearly it is necessary to have some means of "priming" the circuit, or feeding it with at least enough energy to overcome resistance losses; the first scheme utilized a buzzer which connected the oscillatory circuit intermittently to a battery, as in Fig. 1A, so it would recharge the condenser regularly.

*Spark Transmitter.* In order to get greater power output, radio pioneers developed the high voltage oscillatory circuit of Fig. 1B; a step-up transformer having a vibrator in series with its primary (the combination being

called an *induction coil*) primes the oscillating coil-condenser circuit hundreds of times each second. *The frequency* of R.F. oscillation is determined essentially by the values of  $L$  and  $C$ .<sup>\*</sup> Of course there were many later improvements on this early spark transmitter circuit, but these are now "ancient radio history," mentioned simply to show that the coil and condenser were *first* and *still are* the important frequency controlling units.

*Damped Waves.* Each time the condenser in an oscillating circuit is charged, the applied energy immediately starts to oscillate between condenser and coil; the resistance of the oscillatory circuit (essentially the resistance of the coil or of coil and spark gap together) causes the amplitudes of successive cycles to be reduced or damped, as in Fig. 2A. Repeated priming of the oscillatory circuit (at points 1, 2 and 3) give a series of such oscillation groups. Although this form of oscillation serves very well for code transmission, it has the very severe drawback that it creates excessive interference with adjacent-channel stations, often "riding in" with signals of other stations. Reducing the resistance of the oscillatory circuit improves matters by giving the wave form illustrated in Fig. 2B, but still the damped oscillations are unsuited for the transmission of speech and music. The reason for this is simple; the damped wave form is in itself modulated at a frequency which depends upon the number of oscillatory groups per second, and this modulation note would interfere with the program being transmitted.

A continuous, constant-amplitude wave form like that shown in Fig. 2C, free of modulation, is required as a carrier when transmitting voice, music or picture signals. This wave can be produced by a vacuum tube oscillator, a simplified circuit of which appears in Fig. 3.

*Vacuum Tube Oscillators.* In a vacuum tube oscillator the power loss due to the resistance in the L-C oscillatory circuit is *constantly* being compensated for by power which is supplied by the tube acting as an amplifier, whereas in the spark gap transmitter the losses were compensated for *intermittently*.

Let us see how a vacuum tube oscillator operates. Only electrical changes in the circuit, such as connecting the tube to its power supply, will start this vacuum tube oscillator; once oscillation starts, there is produced across the condenser (or coil) terminals an A.C. voltage whose frequency is determined by the values of  $L$  and  $C$ . The vacuum tube, acting as an amplifier, will respond to this varying grid voltage and cause the plate current to vary at the same frequency. Coil  $L_T$ , carrying this A.C. plate current, will induce into coil  $L$  by mutual induction a voltage  $e$ . Coil  $L$ , condenser  $C$  and resistor  $R$  (an apparent resistance, representing the combined resistances of  $L$ ,  $C$ , the grid-cathode path in the tube, and the resistance effects of a load, if used) act as a series resonant circuit. The reactance effects of  $L$  and  $C$  cancel each other, and the voltage  $e$  acting on  $R$  determines the amplitude of the oscillatory A.C. current. As in any simple generator-load

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<sup>\*</sup>The frequency of an oscillatory  $L$  and  $C$  circuit can be determined from the following formula:  $f = 159,000 \div \sqrt{LC}$  where  $f$  = frequency of oscillation in cycles per second,  $L$  = coil inductance in microhenrys, and  $C$  = condenser capacity in microfarads.

circuit, the current assumes a value which will make the A.C. voltage drop in  $R$  equal to the induced A.C. voltage  $e$ .

Because of resonance, the voltage across either  $L$  or  $C$  is greater than the induced voltage; this resonance-amplified voltage, applied to the grid of the tube, drives the plate current through large swings, and the increased current through the tickler coil increases the induced voltage  $e$ . The plate current swing and consequently the induced voltage  $e$  continues to increase up to the limit of the amplifying ability of the tube; the tube thus introduces into the oscillatory circuit, through tickler coil  $L_T$ , enough power to overcome power losses in  $R$  and make the oscillatory current assume the constant amplitude shown in Fig. 2C.

There are many ways of operating the oscillator tube, connecting the oscillatory circuit and producing the feed-back voltage in order to secure improved performance; these will now be studied. Remember, however, that it is essentially the values of coil  $L$  and condenser  $C$  in the oscillatory circuit which govern the frequency of a self-excited vacuum tube oscillator; these values can be adjusted to give frequencies ranging from one cycle to hundreds of kilocycles per second.

### HOW A VACUUM TUBE OSCILLATOR CIRCUIT WORKS

Most tube oscillators work alike, even though the method of feed-back, the position of the oscillatory circuit containing  $C$  and  $L$  (also called the

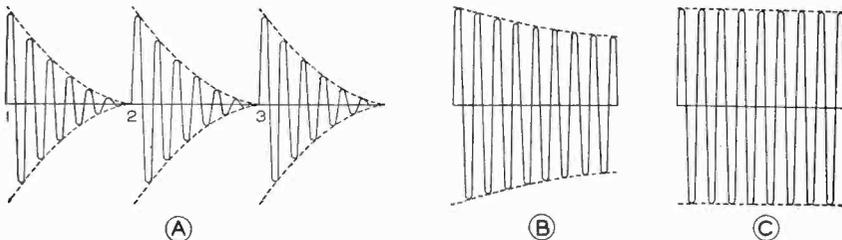


FIG. 2. Priming or recharging of an oscillator circuit gives the series or "trains" of damped oscillations shown at  $A$  and  $B$ , but present-day radio transmission calls for the constant amplitude oscillation shown at  $C$ , which can be obtained from a vacuum tube oscillator. For simplicity, only a few cycles of the oscillatory wave are shown here; actually there may be thousands of cycles of oscillations in each damped wave group, such as between points 1 and 2 in  $A$ .

tank circuit), the method of supplying the D.C. power, or the positions of the circuit components may vary. A typical oscillator circuit is shown in Fig. 4; here, as in most oscillators, the plate feeds a part of its energy back to the grid to make the tube supply an A.C. plate current. All oscillators in which the grid takes a small part of the plate circuit power are said to be self-excited.

**Power Input.** Neglecting tube filament power, the total power fed to the oscillator is the D.C. plate voltage  $V$  multiplied by the D.C. plate current  $I$ . This input power is used in several ways: 1, It overcomes resistance losses in the  $L$ - $C$  tank circuit. If coil  $L$  is coupled to a load, the effective value of  $R$  is increased and the power consumed by the tank circuit also includes useful output power or useful work. 2, The grid in an oscillator circuit is driven positive at times, causing grid current to flow, and the grid conse-

quently draws power from the plate circuit;  $\mathcal{B}$ , a part of the input plate power is used to overcome the A.C. plate resistance, thus heating the tube elements. In spite of all these power losses, from 50% to 85% of the input power can be converted into useful A.C. power in a good oscillator.

An oscillator circuit is unique in that changing any one thing in the circuit changes current and voltage conditions in the entire circuit; the reason for this is that the grid and plate circuits are linked together. A study of Fig. 4 will reveal a few outstanding facts which apply in general to all self-excited oscillators. The A.C. voltage across  $C$  or  $L$  can never have a peak voltage greater than the applied voltage  $V$ ; in fact, the peak value of  $v_T$  will always be less than  $V$  because the charging voltage of  $C$  is equal to the applied voltage *minus* the tube voltage drop. Furthermore, as the A.C. plate current *increases* to furnish more power to the tank circuit, the A.C. voltage across  $C$  or  $L$  becomes less.

A.C. voltage  $v_T$  causes an A.C. current  $i_T$  to circulate in the tank circuit, and this current induces an A.C. voltage in the grid circuit. The amount of voltage induced in the grid circuit will, of course, depend upon the mutual inductance between coils  $L_G$  and  $L$ , upon the A.C. tank current  $i_T$ , and upon the frequency of oscillation. The mutual conductance of the oscillator tube must be such that the grid driving voltage causes enough A.C. plate current to flow to maintain oscillation; in other words, the mutual conductance ( $g_m$ ) of the tube is of as much importance as the A.C. grid voltage. Oscillator action is a sort of "around the circle" affair; the grid must draw enough power from the tank circuit to swing the plate current enough to produce enough A.C. power in the tank circuit to supply the load with power and feed sufficient power back to the grid to maintain oscillation. The amount of output power can be controlled by changing the mutual conductance of the tube or by changing the mutual inductance between the plate and grid circuits.

*Starting an Oscillator.* Getting oscillations to start in a feed-back circuit like that in Fig. 4 is often explained as follows: When the tube is first connected to the power supply, the plate current starts to rise as soon as the cathode begins to emit electrons. A rising plate current flowing through coil  $L$  (see Fig. 4) induces a voltage in coil  $L_G$ . This grid voltage, acting through the mutual conductance of the tube, causes the plate current to vary; *if the voltage to the grid is so phased that the original plate current change is aided*, the plate current will build up to the full capacity of the tube, as determined by the operating voltages. This primes the tank circuit with energy and it starts to oscillate, feeding A.C. voltage to the grid and causing the plate current to increase and decrease alternately; this is the desired condition for a tube oscillator. Tank current and grid voltage continue to increase until a balance is reached, this occurring when the A.C. current in the tank circuit produces (through coupling to the grid) sufficient A.C. grid voltage for the tube amplifier to supply the tank power required by the tank circuit load and to overcome the grid and plate circuit power losses. All power is obtained from the plate supply source.

*Automatic C Bias.* Now let us concentrate our attention on grid resistor  $R_C$  and grid condenser  $C_C$  in Fig. 4. As you will shortly see, condenser

$C_C$  has a double purpose. The A.C. grid voltage  $e_G$  acts directly upon the grid-cathode of the tube because  $C_C$  has a low reactance. Thus the tube is fed with an A.C. signal which makes the grid alternately positive and negative with respect to the cathode. When the grid is positively charged, it will draw electrons from the cathode, these electrons flowing through resistor  $R_C$  on their way back to the cathode since they cannot pass through condenser  $C_C$ ; a D.C. voltage therefore appears across resistor  $R_C$ .

From the basic fact that electrons flow through a load from the negative to the positive terminal, point 1 of resistor  $R_C$  is negative with respect to point 2. If condenser  $C_C$  were absent, the current flow through  $R_C$  would be of a pulsating nature since the grid is positive for only short periods of time; this condenser, however, is charged by the voltage drop across the resistor so that when the electron flow through  $R_C$  starts to decrease, condenser  $C_C$  begins feeding electrons into  $R_C$ . The result is a steady flow of electrons through  $R_C$ , and the voltage drop across it becomes constant. Thus, in addition to the A.C. voltage supplied by  $L_G$ , a steady D.C. bias voltage is automatically supplied to the grid by grid condenser  $C_C$  and grid resistor  $R_C$ ; this voltage is called the *automatic C bias*. This makes the operating plate current set itself at a lower value.

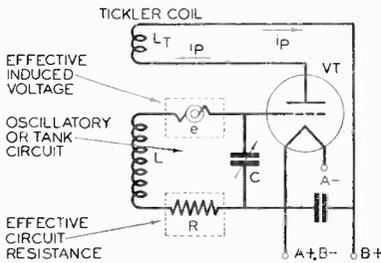


FIG. 3. Simple feed-back type vacuum tube oscillator circuit.  $R$  is not an actual resistor; it simply represents the resistance of coil  $L$  and all other resistances, apparent or otherwise, which may cause losses in the oscillatory circuit. Likewise  $e$  represents the voltage induced in coil  $L$  by the tickler coil.

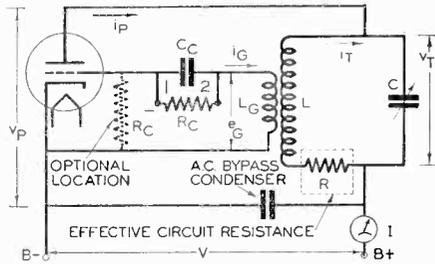


FIG. 4. An understanding of this basic self-excited oscillator circuit will simplify your study of other oscillators.  $L$  and  $C$  form the oscillatory tank circuit.  $L$  induces into  $L_G$  the required feed-back voltage;  $C_C$  and  $R_C$  provide automatic C bias for the tube.

You may consider the A.C. voltage and the automatic C bias voltage as acting independently on the tube, or you may consider both effects at the same time, as you prefer. The presence of the C bias lessens the effect of the A.C. grid voltage on the tube, actually reducing the time and the amount by which the grid is driven positive; the C bias cannot, however, completely prevent the grid from being positive, for if it could, there would be no grid current to produce the automatic C bias.

The automatic C bias can never be greater in value than the peak A.C. grid voltage. The average value of grid current multiplied by the ohmic resistance of  $R_C$  determines the C bias in the circuit. It should be pointed out that the grid bias resistor may be connected from grid to cathode (instead of across the condenser) without changing the automatic C bias.

*Operating Efficiency.* It is true that the circuit in Fig. 4 will oscillate without any form of C bias, but in this case the D.C. plate current will be so high that losses will be excessive, giving very poor tube efficiency at normal

plate voltage, and the resulting heat may melt the tube elements. The use of an automatic C bias keeps D.C. plate current at a minimum and limits the effects of the A.C. grid voltage, allowing the tube to work more efficiently. The value of  $R_C$  and the feed-back voltage are so selected that the average C bias at least equals the value for plate current cut-off (this C bias approximately equals the plate voltage divided by  $\mu$ , the amplification factor of the tube). This means that plate current flows for one-half of each cycle, and grid current flows during a much smaller part of the cycle.\* The coil and condenser tank circuit stores up energy, and because of its oscillatory nature continues to oscillate during those parts of the cycle when there is no plate current; the resulting tube efficiency is at least 50%. By making the C bias even greater than cut-off value, the time of plate and grid current flow is reduced and greater efficiencies, which may be as high as 85%, are obtained. Although increasing the C bias does increase efficiency, it naturally reduces the power output; a compromise must therefore be made between maximum efficiency and maximum power, to suit a particular need.

A fixed C bias voltage (secured from a tap on the voltage divider of the D.C. power supply) could be introduced in place of  $R_C$  and  $C_C$ , but if it is made greater than the plate current cut-off value, the oscillator will not be self-starting (the original rise in plate current will not occur). Automatic C bias is a very practical solution, for the C bias generally sets itself automatically for best operation.

*Blocking.* The ohmic value of the grid resistor in a self-excited vacuum tube oscillator (such as  $R_C$  in Fig. 4) is not critical, but if it is made too large, its D.C. voltage will build up to such a high value that the oscillator will *block* (stop oscillating momentarily) at regular intervals. Values of  $R_C$  above about one megohm cause the oscillator to block, producing a tone modulation whose frequency depends upon the time constant of the resistor-condenser combination; the value of  $R_C$  in megohms multiplied by the capacity of  $C_C$  in microfarads gives the approximate time of one blocking and starting cycle. When the value of  $R_C$  is low (5,000 to 250,000 ohms), the bias is never driven so far negative that intermittent oscillation is obtained. The grid condenser  $C_C$  can be any size sufficiently large to make its A.C. reactance in the grid circuit negligible.

In a properly adjusted oscillator, the C bias is zero at the instant of starting the oscillator, and a definite value of plate current flows. This plate current *decreases* quickly to the operating value  $I_P$  as automatic C bias comes into action, for the operating bias is now more negative. This means that if the oscillator is stopped by shorting one of the coils (or by an internal short in tank condenser  $C$ ), the plate current will *increase*; in some low power oscillators this is actually done to determine if the circuit is operating, but in high power oscillators the high plate current resulting from such a procedure might damage the tube.

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\*These pulsating grid and plate currents produce many harmonics, but the tank circuit accepts only the fundamental. Coil  $L$  suppresses and condenser  $C$  by-passes the harmonics; by using a large value for  $C$  and a small value for  $L$ , the ability of the tank circuit to reject harmonics will be improved even more.

To adjust an oscillator to best operating conditions once the load to the oscillator is fixed, it is customary to adjust the different circuit parts to the values which give minimum D.C. plate current at the desired frequency. When an oscillator is to be tuned over a wide range of frequencies by varying the tank circuit capacity, as is the usual procedure, an optimum (best) circuit adjustment for one frequency may be unsatisfactory at some other frequency. For example, if the circuit is adjusted for best operation at a high frequency in its range, the resetting of the tank circuit to a low frequency may result in too low output or no oscillation due to too low a feedback voltage. When a wide range of frequencies is to be covered, it is best to make optimum adjustments at a mid-frequency; this should provide ample feed-back excitation at the lowest frequencies.

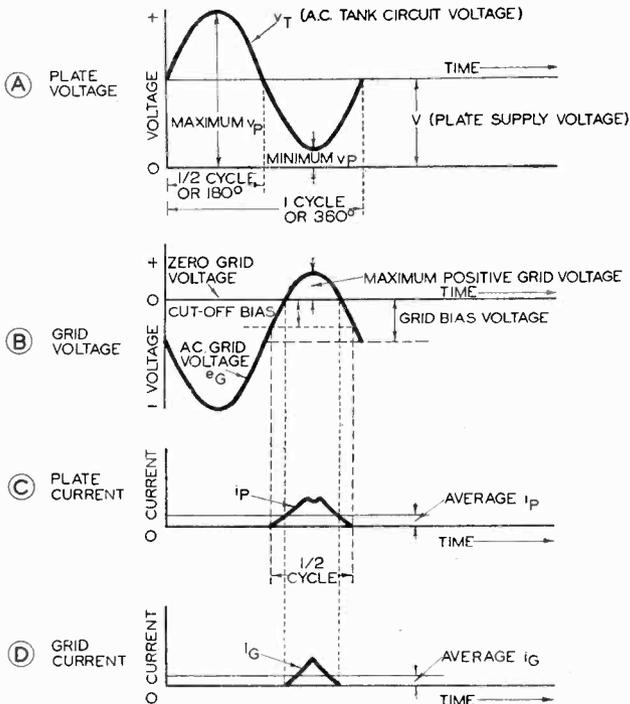


FIG. 5. These plate and grid currents and voltages represent operating conditions in the oscillator circuit of Fig. 4. Remember that graphs like these are always read from left to right. When comparing two voltages, that one which reaches a positive peak *closest* to the vertical reference line is said to *lead* the other; thus,  $v_T$  in *A* leads  $e_g$  in *B*.

**Current and Voltage Relations.** The curves in Fig. 5 present a picture of the grid and plate voltages and currents in the oscillator circuit of Fig. 4 during operating conditions; these curves are characteristic of all types of oscillators. Notice that although the applied plate voltage  $V$  in Fig. 4 is a D.C. voltage, the voltage  $v_T$  across the tank circuit (the plate-to-cathode A.C. voltage component) has essentially the sine wave form shown in Fig. 5A. Although the A.C. plate voltage swings above and below the D.C. plate supply voltage value, it never is driven to zero because of the A.C. plate resistance.

In Fig. 5B is shown the A.C. grid voltage curve; you will notice that the A.C. grid voltage reaches its maximum positive value half a cycle (180 degrees) after the A.C. plate voltage reaches its maximum, or in other words, the A.C. grid voltage lags 180 degrees behind the A.C. plate voltage; coils  $L_G$  and  $L$  in Fig. 4 are so connected that this phase relation is produced. In this way the grid can be driven positive during the time when the plate-to-cathode voltage is low; current thus flows in the plate circuit when the grid swings positive, and the tank circuit absorbs or stores up energy. This energy is released because of the oscillatory action (also called the flywheel effect) of the tank circuit when plate current drops to zero, completing the A.C. cycle and producing the sinusoidal voltage wave shown in Fig. 5A.

Looking at Fig. 5C, note that plate current  $i_p$  begins to flow when the instantaneous A.C. grid voltage becomes more positive than the cut-off value of C bias. Grid current (Fig. 5D) flows only when the A.C. grid voltage overcomes the effective C bias, and makes the grid positive with respect to its cathode. Thus plate current flows for a longer part of a cycle than grid current. Since the flow of grid current robs the plate circuit of current, a dip appears in the peak of the plate current curve; a dip will also appear in the grid current curve if a large amount of secondary emission takes place at the grid of the tube. The average value of plate current controls the amount of power taken from the supply; in the grid circuit the current through the grid resistor is averaged out by the smoothing condenser. By studying curves A, B and C together, you can see that the minimum plate voltage value is very important in controlling plate current; the larger this minimum voltage, the larger the plate current.

*An Important Oscillator Requirement.* Any self-excited oscillator *must* have the same phase relationship between grid and plate A.C. voltages that exist in an ordinary amplifier—the A.C. grid voltage must be 180 degrees out of phase with the A.C. plate voltage. For example, in an amplifier having a resistance load, a positive swing in grid voltage makes the plate current increase, increasing the IR drop in the load and therefore decreasing the plate-to-cathode voltage; the tank circuit of an oscillator is in effect a resistance load at resonance, and the same relation must therefore exist—grid voltage *increase* causes plate voltage *decrease*.

## EFFECTS OF CIRCUIT CHANGES

Changes in the adjustments of various parts of an oscillator circuit will naturally affect the oscillatory condition; the following discussion of these changes will in general apply to all self-excited oscillators.

*Static and Oscillating Plate Current.* If the coupling between plate and grid in an oscillator is reduced or the grid excitation supply is shorted, the circuit will not oscillate, and the plate current will assume a *static* value which is governed essentially by the plate supply voltage and the D.C. plate resistance (the grid bias being zero for a self-excited circuit which is not oscillating). If the circuit was originally adjusted for optimum efficiency as an oscillator, an unusually high plate current will flow; in high power oscillators the tube will overheat and plate and filament will be destroyed by heat. In the case of low power oscillators such as those found

in testing equipment and in superheterodyne receivers, the excessive plate current may weaken the cathode emission qualities of the tube. When the fault is remedied and oscillation starts again, the D.C. plate current will immediately drop. In the case of high powered oscillators, the plate current should immediately assume its normal value (much lower than the estimated static value); if plate current is above normal the circuit should be disconnected from the supply at once.

*Effect of Plate-to-Grid Coupling.* There can, of course, be no oscillation when plate-to-grid coupling is very loose; as coupling is gradually increased, oscillation will start at a definite point and the D.C. plate current will drop from the static value to the operating value. Further increases in coupling simply increase the grid excitation (increase the grid voltage swing), thus increasing the rectified grid current value and shifting the operating grid voltage more negative, but D.C. plate current remains practically constant for all values of coupling which maintain oscillation. When coupling is increased beyond a certain critical point, the time duration of the plate current pulses becomes such a small fraction of a cycle that the tank circuit can no longer supply sufficient energy to build up the voltage wave for the rest of the cycle, and oscillation stops; plate current then rises to the static value. Any change in coupling also changes the inductance and resistance of the tank circuit, thereby changing the frequency of oscillation slightly.

*Effect of Loading the Tank Circuit.* The tube circuit of an oscillator must supply a certain amount of power to the tank circuit in order to maintain oscillation; this power comes from the D.C. supply, for there is no other source of power in a self-excited oscillator. If a load is coupled (usually inductively) to the tank circuit, the tube must also supply the extra power demanded by the load, and this extra power must likewise come from the D.C. supply. This means that the D.C. plate current *increases* in value when a load is applied to the tank circuit of a self-excited oscillator (since power is equal to voltage multiplied by current, and the D.C. supply voltage is here essentially constant).

The A.C. plate current must increase either in amplitude or in the duration of a pulse in order to produce the increase in average or D.C. plate current; actually both factors increase, for the reduced tank current lowers the grid excitation, making the C bias more positive and allowing a current pulse of greater amplitude to flow for a slightly greater fraction of a cycle.

*Effect of Plate Supply Voltage.* Increasing the D.C. plate supply voltage of a self-excited vacuum tube oscillator increases the A.C. tank voltage, thereby raising the tank current and *increasing the power output*. The higher tank current also serves to increase the grid excitation, the rectified grid current, the automatic C bias and to a slight extent the D.C. plate current. If a self-excited oscillator will not start by itself, increasing the plate supply voltage will generally cure the trouble.

*Effect of Increasing C Bias Resistor.* Increasing the ohmic value of the C bias resistor makes the grid bias voltage more negative, with the result that grid and plate currents flow for smaller fractions of the cycle. Other effects are a lowering of the D.C. plate current, a decrease in the amount

of energy supplied to the tank circuit and an increase in oscillator efficiency. These changes are not very great, for the increased efficiency tends to feed more grid A.C. voltage to the oscillator input, offsetting the increased bias. If the resistor value is increased too much, repeated blocking will take place, and stability (the tendency to remain in oscillation) will be reduced.

*General Circuit Effects.* It is desirable to have a high inductance-to-resistance ratio (high Q factor\*) in the tank circuit of an oscillator in order to improve the flywheel action of the circuit and reduce the generation of harmonics in the tank circuit. On the other hand, a high Q factor gives a high tank current, increasing circuit losses unnecessarily. Circuits which are heavily loaded have a low Q factor and are very unstable, hence a compromise involving a mid-value of Q factor is usually required.

Changing the capacity in the oscillatory circuit naturally changes the frequency of oscillation; as the condenser capacity is reduced, the frequency goes up. The feed-back voltage or grid excitation depends upon the coupling (usually fixed), upon the tank current and upon frequency, hence tuning an oscillator to a higher frequency increases the feed-back voltage. Although the losses in the tank circuit increase with frequency, causing the tank current to reduce, this is more than overcome by increased feed-back. Increased feed-back makes the oscillator generate more power, but this effect is somewhat reduced by the resulting increased grid current and increased C bias voltage. By limiting feed-back, it is possible to secure an oscillator which can be tuned over a wide range of frequencies.

On the other hand, an oscillator set for optimum conditions at a high frequency may stop oscillating at the lower frequencies because of insufficient feed-back. In superheterodynes an oscillator may fail at lower frequencies because of poor tube emission, low plate voltage, or too high a C bias resistor value. Remedies include trying a new tube at normal voltage or adjusting the plate voltage and C bias resistor values to give oscillation at the lowest frequency desired with a given tank circuit coil and condenser.

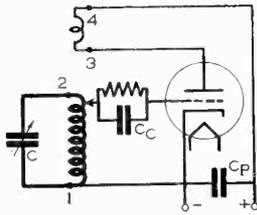
## TYPES OF SELF-EXCITED OSCILLATORS

As I pointed out before, the basic oscillator ideas which you have been studying apply in general to all types of self-excited oscillators. Now we will take up the common variations of the self-excited oscillator circuit.

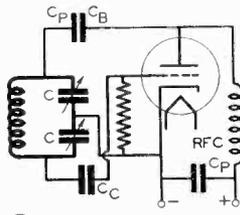
*Tuned Grid Oscillator.* This is of the form shown in Fig. 6A. The tank circuit is in the grid circuit, plate A.C. power being fed to it by inductive coupling. The circuit as shown here is for low power oscillators, such as those found in superheterodynes and in test oscillators. To prevent over-excitation and to permit greater tank current, the grid is often connected to a tap on the tank coil; in this way the tank circuit can be inductively loaded, and the grid excitation can be adjusted to a satisfactory minimum value.

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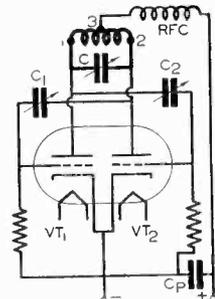
\*A parallel resonant circuit acts as a large resistance, even though the circuit parts themselves have comparatively low resistances. The number of times the resistance of a parallel resonant circuit has been increased by tuning is called the *Q factor* of the circuit; this Q factor is equal to  $\omega L \div R$ , where  $\omega$  is 6.28 times the frequency in cycles,  $L$  is the inductance of the coil in henrys and  $R$  is the resistance of the resonant circuit in ohms.



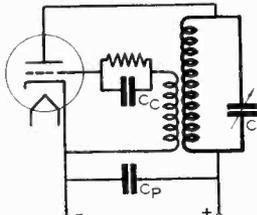
(A) TUNED GRID OSCILLATOR



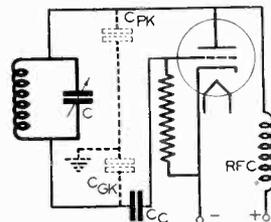
(E) COLPITTS OSCILLATOR



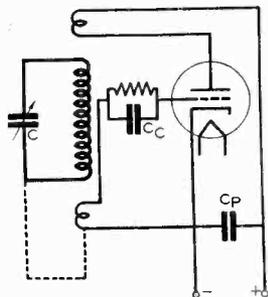
(I) TUNED PLATE, CROSS-  
FED PUSH-PUSH  
OSCILLATOR



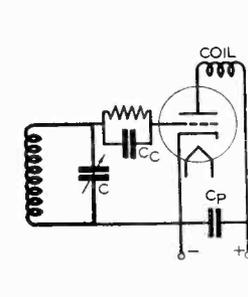
(B) TUNED PLATE OSCILLATOR



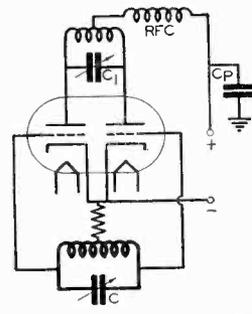
(F) ULTRA-AUDION OSCILLATOR



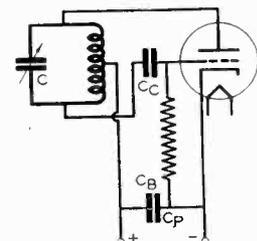
(C) MEISSNER OSCILLATOR



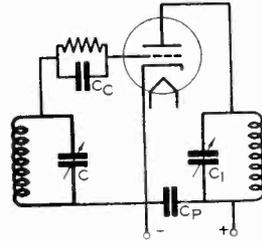
(G) TUNED GRID, INDUCTIVE  
PLATE OSCILLATOR



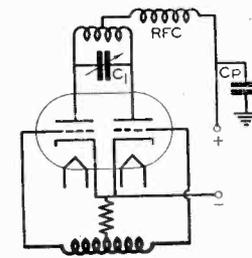
(J) TUNED GRID, TUNED PLATE  
PUSH-PUSH OSCILLATOR



(D) HARTLEY OSCILLATOR



(H) TUNED GRID, TUNED PLATE  
(ARMSTRONG) OSCILLATOR



(K) TNT PUSH-PUSH  
OSCILLATOR

FIG. 6. Simplified schematic circuit diagrams of eleven different types of self-excited vacuum tube oscillators. Heavy lines indicate the coil and condenser tank circuit combination which controls the frequency of the oscillator. Although there are many possible variations of each of these circuits, you can identify a particular oscillator by the way in which its frequency-controlling tank circuit is connected to the grid and plate electrodes of the tube. Condensers marked  $C$  are frequency-controlling condensers;  $C_1$  and  $C_2$  are oscillation-controlling condensers;  $C_c$  is the low A.C. reactance condenser, which also smooths the rectified grid current;  $C_p$  is a low reactance A.C. by-pass condenser;  $C_B$  is a D.C. blocking condenser. When a condenser is double-marked, as  $C_B$  and  $C_p$ , it acts as a blocking condenser for D.C. and as a low reactance condenser for A.C.

*Tuned Plate Oscillator.* You are already quite familiar with this simple but effective oscillator circuit, shown in Fig. 6B and also in Fig. 4. The tank tuning condenser is in the plate circuit; changing the mutual inductance between grid and plate coils changes the grid excitation. This circuit is better than that in Fig. 6A for high frequency power generation. Often the plate is connected to a tap on the tank coil to get a better impedance match between the tank circuit and the A.C. plate resistance of the tube;\* this gives greater efficiency.

*Meissner Oscillator.* This circuit, named after its inventor, is shown in Fig. 6C; the fact that feed-back and grid excitation are essentially independent of each other is an outstanding feature. Though not intended for high power requirements, the circuit is easily adjusted for optimum conditions, and gives highly stable operation. The tank circuit is usually coupled loosely to the grid and plate coils, giving the tank coil and condenser practically perfect control of the oscillating frequency. The isolation of the tank circuit also permits grounding of the condenser without affecting the supply circuits or R.F. circulating currents.

*Hartley Oscillator.* Also named after its inventor, the Hartley is perhaps the most widely used oscillator in the radio field. The basic circuit is given in Fig. 6D. The Hartley oscillator can always be identified by the fact that *the tank circuit is in both the grid and plate circuits, the division of voltages being made by a tap on the tank coil.* This tank coil tap, which connects through an R.F. by-pass condenser to cathode and ground, divides the relative plate and grid R.F. voltages without disturbing the frequency of oscillation. Moving the tap away from the plate reduces the grid excitation. The optimum condition, when D.C. plate current is at a minimum value, is easily obtained. Additional improvement in efficiency is often realized by connecting the plate to a movable tap on the tank coil, especially when low plate impedance tubes are used.

*Colpitts Oscillator.* Where highly stable oscillators are needed, this circuit (shown in Fig. 6E and named after its originator) is widely used. A unique feature is the use of two tuning condensers to divide the R.F. voltages between plate and grid circuits; individually the condensers determine the plate and grid A.C. voltages, but together in series they determine the frequency of oscillation. Decreasing the grid to cathode capacity increases the grid excitation, but the plate to cathode capacity must then be increased to maintain the frequency of oscillation. These adjustments are usually carried out until minimum D.C. plate current is obtained. A low-capacity variable condenser is often connected in parallel with the tank coil to make the final frequency adjustment. Because the division of A.C. plate and grid voltages is independent of frequency, the inductance can be changed from one frequency band to another without affecting the adjustment for best operating conditions. In a few of the low powered Colpitts oscillators the correct distribution of plate and grid capacity is determined beforehand and the two units ganged (their rotors mounted on the same shaft) for quick variation of frequency in a given band.

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\*With this connection the peak amplitude of the tank voltage will be greater than the applied D.C. plate voltage.

*Ultra-Audion Oscillator.* As one of the oldest of oscillator circuits, this seems to have been used long before the reason why it worked was known. The tank circuit is connected between grid and plate, the D.C. plate current being kept out of the grid by a blocking (grid) condenser, as shown in Fig. 6F. This circuit oscillates because the plate-to-cathode and grid-to-cathode tube capacities (indicated as  $C_{PK}$  and  $C_{GK}$ ) form an A.C. voltage divider which makes the circuit a variation of the Colpitts oscillator. At very high R.F. values the tank tuning condenser may be eliminated, the tube capacities serving both for voltage division and for tank tuning. The circuit may be adjusted for the optimum operating condition by varying the plate voltage and the grid resistor value. A midget type variable condenser is often connected between the grid end of the tank circuit and the cathode, to give control of the intensity of oscillation.

*Tuned Grid, Inductive Plate Oscillator.* This circuit, shown in Fig. 6G, is on the whole an undesirable (parasitic) oscillator, more often eliminated from than introduced into a system; it is one of the reasons for oscillations in amplifying stages. The operation of the circuit is such that when the plate load is an inductance, the A.C. voltage which is fed back to the grid circuit through the grid-to-plate capacity of the tube is *in phase* with the A.C. grid voltage. This feed-back voltage thus *aids* the A.C. grid voltage, resulting in large sustained plate current swings and a transfer of power from the plate circuit to the grid circuit. If the plate inductance is large (high reactance), sufficient voltage will be fed back to overcome grid circuit losses and sustain oscillation. The frequency is, of course, controlled by the oscillatory circuit connected to the grid. The oscillations can be reduced by shunting the plate coil with a resistor, by introducing a resistor into the grid load, by reducing the plate voltage, by introducing plate-to-grid capacity-bucking components (so called neutralizing circuits), and by using screen grid tubes. However, the tuned grid, inductive plate oscillator does serve a highly useful purpose, when the tank circuit is replaced with a crystal oscillator.

*Tuned Grid, Tuned Plate Armstrong Oscillator.* The so-called Armstrong (the inventor) tuned grid, tuned plate oscillator is shown in Fig. 6H. This circuit is essentially like that shown in Fig. 6G, except that a parallel resonant plate circuit is used to create the inductive plate load which is necessary for oscillation. When a parallel resonant circuit is tuned slightly above the frequency of the R.F. source, its impedance becomes highly inductive. In this circuit the grid is tuned to the desired frequency and the plate tuned for minimum D.C. plate current, which automatically sets the plate tank circuit at a high inductive value. The grid tank circuit is often replaced with a crystal oscillator and the plate circuit adjusted approximately to minimum plate current; this gives a widely used and very important crystal circuit, to which we will return shortly.

*Tuned Plate, Cross-Fed, Push-Push Oscillator.* In Fig. 6I you have a circuit which is often used where both high efficiency and stability are desirable. Although a double triode tube is shown, two similar triode tubes are more often used. Each tube is automatically biased to cut-off value and each grid is fed 180 degrees out of phase with the other, so that when one tube is drawing plate current, the other tube is idle. In this way the

tank circuit, which is connected directly to the two plates, is receiving energy for both halves of the cycle. If during one-half of the cycle point 1 is positive with respect to point 2, then point 2 will be positive with respect to point 1 for the next half cycle. Point 1 being coupled to the grid of tube  $VT_2$  through a small variable condenser  $C_2$ , this tube receives an A.C. grid voltage which is *in phase with* its normal excitation. Point 3 and the cathode are at the same R.F. potential. Another way of looking at this circuit is to remove one tube; now we have in effect an ultra-audion circuit. Increasing the capacity of feed-back condensers  $C_1$  and  $C_2$  increases grid excitation; both condensers are usually adjusted for minimum D.C. plate current. This circuit is widely used at high and ultra-high frequencies, because the tank circuit is shunted by only one-half the plate-to-cathode capacity; this allows operation at higher frequencies without encountering undesirable tube capacity shunting effects.

*Tuned Grid, Tuned Plate Push-Push Oscillator.* Here is another favorite dual tube circuit, relying on grid-to-plate tube capacity for feed-back; its circuit appears in Fig. 6J. This circuit is essentially like the Armstrong circuit, except that the plate tank circuit is being fed with energy on both halves of the A.C. cycle, giving greater efficiency. Like all twin tube oscillators, it is a valuable circuit at high frequencies where tube capacities become important. The circuit frequency depends upon the grid tank circuit; the plate tank circuit is adjusted for minimum plate D.C. current, which sets the plate load to an inductive condition.

*TNT Push-Push Oscillator.* This circuit, shown in Fig. 6K, uses a coil alone in the grid circuit, relying on distribution coil capacity for grid resonance; the frequency of oscillation is therefore limited to a narrow band. This gives a high L/C ratio for the tuning circuit in the grid, which is excited to large A.C. voltage values by the limited tube capacity feed-back. The plate tank circuit is tuned to an inductive value.

*Direction of Plate and Grid Coil Windings.* When the plate is inductively coupled to the grid, the feed-back voltage may be either in or out of phase. If out of phase, the oscillator will not work; in this case, reversing connections to one of the coils will make the oscillator start. A good rule to follow in connecting oscillator coils (assuming that the two coils are wound in the same direction) is to make the two outer coil terminals or leads (1 and 4 in Fig. 6A) the cathode and B+ terminals, leaving the grid and plate connections for the inner two coil ends, 2 and 3.

## APPLYING OSCILLATOR LOADS

Oscillator loads must be applied in such a manner that load changes have a minimum effect on frequency change, and do not destroy the stability of the oscillator. As you know, the tank circuit current and voltage are sinusoidal in wave form, and therefore are of the fundamental oscillator frequency. To feed the fundamental oscillator signal to the load, then, it is best to couple the load to a *tank circuit inductance*; several different methods are used.

*Inductive Coupling to Load.* This method, shown in Fig. 7A, is very commonly used when the effective load resistance is low (is 500 ohms or

less). The effect of inductive coupling is that of a resistance in series with the tank coil and condenser, the resistance increasing in value as the mutual inductance is increased. Of course, this method of coupling affects the resonance of the tank circuit, shifting the frequency of oscillation slightly when load is applied or varied.

*Capacity Coupling to Load.* To be sure, high resistance loads can be coupled inductively to the oscillator if a large mutual inductance is provided; in cases like this, however, the capacity coupling method shown in Fig. 7B is more often employed. The effect of capacity coupling is that of a high resistance placed in parallel with all or a part of the tank circuit. Any parallel resonant circuit acts as a pure resistance at resonance; connecting a resistance load in parallel merely decreases the net value of resistance. This loading may be increased (by reducing the resistance of the load) nearly to the point where the circuit will no longer maintain its self-excitation. The electrical value of coupling condenser  $C$  is such that its reactance is negligible with respect to the reactance of the load, and hence  $C$  does not affect the loading on the oscillator. The loading can be increased very effectively, in cases where the value of the load is fixed, by moving the load tap on the tank coil away from the R.F. ground.

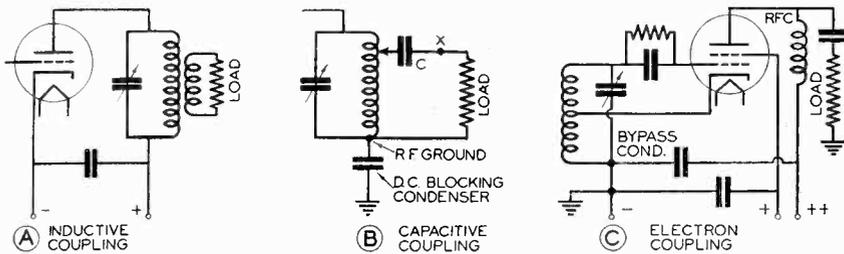


FIG. 7. Three common methods of coupling a load to an oscillator; as a rule it is best to apply the load to a circuit which has a minimum of effect upon oscillator stability and frequency.

After an oscillator is loaded, it must be adjusted for optimum operation by changing the grid bias resistor value and by adjusting the grid excitation until minimum D.C. plate current is obtained, making sure that the expected power output is obtained at all times. Where the effects of the load are to be minimized, as when the oscillator is being tuned over a given band, a series resistor of high value is placed at point X in Fig. 7B, or the inductive coupling to the load is reduced. Of course, this cuts down the available output power, but stability is far more important than high power in the superheterodyne receiver and the test oscillator, where capacity coupling is often used.

*Electron Coupling to Load.* Practical electron coupled oscillator circuits like that shown in Fig. 7C were first introduced by Dow, and hence are sometimes called *Dow oscillators*. These circuits are readily identified by their use of a multi-element tube in which the cathode, first and second grids are the basic oscillator elements (acting as the oscillator cathode, control grid and anode); these oscillator elements produce, just beyond the second grid, a pulsating space charge which acts as a virtual cathode. The plate of the tube, which is at a high positive potential and connects

to the load through a coupling condenser, attracts electrons from the virtual cathode and passes them on to the load. The term "electron coupling" comes from the fact that a stream of electrons is the only coupling between the anode (second grid) of the oscillator circuit and the plate in the load circuit. The circuit in Fig. 7C is that of a Hartley oscillator with the second grid placed at an R.F. ground potential so it acts as a shield between the plate (or load) and the oscillator. Electron coupling gives a high stability with a minimum of load reaction on the oscillator. Although the Hartley oscillator is shown in Fig. 7C, any of the other oscillator circuits may just as well be used.

*Best Position for Load.* In loading one of the standard oscillator circuits, load that tank circuit which least affects the oscillator. In Figs. 6A to 6G, of course, the load is applied to the only tank circuit present; in Fig. 6G, however, the load might be coupled to the plate coil if this did not reduce the reactance of the coil too much. In Figs. 6H to 6K inclusive, the plate coil would be loaded, the plate tank being retuned for minimum D.C. plate current.

## OSCILLATOR POWER SUPPLY CONNECTIONS

Oscillator power supplies can be considered to be essentially at ground potential, for batteries or A.C. power packs have a fairly high capacity with respect to ground. If the oscillator is not "tied down" (connected) to the same ground as the power supply, anything coming near the oscillator chassis and any slight circuit change may cause undesirable changes in frequency and power output. The power supply circuits for the various tube electrodes may introduce undesirable paths for the A.C. currents; filters and chokes are therefore used extensively to keep A.C. power where it belongs. Each type of oscillator can be fed with power from the power supply either through some signal component (known as the series feed method) or directly through a choke which keeps signal currents out of the supply (known as the shunt or parallel feed method). To get definite phase relationships the plate, the cathode or the grid may be tied to ground.

Oscillators shown in Figs. 6A, 6B, 6C, 6D, 6G, 6H, 6I, 6J, 6K, 8A, 8C and 8D have series fed plate supplies; note that in each case the D.C. plate current flows through a circuit part which is essential for oscillation. Oscillators shown in Figs. 6E, 6F and 8B are parallel or shunt fed, the D.C. plate current being fed through an R.F. choke for high frequencies, and through an A.F. choke for low frequencies. The choke allows the plate to assume an A.C. potential without feeding A.C. into the plate supply.

A study of the power supply connections in the various forms of the widely used Hartley oscillator will bring out some important principles which apply to all oscillators. Although the circuits shown in Figs. 8A, 8B, 8C and 8D may at first glance appear different, they are all Hartley oscillators because the tank coil is in both grid and plate circuits in each case, with the mid-tap of the tank coil connected to the cathode either directly or through a condenser to divide the two circuits.

*Plate Grounded.* The Hartley circuit in Fig. 8A contains a filament type tube with the plate connected directly to ground; the grid and cathode

alternately acquire positive and negative potentials, which are always out of phase with each other, and oscillation is therefore maintained. Since the filament must receive D.C. power from its supply, a choke is introduced to make the filament assume an A.C. potential and to keep the A.C. signal from feeding into the supply. The use of a heater type tube would eliminate the need for the cathode choke, as then the cathode would definitely be isolated from the A supply.

*Filament Grounded.* Figure 8B shows another filament type tube in which the filament is fed with A.C. and has its mid-point tied to ground. Feed-back condenser  $C$  provides an A.C. path between grid and plate circuits but blocks the flow of D.C. current to ground. The tank coil center tap is grounded directly to the mid-tap of the filament resistor; each section of this resistor is shunted by a condenser to allow signal currents to flow to the filament proper.

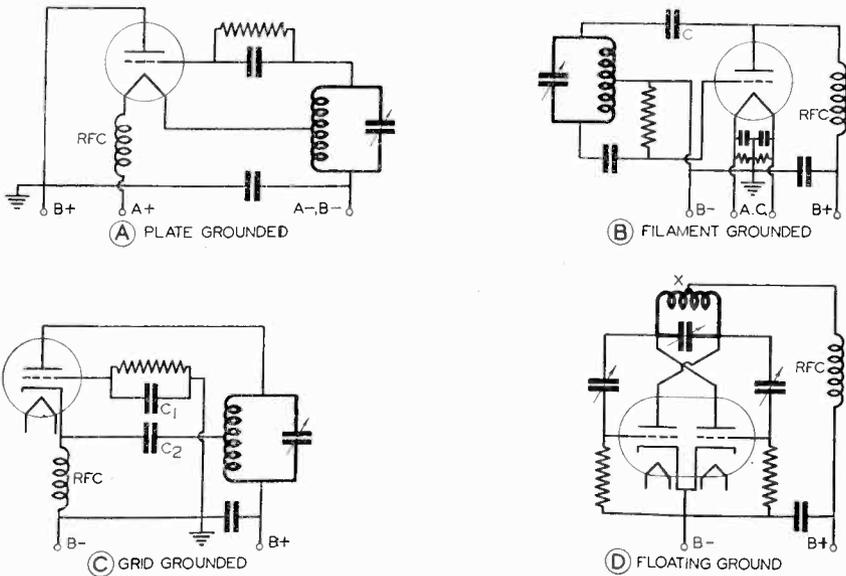


FIG. 8. Four methods of making ground connections in Hartley oscillator circuits are shown here. The frequency-controlling circuit is shown in heavy lines in each case.

*Grid Grounded.* Rarely is the grid grounded, but if it were necessary, the circuit shown in Fig. 8C could be used. As far as A.C. signals are concerned, the grid is tied to ground through  $C_1$  (shunting the grid resistor), and the mid-tap of the tank coil is connected to the cathode through condenser  $C_2$ , which prevents D.C. from flowing back to ground. The cathode circuit choke coil  $RFC$  allows the cathode to assume an A.C. potential without interfering with the flow of direct current from ground to cathode to plate to  $B+$ .

*Floating Ground.* Figure 8D illustrates a rather well-known circuit, often called the *push-pull* or *push-push Hartley oscillator*, in which a floating R.F. ground connection to the tank coil is used. Tap  $X$  is placed as close as possible to the electrical center of its coil, making the voltages

across each half of the coil practically equal at all times and opposite in polarity. But these voltages must be *exactly* equal, each section of the double triode tube getting equal excitation and equal plate currents, if the optimum operating condition is to be obtained; here is where choke RFC enters into the picture. Any slight difference in these tank coil voltages results in a small R.F. voltage drop across the choke, and this drop has the effect of swinging point X to the exact electrical mid-point of the tank coil.

## FREQUENCY STABILITY

When an oscillator circuit is adjusted to a desired frequency shortly after it has been placed in operation, and a check of frequency is made an hour or so later, you will generally find that the frequency has changed or *drifted*. This is highly undesirable, for an oscillator should maintain as near constant frequency as possible. There are three causes for frequency drift in a self-excited oscillator: 1, *Changes in the electrical values of parts in the oscillator circuit*; 2, *changes in load*; 3, *changes in tube constants*, due to power supply voltage changes or to the effects of causes 1 and 2.

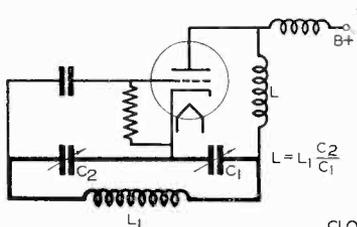


FIG. 9A. Coil  $L$  has been added to this Colpitts oscillator to minimize frequency drift which is caused by variations in the spacings between electrodes as tube temperature varies.

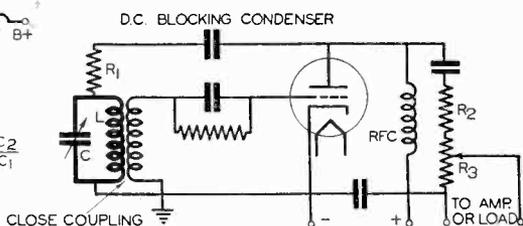


FIG. 9B. The insertion of resistor  $R_1$  in this tuned plate oscillator improves the frequency stability of the circuit, although its use cuts down the output power.

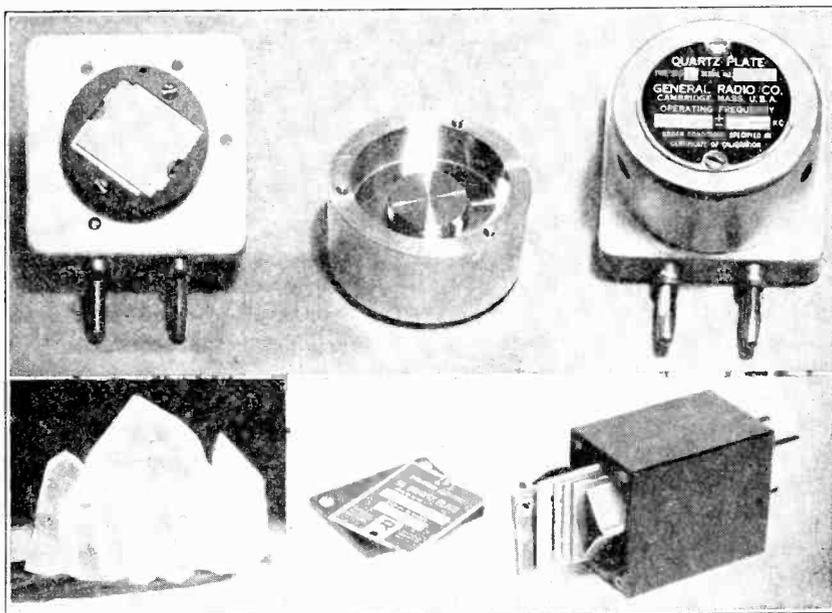
Assuming that sturdy, well designed coils, condensers and resistors which will not change in value of their own account are used, the greatest amount of frequency drift will be caused by *electrical changes* due to temperature changes; room temperature may fluctuate, and parts may heat up because of current passing through them. The solution lies in selecting parts on which temperature has little or no effect; if this proves insufficient, those parts which are most sensitive to temperature can be placed in a temperature-controlled oven.

Effects of *variations in load* can be reduced by keeping the coupling to the load as loose as possible or by using an ordinary amplifier, often called a *buffer* in this particular case, to separate the oscillator tube from its load.

*Changes in tube constants* are not so easily remedied; the A.C. grid-to-cathode resistance ( $R_g$ ) and the A.C. plate-to-cathode resistance ( $R_p$ ) have been mathematically proven to be the important factors. For a given oscillator tube and circuit the value of  $R_g \div R_p$  is always a definite value regardless of the condition in the circuit; keeping the change in one as low as possible thus keeps the other one fixed and insures good stability. Frequency drift is further reduced by using a low loss tank circuit (with high  $Q$ ), which in turn calls for a small load and a low loss coil and condenser.

The use of high values of grid bias resistors (but not so high that they block the circuit) makes  $R_g$  assume a high and practically constant value. If a series resistance of high value is connected between the plate and the B supply (or between the plate tank circuit, if used, and the B supply), this resistor will tend to absorb any changes in plate voltage, leaving  $R_p$  constant.

When high Q tank circuits are used, the frequency can be made practically independent of the values of  $R_p$  and  $R_g$  by inserting coils or condensers (or both) in series with the grid or plate leads (or both) of the tube. An example is shown in Fig. 9A for a Colpitts oscillator; the electrical value required for the added coil  $L$  is determined by the values of  $C_1$ ,  $C_2$  and  $L_1$ .



*Top row:* Three views of a General Radio type 376-L quartz plate (crystal) holder. The quartz plate itself can be seen inside its fiber mounting ring in the view at the left. This unit is of the plug-in type; one prong connects to a metal electrode beneath the quartz plate, while the other prong makes contact with the machined metal cover (top center) which serves

as the upper electrode. *Lower left:* Natural crystals of Brazilian quartz, from which quartz plates used in crystal oscillators will be cut. (Photo supplied through courtesy of Bliley Electric Co.). *Lower right:* A Western Electric quartz plate in its holder; this unit is also of the plug-in type. The quartz plate is here held in position between the two metal electrodes by a spring-steel plate bent at a right angle.

A more practical method, intended for use in cases where the tube capacities are relatively unimportant, is given in Fig. 9B; there is a low loss (high Q) tank circuit ( $LC$ ), a high resistance ( $R_1$ )\* in the plate-to-tank circuit to minimize the effects of voltage and A.C. plate resistance fluctuations, and a small load.  $R_2$  being much higher in ohmic value than  $R_3$  and both being high with respect to the A.C. plate resistance, changes in the load have very little effect on circuit stability. This circuit sacrifices power for good frequency stability. In general, however, any oscillator

\*  $R_1$  should not be so large that it stops oscillation.

should be allowed to heat up from one to two hours before use, where minimum frequency shift is essential.

## CRYSTAL OSCILLATORS

*Frequency Stability Requirements.* In radio transmission, frequency stability becomes of extreme importance. Each year governmental agencies, such as the Federal Communications Commission in the U. S. A., are demanding that frequency drift be reduced and that the exact frequency be nearer to the assigned station value. The present permissible drift is 20 cycles in the broadcast band; if a certain station operates at 1,000,000 cycles, this means the transmitter carrier frequency drift must be less than .005 per cent. Frequency-monitoring equipment used in the station must have even greater stability than this. It is difficult to design self-excited oscillators which will hold their frequency this closely but fortunately the quartz crystal, when properly cut and ground to size, mounted in a holder located in a constant temperature oven and connected into a tube circuit, gives an oscillator which has acceptable frequency stability.

*Types of Crystals.* Crystals are solids and therefore have height, width and depth; lines parallel to these three dimensions of a crystal are called the crystal axes. There is the *X or electrical axis*, the *Y or mechanical axis* and the *Z or optical axis*, all at right angles to each other. By cutting a quartz crystal into small slabs across the *X axis*, a so-called *X-cut crystal* is obtained; by cutting across the *Y axis* a *Y-cut crystal* is obtained; *Z-cut crystals* are of little importance in radio. When one of these crystals is placed between two metal plates, one surface of the crystal being about .003 inch away from one of the plates to allow the crystal to vibrate freely, an A.C. voltage applied to the two plates will cause the crystal to vibrate at a frequency determined chiefly by the thickness of the cut crystal. For an *X-cut crystal* the frequency in cycles is approximately 3,000,000 divided by the crystal thickness in millimeters; for a *Y-cut crystal* the frequency in cycles is about 2,000,000 divided by the thickness in millimeters. Crystals which vibrate up to 5,000 kc. are quite easily made.

The really important feature of a crystal, as used in a crystal oscillator circuit, is that it acts as a *tank circuit which has a better frequency stability than any other practical device*; only changes in crystal temperature produce frequency drift, and temperature can be easily controlled.\*

*Study of a Crystal Oscillator Circuit.* The simple crystal oscillator circuit given in Fig. 10A is essentially the circuit of Fig. 6H with a crystal in place of the grid tank circuit. This crystal is cut for a certain desired frequency, and will oscillate when the plate tank circuit is made inductive (by tuning *L-C* to a frequency slightly higher than the resonant frequency of the crystal). The plate tank circuit then has the effect of feeding back to the grid circuit, through the grid-to-plate capacity of the tube, a voltage which is in phase with the crystal voltage and which therefore serves to

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\*When quartz is cut at an angle to both the *Y* and *Z* axes, giving what is known as a *V* or *A.T. cut crystal*, the resulting crystal is little affected by temperature changes. *V-cut crystals* are used where temperature-controlled ovens are not justified because of space or weight limitations.

maintain the vibrations of the crystal. Figure 10B shows that as condenser  $C$  is varied from a low to a high value, oscillation starts at a certain value of  $C$  (point 1) and plate current decreases to a minimum at point 3 as the value of  $C$  is further increased. As  $C$  is increased beyond point 3, plate current suddenly rises to its original value, indicating that oscillation has stopped.

**Operating Point.** Although best efficiency is obtained when D.C. plate current is a minimum (at point 3), the circuit is not stable at this operating point (slight increases in  $C$  will stop oscillation); point 2 is the most practical operating point, for here changes in load or plate voltage have less effect.

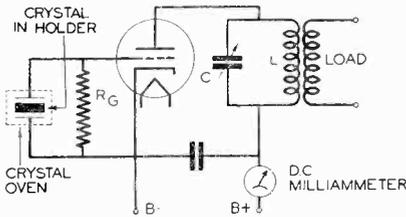


FIG. 10A. Simple crystal oscillator circuit. Rectified grid current flowing through grid resistor  $R_g$  provides the correct bias for the tube, while condenser  $C$  controls the starting, stopping and intensity of oscillation.

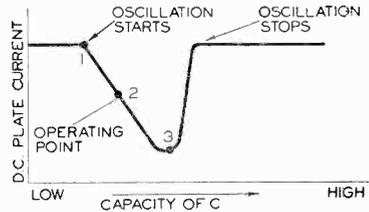


FIG. 10B. This graph illustrates how plate current in the crystal oscillator circuit at the left changes as  $C$  is tuned through the region of oscillation.

Crystal oscillator circuits are of many forms, the greatest variation being in the coupling to the load; electron coupling is widely used. Crystal oscillators have a very low output power, and R.F. amplifiers must therefore be used in practically every instance to get sufficient power output even for test equipment.

## ULTRA HIGH FREQUENCY OSCILLATORS

**Crystal Oscillators are Unsatisfactory for U.H.F. Work.** Oscillators which must generate very high radio frequencies with very low frequency drift introduce many special problems. To be sure, crystal oscillators can be used with one or more special harmonic-producing R.F. amplifiers which are adjusted to double and redouble the oscillator frequency until the desired ultra high frequency (u.h.f.) is obtained. A single amplifier operated with a C bias greater than the plate current cut-off value will, if excited with a signal of some definite frequency, produce in its plate circuit the 2nd, 4th, 6th, 8th, etc. harmonics of this input frequency, the 2nd harmonic being the strongest. By using other similar amplifiers, each tuned to the 2nd harmonic of the preceding stage, the original frequency produced by a crystal oscillator can be doubled many times, but there are serious drawbacks to this doubling method. When the crystal frequency is 5,000 kc. (the practical maximum operating frequency of a quartz crystal), and frequencies of the order of 100,000 kc. are desired (such as for television purposes), it is expensive as well as difficult to build frequency doublers which will operate satisfactorily at such high frequencies.

**Self-Excited U.H.F. Oscillators.** Acorn type tubes are used extensively with very high Q tank circuits for low power, self-excited u.h.f. oscillators. The tank coils are wound with stiff, solid wire which requires no supporting

form, thus keeping the distributed capacity and the losses of the coil at a minimum. Midget tuning condensers can be used, but often the required tank capacity is obtained by varying the spacing of the coil turns. This form of u.h.f. oscillator is used in television receivers.

**Tank Coil Voltage Distribution.** When the mid-tap of the tank coil in a self-excited u.h.f. oscillator is grounded as in Fig. 11A, the coil ends are alternately positive and negative\*; when one end of the coil is grounded, as in Fig. 11B, the other end likewise changes in polarity. The distribution of voltage is sinusoidal, hence Fig. 11A shows half-wave ( $\lambda/2$ ) and Fig. 11B shows quarter wave ( $\lambda/4$ ) voltage distribution. This does not necessarily mean that the coil itself is a half or quarter wave length long in physical size; the physical and electrical lengths of the coil can be made the same, however, by using straight metal wires or pipes, as shown in

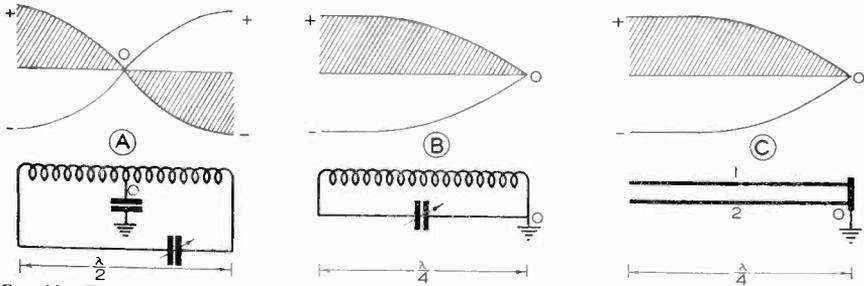


FIG. 11. The curves shown here give the voltage distribution across: A, the coil of an oscillator tank circuit when the coil mid-tap is grounded through an R.F. by-pass condenser; B, the coil of an oscillator tank circuit when one end of the coil is grounded; C, one of the pipes in a u.h.f. tank circuit using two parallel or concentric metal pipes. In each case the horizontal reference line represents ground

potential or zero voltage; in each case the curve surrounding a shaded area represents conditions for that half of a cycle when the left-hand end of the tank coil is positive, and the other curve (the minus curve) portrays conditions for the other half of the cycle. The curves at C are for one pipe (No. 1) only; curves for the other pipe would have exactly the same shape but opposite polarity.

Fig. 11C. Here pipes 1 and 2 are separated from one to four inches by air. When very high frequency, low wave length circuits are desired, simple pipes are used without tuning condensers.

**High Power U.H.F. Oscillators.** For high power u.h.f. oscillators, coils are made of copper tubing; sometimes two straight copper pipes side by side or one inside the other are used to provide the required inductance and capacitance. A single turn of tubing or even less is generally sufficient. Push-push tube circuits like that shown in Fig. 12A are customarily used; to get close grid-to-plate coupling (unity coupling), the grid wire is run inside the length of copper tubing which connects the plates together. Notice how electrode voltages are fed to the mid-points of the plate and grid loops or coils.

The circuit shown in Fig. 12B is a tuned grid, tuned plate oscillator (essentially the same as the fundamental Armstrong circuit shown in Fig. 6H), but using parallel pipes which may or may not be concentric. So-called quarter wave length lines are used to tune the grid and plate circuits. Since the grid tank circuit governs the frequency, it should be mechanically designed so that temperature changes do not affect its physical length. The

\*The coil tap can be grounded directly or through a condenser; either procedure places the coil tap at zero A.C. potential with respect to ground.

plate tuned line can be replaced with a coil and condenser (or coil alone) to conserve space, for it merely needs to be tuned slightly inductive.

Fig. 12C is the ultra-audion circuit shown in Fig. 6F with parallel lines substituted for a coil-condenser tank circuit; whereas the grid and plate in Fig. 12B were tapped along the line and the filament was connected to the end of the line, in Fig. 12C the grid and plate are tapped opposite each other along the line. The latter method gives twice the A.C. tank voltage. The taps are made variable so a better impedance match can be obtained, giving better circuit efficiency and stability. Points 1 and 2 in either circuit are load-coupling points and are usually variable as to position.

The three circuits in Fig. 12 are widely used in television transmitters, where high power and good frequency stability are required. They will operate at from 50 to 200 megacycles, the higher frequencies generally

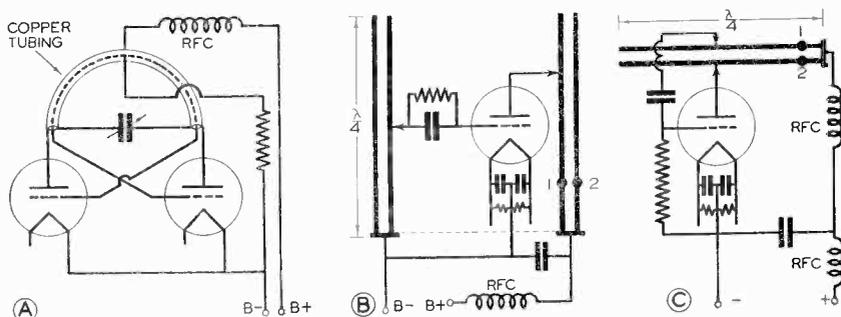


FIG. 12. Three high power ultra-high frequency oscillator circuits which are widely used in television transmitters.

being used to relay a program from a pick-up point to the main studio or from studio to transmitter. Amateurs use these circuits for communication in the 5 meter (56 megacycle) band. Air core coils and midget variable condensers are preferred in television receivers, although lines are occasionally used where space permits.

## DYNATRON OSCILLATORS

When the voltage applied to a resistor is increased, we naturally expect the current through the resistor to increase. Likewise, when the plate voltage of a tube is increased (the grid bias being at less than cut-off), we expect that plate current will increase. But under certain circumstances, especially with vacuum tubes, it is found that the current actually *decreases* when the voltage is increased. This can mean only one thing, that somehow the vacuum tube is feeding energy back into the circuit. A circuit acting like this is said to have *negative resistance*.\* Naturally, when an

\*By definition, resistance is voltage divided by current. A generator can be looked upon as a resistor whose ohmic value is equal to the generated voltage divided by the current flowing; since this voltage is equal and opposite to a real or positive-voltage drop in a resistance, the generated voltage has a *negative resistance* effect; it cancels the effects of real resistance, supplying the power required by the real resistance.

oscillator tank circuit is connected to terminals which are acting as generator terminals and which have this negative resistance effect, the oscillator circuit will be fed with energy to overcome its resistance loss and oscillation will take place.

*A Dynatron Oscillator Circuit.* A screen grid tube is one device which can act as a negative resistance. When the plate is at a lower positive voltage than the screen grid, considerable secondary emission will take place at the plate, and these secondary electrons will be attracted to the screen. An increase in the plate supply voltage (but not above the screen grid voltage) increases the secondary emission, thus making the plate current decrease; the screen therefore gets more electrons than the plate. A typical oscillator circuit using a screen grid tube is shown in Fig. 13A; this is known as a *dynatron* oscillator circuit. For a given C bias voltage (determined by the setting of  $P_2$ ) a plate voltage-plate current characteristic like that in Fig. 13B is obtained. Notice that after  $E_p$  reaches about

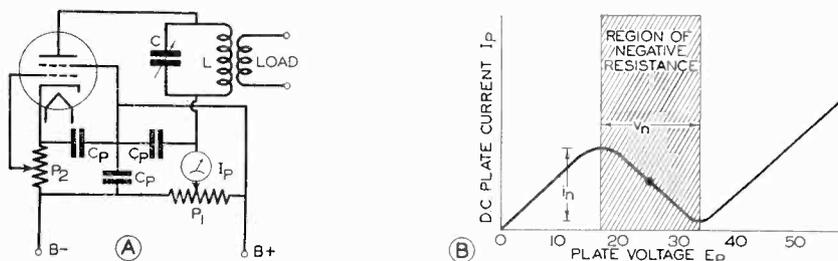


FIG. 13. A dynatron oscillator circuit (A) and its plate-voltage plate-current characteristic (B).

17 volts, further increases up to about 34 volts actually reduce  $I_p$ . This means that when the plate voltage of a screen grid tube varies within the shaded region in Fig. 13B, the tube will behave as if it had a *negative resistance* equal in ohmic value to  $v_n$  divided by  $i_n$ .

In the dynatron circuit of Fig. 13A this negative resistance characteristic makes the tube act as a generator which supplies energy to the  $L$ - $C$  tank circuit to overcome resistance losses. *Oscillation will occur in a dynatron circuit whenever the resonant resistance of the  $L$ - $C$  tank circuit is numerically greater than the negative resistance of the tube.* The minimum oscillator output voltage,  $v_n$ , is obtained when the two resistances are practically equal; increasing the resonant resistance makes the oscillator output voltage swing beyond the shaded area in Fig. 13B. Oscillation is most stable when the D.C. plate voltage is in the middle of the region of negative resistance. A dynatron oscillator delivers a nearly pure sine wave voltage to a load which is inductively coupled to the tank circuit coil.

Oftentimes we get dynatron oscillator action in a vacuum tube circuit where it is not desired, resulting in squeals. The undesired oscillations are then known as *parasitic oscillations*; they can be stopped by increasing the D.C. plate voltage to get it beyond the region of negative resistance in Fig. 13B, by adjusting the C bias to get this same result, or by inserting in the plate lead of the tube a resistor which will make the negative resistance of the tube too high for oscillation to occur.

## RELAXATION OSCILLATORS

The cathode ray tubes used in television pick-up cameras, in television picture reconstructors and in cathode ray oscillographs used for testing purposes, require a special kind of oscillator, one which will produce a "saw tooth" voltage or current characteristic like that shown in Fig. 14A. This saw tooth wave form is needed to sweep the electron beam at a uniform rate across the cathode ray tube screen, then return it almost instantaneously to the starting point for another sweep. Oscillators capable of producing a saw tooth wave form are known as sweep oscillators or as *saw tooth oscillators*. In radio circuit testing work, oscillators capable of producing the *square top* wave form shown in Fig. 14B are required, for this wave form results in strong harmonics.

In general, circuits for these saw tooth and square top oscillators use a condenser which is alternately charged up by a D.C. voltage and discharged by an electrical device such as a gaseous glow tube. The output voltage drops to zero or *relaxes* for a definite fraction of each cycle, and such units are therefore called *relaxation oscillators*.

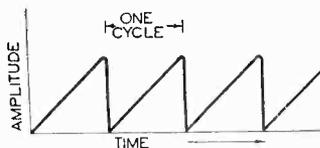


FIG. 14A. One type of relaxation oscillator produces this *saw-tooth* wave form, which gives the sweep voltage required for cathode ray tubes.

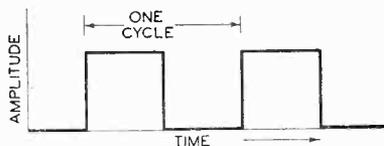


FIG. 14B. Another form of relaxation oscillator produces this *square-top* wave, used in laboratory oscillators because it produces very strong harmonics.

The simplest circuit for generating a saw tooth voltage is given in Fig. 15A. In this circuit  $N$  is a neon glow tube consisting of two electrodes in a glass envelope filled with neon gas. This tube will not conduct electricity until a definite voltage, called its *firing* or *ignition* voltage, is applied; the tube resistance then becomes very low and a large current can flow unless it is limited by a resistance. Thus, when a condenser  $C$  is charged through resistor  $R$ , the condenser voltage gradually builds up (depending on the time constant;  $R$  in megohms times  $C$  in microfarads gives time in seconds), until the voltage across  $C$  is high enough to ignite the neon tube. The resulting flow of current through  $N$  lowers its resistance materially, thus shorting the condenser and dropping its voltage practically to zero (most of the line voltage being taken by  $R$ ); the neon tube then stops glowing or firing and the action starts over again. The frequency is controlled by the characteristics of  $N$ , the values of  $C$  and  $R$ , and the value of the D.C. supply voltage.

*Gaseous Triode Circuit.* A gaseous triode tube is often used with a condenser-resistor charging circuit to produce a saw tooth wave, it being easier to control the firing voltage of this tube than of a diode. In Fig. 15B,  $T$  is a Thyatron tube (mercury vapor-filled) or a grid glow tube (filled with neon gas). Conduction through the tube depends upon the ratio of plate to grid voltages; the more negative the  $C$  bias, the higher the plate voltage must be before the tube will conduct current. Once conduction starts, only the reduction of plate voltage to a very low value

will stop current flow. In this circuit the condenser voltage builds up gradually through  $R$  until sufficiently high to fire the tube; the condenser is momentarily short-circuited by the tube as it discharges, the tube stops conducting and the charging of  $C$  starts over again. The frequency is essentially controlled by the values of  $C$  and  $R$ , for the plate and grid voltages are usually adjusted to correspond to the desired time constant. A controlling signal, such as the line or picture frequency of a television broadcast, can be introduced at point  $X$ , if the  $C$  bias is made sufficiently high so that the tube will discharge only when the controlling impulse is applied. The circuit can thus be made to produce a saw tooth signal output which is in synchronism with an input signal.

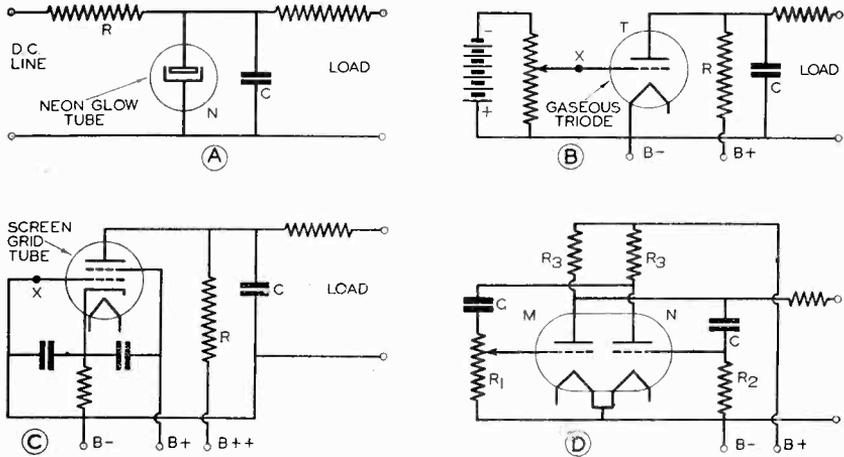


FIG. 15. Relaxation oscillator circuits.

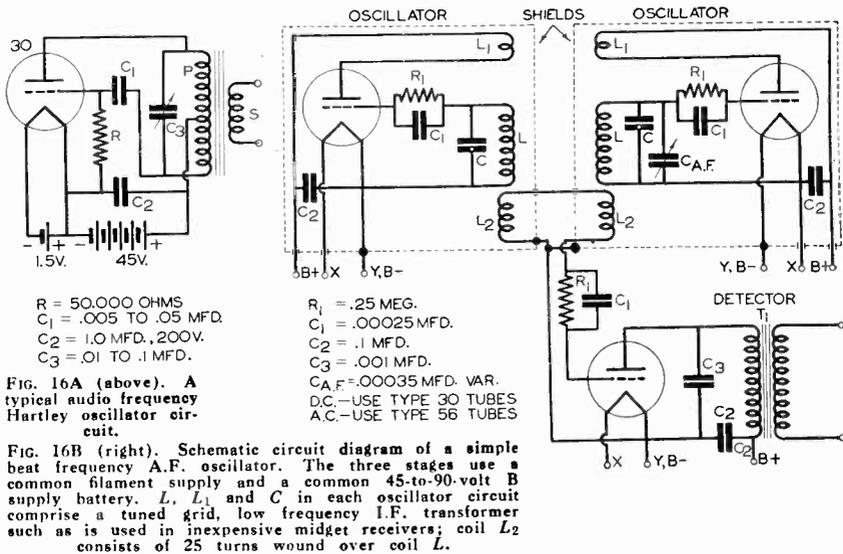
An electron-coupled vacuum tube amplifier may be used instead of the gaseous tube signal circuit; a screen grid tube circuit is shown in Fig. 15C. Space (cathode) current is made to flow by applying a positive voltage to the screen and a negative  $C$  bias voltage to the grid. Only a negligibly small current flows to the plate until it is placed at a very high voltage; at that point the plate-to-cathode resistance drops to a low value. Again a condenser charged through a resistor is used to build up a voltage which will break down the resistance of the tube (which is in parallel with the condenser).

In the circuits of Fig. 15A, 15B and 15C, the desired saw tooth voltage is obtained by connecting the load across condenser  $C$ . A resistor of high ohmic value (10 to 30 times the value of  $R$ ) is placed in series with the load to insure a uniform build-up of voltage across  $C$ . Quite often, charging resistor  $R$  is replaced by a vacuum tube, the cathode-to-plate resistance of the tube serving as a self-regulating high resistance. The tube has the characteristic of making the condenser charge more evenly with time; in either case the presence of the charging resistor prevents the condenser from being charged to line voltage.

*The Multivibrator.* This important type of relaxation oscillator, which produces the square top wave form shown in Fig. 14B, depends for its

operation upon the fact that the input and output voltages of a resistance-coupled amplifier are 180 degrees out of phase. The multivibrator circuit consists of two amplifiers connected as in Fig. 15D, each amplifier using one section of the double triode tube.

The operation of this circuit is a sort of "see-saw" affair in which first one tube and then the other passes plate current to produce the desired square top wave. When the circuit is first placed in operation, an increase in plate current in one tube causes a decrease in plate current in the other (because the two stages are out of phase with each other), and this initial increase is reamplified almost instantly to make the current in that tube rise to a maximum while the plate current in the other tube drops to



zero. Current continues to flow through the tube, forming one square-top portion of the wave, until such time as leakage through the resistor and condenser cause a slight plate current increase in the non-conducting tube; this increase is almost instantly amplified and reamplified, forcing current in the first tube down to zero. The time interval for which the current flows in each tube depends upon the values of  $R_1$ ,  $R_2$ ,  $R_3$ , and  $C$ , unless an A.C. control voltage is applied to one of the grids to set the rate of oscillation. This controlling signal may be the fundamental frequency or any harmonic of the natural frequency of the multivibrator; the input signal, which may come from a crystal oscillator, therefore controls the stability of the multivibrator.

## AUDIO AND BEAT FREQUENCY OSCILLATORS

Oscillators which produce signals in the aural band (from about 30 to 16,000 cycles) differ in no way from R.F. oscillators; it is merely a matter of getting a sufficiently high inductance and capacity in the tank circuit. For a 100 cycle oscillator the inductance should be about 25 henrys if the capacity is .1 mfd., and 2.5 henrys if the capacity is 1 mfd. Air core coils

with inductances of this order would be enormous. Saturation in iron core coils introduces distortion, but if an appreciable air gap exists in the magnetic path an A.F. signal which is entirely satisfactory for radio testing (defect isolation) purposes can be had.

A typical Hartley audio frequency oscillator circuit is shown in Fig. 16A. The iron core coil can be the output transformer of a push-pull amplifier. Grid condenser  $C_1$  must be rather high in value, for it must offer a low reactance to low frequency audio signals. Tuning is accomplished by varying  $C_3$ , the maximum frequency being governed by the distributed capacity of the transformer windings.

*Beat Frequency A.F. Oscillators.* Iron core coils having a low distributed capacity and operating below saturation are costly and even difficult to build in units which will cover satisfactorily the entire A.F. band with a single tuning condenser. For this reason audio oscillators generally employ the beat between two low R.F. signals. For example, if a 100 kc. and

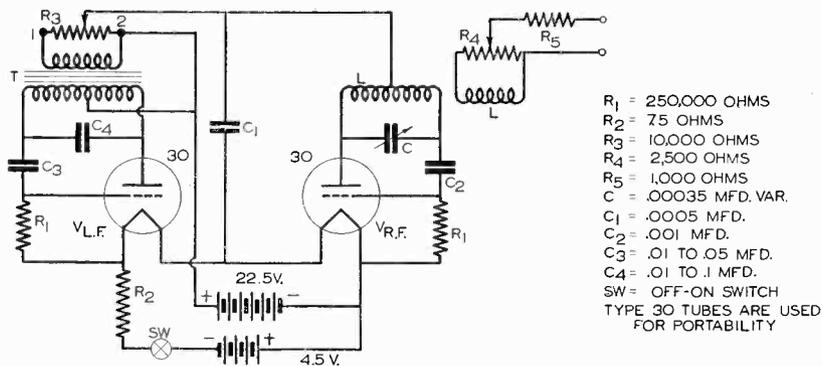


FIG. 17. Schematic circuit diagram of a typical modulated R.F. oscillator.

a 102 kc. signal are fed to a detector, the 2,000 cycle difference frequency will appear in the plate circuit. If the load is coupled to the detector plate circuit with an iron core transformer having a condenser shunted across its primary, the two fundamental frequencies and the 202 kc. sum frequency will automatically be shorted out or by-passed. Beat A.F. oscillators are widely used for testing and for precision checking work. The great difficulty lies in preventing the two R.F. oscillators from drifting in frequency.

The circuit of a beat frequency oscillator, built from parts found in the average radio shop, is given in Fig. 16B. Each R.F. oscillator should be placed in an individual shielded compartment. With  $C_{AF}$  set at minimum capacity, adjust either of condensers  $C$  until the D.C. plate current in the detector is at its no-excitation value; this places both oscillators at the same frequency and there will therefore be no audio output.

Calibration of this oscillator is rather difficult with ordinary equipment, but a rough calibration can be obtained by comparing the audio beat output notes for various settings of  $C_{AF}$  with notes produced by tuning forks of known frequency. For precision results, it is far better to buy a commercial A.F. oscillator than to build your own.

## A SERVICEMAN'S MODULATED TEST OSCILLATOR

All of the oscillators discussed so far produce constant amplitude A.C. signals. In testing radio receivers, however, the modulated radio signal ordinarily picked up must be replaced by an equivalent test signal. An R.F. oscillator which is modulated with an audio or video signal is, therefore, needed by a serviceman. There are many ways of producing this, but in a test oscillator (or signal generator) the easiest way to get this is to vary the D.C. plate voltage applied to the R.F. oscillator by introducing an A.C. voltage of an audio or video frequency. Figure 17 shows a typical modulated R.F. oscillator extensively used by servicemen; the low frequency or modulating signal is produced in the  $V_{LF}$  oscillator, the output from transformer  $T$  being connected in series with the D.C. plate supply to tube  $V_{RF}$  in the R.F. oscillator circuit. The degree or percentage of modulation is varied by varying  $R_3$ , which changes the amount of low frequency voltage.

The low frequency oscillator could just as well be replaced by an electric phonograph reproducer whose output was fed to points 1 and 2. For testing all-wave receivers, coils  $L-L$  should be of the plug-in type; otherwise an arrangement whereby a switching mechanism inserts the proper coils must be used. Calibration of the oscillator is quite a tedious and exacting process, and since test oscillators are inexpensive, commercial equipment is always preferred.

### TEST QUESTIONS

Be sure to number your Answer Sheet 21FR-2.

Place your Student Number on every Answer Sheet.

Never hold up one set of lesson answers until you have another ready to send in. Send each lesson in by itself before you start on the next lesson. In this way we will be able to work together much more closely, you'll get more out of your Course, and you will receive the best possible lesson service.

1. In a *vacuum tube* oscillator, are the power losses in the L-C oscillatory circuit compensated for *constantly* or *intermittently*?
2. What, in a self-excited vacuum tube oscillator, essentially governs the frequency?
3. Give another name for the oscillatory circuit.
4. In the oscillator circuit of Fig. 4, what two parts automatically supply a steady D.C. bias voltage to the grid?
5. What happens if the ohmic value of the grid resistor in a self-excited vacuum tube oscillator (such as  $R_G$  in Fig. 4) is made too high?
6. When a load is applied to the tank circuit of a self-excited vacuum tube oscillator, will the D.C. plate current *increase* or *decrease*?
7. Can the power output of a self-excited vacuum tube oscillator be increased by increasing the D.C. plate supply voltage?
8. How can the Hartley oscillator circuit be identified?
9. What is the purpose of the parallel resonant plate circuit in the tuned grid, tuned plate Armstrong oscillator?
10. To what part of a self-excited oscillator circuit should the load be coupled in order to feed it with the fundamental oscillator signal?

## THOROUGHNESS

Whatever you do, do well if you would stay on the straight road to success. The habits of carelessness and slipshod work are all too easy to acquire; beware of them as you would the plague. Men who are thorough in their work cannot remain undiscovered for long, because the demand for such men is greater than the supply.

Thoroughness is just as important in study as it is in work; what you get out of a lesson depends upon how completely you master the material presented in it. Some books, as fiction, are read hurriedly and only once, then cast away; the enduring works of literature are carefully read and reread many times but always essentially for the pleasure they give; textbooks, however, must be read quickly, to get the basic ideas, then carefully many times until every important principle has been mastered. Textbooks are always saved for future reference; tomorrow you may have urgent need for the information given in a paragraph which today seems so insignificant.

Thoroughness in study habits leads to thoroughness in work habits, and eventually to a thorough success.

J. E. SMITH

**PEAK AND BAND-PASS  
R. F. TUNING CIRCUITS**

22FR-2

**NATIONAL RADIO INSTITUTE**

ESTABLISHED 1914

**WASHINGTON, D. C.**



# STUDY SCHEDULE No. 22

For each study step, read the assigned pages first at your usual speed. Reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind, then answer the Lesson Questions for that step. Study each other step in this same way.

1. Importance of Response Curves; General Analysis of a Modulated R.F. Carrier; Analyzing Typical Response Curves ..... Pages 1-6

This general information on response curves prepares you for speedier understanding of the actual tuning circuits taken up in the rest of the lesson. Study this section thoroughly. Answer Lesson Questions 1, 2 and 3.

2. Factors Controlling Response; Coupling Methods for R.F. Tuning Circuits ..... Pages 6-11

Here is where you begin studying in detail exactly how coils and condensers work together to permit passage of the desired signal while rejecting all other signal frequencies. Answer Lesson Questions 4, 5 and 6.

3. Directly Coupled Resonant Loads; Tuned Secondary Transformer Loads ..... Pages 11-16

This section should not prove at all difficult if you realize right from the start that your goal now is simply understanding the material. Don't try to memorize it, because you can always review this material later, and can refer to it when in need of specific facts. Answer Lesson Question 7.

4. Double-Tuned Transformer Loads; Double-Tuned Capacity-Coupled Loads ..... Pages 16-23

You'll find quite a bit of practical information here, along with more r.f. tuning circuit data. Answer Lesson Questions 8, 9 and 10.

5. Combination Capacitive and Inductive Couplings; Tuned Secondary with Fixed Tuned Primary Circuit; R.F. Tuning Circuits Used in Actual Receivers ..... Pages 23-28

Pay particular attention to the section covering a fixed tuned primary circuit, because this scheme is used in a great many table model receivers. The actual tuning circuits at the end of this lesson illustrate many of the important principles previously studied.

6. Mail Your Answers for This Lesson to N.R.I. for Grading.

7. Start Studying the Next Lesson.

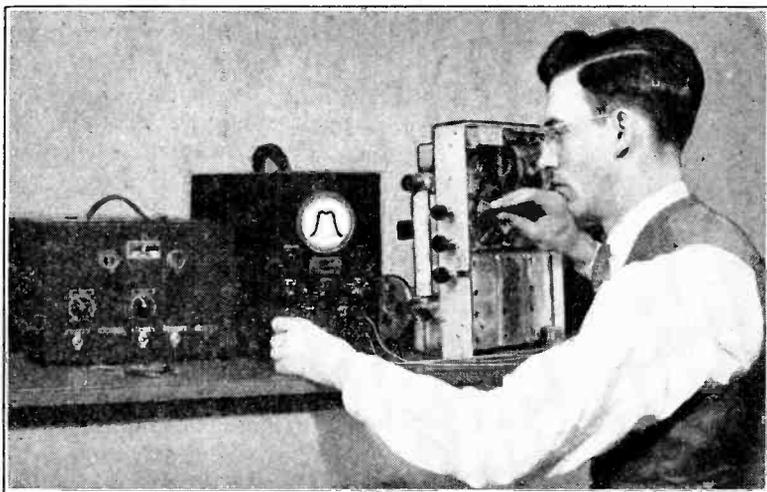
# Peak and Band-Pass R. F. Tuning Circuits

## Importance of Response Curves

**P**RACTICAL radio men are today more concerned than ever before with the shapes of the resonant response curves for R.F. amplifiers, for they have come to realize that these curves reveal the exact characteristics of a receiver or transmitter and tell when undesirable effects have been in-

sired gain and fidelity within the limitations of the tuning circuits. A thorough understanding of the peculiar characteristics of R.F. tuning circuits and an ability to read the story told by each shape of response curve will prove particularly valuable when using a cathode ray oscilloscope for radio receiver testing and servicing.

You are already familiar with peak



All-wave superheterodyne receiver being aligned for band-pass response, using a frequency-wobblated R.F. signal generator (extreme left) and a cathode ray oscilloscope. The final double-peak response curve, secured after all adjustments are made, can be seen on the screen of the cathode ray tube. Note that the receiver chassis is set on end, for convenience in making connections and adjusting under-the-chassis trimmer condensers.

troduced by adjustments or by defects in circuit parts.

Older receivers, as well as a great many modern receivers, use R.F. tuning circuits which are adjusted for peak response. On these receivers a serviceman need only adjust for maximum output, never giving a thought to the shape of the peak response curve. A modern high-fidelity radio receiver has band-pass R.F. tuning circuits, however, and actual viewing of the response curve greatly simplifies the adjusting of the receiver to give the de-

response curves like those shown in Fig. 1A, for they have been discussed in previous lessons. You know that a sharp peak response curve for an R.F. amplifier indicates high gain and high selectivity, while a broad peak response represents somewhat lower gain and lower selectivity but better fidelity. Likewise you are familiar with the band-pass response curves shown in Fig. 1B, and know that R.F. tuning circuits having these curves give better fidelity at the expense of gain. In this lesson we will study in detail the tuning

circuit conditions which give to an R.F. amplifier any of these four response curves or any of the many possible variations of these curves.

### General Analysis of a Modulated R.F. Carrier

If we used a special cathode ray oscilloscope to analyze an R.F. carrier which is 100% modulated with a single sine wave signal (of frequency  $f_m$ ), we would see on the screen of the cathode ray tube a pattern much like that in Fig. 2A (the dotted lines indicating the modulation envelope would of course be absent). Either a mathematical analysis or actual measurements will

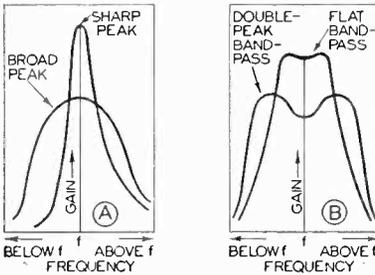


FIG. 1. Typical peak and hand-pass response curves of R.F. amplifiers.

show that we really have three different R.F. signal frequencies in this modulated carrier, as indicated in Fig. 2B:

- $f$ , the R.F. carrier frequency
- $f_1$ , the lower side frequency, which is equal to the carrier frequency minus the modulation frequency ( $f_1 = f - f_m$ )
- $f_2$ , the upper side frequency, which is equal to the carrier frequency plus the modulation frequency ( $f_2 = f + f_m$ )

Furthermore, with 100% modulation the voltage of each side frequency signal will be *exactly one-half* that of the carrier signal. (With less than 100% modulation, the amplitude of each side frequency will be *less than one-half* that of the carrier.) In dealing with R.F. tuning circuits, we must consider all three of these R.F. signals, for

the side frequencies must be amplified the same amount as the carrier frequencies if distortion is to be avoided.

When a 100%-modulated R.F. carrier is sent through an R.F. amplifier which has a perfectly flat top response, the side frequencies will be amplified equally as much as the carrier and the output wave pattern will be identical to the input wave pattern. If, however, we send this 100%-modulated R.F. carrier through a tuned R.F. amplifier which is considerably off tune, severe amplitude distortion occurs because the side frequencies and the carrier are amplified different amounts, and the output wave pattern might be as shown in Fig. 2C (this pattern corresponds to the condition where one side frequency is not amplified at all and the other side frequency is amplified twice as much as the carrier; Fig. 2D indicates the output voltage relationship under this condition). Output wave patterns thus tell directly whether distortion is occurring in an R.F. amplifier.

Unfortunately the average cathode ray oscilloscope used by servicemen is not designed to amplify R.F. carrier voltages sufficiently to give useful modulated R.F. patterns on the screen; an extra R.F. amplifier would have to be used, or a costly and bulky laboratory type oscilloscope secured. A response curve of the R.F. amplifier in question gives essentially the same information about distortion, however, and is easily produced with an ordinary radio servicing oscilloscope. A typical peak response curve is shown in Fig. 2E; this curve tells how much amplification the side frequencies will get at any modulation frequency value. Each response curve has a story to tell you; to show how these stories can be read, we will consider a typical example in which the carrier frequency is assumed to be 1,000 kc., with 100% modulation.

When the modulation frequency is 100 cycles, the side frequencies will be 999.9 kc. and 1,000.1 kc.; by referring to Fig. 2E, where these side frequencies are designated as  $f_3$  and  $f_4$ , we can readily see that these will receive essentially the same amplification (gain) as the 1,000 kc. carrier. This means that after the modulated signal has passed through the R.F. amplifier, the two side frequencies will each be the same fraction of the carrier voltage (one-half in this case of 100% modulation) as they originally were.

With a 5,000-cycle modulation signal, however, the resulting 995 kc. and 1,005 kc. side frequencies ( $f_1$  and  $f_2$  in Fig. 2E) receive considerably less amplification than the carrier ( $f$ ); this means that after the modulated signal has passed through the R.F. amplifier, the two side frequencies will be a considerably lower fraction of the carrier voltage than they originally were (each will be *less than one-half* the carrier voltage in our case of 100% modulation). With only one modulation frequency, this attenuation of side frequencies is simply equivalent to a reduction in the modulation percentage, provided that both side frequencies are equally attenuated; after demodulation, then, the 5,000-cycle audio signal voltage will be lower than if there were no attenuation of side frequencies.

**Frequency Distortion.** When a number of different modulation frequencies ranging from 0 to 5,000 cycles (such as we have in radio receivers which are tuned to sound broadcasts) are present in an R.F. amplifier having the response curve in Fig. 2E, there will be a large number of side frequencies in the range from .995 kc. to 1,005 kc. Those farthest away from the carrier frequency, corresponding to the higher modulation frequencies, will be amplified the least, with the result that a certain amount of frequency distortion

will be present in the audio signal after demodulation. If not too severe, this frequency distortion can be corrected in a receiver by the use of equalizing circuits which make the audio amplifier provide increased amplification for those higher modulation frequencies which were cut down in the R.F. tuning circuits. Servicemen often use this little equalizing trick to com-

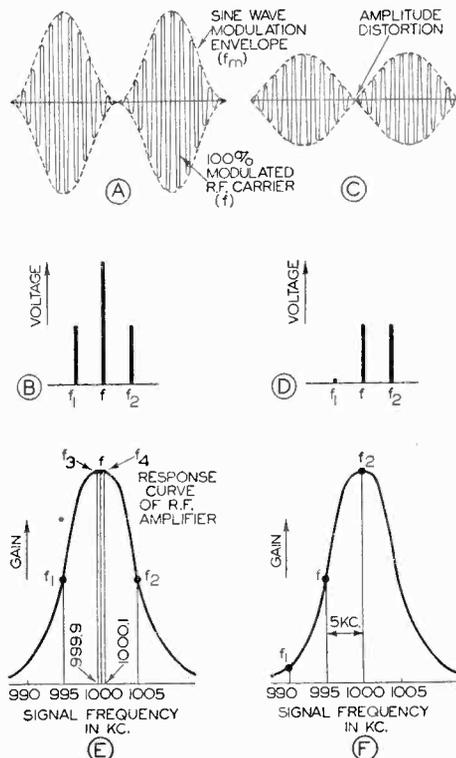


FIG. 2. These diagrams tell you what happens to a modulated R.F. carrier signal when the R.F. amplifier is properly tuned (A, B and E) and when it is improperly tuned (C, D and F.)

pensate for frequency distortion in a highly selective R.F. amplifier. In television circuits the modulation frequencies may range from 0 to over 2.5 megacycles, and the amount of equalization required in picture signal amplifiers may therefore be quite great.

**Amplitude Distortion.** When the side frequencies associated with an R.F. carrier are attenuated or out down

unequally by a tuning circuit (so that with a single modulation frequency, one side frequency will have a greater amplitude than the other), *amplitude distortion* as well as frequency distortion will be present. This fact is not so easily shown without mathematics, but by considering the extreme case where only one side frequency is allowed to pass, the other being cut out entirely by the tuned circuit, we can get some idea as to why this statement holds true.

Suppose that the R.F. amplifier in our previous example is tuned to 1,000 kc. but is fed with a 995 kc. carrier modulated at 5,000 cycles (5 kc.); now we have the condition represented by Fig. 2F, where the upper side frequency (1,000 kc.) is fully amplified, the carrier is amplified about half as much, and the lower side frequency receives hardly any amplification at all. The R.F. amplifier under this condition allows only one side frequency to pass through with the carrier, as was indicated in Fig. 2D; both will have the same amplitude at the output, and the wave form of the amplifier output voltage will be as shown in Fig. 2C. (Originally, as in Fig. 2B, the amplitude of the carrier  $f$  was twice that of the upper side frequency  $f_2$ ; reducing the carrier amplitude one-half without reducing the amplitude of  $f_2$  thus makes both amplitudes equal.

Observe that the outer peaks of modulation in Fig. 2C have sine wave shapes, but the valleys or troughs are V-shaped; this is clearly a case of amplitude distortion, for the modulation envelope no longer corresponds to the sine wave modulation signal (like that in Fig. 2A) at the input of the tuning circuit. (The curve in Fig. 2C was obtained by adding together and plotting the values of the carrier and the side frequency at each instant of time; it can be verified with a cathode ray oscilloscope and suitable special

laboratory equipment.) Lower percentages of modulation than 100% and different off-tune carrier frequencies will of course give slightly different wave forms for the envelope in Fig. 2C, but amplitude distortion will be evident in all cases.

You can expect distortion similar to that in Fig. 2C whenever a tuning circuit is not properly tuned to the input carrier or when it has an unsymmetrical resonant curve, for both conditions result in unequal amplification of upper and lower side frequencies. The resulting amplitude distortion cannot be corrected for in the audio system; it will be present in the receiver output, and will often be annoying to the radio listener. The elimination of amplitude distortion in the R.F. system is therefore a matter of vital importance to the radio serviceman as well as the receiver designer.

### Analyzing Typical Response Curves

Four typical resonant response curves of actual radio receivers, such as might be obtained by using a cathode ray oscilloscope and the necessary associated equipment, appear in Fig. 3. In each case  $f$  represents the carrier frequency to which the R.F. amplifier is tuned, while  $f_1$  and  $f_2$  represent the lowest and highest side frequencies involved. When properly interpreted, these curves reveal considerable information about distortion.

*Sharp Peak.* An R.F. amplifier having the sharp peak response curve shown in Fig. 3A will cause severe attenuation of the higher modulation frequencies (severe frequency distortion). (Remember that low modulation frequencies correspond to side frequencies close to and both above and below  $f$ , while high modulation frequencies correspond to side frequencies near  $f_1$  and  $f_2$ .) Since the gain at  $f_1$  is less than at  $f_2$  in this example, some amplitude dis-

tortion is also to be expected; this may not be severe, since the difference between the two values is not great. In R.F. amplifiers which have peak response curves, the greatest amount of amplitude distortion occurs because of improper tuning, which gives the condition represented by Figs. 2C, 2D and 2F. It is primarily for this reason that highly selective receivers, which naturally have sharply peaked response curves, are equipped with tuning aids.

**Rounded Peak.** Rounding or broadening of the peak of a response curve, by cutting down the amplification in the vicinity of the carrier frequency more than at the extreme side frequen-

than the other, with resulting amplitude distortion. Furthermore, if the valley between the peaks is too deep, the lower modulation frequencies (having side frequencies in this valley) will be attenuated and frequency distortion will be evident.

**Symmetrical Double Peak.** When a serviceman adjusts a band-pass tuning circuit for high fidelity and good selectivity, his goal is the rarely-attained ideal square top response curve; ordinarily, however, he is entirely satisfied if he can secure the symmetrical double peak response curve shown in Fig. 3D, which has a negligible valley between the peaks. He knows that

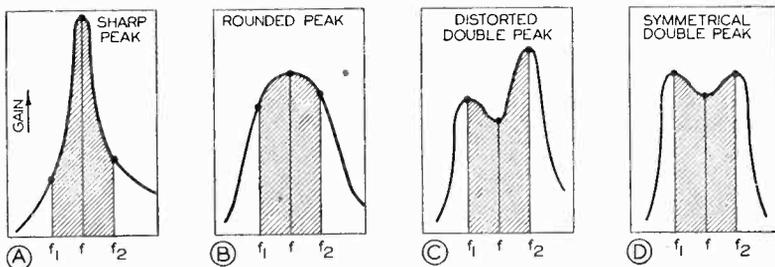


Fig. 3. These four response curves are representative of those which can be viewed on the screen of a cathode ray oscilloscope when actual R.F. and I.F. amplifiers are being tested. The shaded areas and vertical lines are of course not seen on the C.R.O. screen; they have been added here in order to show the range of side frequencies handled by the amplifier along with the carrier in each case.

cies, is easily accomplished by a service technician who understands R.F. tuning circuits. The result is a broad peak response curve similar to that shown in Fig. 3B, which gives considerably less frequency distortion at the expense of selectivity and gain.

**Distorted Double Peak.** Band-pass R.F. tuning circuits will, if properly designed and adjusted, give a double peak response curve with steep sides, insuring good fidelity and selectivity. Unless the adjustments are carefully made, however, there is a possibility that a distorted double peak response curve like that shown in Fig. 3C, in which one peak is higher than the other, will be obtained. Naturally a curve such as this is undesirable, for one side frequency is amplified more

when an R.F. amplifier has a symmetrical double-peak response curve such as this, amplitude distortion will not occur and frequency distortion will be negligible in the R.F. or I.F. amplifier. A properly designed band-pass tuning circuit can give far better selectivity than a circuit having a peak response which has been broadened to give equally as good fidelity. (The steeper the sides of the response curve outside the  $f_1$ - $f_2$  region, the better is the selectivity.)

**Alignment of R.F. Tuning Circuits.** The final factory inspection of a radio receiver generally includes a check of the response curve for the R.F. section, to make sure that it has the desired shape. This is referred to as a check of the alignment. Oftentimes this align-

ment may be disturbed by rough handling during shipment and by general aging of the receiver, making it necessary for the serviceman to realign the tuning circuits.

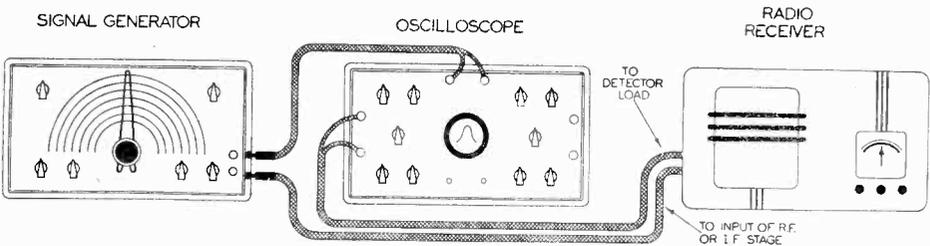
It is a well-known fact that most radio receivers are designed to have a compromise between selectivity, gain and fidelity, so they will please the greatest possible number of listeners. It is when a particular listener wants the highest possible fidelity or wants maximum gain and selectivity for reception of distant stations that the serviceman is called in to change this compromise response characteristic. Realigning a receiver to have a sharp-peak response gives maximum possible gain and selectivity; usually this is easily done with an ordinary all-wave

which control the response characteristic.

### Factors Controlling Response

The shape and height (maximum gain) of the response curve for a tuned R.F. amplifier are essentially determined by one or more of the following factors: 1, *The Q factors of the coils used in the amplifier;* 2, *the L/C ratio of each tuned circuit in the amplifier;* 3, *the types of coupling used to connect the tuned circuits to each other and to vacuum tubes;* 4, *the characteristics of the vacuum tubes.* Although these factors have been discussed to a certain extent in previous lessons, they are so important to our study of tuning circuits that I will review them briefly at this time.

*Q Factor of a Coil.* In previous les-



The response curve of the radio receiver at the right appears on the screen of the cathode ray tube in the radio servicing oscilloscope (center) when proper connections are made between the receiver, the oscilloscope and the frequency-wobulated R.F. signal generator at the left.

signal generator and an output indicator. Correct aligning for high fidelity cannot be easily carried out without additional equipment, however; a frequency-wobulated signal generator\* and a cathode ray oscilloscope are essential in this case. It is not the purpose of this lesson to describe the service procedures followed in realigning radio receivers, but rather to point out the various factors in tuned circuits

sons it was pointed out that any tuning circuit has a certain amount of loss due primarily to the A.C. resistance of the coil (the resistance of the condenser and the circuit wiring is so low that it is usually neglected entirely). The ohmic value of this A.C. resistance of a coil depends not only upon the D.C. resistance of the wire used in making the coil, but also upon "skin effects" associated with high frequency currents, upon losses occurring in the dielectric materials used for the coil form and insulation, and upon the nature of the load which is coupled to the coil. The Q factor of a coil was de-

\*A frequency-wobulated signal generator is a special type of R.F. signal generator whose output frequency can be made to vary regularly and automatically above and below a definite R.F. value to cover any desired range of side frequencies.

$$Q \text{ factor of a coil} = \frac{\text{Coil reactance in ohms}}{\text{Coil A.C. resistance in ohms}}$$

defined as the *coil reactance divided by this coil A.C. resistance*, all values being measured at the same frequency. Furthermore, since it is the coil which controls the tuning circuit losses, the Q factor of the coil can be considered as the Q factor of the entire tuning circuit.

*What Q Factor Tells Us About Tuning Circuits.* The Q factor of the coil in a tuning circuit is a numerical value, often referred to simply as Q; it tells us the following important facts about the two types of tuning circuits:

**Series Resonant Circuits—**

1. At resonance, the A.C. voltage across the coil is Q times the source voltage.
2. At resonance, the impedance of the tuned circuit is entirely resistive, and is equal to the impedance of the coil in ohms divided by the Q factor of the coil.

**Parallel Resonant Circuits—**

1. At resonance, the current through the coil is Q times the source current.
2. At resonance, the impedance of the tuned circuit is Q times the coil impedance, and is entirely resistive.

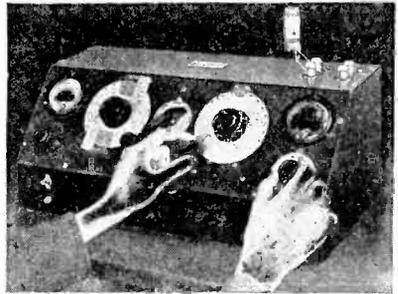
Up to a few years ago, engineers and scientists discussed the behavior of tuned circuits in terms of the A.C. resistance of the coil; this practice is quite correct, and may still be found in many text-books. Modern engineers

\*Q factor is actually the reciprocal of power factor. You will remember from previous Lessons that the power factor of a device is equal to its resistance divided by its impedance; where the impedance is essentially reactance, power factor can be considered equal to resistance divided by reactance, just as Q factor is equal to reactance divided by resistance under the same conditions. You can always find the Q factor of a device (if the power factor is known) by dividing the number 1 by the power factor. Good coils and condensers have a high Q factor and a low power factor; good resistors have a low Q factor and a high power factor.

prefer to think in terms of Q factor, however, since they now have instruments with which they can measure the Q factor of a coil directly.\*

One of the instruments available for measuring the Q factor of a coil is shown in Fig. 4; it is known as a Q factor meter, and can also be used for measuring the Q factor of any resistor or condenser.

In a coil or condenser the radio engineer desires pure reactance at any frequency, with no resistance to cause loss of useful power, and consequently he wants the Q factors of these parts (the ratios of reactance to resistance)



*University of Dayton Radio Corp.*

FIG. 4. This Q-factor meter is typical of the instruments used by radio engineers for measuring the Q factor of coils, condensers and resistors. The device being measured (a coil in this case) is connected to two of the terminals at the top of the instrument.

to be as high as possible in most cases. The Q factor meter reveals that the Q factors of condensers are very high (resistance is very low) in comparison with coils; the meter can also be used to compare various coils (or condensers) as to quality.

Although a resistor is ordinarily thought of as a pure resistance, it often has appreciable inductance, particularly when of the wire-wound type. The Q factor of a resistor (the ratio of

## REVIEW DATA FOR A.C. CIRCUITS

**Resistance.** That opposition to current flow in an A.C. circuit which results in power loss; it is often called A.C. resistance.

**Reactance.** That opposition to current flow in an A.C. circuit which does not result in power losses. Reactance may be either inductive (due to a coil or inductance) or capacitive (due to a condenser or capacitance).

**Impedance.** The total opposition to current flow in an A.C. circuit. Impedance combines the effects of both resistance and reactance, and therefore determines how much alternating current will flow.

When the resistance of a device is very small with respect to its reactance, as in coils and condensers, the impedance will be just slightly larger than the reactance, and for all practical purposes we can consider the impedance and reactance to be equal.

reactance to resistance) should therefore be as low as possible in circuits where only resistance is desired.

*How to Increase the Q Factor of a Coil.* For a coil of given inductance, keeping the losses in the coil at a minimum insures a high Q factor. Losses which are due to capacity between the turns of the coil can be reduced by using insulating materials and coil coatings which have low dielectric losses. Losses due to the coil form can be reduced by improving the quality of the material used in the coil form. Losses due to skin effects in the wire at high frequencies can be reduced by using a large number of enamel-covered wires which are braided together to form what is known as "litz" wire. (Unfortunately, litz wire is valuable only at frequencies between about 200 and 900 kc.) At high frequencies, losses can be kept down by making the coil with large solid wire, with flat copper ribbon or with copper tubing, all turns being equally spaced.

The shape of a coil has considerable effect upon its Q factor, for coil losses vary with the shape of the coil, particularly when the winding is in several layers. When designing multi-layer R.F. coils, radio engineers generally test out several shapes and select

that which gives the highest Q factor for the coil.

Shielding an R.F. coil by placing it in a metal can or compartment increases the losses in the coil, reduces the coil inductance, and also reduces the coil Q factor. With a Q factor meter like that shown in Fig. 4, the designer can select a shield for a given coil which will not excessively reduce the Q factor.

Large coils which are made from heavy copper wire, tubing or ribbon can have Q factors of over 500. In radio receivers, where small coils must be used because of space limitations, Q factors of 150 are considered excellent; in order to obtain this high value for radio receiver coils, it is necessary to use coil forms at least two inches in diameter and use heavy solid wire or litz wire for the windings.

Even though an engineer is able to choose a coil size, shape and type of wire which will keep losses low and thus give a high Q factor at a particular frequency, this is no guarantee that the Q factor will remain high for other frequencies. The graph in Fig. 5 shows that *the Q factor of a practical coil varies with frequency*. This graph tells us that in general, a coil which has a very high Q factor at a low frequency

will lose its  $Q$  factor rapidly at higher frequencies (curve 1), whereas a coil with a reasonably high  $Q$  factor at low and medium frequencies will tend to retain this  $Q$  factor value as frequency is increased (curve 2 in Fig. 5). Naturally these facts about how the  $Q$  factor for a coil varies with frequency are of extreme importance in connection with the tuning circuits of radio receivers, for these circuits are made to respond to a wide range of carrier frequencies.

**L/C Ratio for Tuning Circuits.** In a tuning circuit the ratio of coil inductance to condenser capacity, commonly known as the  $L/C$  ratio, oftentimes has an important effect upon the selectivity of the circuit, determining the ability of the circuit to reject frequencies which differ from the resonant frequency. In series resonant circuits a large  $L/C$  ratio (secured by using a high-inductance coil) gives best selectivity, but in parallel resonant circuits it is the lowest  $L/C$  ratio which gives the best selectivity (assuming, of course, that the losses in the coils are the same for all  $L/C$  ratios).

When comparing the  $L/C$  ratios for two different tuning circuits, it is important that the same unit of inductance be used for each coil and the same unit of capacity be used for each condenser. For example, a typical 500-to-1,500 kc. tuning circuit uses a 250 microhenry coil and a 400 micro-microfarad (mmfd.) maximum capacity variable condenser; with this condenser set at 100 mmfd., the  $L/C$  ratio is  $250 \div 100$ , or 2.5. For comparison purposes, then, the capacity and inductance used in another circuit should be expressed in these same units when figuring out the  $L/C$  ratio.

### Coupling Methods for R.F. Tuning Circuits

The method used for coupling an R.F. tuning circuit to a tube in an R.F. amplifier or for coupling two tuning

circuits together naturally has a great deal to do with the operation of the amplifier. Four basic coupling methods are in general use:

**Method 1. Directly coupled resonant load** like that in Fig. 6A, where a parallel resonant circuit is directly connected to the plate of the amplifier tube.

**Method 2. Tuned secondary transformer load** like that in Fig. 8A, where a series resonant circuit is inductively coupled to the plate of the amplifier tube.

**Method 3. Double-tuned transformer load** like that in Fig. 9A, where two resonant circuits which are mutually coupled inductively serve as the plate load for the amplifier tube.

**Method 4. Double-tuned capacity-coupled load**, like that in Fig. 11A, where two resonant circuits which are mutually coupled capacitively serve as the amplifier tube plate load.

**Coefficient of Coupling.** We are particularly interested in the amount of coupling provided between the resonant load circuit and the plate circuit of the R.F. amplifier tube by each of the basic coupling methods. Direct coupling such as that in method 1 pro-

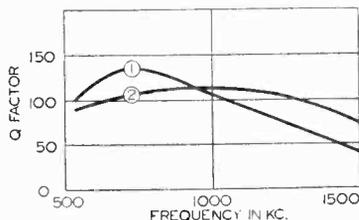


FIG. 5. Chart showing how  $Q$  factor varies with frequency for two representative broadcast band coils

vide maximum possible coupling between the source (the vacuum tube plate circuit) and the load (the resonant circuit).

With transformer loads as in methods 2 and 3, where mutual inductance ( $M$ ) provides the coupling between circuits, we can change the amount of coupling by changing the position of either  $L_1$  or  $L_2$ . When all of the flux produced by current flowing in the primary coil ( $L_1$ ) links with the sec-

ondary coil ( $L_2$ ), we have maximum coupling and maximum possible mutual inductance. When the primary and secondary coils are so positioned with relation to each other that only a part of the flux produced by one coil links with the other coil, the mutual inductance between them will be low and we have the condition of weak coupling. Under this condition the ratio of the actual mutual inductance to the maximum obtainable mutual inductance with very close coupling is a measure of the amount of coupling; this ratio is always a number less than 1, and is known as the *coefficient of coupling*. When two resonant tuning circuits are coupled, as in methods 3 and 4, the coefficient of coupling is quite important in determining the response characteristic of the circuit.

*Effect of Vacuum Tube Characteristics.* At the present time the use of pentode tubes in the R.F. amplifiers of radio receivers is becoming almost universal. Screen grid tubes are still to be found in some receivers, but triode tubes will be found only in the very old R.F. amplifier circuits. There are two features of pentode tubes which are significant in connection with tuning circuits and which should therefore be considered before we study coupling methods: 1, *pentode tubes have high A.C. plate resistance values*, because the plate is farther away from the cathode than in an ordinary triode tube; 2, *pentode tubes have high amplification factors*, because the control grid is considerably closer to the cathode than in an ordinary triode tube. For example, a 6K7 super-control pentode tube has an A.C. plate resistance of about 1,000,000 ohms and an amplification factor ( $\mu$ ) of about 1,000; a 6J7 pentode tube has an A.C. plate resistance greater than 1,500,000 ohms and a  $\mu$  of over 1,500.

In a practical modern R.F. amplifier the effective load resistance rarely

reaches a value higher than about one-tenth the A.C. plate resistance of the pentode tube. This condition corresponds to that of a generator which has an internal resistance of at least 1,000,000 ohms, connected to a load resistance of not more than 100,000 ohms; you can readily see that variations in the load resistance will have little effect upon the A.C. plate current. You could actually short out the load resistance without affecting the A.C. plate current more than 10%, since the resistance of the generator itself has the greatest amount of control over circuit current. Engineers say that under this condition we have a *constant-current generator*.

Because the A.C. plate resistance of a pentode tube in an R.F. amplifier stage of a receiver is extremely high with relation to its plate load resistance, we can consider a pentode tube as a *constant-current generator* for any given grid input signal voltage, and thus simplify greatly our study of resonant circuits. The constant-value A.C. plate current which the tube will deliver is easy to determine if the mutual conductance of the tube is known. Here is the rule: The A.C. plate current in microamperes in a pentode tube is equal to the *grid A.C. voltage in volts multiplied by the mutual conductance of the tube in micromhos*. Remember that with pentode tubes, whatever changes you make in the tuning circuit or circuits which serve as the load will have only negligible effect upon the A.C. plate currents.

The high amplification factor of a pentode tube produces a reasonably high output voltage despite the great loss in signal voltage due to the fact that the load resistance is less than the A.C. plate resistance.

*Importance of Studying Coupling Methods.* Now let us make a more detailed study of each of the basic coupling methods in order to see what the

engineer can do to secure the desired amount of selectivity, gain and fidelity for a particular purpose when designing radio apparatus using one of these circuits, and in order to see what the serviceman can do to alter the selectivity, gain or fidelity characteristics of a receiver. This discussion will show you the importance of making exact tube and parts replacements in R.F. circuits. Substituting some newly developed tube or part for older equipment may lead to serious trouble, unless the substitution is made with full knowledge

grid input voltage  $e_g$  multiplied by the amplification factor ( $\mu$ ) of the tube; this is simply the equivalent vacuum tube circuit idea which you have already studied, and which can be proven correct either by mathematics or experiment. In the simplified circuit of Fig. 6B, this A.C. voltage  $e_p$  is divided between the load circuit (across points 1 and 2) and the A.C. plate resistance  $r_p$ ; only when the resonant resistance of this load is many times greater than  $r_p$  will the load get practically all of the source voltage  $e_p$ .\*

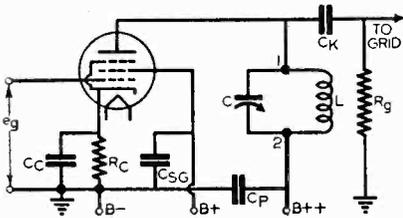


FIG. 6A. Directly coupled resonant load (a parallel resonant circuit whose terminals are 1 and 2) as commonly used in a pentode R.F. amplifier stage.

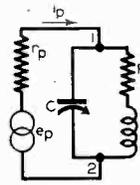


FIG. 6B. Simplified equivalent circuit of the directly coupled resonant load arrangement in Fig. 6A.

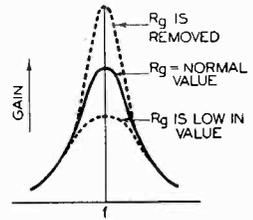


FIG. 7. Effects of grid resistor values upon the shape of the response curve for the circuit of Fig. 6A.

of the design problems involved in the change.

### Directly Coupled Resonant Loads

**Simplified Circuit.** We can simplify our study of the directly coupled parallel resonant load circuit in Fig. 6A by omitting all parts which have no effect upon the performance of the amplifying circuit and redrawing our circuit in the form shown in Fig. 6B. Observe that the R.F. by-pass condensers, the D.C. voltage supply leads for the tube electrodes and the automatic C bias resistor have been omitted, and the resistance of coil  $L$  is now represented by a separate resistor  $R$ . The vacuum tube has been replaced by an A.C. voltage source  $e_p$  (shown as an A.C. generator) in series with the A.C. plate resistance  $r_p$ , with the value of  $e_p$  being equal to the A.C.

**Effect of Coil Inductance and Q Factor on Over-all Amplification.** We know that the resonant resistance of the parallel resonant circuit in Fig. 6B depends upon the coil reactance and upon the Q factor of the coil; for any given frequency, then, increasing the inductance of the coil will increase its reactance, will increase the resonant load resistance, and will therefore increase the over-all amplification. Likewise, increasing the Q factor of the coil will increase the over-all amplification.

\*The voltage produced across the load for a one-volt grid input signal is a measure of the true amplification of a stage. In any pentode circuit such as we have here, we simply multiply the mutual conductance of the tube in micromhos by the resonant resistance of the load in ohms and divide the result by 1,000,000 to get the true or over-all amplification. Any change which increases the load resistance will therefore increase the over-all amplification.

For circuits which handle only a single frequency, such as I.F. amplifier stages, the designer endeavors to select a coil which has the highest usable inductance and at the same time has a high Q factor. From this it should be obvious to you that when a coil in the tuning circuit of an R.F. amplifier becomes defective, the mere substitution of another coil having the same inductance is not a guarantee that the correct over-all amplification will be obtained. The practical radio man uses exact duplicate replacement coils in order to make sure that he is using a coil with the correct Q factor as well as the correct inductance.

*Effect of Frequency upon Over-all Amplification.* Suppose we have a parallel resonant load circuit which tunes over the frequency range from 500 kc. to 1,500 kc.; will the over-all amplification remain the same at all frequencies in this range? Curve 2 in Fig. 5 shows that as frequency is increased above the middle-frequency range, the Q factor of a practical radio coil decreases: this reduction in Q factor would tend to reduce the over-all amplification at the higher frequencies. On the other hand, increasing the frequency three times (from 500 kc. to 1,500 kc.) would increase the reactance of the coil three times, thus increasing the resonant load resistance three times and consequently increasing the over-all amplification about three times. In most cases this increase in amplification due to an increase in coil reactance will completely overshadow the decrease due to a reduction in Q factor at high frequencies. Consequently we can say that increasing the signal frequency being fed to a tuned R.F. amplifier which uses a parallel resonant circuit as a plate load will in most cases *increase the over-all amplification* of the amplifier. Only when the Q factor drops rapidly with frequency, as in curve 1 in Fig. 5, will

the over-all gain remain constant or drop when frequency is increased.

*Effect of Frequency upon Selectivity.* The selectivity of an R.F. amplifier circuit is defined as the ratio of the amplification provided at the desired signal frequency to the amplification provided at the nearest undesired signal frequency. The effect of frequency upon selectivity can be demonstrated by considering an actual case, that where a broadcast band receiver using the circuit represented by Figs. 6A and 6B is first tuned to 500 kc. and then to 1,500 kc. For simplicity we will assume that in each case the nearest undesired frequency is 100 kc. away.

Remember that lowering the resonant resistance (or reactance if off resonance) of the tuned load circuit in this amplifier will lower the over-all amplification of the amplifier. Let us first assume that the Q factor remains constant over the range from 500 kc. to 1,500 kc.\* Under this condition we know that the amplification at resonance for 1,500 kc. will be three times the amplification for 500 kc.

At frequencies considerably off resonance, a parallel resonant circuit like that in Fig. 6B acts like that reactance (capacitive or inductive) which is lowest in ohmic value. At 400 kc. (100 kc. below the desired 500 kc. signal), then, the circuit will have a reactance essentially equal to the reactance of the coil, and this reactance will determine the amplification at this nearest undesired signal frequency in our example. Selectivity at 500 kc. will then be the ratio of the amplification at 500 kc. to the amplification at 400 kc. At 1,400 kc. (100 kc. below the desired 1,500 kc. signal), the reactance of the coil will be approximately three times its value at 400 kc. and there-

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\*The Q factor of a practical coil actually varies considerably with frequency; a constant Q factor is assumed here in order to simplify this discussion.

fore the amplification at the nearest undesired signal in this case will be three times what it was for the nearest undesired signal in the previous case. Since the amplification at 1,500 kc. is likewise three times the value at 500 kc., we will get the same selectivity ratio in both cases. This means that *when the Q factor is assumed to be constant over the tuning range, the selectivity of a parallel resonant circuit will likewise remain essentially constant over the tuning range.*

With the coils generally used in the tuning circuits of radio receivers, however, the Q factor will be found to decrease considerably as frequency is increased. This decrease in Q factor lowers the amplification at resonance but has no effect upon amplification at off-resonance frequencies; consequently the selectivity ratios for higher frequencies will be reduced. In actual circuits the selectivity will vary in much the same manner as the Q factor of the coil varies; the curves in Fig. 5 thus can tell us how selectivity varies with frequency. In general, you will find it easier to separate stations in the middle region of a radio receiver tuning range than at the extreme high or low frequency ends, for coil Q factors are generally highest in the middle region.

#### *Effect of C Bias Voltage Variations.*

It is a known fact that in super-control pentode tubes, increasing the negative C bias voltage has the effect of decreasing the mutual conductance of the tube, thereby lowering the A.C. plate current and reducing the over-all amplification of the stage. Variations in C bias voltage have little effect upon the selectivity of R.F. amplifiers using pentode tubes, however, for the resonant load characteristics are not affected by C bias voltage variations.

#### *Effect of Loading the Tuned Circuit.*

There is one simple way of changing the selectivity and gain of a tuned cir-

cuit such as that shown in Fig. 6A, and this is to change the load on the tuned circuit by changing the value of grid resistor  $R_g$ . Obviously the gain will be lowered when the value of  $R_g$  is reduced, for this grid resistor acts in parallel with the resonant circuit and therefore reduces the load resistance in the plate circuit of the tube (assuming that coupling condenser  $C_K$  has negligible reactance). Off resonance, however, the value of  $R_g$  has little effect upon the amplification of undesired



All-wave superheterodyne receiver being aligned for peak response. An R.F. signal generator and an output meter are the only instruments needed. The output of the R.F. signal generator section is fed into the receiver, and the multimeter section is used as an output meter.

frequencies, for now the reactance of the tuned circuit will be considerably lower than the value of  $R_g$  ordinarily used, and the reactance of the tuned circuit will control amplification. Loading the tuned circuit of an R.F. amplifier by reducing the value of  $R_g$  thus *lowers selectivity* by lowering the amplification ratio for desired and undesired signals.

The effects of various values of  $R_g$  are shown graphically in Fig. 7; the middle response curve is for the condition where the usual fairly high value of  $R_g$  is in the circuit of Fig. 6A. The lower dotted curve is for a lower value of  $R_g$ , and clearly shows that both the

gain and the selectivity of a tuned circuit in an R.F. amplifier are lowered when the tuned circuit is loaded by reducing the ohmic value of the grid resistor for the following stage; fidelity is considerably improved, however, for the broad peak insures uniform amplification of all side frequencies. The uppermost dotted curve is for the condition where  $R_g$  is removed entirely; now amplification is very high and selectivity is good, but fidelity is very poor because side frequencies are amplified very much less than the carrier frequency. Thus you can see that the fidelity of a receiver using a tuning circuit like that in Fig. 8A could be improved by reducing the value of  $R_g$ , provided that a loss in amplification and selectivity is permissible. Likewise, DX (distance-getting) performance could be increased by using a higher value of  $R_g$ .

*High-gain Directly Coupled Resonant Load Circuit.* In some inexpensive receivers which have few tubes, the circuit in Fig. 6A is modified slightly to eliminate the amplification-reducing effect of  $R_g$ . Instead of coupling to the next tube through  $C_K$  and  $R_g$ , a second coil is inductively coupled to coil  $L$  and connected to the grid and cathode of the following tube. (The circuit arrangement is exactly as in Fig. 9A with tuning condenser  $C_2$  removed.) By using a large mutual inductance between the two similar coils, the entire resonant circuit voltage can be transferred to the following stage without appreciable loss. If there are more turns on the secondary than on the primary, a step-up in output voltage can be secured. Fidelity is somewhat poor with this arrangement, however, for a sharp peak response curve is secured. The peak can be broadened by shunting the tuning condenser ( $C$  in Fig. 8A with a 20,000 to 200,000 ohm resistor, but this will, of course, lower the gain.

## Tuned Secondary Transformer Loads

The R.F. amplifier circuit arrangement shown in Fig. 8A, which uses a tuned secondary transformer load, is widely used in tuned radio frequency receivers and in the station selector (preselector) circuits of superheterodyne receivers. Since it is modern practice to use pentode or screen grid tubes in this circuit also, the A.C. plate current will be essentially independent of conditions in the resonant load circuit; we will assume this condition during our discussion of this circuit.

A simplified version of the tuned secondary transformer load circuit in Fig. 8A appears in Fig. 8B. Observe

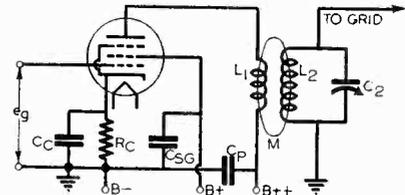


FIG. 8A. Tuned secondary transformer load circuit as commonly used in a pentode R.F. amplifier stage.

that the primary and secondary coils,  $L_1$  and  $L_2$ , are each divided into two parts in this equivalent circuit. Those sections on each coil which link each other completely through mutual inductance  $M$  provide the only coupling between the two circuits; the remaining sections, which do not link each other at all, are known as the primary and secondary leakage inductances respectively. The primary leakage inductance is always equal to the original primary inductance minus that inductance which totally links with the secondary coil, and this totally-linking portion is in turn equal to the primary inductance multiplied by the coefficient of coupling of the original circuit. This same reasoning also applies to the sections of the secondary inductance.

Resonance exists in the secondary circuit for a desired signal frequency when the reactance of tuning condenser  $C_2$  is equal to the reactance of the secondary leakage inductance at that frequency; the reactance of the other secondary coil section can be neglected, for it is cancelled out through mutual inductance  $M$  by the corresponding primary coil section. At resonance, then, the secondary resistance  $R_2$  is the only factor which limits secondary current. This secondary resistance has an effect upon the primary circuit; engineers say that it is reflected into the primary circuit, with the reflected value of resistance being determined by the value of mutual inductance  $M$ , by the signal frequency and by the original value of  $R_2$ . Increasing the mutual inductance, increasing the frequency or decreasing the ohmic value of  $R_2$  will increase the value of reflected resistance in the primary circuit.\* When the ohmic value of the reflected resistance equals the A.C. plate resistance of the tube, maximum gain is obtained in this tuned secondary transformer load circuit. It is almost impossible to secure this condition with screen grid and pentode tubes because of their high A.C. plate resistance values, but it can be done with triode tubes which have low A.C. plate resistance values.

In all practical circuits which use screen grid or pentode tubes, the reflected resistance in the primary circuit is negligibly small in comparison to the A.C. plate resistance of the tube. The primary signal current is therefore

\*Although undoubtedly you will never have to determine exactly what the reflected resistance value is, the formula for doing this is presented here for reference purposes: Multiplying the mutual reactance of  $M$  by itself once and then dividing by the secondary circuit resistance  $R_2$  gives the reflected resistance in the primary circuit. (Reflected resistance in ohms =  $2\pi fM \times 2\pi fM \div R_2$ , where  $\pi = 3.14$ ,  $f$  = frequency in cycles and  $M$  = mutual inductance in henrys.)

unaffected by any conditions in the secondary circuit which might change the value of reflected resistance. Furthermore, the reactance of the primary leakage inductance is also negligibly small in comparison to the A.C. plate resistance, and consequently the changes in this reactance with frequency can be neglected. Thus we find that the only two factors which control the A.C. plate current in a practical R.F. amplifier of the tuned secondary transformer load type are the mutual conductance of the tube and the applied A.C. grid voltage. The signal voltage  $e_s$  which is induced in the secondary depends upon this A.C. plate current and mutual reactance of  $M$ .

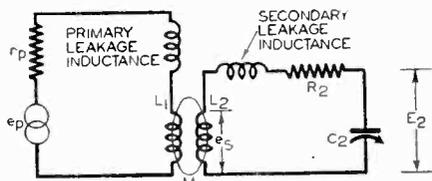


FIG. 8B. Simplified equivalent circuit of the tuned secondary transformer load circuit in Fig. 8A.

This means that desired as well as undesired signal voltages applied to the grid of the tube in Fig. 8A will produce the same induced voltage in the secondary circuit of the plate load; it remains for the series resonant secondary circuit to tune out all but the desired frequencies.

*Effect of Tuned Secondary Circuit on Amplification.* The voltage across secondary tuning condenser  $C_2$  in Fig. 8A is applied directly to the grid of the following tube, and therefore anything which increases the value of this voltage at resonance will make the over-all amplification of the stage greater. Since the tuned secondary is a series resonant circuit, this voltage is equal to the source voltage  $e_s$  multiplied by the Q factor of the secondary coil. Anything which increases the Q factor of the coil therefore increases

the over-all amplification. This is a good reason for using high  $Q$  coils in circuits of this type.

For a given grid input voltage, increasing the secondary induced voltage will also increase the over-all amplification of the stage; this can be done by using a tube which has a higher mutual conductance, by increasing the mutual inductance  $M$  between the primary and secondary, and by increasing the frequency of the desired signal.

To reduce the gain of an R.F. amplifier circuit like this, as is often necessary in actual receivers in order to prevent overloading of one or more following stages or to reduce the output volume to a desired lower level, the mutual conductance of the tube can be reduced. In a super-control pentode tube, the usual procedure for doing this involves increasing the negative C bias voltage on the tube.

When tuning the secondary circuit over a given frequency range, the amplification of the circuit will depend upon the manner in which the  $Q$  factor of the coil varies with frequency.  $Q$  factor normally decreases with frequency, but the reduction in gain due to this effect will be offset by an increase in gain due to the increased mutual reactance\* of  $M$  at higher frequencies; the result is that the amplification of a typical tuned secondary transformer load circuit varies slightly over its tuning range.

*Effect of Tuning Circuit on Selectivity.* If the  $Q$  factor of the coil in a tuned secondary transformer load circuit remained constant over a given frequency range, the selectivity would also remain constant;  $Q$  factor decreases at the higher frequencies, however, so selectivity will likewise decrease.

\*Mutual reactance equals mutual inductance times frequency in cycles per second times the number 6.28.

The peak of the response curve can be broadened or rounded by shunting the tuning condenser in Fig. 8A with a 20,000 to 100,000 ohm resistor. This will reduce the gain at resonance by reducing circuit current but will have little effect at off-resonance frequencies.

## Double-Tuned Transformer Loads

The double-tuned transformer load circuit shown in Fig. 9A is widely used in the I.F. stages of superheterodyne receivers. One advantage of this circuit is that it can provide high selectivity while keeping the number of tubes at a minimum; another advantage is that the circuit can be adjusted to give an almost flat-top response curve for high fidelity. We will consider first the adjustment of this circuit for peak response and for band-pass response, and will then analyze the factors which

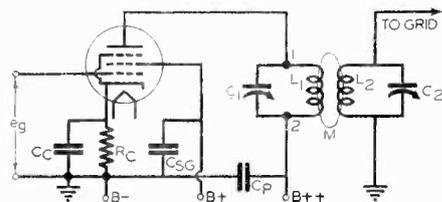


FIG. 9A. Double-tuned transformer load circuit as commonly used in a pentode R.F. amplifier stage.

control the circuit gain, selectivity and fidelity.

The circuit of Fig. 9A has been redrawn in simplified form in Fig. 9B in order that we can concentrate our study upon those parts which affect the performance of the tuned circuit. Again we have the coils divided into totally coupled sections and leakage inductance sections, as before.  $R_1$  represents the A.C. resistance of the primary coil, while  $R_2$  represents the A.C. resistance of the secondary coil.

*Adjusting for Peak Response.* With the circuit of Fig. 9B, a single-peak response characteristic can be obtained at any desired I.F. value by adjusting tuning condenser  $C_2$  until its reactance

exactly equals the secondary leakage reactance of the secondary winding at that I.F. value. Condenser  $C_1$  is then adjusted in the same way in order to tune the primary circuit to resonance at the desired I.F. value. In a practical case this adjustment is made by connecting an R.F. voltmeter across  $C_2$  to measure the secondary circuit output voltage  $E_2$ ;  $C_1$  and  $C_2$  are then adjusted for maximum voltmeter reading.

At resonance, there is only the secondary circuit resistance  $R_2$  to be reflected into the primary tuned circuit through mutual inductance  $M$ . This increases the resistance in the primary circuit, and therefore decreases the  $Q$  factor of the primary circuit. The presence of the secondary circuit thus reduces the voltage across primary tuning condenser  $C_1$ , thereby reducing the amount of resonant stepped-up current through coil  $L_1$  and reducing the amount of voltage induced in the secondary winding for resonance step-up by the secondary series resonant circuit. A double-tuned transformer load circuit which uses identical coils in both tuning circuits always gives less gain than a single parallel resonant load circuit using only one of these coils.

Increasing the mutual inductance  $M$  by increasing the coupling between primary and secondary coils tends to make the resistance which is reflected into the primary circuit larger, thus reducing  $E_1$ , but at the same time the increased mutual inductance serves to increase the voltage which is induced in the secondary. Since the two effects tend to offset each other, there is naturally a particular value of mutual inductance which will give the highest possible gain. By experiments as well as calculation, engineers have determined that this optimum condition occurs when the mutual inductance  $M$  is such that the resistance which is re-

flected into the primary circuit is exactly equal to the primary circuit resistance  $R_1$ .

In practical radio circuits, condensers  $C_1$  and  $C_2$  in Fig. 9A are usually of the same capacity, and consequently the primary and secondary coils must be alike in order to secure resonance at the same frequency. Since double-tuned transformer load circuits are ordinarily found in I.F. amplifier stages, these condensers will be of the trimmer type, independently adjustable. With any given coils, the condenser settings required for resonance are definitely fixed, and only the coupling between coils can be varied in

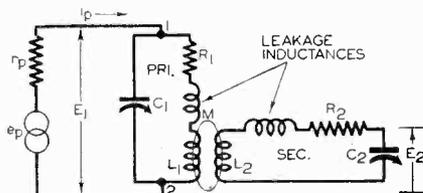


FIG. 9B. Simplified equivalent circuit of the double-tuned transformer load circuit in Fig. 9A.

order to secure optimum conditions. That coupling which gives the maximum possible gain is called the optimum or critical coupling. With identical coils, optimum coupling is obtained when the coefficient of coupling is exactly equal to 1 divided by the  $Q$  factor of the coil. For example, if the coil has a  $Q$  factor of 100, the coefficient of coupling for optimum gain will be  $1 \div 100$ , or .01.

Once a double-tuned transformer load circuit is adjusted for optimum coupling, either increasing or decreasing the coupling from this value will reduce the over-all gain. Increasing the coefficient of coupling also reduces the selectivity, broadening the peak of the response curve because of the increase in the resistance reflected into the primary circuit, but decreasing the coupling serves to increase the selectivity. When double-tuned transformer load

circuits are used to give a single-peak response, the coupling is kept less than the critical value (the coils are under-coupled) in order to improve the selectivity at a sacrifice of gain.

*Adjusting for Double-Peak Response.* Let us assume first that there is no coupling whatsoever between the secondary circuit and the primary circuit, both being tuned to the frequency of the source. Clearly there will be no voltage induced in the secondary coil under this condition, and consequently there will be no output voltage. Now as we bring the secondary coil closer to the primary coil, energy will be transferred through the mutual inductance

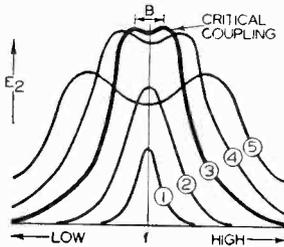


FIG. 10. Effect of variations in coupling upon the shape of the response curve for the double-tuned transformer load circuit in Fig. 9A.

between the two coils and an output voltage will be produced across  $C_2$ . As we gradually increase the coupling up to the critical coupling value, this output voltage will continue to increase. Increasing the coupling beyond the critical value will at first have no effect upon the output voltage and will then gradually cause the output voltage to lower. What actually happens is shown in Fig. 10; curve 1 represents a very small amount of coupling, while curves 2, 3, 4 and 5 represent increasing greater amounts of couplings, with curve 3 representing the conditions for critical coupling.

Let us see why double peaks occur in curves 4 and 5 in Fig. 10. When coupling is below the critical value,

both primary and secondary circuits have the same resonant frequency (assuming correct tuning) and single-peak response is secured. Increasing the coupling beyond the critical value without changing the tuning condenser settings causes the leakage inductance in each circuit to decrease, and consequently the resonant frequency of the secondary circuit becomes higher than before (the lower the effective inductance for a given capacity value, the higher is the resonant frequency of a resonant circuit). Increasing the coupling beyond the critical value likewise lowers the primary leakage inductance, making the primary circuit resonate also at a higher frequency. A signal at this higher frequency will thus undergo resonant step-up in both the primary and secondary circuits, giving a high output voltage.

When we exceed the critical coupling value, we have another interaction between the two circuits to consider. Consider conditions for a signal which is lower than the original resonant frequency of the secondary. At this lower frequency the series resonant secondary circuit will act as a capacity and will reflect into the primary circuit as an inductance which increases the effective primary inductance, bringing the primary circuit to resonance at this lower frequency. Likewise, the parallel resonant primary circuit alone will act as an inductance at the lower frequency and will reflect into the secondary as a capacity which brings the secondary to resonance at the lower frequency also. Primary and secondary circuits are thus resonant to both a higher and a lower frequency than the resonant frequency of either circuit alone, and as a result we have double-peak or band-pass response.

At critical coupling, an essentially flat-top response curve is obtained, with fairly steep sides and double peaks just beginning to form. The ap-

## RESONANT CIRCUIT DATA

At frequencies below the resonant frequency:

A series resonant circuit acts as a capacity.

A parallel resonant circuit acts as an inductance.

At frequencies above the resonant frequency:

A series resonant circuit acts as an inductance.

A parallel resonant circuit acts as a capacity.

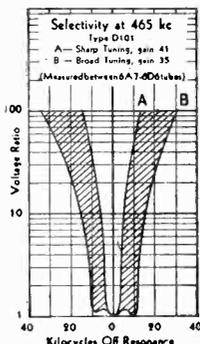
General rule: At any off-resonant frequency, a series resonant circuit acts like that part which has the highest reactance, while a parallel resonant circuit acts like that part which has the lowest reactance.

proximate distance between these peaks, as measured in kc., is easily computed; it is equal to the coefficient of coupling multiplied by the carrier frequency to which the R.F. amplifier is tuned. For example, suppose that the coils in Fig. 9A each have a Q factor of 150, and the two resonant circuits are tuned to 460 kc. Critical coupling will be obtained when the coefficient of coupling is 1 divided by 150, which equals .0067; the separation between peaks at this condition of critical coupling will then be equal to 460 (the frequency in kc.) multiplied by .0067, or about 3 kc. The actual practical band width for this particular case will be somewhat wider (about 4.5 kc. in this example), for it is determined by the distance between the steep sides of the response curve rather than by the distance between the peaks.

In the above example, we can triple the amount of separation between peaks (thereby tripling the band-width) by tripling the amount of coupling, but this will give a valley between peaks as in curves 4 and 5 in Fig. 10. If, in addition to tripling the coupling, we reduce the Q factors of the coils to one-third of their original values, we can secure this same tripling of band-width without having a valley between peaks, but the gain will be considerably less when this is done. Clearly we

must sacrifice one advantage in order to secure another, in this particular case.

When an engineer designs band-pass circuits like that in Fig. 9A, he endeavors to choose coils which will give the desired band-width at critical



*Courtesy Aladdin Radio Industries, Inc.*

This Aladdin type D I.F. transformer provides any desired performance characteristics from high fidelity to extreme selectivity without appreciable variation in gain, at the option of the user. Rotating the center coil by means of a knob on the front panel of the receiver serves to vary the coupling between the other two coils. In the under-coupled position, as represented by curve A, we have sharp tuning and severe cutting of side frequencies above about 4,000 cycles, while curve B corresponds to over-coupling, with practically uniform amplification of all side frequencies up to 10,000 cycles off resonance. Three built-in trimmer condensers, one for tuning each coil, give additional control over transformer characteristics.

coupling. If he finds that this is impossible, he increases the coupling beyond the critical value enough to secure the desired band-width. This gives somewhat lower gain (curve 5 in

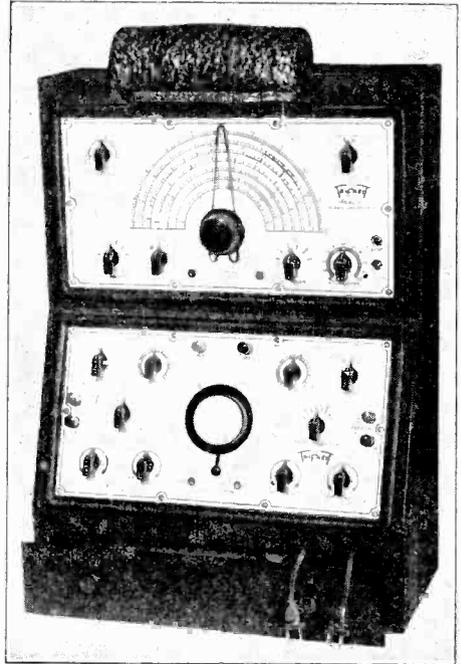
Fig. 10 represents lower gain than curves 3 and 4).

*Adjusting Actual Band-Pass Circuits.* Band-pass circuits in practical radio frequency amplifiers are easily adjusted with the aid of a cathode ray oscilloscope which is connected to reproduce the response curve of the amplifier. By watching the effects of each adjustment upon the shape of this response curve, the radio serviceman knows definitely when he has secured the desired shape.

Ordinarily a serviceman does not know whether a particular band-pass circuit is over-coupled (greater than critical coupling), or is under-coupled (less than critical coupling). For this reason he must always try to adjust the circuit for peak response before making band-pass adjustments. In the case of Fig. 9A, this is done simply by feeding into the amplifier an R.F. signal of a definite frequency and adjusting tuning condensers  $C_1$  and  $C_2$  for maximum output voltage at the detector load, as indicated either by an output meter or by a cathode ray oscilloscope. It does not matter which condenser is adjusted first. The adjustments are repeated several times. If there are several band-pass circuits in an amplifier, each is adjusted in this same way for peak response. Inability to secure a single-peak response when this is done means that the coils are over-coupled; in this case the preliminary adjustment is omitted.

A cathode ray oscilloscope and a variable frequency signal generator are now connected to the amplifier in the proper manner to produce on the cathode ray oscilloscope screen the actual response curve of the entire amplifier. Assuming that the circuit in Fig. 9A is the only band-pass circuit in the amplifier, the capacity of  $C_1$  is increased slightly and the capacity of  $C_2$  is decreased the same amount (or  $C_1$  may be decreased and  $C_2$  increased) to

make the flat top appear. If the coils have approximately critical coupling, only small changes in the condenser settings will be needed to secure double peaks. When coupling is considerably less than the critical value, however, the two condensers may have to be changed considerably from their peak response settings before the two peaks will appear. If the coils in a double-



*Courtesy Triplett Electrical Instrument Co.*

A cathode ray oscilloscope (lower section) and a frequency-modulated signal generator (upper section) are here mounted in a single cabinet. They can be connected to a radio receiver in a few minutes for producing the response curve of the receiver, and have a host of other radio servicing and testing uses as well.

tuned circuit are over-coupled, it will be impossible to adjust for a single-peak response; the two peaks will always be present, and changes in condenser settings will merely serve to change the distance between peaks and alter the symmetry of appearance.

The more the condenser settings are changed, the greater will be the distance between the two peaks and the deeper will be the valley between the peaks. If the peaks of the response

curve of a double-tuned circuit are excessively high with respect to the valley between them when the desired band-width is secured, they can be reduced by *shunting the primary and secondary circuit tuning condensers with 20,000 to 100,000-ohm resistors*. This actually pulls the two peaks closer to the level of the valley, thus making the response more uniform over the entire band-width. It may be necessary to experiment with different values of resistors in the primary and secondary circuits in order to reduce both peaks equal amounts, if condenser adjustments alone are not sufficient to give a symmetrical response curve.

## Double-Tuned Capacity-Coupled Loads

Two resonant circuits may be coupled together by capacity coupling, as illustrated in Fig. 11A, instead of inductive coupling. Circuit action is much the same with either type of coupling, so there should be no difficulty in understanding how the simplified version of this capacity-coupled arrangement, shown in Fig. 11B, behaves under various operating conditions.

When both tuned circuits are at resonance, the secondary tuned circuit made up of  $L_2$ ,  $C_2$  and  $C_M$  will act as a resistance. If we measure the resonant resistance between points 2 and 3, in

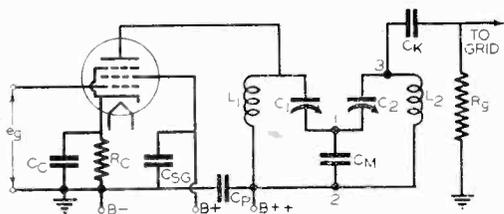


FIG. 11A. Double-tuned capacity-coupled load circuit as commonly used in a pentode R.F. amplifier stage.

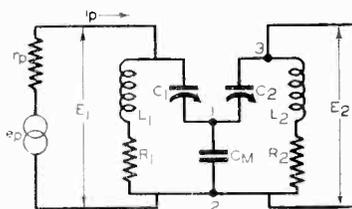


FIG. 11B. Simplified equivalent circuit of the double-tuned capacity-coupled load circuit in Fig. 11A.

In many modern high-fidelity receivers, controls are provided which vary the coupling between the primary and secondary coils of one or more double-tuned transformer load circuits in order to permit a choice between peak response and band-pass response. These transformers will have critical coupling when this control is set for band-pass performance. In some circuits which use variable coupling, you may find that the variable coupling control is labeled "volume control" on the schematic circuit diagram and on the panel of the receiver. In this case the coupling will always be less than the critical value, and under this condition any changes in coupling will affect the receiver gain far more than it will the receiver selectivity. We thus have simply a unique volume control.

the circuit of Fig. 11A, we find it to be quite high.

You will note that tuning condenser  $C_2$  and coupling condenser  $C_M$  are in series between points 2 and 3. A part of the total high resistance between these points will therefore exist across the terminals of coupling condenser  $C_M$ , and experience has shown that the actual value of this resistance will be proportional to the reactance of  $C_M$ . Increasing the reactance of coupling condenser  $C_M$  will therefore increase the effective resistance across it and thereby increase the resistance which is reflected into the primary circuit.

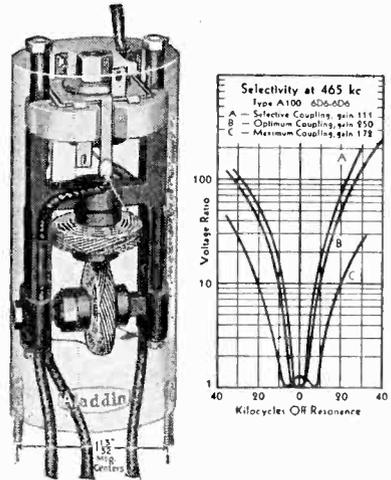
At resonance, the resonant resistance of the primary tuning circuit made up of  $L_1$ ,  $C_1$  and  $C_M$  determines the value of the signal voltage  $E_1$  which will be produced across the primary coil,

Again we assume that the pentode tube maintains the plate current  $i_p$  essentially constant; the resonant circuit current through the primary coil is of course greater than  $i_p$  and varies in value as circuit conditions are changed. The greater the resistance reflected into the primary circuit by the secondary, the lower will be the resonant circuit current through the primary coil and the lower will be the voltage across the primary. Just as with inductive coupling, maximum secondary circuit output voltage is obtained with critical coupling between the primary and the secondary, under which condition the resistance reflected into the primary is equal to the primary circuit resistance  $R_1$ . When  $C_1$  is equal to  $C_2$ , as is usually the case in a practical circuit, the coefficient of coupling is the ratio of  $C_1$  to  $C_M$ .

Just as with other band-pass circuits, increasing the coefficient of coupling beyond the critical value results in a double-peak response curve. The reasons for the existence of a double peak under these conditions are quite easily understood. When the capacity of coupling condenser  $C_M$  is made lower than the critical value, thereby raising its reactance, raising the voltage developed across it for transfer from primary to secondary, and increasing the coupling, the combined capacity of condensers  $C_2$  and  $C_M$  in series will be less than before, making the resonant frequency of the secondary higher than before. Likewise the lower capacity of  $C_M$  in series with  $C_1$  will raise the resonant frequency of the primary to the same high value. We have thus accounted for the higher-frequency peak on the band-pass response curve.

Now consider the effects of interaction between the two tuned circuits when the coefficient of coupling is beyond the critical value and the incoming signal is below the resonant frequency. Looking at the primary cir-

cuit first, we note that secondary circuit components  $L_2$  and  $C_2$  in series are shunted across  $C_M$ ; the reactance of  $C_2$  being greater than that of  $L_2$ , this shunt combination will act as a condenser in parallel with  $C_M$ , increasing the effective capacity of  $C_M$  in the primary circuit. This higher value of  $C_M$  acting in series with  $C_1$  increases the effective capacity in the primary circuit and therefore causes the primary



Courtesy Aladdin Radio Industries, Inc.

Here is an adjustable coupling I.F. transformer (Type A Aladdin Poliviron transformer) which you may encounter in high quality radio receivers and particularly in commercial receivers. The curves at the right show its performance at different degrees of coupling. Note that the coils are at right angles to each other; moving the lower coil in either direction along its shaft by means of adjusting screws changes the coupling. Critical or optimum coupling is secured when the lower coil is not directly under the upper coil; curve *B* represents this condition, giving the ratio of the voltage output at resonance to the voltage output at frequencies up to 40 kc. above and below resonance. Amplification at the resonant frequency (465 kc.) is 250 for this condition, an unusually high value. Moving the coil more nearly under the upper coil gives under-coupling, with reduced gain but improved selectivity, as indicated by curve *A*. Moving the lower coil out from the critical coupling position gives over-coupling, with double peak response for high-fidelity band-pass results.

circuit to resonate at a lower frequency than the original value. Exactly the same analysis will show that the secondary circuit will also resonate at this lower frequency. Thus we have accounted for the lower-than-normal peak in the band-pass response curve.

The reason why double peaks are not secured when the coefficient of coupling is less than the critical value is quite simple. Under this condition the capacity of  $C_M$  is so high that interaction between the two circuits is negligible. As a result, each circuit resonates at only one frequency, and a single peak response curve is secured.

With capacitive coupling between two resonant circuits, the separation between peaks at critical coupling is equal to the operating frequency multiplied by the coefficient of coupling, just as in the case of inductive coupling. Increasing the  $Q$  factor of the coils in the circuit of Fig. 11A increases the value of coupling required for critical coupling, thereby lessening the separation between peaks at critical coupling, but higher gain is secured. Increasing the coupling (by reducing the capacity of  $C_M$ ) increases the separation between peaks and therefore increases the band-width.

Just as with any double-tuned circuit, the separation between the two peaks may be increased, giving greater band-width, by detuning the primary and secondary circuits (by increasing the capacity in one circuit and decreasing the capacity an equal amount in the other). This deepens the valley between the peaks, but loading of the primary and secondary circuits with resistors will flatten the peaks and give a more uniform response curve over the entire band-width.

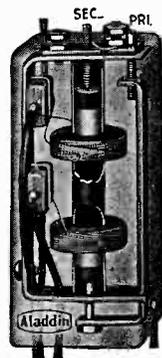
The procedure just described for widening the band-width and loading the tuning circuit is commonly used in the tuned circuits of television receivers, where a band-width of the order of 6 megacycles (6,000,000 cycles) is generally required. This procedure reduces the gain considerably, and consequently television receivers require more amplifier stages than broadcast band receivers. Both

capacitive and inductive coupling are used in the double-tuned circuits of television receivers.

### Combination Capacitive and Inductive Coupling

In tuned circuits which are to operate over a definite range of radio frequencies, such as the preselector circuits in superheterodyne receivers, the amount of coupling between tuned circuits is not entirely independent of frequency. As a result, the response curve of the circuit will vary in shape and

Fixed mica condensers are permanently connected across the coils in this type L inductance-tuned Aladdin I.F. transformer. Tuning is accomplished by changing the positions of the Polytiron pulverized iron cores inside each coil; this is done by adjusting the set screws at the top, labeled *PRI.* and *SEC.* Coupling is always less than the critical value. A gain of about 100 is secured when the coils are adjusted for peak response.



Courtesy Aladdin Radio Industries, Inc.

size at different frequencies, with the nature of the variation being dependent upon the type of coupling used.

Consider the double-tuned transformer load circuit in Fig. 9A first. Assuming that the coil  $Q$  factors remain the same as we tune from a low to a high radio frequency, we see that the reactance of mutual inductance  $M$  increases with frequency, producing a greater induced voltage in the secondary circuit and therefore giving a greater gain at the higher frequencies.

With the capacity-coupled circuit shown in Fig. 11A, on the other hand, the reactance of coupling condenser  $C_M$  decreases as we tune from a low to a high frequency, with the result that less voltage is fed into the secondary tuned circuit at higher frequencies, and

less gain is realized at higher frequencies.

Capacitive and inductive coupling are often used in the same circuit in order to make an amplifier have the same gain at both low and high frequencies. A typical band-pass R.F. amplifier circuit which uses both types of coupling is given in Fig. 12; as you can see, primary coil  $L_1$  is inductively coupled (through mutual inductance  $M$ ) to secondary coil  $L_2$ , and these two coils are also coupled together capacitively by coupling condenser  $C_M$ . The antenna is inductively coupled to coil  $L_1$  in the first tuned circuit through mutual inductance  $M_A$ . Resistor  $R_g$ , having a high ohmic value, provides

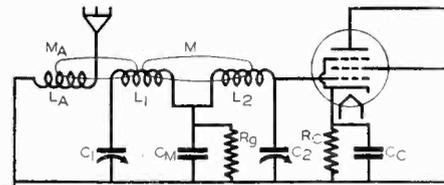


FIG. 12. Double-tuned R.F. amplifier input circuit using both inductive ( $M$ ) and capacitive ( $C_M$ ) coupling to equalize the gain at all frequencies in the tuning range.

a conductive path around coupling condenser  $C_M$ , so that the negative C bias voltage across  $R_C$  and  $C_C$  will be applied to the grid of the tube.

### Tuned Secondary with Fixed Tuned Primary Circuit

When a single resonant circuit gives more gain at high frequencies than at low frequencies, the circuit illustrated in Fig. 13 is often used to equalize the gain over the entire tuning range. The primary winding  $L_1$  has a distributed capacity between turns which is in effect equivalent to a condenser  $C_1$  connected across the winding. The primary coil can be so designed that the coil and its distributed capacity form a parallel resonant circuit which reso-

nates at a low frequency in the tuning range, giving resonant current step-up at low frequencies and thereby inducing larger voltages than normal in the secondary circuit at the lower frequencies in the tuning range. At the higher frequencies, however, the primary circuit is off resonance and the value of mutual inductance  $M$  alone determines the amount of voltage induced in the secondary.

Unfortunately, this use of a fixed tuned primary circuit to boost the gain at low frequencies works entirely too well, boosting the gain at low frequencies so much that we have the reverse of the initial unequal gain condition. It is for this reason that a small amount

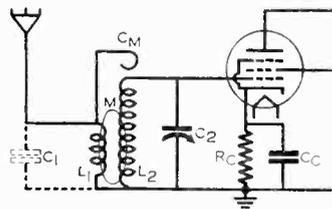
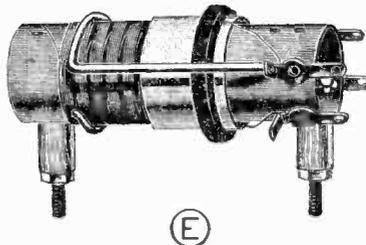
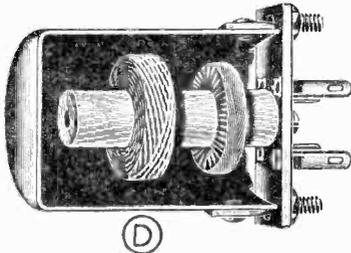
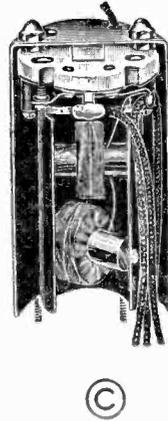
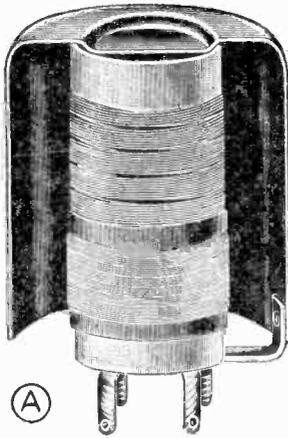


FIG. 13. Another R.F. amplifier input circuit which uses both inductive and capacitive coupling to equalize the gain over the tuning range. Fixed tuning of the primary coil is provided by the distributed capacity ( $C_1$ ) between turns of the coil.

of capacity coupling is used between the high R.F. terminal of the primary coil and the high R.F. or grid terminal of the secondary coil; a stiff copper wire connected to the primary and curled partly around the secondary winding at the grid terminal end provides sufficient capacity coupling for the purpose. This wire does not connect to the secondary coil, as only capacity coupling is desired.

The circuit diagram in Fig. 13 illustrates the use of this capacity-coupling wire in the antenna system of a receiver. Antenna coil  $L_1$ , along with its distributed capacity, tunes to about the lowest frequency in the tuning range of the secondary circuit (this tuning range being controlled by  $L_2$



Typical R.F. transformers which can be used in the tuning circuits described in this lesson. Getting the required inductance is only a small part of the coil designer's work; he must also consider such important things as the Q factor and how it varies with frequency, the degree of coupling needed for desired results, effect of the shield, and gain-equalizing methods. The examples shown here are all Gen-Ral coils, made by General Mfg. Co.

A—Shielded R.F. transformer having a bank-wound secondary made of litz wire, with an ordinary single-layer primary wound over the lower end of the secondary. With the average pentode tube this transformer gives a gain of 42 at 550 kc. and a gain of 62 at 1,500 kc.; this nearly uniform gain over the tuning range is secured by proper design of the secondary coil. You will find this coil used in circuits like the tuned secondary transformer load arrangement of Fig. 3A.

B—Shielded I.F. transformer. Triple-section cross-wound primary and secondary coils are used to give a high Q factor. The coupling (spacing between coils) is adjusted during manufacture for optimum results. Tuning condensers are built into the housing, one being connected across

each coil. Look for units like this in double-tuned transformer load circuits such as that in Fig. 9A.

C—Shielded I.F. transformer. The cross-wound coils are made with litz wire and are weakly coupled (coefficient of coupling is less than the critical value) to give sharp peak response. Also used in circuits like Fig. 9A.

D—Shielded R.F. transformer having cross-wound primary and secondary coils mounted permanently on a wood dowel; coupling cannot be adjusted. This construction gives fair gain, but this varies considerably over the tuning range. Used in circuits like Fig. 3A.

E—Unshielded antenna coil for broadcast band. Secondary is made of litz wire, bank-wound directly on coil form, while the lattice or cross-wound primary is located over one end of the secondary. The primary is self-tuned (by its distributed capacity) to about 550 kc., and a heavy-wire coupling ring provides capacity coupling between primary and secondary, so that the gain is very nearly uniform throughout the broadcast band. Fig. 13 illustrates how the coil is used.

and  $C_2$ ). A stiff wire attached to the antenna post (the high R.F. terminal) of primary coil  $L_1$  loops around the grid end of secondary coil  $L_2$ . This arrangement gives practically uniform gain over the entire tuning range. You will find this stiff wire scheme used for

capacity coupling between coils in practically all of the small universal A.C.-D.C. tuned radio frequency receivers. (In some receivers this stiff wire is replaced by several turns of insulated wire or by two short insulated leads twisted together.)

## R.F. Tuning Circuits Used in Actual Receivers

Although the R.F. tuning circuit diagrams already presented in this lesson are typical of those used in most radio receivers, there are a number of minor variations of these circuits which will occasionally be encountered. Three different receivers which use out-of-the-ordinary R.F. tuning circuits have been selected for study. The circuit diagrams of the R.F. sections of these sets are shown in much the same way as they appear in the re-

ly does not have A.V.C.; the volume is controlled by a dual rheostat which simultaneously varies the resistance across the antenna input coil (this is done by  $R_1$ ) and varies the C bias voltage on the R.F. tubes (this is done by  $R_2$ ). The cathode, screen grid and plate by-pass condensers for each tube are mounted in a single case (indicated by dotted lines and labeled  $C_p$ ).

You will readily recognize  $L_1-C_1$  as the first tuning circuit, this being fed with the antenna signal by a direct connection from the antenna to a tap

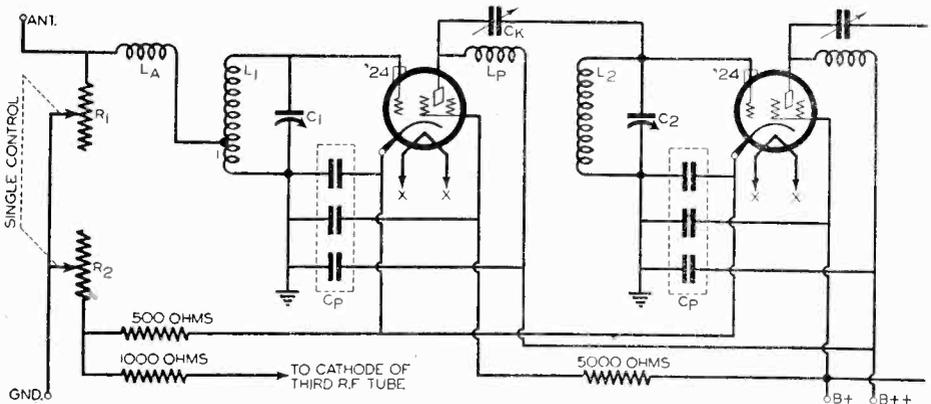


FIG. 14. A portion of the circuit diagram of the General Motors model MA T.R.F. receiver, illustrating an interesting method used to equalize the gain over the entire tuning range.

spective service manuals, in order to make you familiar with the different drawing techniques used in actual practice.

*General Motors Model MA T.R.F. Receiver.* The first two stages of this receiver are shown in Fig. 14; in the actual receiver there are three R.F. stages, giving four tuned circuits which can be tuned over the 550 kc. to 1,500 kc. broadcast band by a single tuning control. Observe that type 24 screen grid tubes are used; since these have very high A.C. plate resistance values, their A.C. plate currents can be considered essentially constant under all circuit conditions. This receiver clear-

at point 1 on  $L_1$ . But notice that the antenna signal must first pass through coil  $L_A$ , which is *not* inductively coupled to  $L_1$ ; we suspect immediately that this is used by the designer to equalize the gain over the tuning range. Coil  $L_A$  actually serves to tune the antenna at the low frequency end of the tuning range, reducing the antenna impedance to a minimum value and thereby causing a large signal current to flow into the portion of coil  $L_1$  between point 1 and ground. At the higher frequencies in the range, the antenna is off tune, antenna current is lower and the gain of the input tuning circuit is consequently lower also. Thus coil  $L_A$  can



is both inductive and capacitive coupling, giving equalization of gain over the entire tuning range. Tuning condenser  $C_5$  tunes the portion of  $L_3$  between point 1 and the grid of the first tube over the entire tuning range;  $C_4$  is simply a trimmer condenser used for alignment purposes. The three vertical lines and one diagonal line above coil  $L_3$  symbolically indicate that the coil uses a pulverized iron core, with the diagonal line indicating that the position of the core and therefore the inductance are adjustable. This core thus provides an aligning adjustment at the low frequency end of the tuning

to equalize the gain; in a circuit of this type, we are safe in assuming that the primary will tune to a lower frequency in the range to counteract increasing gain provided by the secondary circuit at the higher frequencies. The low resistance of the secondary coil indicates that this coil has a high Q factor.

*Conclusion.* I do not expect you to be able to analyze in a few minutes any R.F. tuning circuit which you may encounter after your study of this lesson, for there apparently is no limit to the number of variations which are being used by circuit designers. I do say, however, that after you have studied a

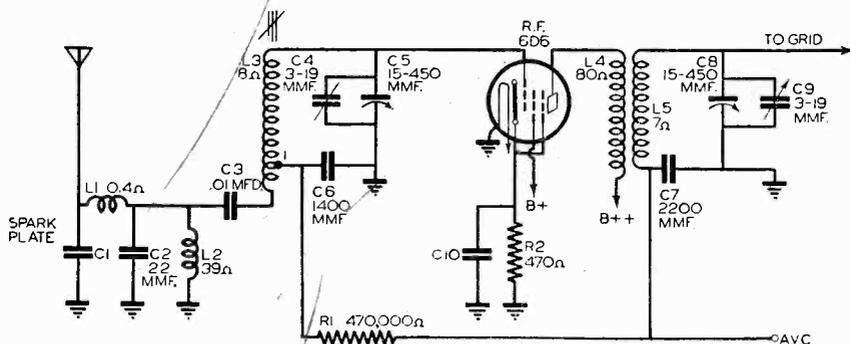


FIG. 16. Input circuits of the model 5M RCA Victor Auto Radio. Condenser  $C_1$ , labeled SPARK PLATE, provides a shunt path to ground for high-frequency interference signals produced by the spark plugs and ignition system of the automobile.

range, while high frequency trimmer  $C_4$  permits alignment at the high frequency end of the range.

Turning now to the plate circuit of the first tube, you might say at first glance that this contained a conventional tuned secondary transformer load. Closer examination would show, however, that while the D.C. resistance of the primary coil was 80 ohms, the secondary coil had a D.C. resistance of only 7 ohms. These figures tell us that the primary coil must be made of many turns of fine wire and hence must have a high inductance and high distributed capacity. This distributed capacity tunes the primary to some frequency in the tuning range in order

number of other schematic circuit diagrams in the same way as we have analyzed the three diagrams at the end of this lesson and after you have acquired practical experience with the behavior of tuning circuits in actual radio receivers, you will find yourself in a far better position to service radio receivers and know what to expect from them than if you simply made adjustments blindly without knowing exactly what you were doing. You will know *why* certain things happen when certain screws and knobs are turned in a receiver, and will be able to determine whether or not the fidelity or selectivity of a particular receiver can be improved.

# Lesson Questions

Be sure to number your Answer Sheet 22FR-2.

Place your Student Number on every Answer Sheet.

Never hold up one set of lesson answers until you have another ready to send in. Send each lesson in by itself before you start on the next lesson. In this way we will be able to work together much more closely, you'll get more out of your Course, and you will receive the best possible lesson service.

1. What does a sharp peak response curve for an R.F. amplifier indicate as regards gain and selectivity?
2. Will an R.F. amplifier which has a sharp peak response curve cause severe attenuation of the higher modulation frequencies?
3. Will amplitude distortion occur when an R.F. amplifier has a symmetrical double-peak response curve?
4. How is a high Q factor obtained for a coil which has a given inductance?
5. Does the Q factor of a practical coil vary with frequency?
6. Why can a pentode tube in the R.F. amplifier stage of a receiver be considered as a constant-current generator?
7. What effect does the loading of a tuned circuit in an R.F. amplifier have upon its selectivity and gain?
8. Name two advantages which are secured by using a double-tuned transformer load circuit.
9. Is it possible to adjust the trimmer condensers for a single-peak response when the two coils in a double-tuned circuit are over-coupled?
10. If the peaks in the response curve of a double-tuned circuit are excessively high with respect to the valley between them, how can they be reduced without changing the band width?

## THE VALUE OF REVIEW

Man has acquired so much new knowledge in recent years that it has become impossible for one person to know even a small fraction of the available information. Educational authorities realize this fact, and the colleges of today consider a man well-educated if he knows the elementary ideas *and knows where to find other information when he wants it.*

Radio, along with the other fields of endeavor, has outgrown the memorizing ability of the human mind. Also, radio is such a comprehensive field that occasionally you cannot recall important facts previously studied. Review is obviously the solution to this problem.

Time spent in review several weeks or months after a book is studied will be far more profitable than an equivalent amount of extra time spent on the book initially, for your mind has then had a chance to file and store away the information secured from the first study. Each review results in more information being transferred from the textbook to your mind, and soon, with no conscious attempt to memorize, you will find yourself able to recall an amazing number of valuable facts.

J. E. SMITH

**HOW BROADCAST, ALL-WAVE,  
AND TELEVISION SUPERHET-  
ERODYNE RECEIVERS WORK**

23FR-3

**NATIONAL RADIO INSTITUTE**

ESTABLISHED 1914

**WASHINGTON, D. C.**



# RULE NO. 23

practical facts, practical circuits and practical superheterodyne receivers, most of these being presented in previous lessons. This means that this lesson gives a bit of difficulty, you can clear up by reviewing the previous lesson which covers the theory of oscillators, if the operation of the local oscillator in a superheterodyne is not quite clear to you, review the lesson on oscillators.

1. The Superiority of the Superheterodyne . . . . . Pages 1-5  
A brief comparison of the superheterodyne and the t.r.f. receiver. A review of the parts of the superheterodyne. Answer Lesson Questions 1 and 2.
2. The Importance of the Preselector . . . . . Pages 6-13  
By presuming that there is no preselector, we show the number of undesirable interference conditions which can exist. There are a number of service hints in this section, so study it carefully. Answer Lesson Questions 3, 4, 5 and 6.
3. The Local Oscillator . . . . . Pages 14-17  
There are a number of requirements placed on the local oscillator. It has to supply a voltage at the proper frequency, without frequency drift, and without fluctuations in amount. Typical circuits are shown.
4. The Mixer-First Detector . . . . . Pages 18-24  
This is the stage where the frequency change takes place. Today, combination oscillator and detector stages are the most common, so be sure to study these circuits carefully. Answer Lesson Question 7.
5. Oscillator-Preselector Tracking . . . . . Pages 25-30  
To the serviceman, this is a very important section. Alignment of receivers to obtain the proper tracking is a frequent service step, so read this section a number of times. Answer Lesson Questions 8 and 9.
6. The Intermediate Frequency Amplifier . . . . . Pages 31-34  
The importance of the choice of the i.f. value; typical circuits; requirements for high fidelity; variable selectivity. Answer Lesson Question 10.
- No. 7 The Television Superheterodyne . . . . . Pages 35-36  
This short section shows how television sets use standard circuits, with certain modifications, for video signal amplification. This is a preview of this branch of radio.
8. Mail Your Answers for this Lesson to N.R.I. for Grading.
9. Start Studying the Next Lesson.

# HOW BROADCAST, ALL-WAVE, AND TELEVISION SUPERHETERODYNE RECEIVERS WORK

## The Superiority of the Superheterodyne

**T**HE superheterodyne receiver is by far the most common type of radio set in use today, so it is well worth further study. You have learned, in previous lessons, how almost all the components of a "superhet" function and have been given a preview of this circuit. Now we will collect this information and show you in more detail how the various components act when they are combined. If you find you are a little hazy on the operation of some individual circuit component, by all means make a quick review of the lesson in which it was discussed. Doing so will make it much easier for you to grasp all the details of the operation of the superheterodyne.

Basically, the superheterodyne principle of r.f. amplification involves the conversion of each incoming signal within the receiver tuning range to one definite fixed radio frequency which is known as the intermediate frequency or the i.f. value.\* It is much easier to get optimum (best) results from an r.f. amplifier which always works at the same frequency, as in the i.f. amplifier in a superheterodyne receiver, than from an amplifier which is tuned through a wide range of radio frequencies, as in t.r.f. receivers.

\*When a signal demodulator (second detector), audio amplifier, loudspeaker, and power pack are added to a superheterodyne r.f.-i.f. amplifier, the result is a complete superheterodyne receiver. The above-mentioned sections of the receiver are of conventional design and are all covered thoroughly elsewhere in the Course.

► This is true for frequency modulation receivers as well as for amplitude modulation receivers. Incidentally, both use the same *sections*—the important differences are that an FM receiver has a slightly different i.f. design, uses a limiter stage, and has a different demodulator or second detector. (FM receivers are described fully in



Courtesy Philco

This large console radio uses the superheterodyne circuit.

another lesson.) This lesson covers AM receivers; however, you will need all this information to gain a complete understanding of FM receivers.

**Review of R.F. Amplifier Operation.** If you will review in your mind the action of an r.f. amplifier, you will recall that the gain and selectivity of

any stage are dependent upon the resonant resistance of the plate load; this resonant resistance depends upon  $L/C$  (the ratio of coil inductance to condenser capacity), and upon the losses in the tank circuit. In tuning to different frequencies, we vary the condenser capacity and thus vary the  $L/C$  ratio. For this reason, and because tank circuit losses are different at each frequency, uniform amplification at all frequencies is not obtained readily. Thus the amplification and the selectivity of a t.r.f. receiver are different at each setting of the tuning dial.

► In the superheterodyne receiver, on the other hand, practically all of the gain and selectivity are produced at one fixed i.f. value, regardless of the

frequency of the incoming signal; selectivity and sensitivity are therefore uniform at all tuning dial settings.

A comparison of the performance curves of a superheterodyne receiver with those of a t.r.f. receiver having approximately the same number of tubes will show very clearly the superiority of the superheterodyne. Such curves are given at *A*, *B*, *C* and *D* in Fig. 1. In these, sensitivity is measured in terms of microvolts of signal input required to get a 50-milliwatt output; thus, a lower input value means a more sensitive receiver. The sensitivity curve for the superheterodyne (Fig. 1*A*) is practically constant for all frequencies in the tuning band, while the same curve for the t.r.f. set (Fig. 1*B*) varies

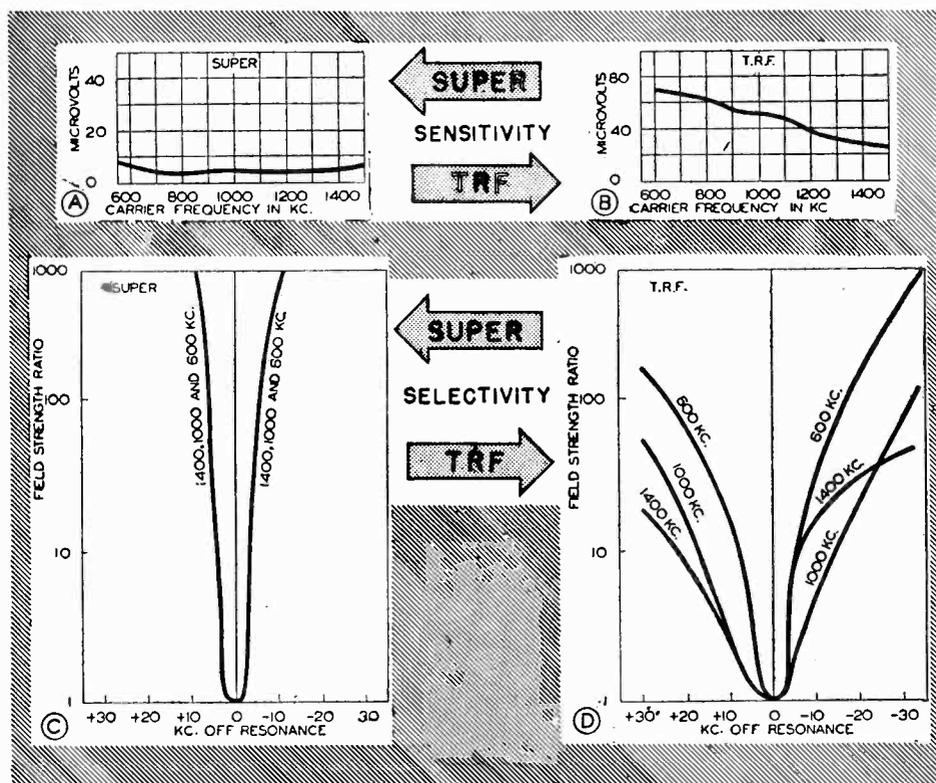


FIG. 1. These sensitivity and selectivity curves for a superheterodyne receiver and for a t.r.f. receiver show clearly the superiority of the superheterodyne.

considerably and represents much lower sensitivity. The selectivity curves in Figs. 1C and 1D give, for various frequency values off resonance, the ratio of signal input at resonance to the signal input required to give the same output at the off-resonance frequency: the steeper the curve, the more selective the receiver. The curve in Fig. 1C (for the super) thus represents very good selectivity which is uniform at all frequencies in the band, while the curves in 1D (for the t.r.f. receiver) show poor selectivity at 600 kc., and show increasingly poorer selectivity at higher frequencies in the band.

► If a t.r.f. receiver were designed for all-wave reception, it would be necessary to change every coil in the set each time the band\* was changed. In a superheterodyne receiver, however, only the coils in the preselector and oscillator sections need be changed. Because of this, every practical all-wave receiver is a superheterodyne. In addition, the public demand for high-fidelity reception, which entails amplification of wide side bands over the entire range of carrier frequencies, is easier to meet with the fixed i.f. amplifiers used in supers. It is no wonder that the superheterodyne is the most widely used system today; only a few of the compact and inexpensive midget receivers are manufactured now with t.r.f. circuits.

### PREVIEW OF SUPERHETERODYNE PRINCIPLES

Before the i.f. amplifier can do its job, the incoming signal with its picture or sound modulation must be converted to the intermediate frequency selected. This is done by combining the *incoming modulated r.f. carrier* with a

\*The range of frequencies which can be covered by a given combination of a coil and a condenser when one of them is variable is called a *band*.

*local r.f. oscillator signal*—a process called “frequency conversion.”

Now, when two signals of different frequencies are mixed together and sent through a detector or demodulator tube circuit, the plate current will consist of many components. There will be signals at the two original frequencies, a signal whose frequency is the



Courtesy Emerson Radio and Television Corp.

Battery-operated portables like this use the same basic superheterodyne circuit as is used in power-line operated sets, except for the use of battery type tubes.

difference between the original frequencies, a signal whose frequency is the sum of the original frequencies, and harmonics of all four of these frequencies. Each of these plate current components will carry the original modulation. By placing a highly selective resonant or tank circuit in the plate circuit of the detector, we can tune to any signal component and thus separate it from the others. Inasmuch as it is easier to make selective high-gain amplifiers for low r.f. values, the *difference* between the two original frequencies is always selected as the i.f. value.

If, for example, the incoming frequency is 1000 kc. and the local r.f. os-

cillator signal frequency is 1175 kc., the plate current of the detector circuit will contain these two frequencies, as well as 2175 kc. (the sum frequency) and 175 kc. (the difference frequency). From what we just said, the 175-kc. frequency will be selected as the i.f. value. The following stage (the i.f. amplifier) will be designed to pass and amplify only this 175-kc. signal and such side frequencies as are required to give the desired fidelity characteristic to the receiver.

Now, when a 500-kc. signal is tuned in, a local oscillator signal of either 325 kc. or 675 kc. will produce a difference frequency of 175 kc. If the incom-

it is difficult to tune a coil and condenser combination over a range having a frequency change which is greater than about 3.3 to 1. For this reason, the local oscillator in a superheterodyne is generally made to produce a frequency *higher than that of the incoming signal*. In our example, then, the oscillator would vary in frequency from 675 kc. to 1675 kc. as the receiver was tuned from 500 kc. to 1500 kc.

► Turning now to a block diagram of a superheterodyne type r.f. amplifier (Fig. 2), we find that there are four important sections: 1, the preselector; 2, the local oscillator; 3, the mixer-first detector; and 4, the i.f. amplifier. Be-

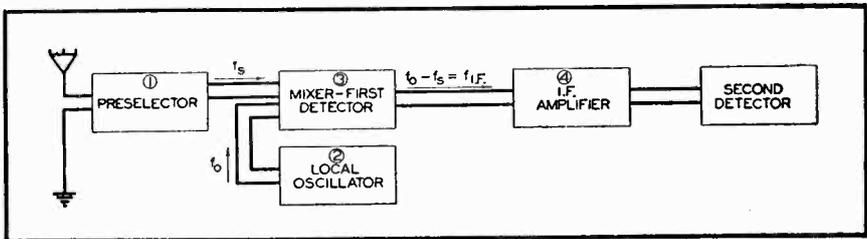


FIG. 2. A block diagram showing the four important r.f. sections of a superheterodyne; the second detector and the stages following it are identical for both t.r.f. and superheterodyne receivers, and hence are not discussed in this book.

ing signal is 1500 kc., a local oscillator signal of either 1675 kc. or 1325 kc. will produce the required i.f. value. This brings up the question of whether we should make the oscillator frequency lower or higher than the signal frequency.

► Let us consider this problem for a receiver which is to tune from 500 kc. to 1500 kc., with the i.f. at the fixed value of 175 kc. Clearly, the local oscillator must vary in frequency either from 325 kc. to 1325 kc. or from 675 kc. to 1675 kc. In the first case (325 to 1325) there is a 4 to 1 change from the highest to the lowest frequency, and in the second case (675 to 1675) there is a 2.5 to 1 change. As a practical matter,

fore we study these sections in detail, let us see what the important functions of each are.

**Preselector.** The preselector consists of one or more resonant circuits which can be adjusted to the frequency of the desired r.f. signal; it may or may not have r.f. amplifier stages. Theoretically, the preselector is not necessary in order to convert an r.f. signal to a lower frequency value. If you omit it, however, interfering signals will get into the mixer-first detector and they will react with each other or with the local oscillator signal to produce undesired signals at the i.f. value. Some gain (amplification) in the preselector is always desirable, because the mixer-

first detector creates inherent noise which can be over-ridden only by a strong r.f. signal.

**Local Oscillator.** Frequency conversion cannot take place unless a local signal is produced, which differs from the incoming r.f. value by the i.f. value. A local oscillator is therefore of fundamental importance in the superheterodyne circuit. The preselector tuning condenser and the oscillator tuning condenser either can be controlled independently or can be ganged together and controlled by a common tuning dial. The latter practice is now universally followed, for with it the oscillator is always at the correct setting to deliver a signal differing from the incoming signal by the i.f. value. Besides giving single-dial tuning, ganging together the oscillator and preselector tuning condensers reduces what is known as *repeat point reception*—a phenomenon which we will discuss a little further in this lesson.

**Mixer-First Detector.** This is the actual point of frequency conversion. Of great importance is the ability of this section to act as a detector for both the incoming and local oscillator signals, provided they are of reasonable intensity. Its plate circuit must contain a highly selective, high Q tank

circuit having a resonant frequency equal to the i.f. value (the difference frequency), so this frequency can be accepted and the other plate circuit components rejected. This tank circuit should have a low L/C ratio, so the resulting high capacity value will then act as a by-pass for the unwanted signals.

**I.F. Amplifier.** Here the i.f. signal, modulated with the original picture or sound signal, gets its real boost in gain. The i.f. amplifier must be able to amplify the i.f. signal and some or all of the important side-band frequencies, depending upon the type of receiver performance desired. The i.f. amplifier can be made highly selective, thus cutting out undesirable signals as well as unimportant side-band frequencies of the radio signal being received, or it can be made to have broad band-pass characteristics, thus passing a wide range of side frequencies. In the first case the i.f. amplifier is said to have good *adjacent channel selectivity*, and in the second case it has *high-fidelity* response characteristics.

Now that you've had a "preview" of the action of a superheterodyne r.f. amplifier, let's take up each section in detail. We will start with the pre-selector.

# The Importance of the Preselector

It is possible to impress a desired signal directly (without tuning) on the grid-cathode terminals of the mixer-first detector tube in a superheterodyne r.f. amplifier circuit and to secure, with the aid of the local oscillator, a beat frequency output signal having the desired i.f. value. Of course this is never done in practice, because there are a number of undesirable results. By considering first the effects encountered in a theoretical direct-input circuit (one having no preselector), we can learn a great deal about the importance of the preselector and about the problems which may be met in service work.

**Direct Input Circuit.** The mixer-first detector section of a superheterodyne circuit which has no preselector is given in Fig. 3; any signal  $f_s$  which is picked up by the antenna flows through resistor  $R$  to ground, and the r.f. voltage developed across this resistor is fed *directly* to the grid of the tube. The local oscillator feeds into the cathode circuit of the tube a signal which we will designate as  $f_o$ . Assume that the frequency of this signal can be independently controlled by varying the setting of oscillator tank condenser  $C_o$ . The resonant circuit in the plate lead of the tube is adjusted to the desired i.f. value  $f_{i.f.}$ , so that only the i.f. current produces a voltage drop across the i.f. resonant circuit for further amplification. For the present, we need not consider any other parts or sections of this superheterodyne circuit.

## REPEAT POINTS (DOUBLE-SPOT TUNING)

Assume that only one signal, having a frequency of 1000 kc., is being picked up by the antenna in Fig. 3, and that

the i.f. resonant circuit is adjusted to an i.f. value of 100 kc. Under these conditions the required 100-kc. beat frequency  $f_{i.f.}$  will be produced when the oscillator is tuned to 1100 kc. But we can also secure this 100-kc. beat frequency by setting the oscillator to 900 kc. Thus, there are two oscillator tuning dial settings at which the 1000-kc. incoming signal will be passed on to the i.f. amplifier. This condition is called *repeating* or *double-spot tuning*, for we have *repeat points*—two different settings at which the same station can be received—on the oscillator tuning dial. These repeat points are present in any superheterodyne circuit *when the oscillator can be separately tuned*, even if resistor  $R$  is replaced with a highly selective preselector circuit. The *repeat point for any one station is always separated from the correct oscillator dial setting by twice the i.f. value.*

► Of course, superheterodynes with separately tuned oscillators have long been obsolete. Ganging the preselector and oscillator will give single dial control and, in addition, will serve to eliminate repeat-point reception provided a good preselector is used.

**Service Hints.** A repeat point can occur in a single-dial receiver if the selectivity of the preselector is too low or is impaired by a circuit defect or improper adjustment, or if the signal from the station is exceedingly strong. In each case, you would hear a station at its correct dial setting and at a repeat-point setting which will be *below* the correct dial frequency setting by twice the i.f. value (assuming that the oscillator tunes *above* the frequency of the incoming signal). Examine the set for defects and check the alignment. If

there appear to be no defects and the alignment is normal, then the condition may be due to the design of the receiver. If no interference occurs with another desired signal, then you can ignore the condition, as it is impractical to change the receiver design. (Explain it to the customer if he remarks about it.) However, if there is interference, you should follow the hints in the next section of this lesson.

### IMAGE INTERFERENCE

From your own experience you know that signals of many different frequen-

the desired signal by *twice the i.f. value*, is heard along with the desired signal. This condition is called *image interference*.

► Notice that image interference is caused by repeat-point reception of an *undesired* station—in fact, a repeat-point signal could be called an image. The only difference is that now the undesired signal is interfering with a desired signal.

► The obvious solution to this problem is to use a highly selective preselector which is capable of tuning to the desired signal and of rejecting the

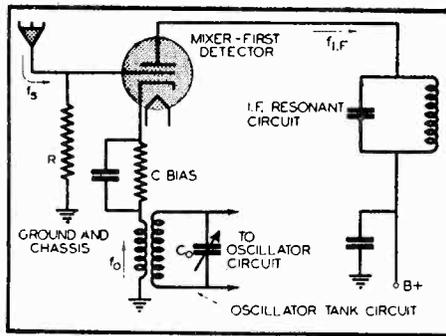


FIG. 3. Schematic diagram of the mixer-first detector section of an imaginary superheterodyne receiver which has no preselector.

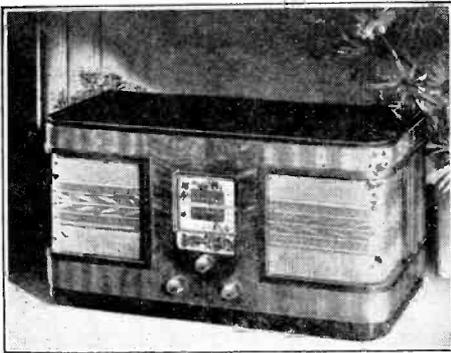
cies are always present in the antenna circuit of a receiver. Let us, therefore, assume that in addition to the desired 1000-kc. signal, there is an undesired 1200-kc. signal in the antenna circuit of Fig. 3. The oscillator is set at 1100 kc. in order to convert the desired signal to the i.f. value of 100 kc., but the undesired 1200-kc. signal can also mix with the 1100-kc. oscillator signal and produce a 100-kc. beat frequency. Both the desired 1000-kc. signal and the undesired 1200-kc. signal will then get through the i.f. section, be reproduced by the loudspeaker, and cause interference. This occurs when an interfering signal, whose frequency is above

image signal. However, the ideal preselector, which will allow only a single frequency or a narrow band of frequencies to pass and will absolutely reject all other frequencies, does not exist. An engineer is quite satisfied if he can design a preselector which *reduces* the strength of the interfering station (at the image frequency) 1000 times.\* This number is called the *image interference ratio* and, in this case, means

\*This ratio value must be considerably higher in receivers which have high-gain i.f. amplifier sections. In all-wave receivers, however, ratios as low as 100 to 1 for the higher frequency bands are considered acceptable because it is difficult to design all-wave sets with better ratios.

that an undesired image signal (of a strength equal to that of the desired signal) will be heard 1000 times weaker than the desired signal.

► Naturally, the closer the desired and undesired frequencies are to each other, the greater the problem is of separation. When a low i.f. value is used, it often takes two or even three preselector-tuned circuits in cascade to get an image interference ratio of 1000, and



Courtesy General Electric

A typical table-model superheterodyne receiver.

this increases the cost of building the receiver. Special image-rejecting circuits have been developed, but a simpler solution involves the use of a high i.f. value, somewhere between 250 and 500 kc.

Let us see what advantage is secured by using a high i.f. value. Assume that a certain receiver which is tuned to a 1000-kc. signal has an i.f. of 500 kc. The oscillator will therefore be at 1500 kc., and the image interference frequency will be 2000 kc. Only a simple preselector tuned to 1000 kc. is needed here to reject the undesired 2000-kc. signal, since the difference between the frequencies of desired and undesired signals is so great. On the other hand, if the i.f. value is 150 kc., the oscillator will be at 1150 kc., and the image will be at 1300 kc. This is much closer to 1000 kc., and will require a far better

preselector. Thus, superheterodynes having high i.f. values will have simple preselectors, while highly selective preselectors are imperative for sets with low i.f. values (values between about 135 kc. and 250 kc.).

Local stations of image frequency are sometimes so powerful, however, that even an image interference ratio of 1000 is insufficient to prevent interference with some desired station.

**Service Hints.** When a receiver is to be serviced for image interference, the first step would be to realign the preselector. Shortening the antenna is another remedy, for a very long antenna broadens the response characteristic of the first resonant circuit in the preselector, thus letting the undesired signal get through. If only one station is causing image interference, place in the antenna circuit a wave trap which is tuned to the frequency of this station.

Another possible solution would be to shift the i.f. value about 10 kc. This would move the repeat point to another frequency, away from the desired frequency. In this case, the repeat-point still exists but as long as interference with a *desired* signal is eliminated, it will not matter greatly. However, it is not always possible to make this change. It will interfere with the tracking when the oscillator tuning condenser has specially-cut plates, but it can be carried out if padder condensers are used. (You will study oscillator tracking later in this lesson.)

## INTERMODULATION INTERFERENCE

When *any two* signals whose frequencies differ by exactly the i.f. value exist in the antenna circuit of Fig. 3, they can beat *with each other*, producing the i.f. value without the aid of the local oscillator. This condition is known as *intermodulation interference*.

Without a preselector, these mixed signals cause an interference sounding like garbled (unintelligible) speech which would be heard regardless of the oscillator dial setting. When a preselector is used, only those r.f. signals which get through the preselector can produce this trouble, and this reduces the number of possible interfering signals greatly.

**Service Hints.** If garbled speech continues even when the local oscillator is blocked or is detuned, intermodulation interference is present. Improving the selectivity of the preselector blocks out the interfering stations and thus eliminates the trouble. If this is impractical, try cutting out one of the interfering stations with a wave trap; try a shorter aerial; and try changing the i.f. value of the receiver about 10 kc. (or to a value ending in 5).

### OSCILLATOR HARMONIC INTERFERENCE

We have assumed up to this time that the local oscillator is feeding only its fundamental frequency to the mixer-first detector. In many cases, especially where the oscillator and its coupling circuit are poorly designed, harmonics of the oscillator (usually only the second harmonic) may reach the mixer-first detector and react with an undesired incoming signal to produce an undesired i.f. beat signal. This condition is called *oscillator harmonic interference*. For each oscillator setting there may be two frequencies which incoming signals can have in order to beat with the second harmonic of the oscillator and produce an undesired i.f. signal. An example will explain how this occurs.

Suppose that a receiver which has a 260-kc. i.f. value is tuned to an 1160-kc. station. The oscillator fundamental frequency will be  $1160 \div 260$ , or 1420

kc. and the second harmonic of this will be 2840 kc. Now any signal, which differs from 2840 kc. by 260 kc. and which is strong enough to get through the preselector, will produce an i.f. beat frequency which can cause interference. Thus either an aircraft radio station at a frequency of 3100 kc. ( $2840 + 260$ ) or a commercial station on 2580 kc. ( $2840 - 260$ ), or both, could be heard on this set with the desired 1160-kc station.

You can identify oscillator harmonic interference if you can identify the interfering station as being one with a frequency either above or below the second harmonic of the oscillator by the i.f. value. (Oscillator harmonics higher than the second are so weak that they can be neglected.) The frequency difference between interfering and desired signals is so great in the case of harmonic interference that generally only strong local stations, such as amateur, police, commercial, or government code or phone stations, can ride through the preselector.

**Service Hints.** Either improving the selectivity of the preselector by realigning it to keep out the interfering signals, or adjusting the voltages of the oscillator to suppress its second harmonic are possible remedies for harmonic interference. However, the installation of a wave trap which is tuned to the frequency of the offending station is the simplest cure. When the interfering signal is especially strong, it may be necessary to shield the mixer-first detector to prevent the signal from acting on it directly without going through the preselector and wave trap; a filter in the power line may also be needed.

### CODE INTERFERENCE

Trouble can be caused also by an undesired signal *having a frequency equal*

to the i.f. value of the receiver. If this signal gets through the preselector, it will be passed on by the mixer-first detector to the i.f. amplifier, as there is no need for frequency conversion. Since most transmitters below 500 kc. (in the range of i.f. values) are code stations, this trouble is commonly referred to as code interference. It may be heard at any point on the tuning dial, but is strongest at the lower end of the receiver dial, around 550 kc.

Occasionally the second harmonic of a powerful local long-wave transmitter

tuning dial can be eliminated by installing a wave trap which is tuned to the interfering code station, by shortening the antenna, or by changing the i.f. value of the receiver.

► Bear in mind that the cures suggested here and in the previous service hints are necessary because of the inability of the preselector in the receiver to keep out undesired signals. These cures were suggested only because the fundamental cure, *redesigning* the preselector for greater selectivity, is usually impractical.

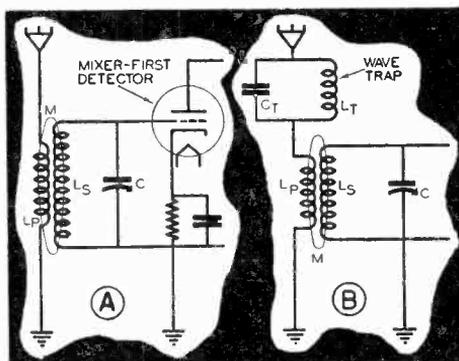


FIG. 4. A simple tuned r.f. transformer pre-selector is shown at A. A wave trap is added in B.

may produce code interference. For example, if a receiver has an i.f. of 480 kc., a local 240-kc. station might be heard at all points on the tuning dial. This could be produced because the second harmonic of the 240-kc. carrier is getting through the preselector, but a more likely condition is that the 240-kc. signal gets through the preselector and harmonics of it are created by the rectifying action of the mixer-first detector tube. The second harmonic, being of the i.f. value and carrying the modulation of the original code signal, would pass through the i.f. amplifier and produce interference.

**Service Hints.** Code interference which is heard at all settings of the

► Having studied the importance of the preselector, let us take up a few of the common preselector circuits.

#### R.F. TRANSFORMER WITH A TUNED SECONDARY

The simplest of all preselectors is the tuned-secondary r.f. transformer shown in Fig. 4A. It is often quite satisfactory in a receiver which uses a high i.f. value. The selectivity of this circuit is essentially dependent upon the mutual inductance  $M$  and upon the frequency of the desired incoming signal; increasing either reduces the selectivity. In a receiver which is to be used with a short antenna, the mutual inductance usually is made quite large

so that a strong input signal can be obtained, but such a receiver tunes broadly (has poor selectivity) when coupled to a long antenna. Even with a high i.f. value, such a circuit will tune broadly at very high frequencies (around 20 megacycles). It is for this reason that all-wave receivers ordinarily require better preselector circuits than this.

**Wave-Trap Circuit.** A wave trap is often connected in series with the primary of the r.f. transformer, as shown in Fig. 4B, to eliminate the interference caused by a particular station.  $C_T-L_T$  is the wave trap; it is tuned to the frequency of the interfering station by adjusting trimmer condenser  $C_T$ . Factory-installed traps usually tune to the i.f. value, to eliminate code interference, which is one of the most annoying and prevalent of the interferences. (A serviceman can connect a similar trap in the antenna lead of any radio receiver having image interference, harmonic interference, intermodulation, or code interference troubles caused by one station. The trap is tuned to the frequency of the interfering station.)

## BAND-PASS PRESELECTORS

In the preselector circuit just considered, only one tuned circuit contributes to the selectivity of the receiver. An extra tuned circuit connected in cascade as shown in Figs. 5A, 5B, and 5C will greatly improve the selectivity and, consequently, will reduce interference troubles. This extra circuit also permits adjustments which give band-pass characteristics to the preselector. The circuit shown in Fig. 5A is simply the circuit of Fig. 4A with an additional resonant circuit, made up of  $L_{S2}$  and  $C_2$ , coupled by mutual induction to  $L_{S1}$ . Fig. 5B shows two resonant circuits which are *directly*

coupled inductively, with coil section  $L_{M2}$  common to both resonant circuits. Capacitive coupling is used in the circuit of Fig. 5C, with condenser  $C_K$  common to both resonant circuits. For a fixed value of coupling, a single peak resonance characteristic is obtained when condensers  $C_1$  and  $C_2$  are tuned for maximum receiver output. Increases

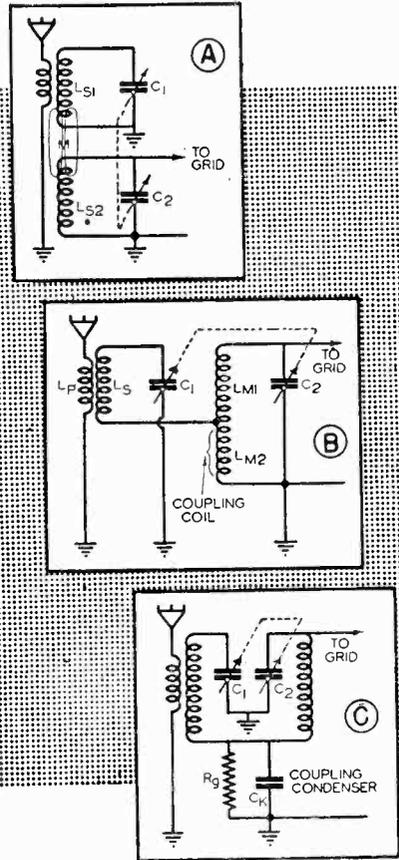


FIG. 5. Typical band-pass circuits. At A there is a band-pass preselector circuit with mutual inductance coupling. The dotted lines indicate that the two tuning condensers are ganged together to give a single-dial tuning control. B shows a band-pass preselector circuit with direct inductive coupling, coil section  $L_{M2}$  being common to both tank circuits. At C there is a band-pass preselector circuit with capacitive coupling.  $R_g$  is a .5-meg. to 1-meg. resistor which provides a d.c. grid return path to ground for the application of the C bias.

ing the coupling between the resonant circuits or tuning  $C_1$  above and  $C_2$  below the resonant frequency gives a flat or double-peak response characteristic. When this is done, the circuits are referred to as *band-pass preselectors*.

► Although band-pass preselectors are quite effective in eliminating (or at least reducing) the many types of interference troubles, they have one important drawback in that they considerably reduce the strength of the incoming signal. One way of overcoming this loss in signal strength is to step up the gain of the intermediate frequency

resonant circuits which have a high Q factor, which operate at a high temperature, and which have a wide response characteristic. A hissing or frying noise which is heard in the loudspeaker and is especially loud when the receiver is tuned between stations is an indication that thermal agitation is present. Because of this effect, the practical limit to the sensitivity of a receiver is about 1 microvolt (which means that the smallest signal voltage which can be made to give 50 milliwatts output to the loudspeaker is approximately 1 microvolt).

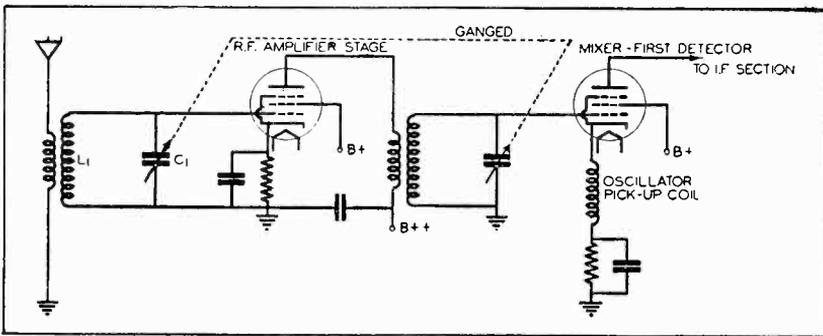


FIG. 6. A widely used preselector circuit, in which one stage of r.f. amplification boosts the strength of the incoming signal before it reaches the mixer-first detector. This additional amplification makes the desired incoming signal over-ride any noise which may be present in the mixer-first detector, and also lessens interference troubles.

amplifier section, but two undesirable effects, thermal agitation and converter noise, become annoying when this is done. Let us consider them:

**1. Thermal Agitation.** Free electrons are continually moving around in any conductor, producing tiny pulses of electron current. Some component of these pulses will be at the same frequency as that to which the receiver is tuned. Hence, these pulses of electron current undergo resonance step-up, along with the signal currents. The effect of these pulses of electron current is commonly designated as *thermal agitation* of electrons. It is greatest for

**2. Converter Noise.** Even more troublesome than thermal agitation is an effect which occurs in the mixer-first detector tube. Although the average plate current value is controlled accurately by signal voltages, nonetheless, there are current variations caused by irregularities in the electron emission. As a result, electrons emitted by the cathode arrive at the plate in spurts or "shots." These variations are amplified by succeeding stages along with the desired signals, and are heard in the loudspeaker as a "frying" noise. This action is known as the *shot effect*, as the *electron grain effect*, or as *frequency*

*converter\* noise.* Although the shot effect is present in practically all vacuum tubes, the variation in plate current due to it is ordinarily so small in comparison to the average plate current that the effect is negligible. In a tube which operates as a detector, however, the average plate current is so low, because of the high negative C bias, that the variations in current affect an appreciable part of the total plate current. Frequency converter noise is most noticeable when a receiver is tuned to a weak signal; strong signals tend to "drown out" or over-ride the noise. The strength of the signal fed to the input of the mixer-first detector must be large enough to make the signal-to-noise ratio at the output of this section as great as possible and thus minimize the effects of frequency converter noise.

► It is highly desirable to have a stage or two of r.f. amplification ahead of the frequency converter section in order to build up the strength of the incoming signal so that it will over-ride any converter noise which is present in the mixer-first detector tube. The greater the signal strength with respect to the noise, the less disturbing will be the noise.

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\*The mixer-first detector and the local oscillator together constitute the frequency converter section.

**R.F. Amplified Type of Preselector.** A widely used preselector circuit which contains a stage of r.f. amplification to increase the signal strength at the input to the mixer-first detector is shown in Fig. 6. The amplification of this stage must be made high enough to eliminate frequency converter noise, yet not so high that it increases the total gain of the receiver to the point where thermal agitation effects in the r.f. antenna transformer will come through. The first resonant circuit, consisting of  $L_1$  and  $C_1$  is sometimes replaced by a band-pass resonant circuit of the form shown in Figs. 5A, 5B, and 5C, in order to increase further the image-interference ratio.

This amplifier stage is not intended to increase the gain greatly—the i.f. amplifier still is being depended upon for most of the amplification. However, even a small increase is helpful in increasing the sensitivity and in overcoming the frequency-converter noise. An r.f. stage like this is frequently found in all-wave receivers.

Another advantage of the r.f. stage is the extra resonant circuit it provides in the preselector. (Both the r.f. and the mixer stages have resonant input circuits.) This extra selectivity helps to improve the image rejection ratio, thus reducing the interference.

# The Local Oscillator

If the local oscillator in a superheterodyne receiver is to perform its job satisfactorily, it must meet the following requirements:

1. At any dial setting, the frequency of the oscillator must be constant in value (there must be very little *frequency drift*).
2. The voltage which the local oscillator supplies to the mixer-first detector must be at least ten times *greater* than the voltage of the most powerful signal which is fed to the mixer-first detector by the preselector.
3. The variation in oscillator voltage output must be as small as possible as the frequency of the oscillator is changed by tuning (since absolute constancy is practically impossible, a maximum variation of 3 to 1 is considered by engineers to be satisfactory).
4. The output of the oscillator must have negligible harmonic content.
5. The oscillator itself should not radiate radio waves which would interfere with nearby receivers.
6. The oscillator must be coupled to the mixer-first detector in such a way that the frequency of the oscillator is not affected by changes in other receiver circuits.

The reasons for some of these practical requirements are important enough to warrant further explanation, so let us discuss these points before taking up typical circuits.

## FREQUENCY DRIFT

The frequencies of transmitting stations are kept within extremely close

tolerances by special control devices; Government laws require that the frequency of a broadcast station shall not vary to any appreciable extent. Therefore, the signal which the preselector handles is quite constant in frequency. However, any variations in the local oscillator frequency while the receiver is tuned to a station will cause the beat frequency output of the first detector to vary in frequency. If this should occur and if the i.f. amplifier is sharply tuned, the i.f. amplifier will cut off varying amounts of side frequencies, which will result in distortion and weakened receiver output. This condition can be corrected by retuning, as this changes the oscillator frequency. However, if the drift is large, you will soon reach a point where retuning has so changed the preselector resonant frequency that it cuts sidebands. Hence, this correction will work only over a limited range.

► When the oscillator operates at very high frequencies, as in an all-wave superheterodyne receiver, very slight circuit changes produce large amounts of oscillator frequency drift. This causes much trouble unless the response of the i.f. amplifier is broadened. Since this broadening will cause a loss in the adjacent channel selectivity of the receiver, it is better to use an oscillator which does not vary appreciably in frequency at any setting. Good frequency stability is secured by using in the oscillator a tank circuit which has a high Q factor and by limiting the loading of the tank circuit. Other frequency stability requirements include constant d.c. supply voltages, the locating of all oscillator parts away from any sources of heat, and the mounting of parts in such a way that

they will not be set into vibration by the loudspeaker.

## OSCILLATOR OUTPUT VOLTAGE VALUES

To understand why the voltage which the local oscillator feeds into the mixer-first detector must be at least ten times greater than the signal input voltage to the mixer-first detector, we must consider the action of the two types of first detectors (square law and linear) which are commonly used in superheterodynes.

**1. Square Law Type of First Detector.** The incoming signal in a superheterodyne, as you know, consists of an r.f. carrier frequency and many r.f. side frequencies. Each of these must beat with the local oscillator to produce, after detection, the desired i.f. carrier frequency and its side frequencies. When a square law type of first detector is used, the strength of the beat frequency depends upon the *product* of the strengths of the local and incoming signals. (This statement is the result of a mathematical analysis of the problem.) The greater the voltage supplied by the local oscillator, then, the stronger will be the desired beat frequencies. However, the oscillator output voltage must not be made so large that the detector operates outside the square law region of its curve. This is why a ratio of about 10 to 1 between the voltages which the oscillator and the preselector feed into the mixer-first detector is highly desirable.

**2. Linear Type of Detector.** When a linear first detector is used, mathematics and experimental tests show that maximum i.f. output is obtained when the two signals (incoming and local) which are fed to the mixer-first detector are equal in strength. On the other hand, many harmonics of the

beat (i.f.) frequency are produced under this condition. These harmonics, being two, three, four, etc. times the resonant frequency of the first detector plate load circuit will be tuned out here. But, they still may feed back into the input of the mixer-first detector through the plate-to-grid capacity or through stray coupling. This feedback energy will produce annoying audio beat notes (squeals) when the receiver is tuned to a station which has approximately the same frequency as one of these harmonics.

To eliminate this problem, it was found that if either one of the signals which are fed into the mixer-first detector is many times stronger than the other, the harmonic frequencies associated with linear detection become negligible. A ratio of 10 to 1 has been found sufficient in actual practice. Since it is easier to control the local oscillator than the carrier signal input strength, it is customary to make the signal voltage which the local oscillator feeds into the mixer-first detector at least ten times *greater* than the signal voltage fed into this section by the preselector *regardless of the type of first detector used.*\*

**Summary.** With both types of detectors, then, satisfactory performance is obtained when the ratio of the local oscillator output to preselector voltage is greater than 10 to 1. When this ratio is less than 10 to 1, there will be a lowering of receiver gain (but no other undesirable effects) in the case of a square law detector. Also there will be

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\*Since the incoming signal will vary considerably in strength if an r.f. amplifier stage is used in the preselector, it is important that when the first detector is of the linear type, an automatic volume control (a.v.c.) circuit is used to reduce the gain of the r.f. amplifier on strong signals and thus keep the ratio of the mixer input voltages higher than 10 to 1 at all times. A.V.C. circuits are taken up elsewhere in the Course.

an increase in receiver gain, accompanied by annoying squeals when a linear detector is used.

### Variations in Oscillator Voltage.

Once the amount of oscillator output has been chosen, it is desirable that this same amount be delivered, as near as possible, over the tuning range and in spite of operating voltage changes. Should the oscillator output increase

still keep the sum of these signals low enough so the grid cannot swing positive, is to be sure that fluctuations in oscillator voltage do not exceed a ratio of about 3 to 1. This means that the ratio of maximum to minimum oscillator output should not be more than 3.

**Harmonics.** The importance of preventing the local oscillator from feeding harmonics of its fundamental frequency to the mixer-first detector was considered earlier in this lesson, in connection with the study of harmonic interference.

**Radiation.** An oscillator needs only an antenna of some sort to become a midget radio transmitter. Obviously, it is the job of the radio engineer to see that such an antenna is not provided in a receiver. He does this by proper layout of circuit leads, and by using bypass condensers to prevent oscillator currents from leaking into any open wires which might serve as antennas and, in some cases, by shielding the oscillator coil.

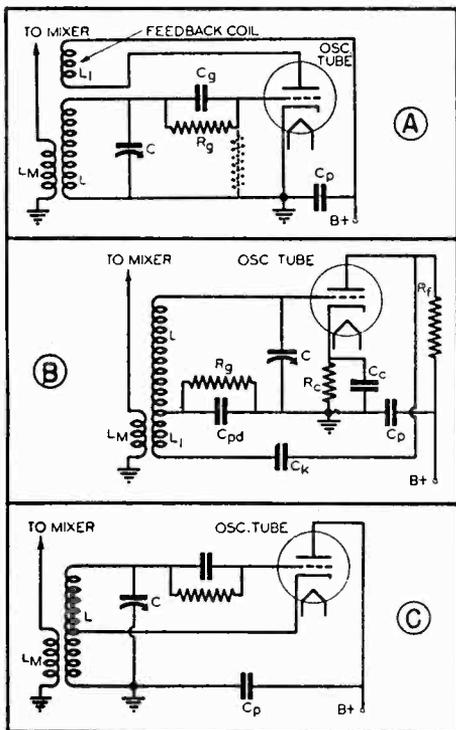


FIG. 7. Typical oscillator circuits.

greatly, the combined effect of local and incoming signals may be sufficient to swing the grid of the first detector positive. This would cause the grid to draw current and load the mixer input circuit, cutting down the peaks of the incoming modulated signal and thus producing distortion. About the only way we can have the local oscillator signal at least ten times as strong as the incoming signal at the mixer, and

### TYPICAL OSCILLATOR CIRCUITS

Almost any oscillator circuit which has a fair degree of frequency stability will provide the local r.f. signal required in a superheterodyne receiver, but for practical reasons those circuits are used which require only one variable condenser, and which permit grounding of the rotor of the variable condenser. It is also desirable to have a circuit in which there is no high d.c. voltage across the variable condenser. Tuned grid and Hartley circuits are by far the more common, both in receivers having separate oscillator tubes and in those having combination oscillator-first detector stages. As a brief review, here are typical oscillators.

**Tuned Grid Oscillators.** Typical tuned grid oscillator circuits are shown

in Figs. 7A and 7B, and in each, the frequency of oscillation is controlled by the values of  $L$  and  $C$ .  $L$  is varied only when changing from one band to another, as in an all-wave receiver, but  $C$  is ganged to the preselector tuning condensers and therefore is varied each time a new station is tuned in. Notice that in each case the rotor of  $C$  is grounded; this is done to simplify the construction of the ganged variable condenser of which  $C$  is one section.

The circuit in Fig. 7A employs inductive feedback (coil  $L_1$ ). Automatic C bias is provided by  $C_g$  and  $R_g$  ( $R_g$  often will be found connected between grid and cathode, as indicated by the dotted lines), while  $C_p$  is a by-pass condenser which keeps r.f. currents out of the plate supply. One way to couple energy into the first detector is to use a pick-up coil like  $L_m$ , which is inductively coupled to oscillator tank coil  $L$ .

► In the circuit of Fig. 7B the feedback is still inductive in nature, but for convenience in manufacture, coils  $L$  and  $L_1$  have been combined into a single tapped coil. Resistor  $R_c$  and condenser  $C_c$  supply the customary form of automatic C bias.  $R_g$  and  $C_{pd}$  increase the C bias when the oscillator tends to develop excess tank circuit power (as it does at the higher frequencies), thus smoothing out or equalizing oscillator power variations.

$C_{pd}$  also serves as a padding condenser, the function of which will be considered later. This oscillator is shunt fed. R.F. current flows from the plate through condenser  $C_k$ , while by-pass condenser  $C_p$  and resistor  $R_r$  keep r.f. currents out of the plate supply.

**Hartley Oscillator.** The form of Hartley oscillator circuit in which the plate is grounded (to permit grounding of the rotor of variable condenser  $C$ ) is used in superheterodyne receivers; a typical circuit is given in Fig. 7C. Condenser  $C_p$  here grounds the plate for r.f. currents. Automatic C bias is used; there is no d.c. voltage across the variable condenser.

**Service Hints.** The oscillator stage in a superheterodyne rarely gives trouble but when it does, it usually goes dead and kills reception. The feedback is adjusted to give satisfactory results with average tubes; however, as the tubes age, its output may drop so that oscillation will not be sustained. Usually, a new tube will clear up the trouble.

Sometimes the grid resistor  $R_g$  will be at the wrong value or will change in value with age. Check its resistance. If it has changed from the rated value, use a new resistance. The resistor can be reduced in value somewhat—the tube is just made to work harder and its life is slightly reduced. Too large a value will cause self-modulation or may even stop operation.

# The Mixer-First Detector

The method used for mixing the output of the local oscillator with the incoming signal is highly important, for this mixing must take place with a minimum of reaction on the oscillator. There are two methods of mixing; electronic and external. Electronic mixing occurs in the electron stream within the mixer tube, while external mixing occurs in a grid or a plate circuit of the mixer tube (external or outside the tube). Today, the electronic method has largely replaced all others, particularly when the same tube is used as both the oscillator and the mixer-first detector.

However, we should study the methods of external mixing, as they have been widely used, and receivers using these systems, are still being serviced. Typical circuits for this purpose are shown in Fig. 8. One of the most widely used mixer connections of the external type is shown in Fig. 8A, where the oscillator output coil  $L_M$  is connected into the cathode lead of the mixer-first detector tube.  $R_g$  and  $C_g$  furnish automatic C bias voltage. The incoming signal acts directly upon the grid, changing its potential with respect to ground, while the oscillator signal changes the potential of the cathode with respect to ground. Both, therefore, affect the plate current which produces the desired mixing.

► Another widely used mixer connection is that in Fig. 8B, where oscillator pickup coil  $L_M$  is in series with coil  $L_P$  (which feeds the incoming signal to the detector). The disadvantage of this circuit is that coil  $L_M$  is a part of the mixer input resonant circuit. Also, changes in  $C_p$ , as the set is tuned may affect the oscillator.

► Some time ago, the oscillator tank

coil  $L$  and the preselector tuning coil  $L_p$  often were mutually coupled as in Fig. 8C, eliminating the need for coil  $L_M$ . Both oscillator and preselector tank circuit coils were wound then on the same form and both often were placed in a shielded housing to prevent radiation. Today, however, this circuit is rarely encountered.

► Capacitive-resistive coupling between a high r.f. potential point on oscillator tank coil  $L$  and the control grid of the mixer tube occasionally is used, as shown in Fig. 8D. A small coupling capacity (from 10 to 100 mmfd.) is usually sufficient. Resistor  $R_k$ , of high ohmic value, is placed in series with this capacity to help isolate the circuits and prevent interactions as they are tuned.

Coil  $L_M$  in the oscillator circuits of Figs. 7A to 7C can be connected also in the screen grid, suppressor grid, or plate lead if the mixer-first detector tube is a pentode, as shown in Fig. 8E. The plate lead connection is very rare, however, for changes in plate voltage which ordinarily can be produced by oscillators have relatively little effect upon plate current. These connections are used when a large oscillator output voltage is available.

► A super-control pentode tube often is used as a mixer-first detector, for it closely approximates a square law detector on weak signals, giving distortionless frequency conversion, and it approximates a linear detector on large swings. It is necessary to use automatic volume control to make the operating point move automatically to a linear portion of the characteristic curve when a strong signal is obtained. When this is done, strong signals cannot swing the grid positive. There is

some generation of harmonics with linear operation, but this is considerably less than would occur with a tube acting as a linear detector at all times.

### COMBINATION OSCILLATOR-MIXER-FIRST DETECTOR

The introduction of multigrid tubes permitted a single tube to serve the function of oscillator, electron mixer, and first detector. The screen grid tetrode tube and the pentode tube were the first to be used for these combined functions.

A multi-function circuit using a pentode tube is shown in Fig. 9; this circuit was widely used in midget superheterodyne receivers. The oscillator section of the circuit is of the Meissner type, with coil  $L$ , variable condenser  $C$ , and padding condenser  $C_{pd}$  forming the oscillator tank circuit for tube  $VT$ . The plate of the tube is connected to (loaded by) the oscillator tank circuit  $L$ - $C$  through two forms of coupling: 1, *Inductive*, through the mutual inductance between coils  $L$  and  $L_1$  (condenser  $C_{IF}$  has negligible reactance at the frequencies generated by the oscillator); 2, *Capacitive*, through condenser  $C_{pd}$ , with the oscillator currents flowing from the plate through  $C_{IF}$ ,  $L_1$  and  $C_{pd}$  to ground. With this coupling arrangement the load on the oscillator tank coil is equalized over the entire band, resulting in more uniform oscillator output.

Feedback of the oscillator tank circuit voltage into the control grid circuit is obtained simply by connecting feedback coil  $L_M$  into the cathode circuit. This arrangement varies the potential between the cathode and ground. But since the grid circuit also eventually goes to the cathode, changing the potential of the cathode with respect to ground also changes the

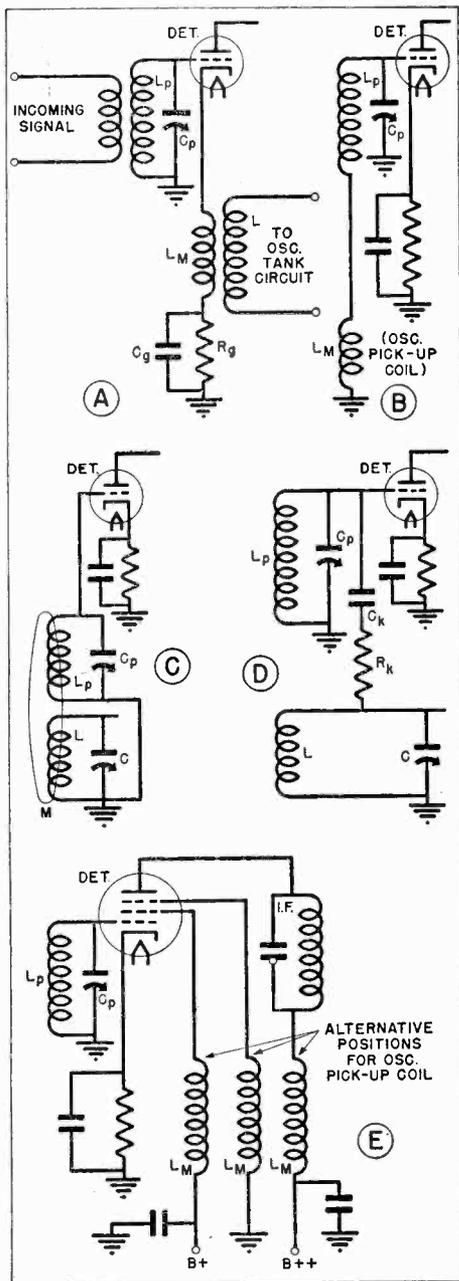


FIG. 8. Methods which have been used for feeding the local oscillator signals into the mixer-first detector of a superheterodyne. When  $L_M$  is shown, it is assumed to be coupled inductively to the oscillator tank circuit. In other cases, either direct inductive coupling or capacitive coupling is used.

potential of the grid with respect to the cathode.

Having analyzed the oscillator and mixer functions of this circuit, we shall now see how it performs the duties of first detector. First of all, observe that the plate of the tube is fed (with d.c.) through coil  $L_p$ , which serves as the primary of the first i.f. transformer and as the plate load coil for the first detector. The i.f. currents flowing out from the plate of tube  $VT$  thus flow through coil  $L_p$  and thence through bypass condenser  $C_p$  to ground. In addition, i.f. currents flow through the path formed by  $C_{1F}$ ,  $L_1$  and  $C_{pd}$ . The reactance of  $L_1$  to i.f. currents is negligible, so  $C_{1F}$  and  $C_{pd}$  together tune coil  $L_p$  to the i.f. value. (This circuit is completed from  $C_{pd}$  through the chassis and through  $C_p$  to  $L_p$ .)

Automatic C bias is furnished by  $C_g$

high  $G_M$  tubes. As a result, the bias may be too high for an average tube, and the original tube may stop oscillating when its emission weakens with age. If a receiver using this circuit fails to work when all parts are in good condition, try lowering the value of  $R_g$  by about one-third; this often will allow the original tube and, surely, all other new tubes to oscillate. If  $R_g$  is reduced too much, however, frequency conversion will not take place.

### PENTAGRID CONVERTER TUBE

So much difficulty was experienced with screen grid and pentode tubes operating as combination oscillator-detectors that tube engineers developed a special tube which would permit independent biasing for optimum oscillator and optimum detector action. The pentagrid (five grid) tube was the re-

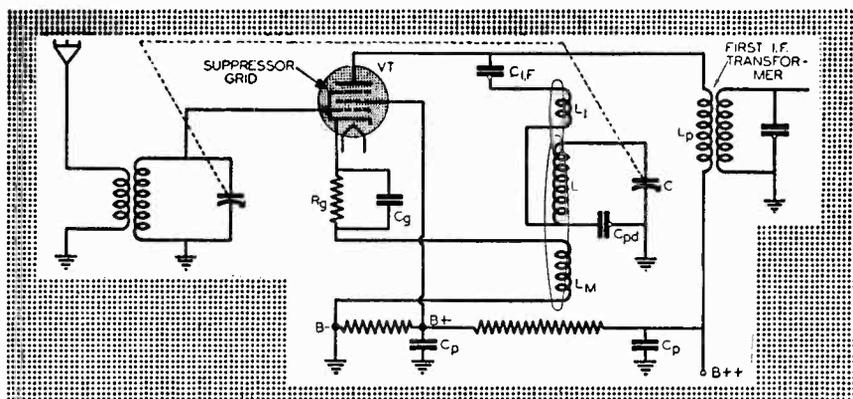


FIG. 9. Combination oscillator-mixer-first detector circuit using a pentode tube. If the suppressor grid is omitted, this circuit will apply to the screen grid tubes which were once widely used for the same purpose.

and  $R_g$ , the value of  $R_g$  being carefully selected to make tube  $VT$  function as both oscillator and detector. Optimum operation as an oscillator is more important than as a detector, because the detector characteristics are less critical.

**Servicing Hint.** Many of these circuits were built for specially selected,

result. Since this tube provides all the functions of the frequency converter section, it is called a *pentagrid converter tube*.

A practical frequency converter circuit employing a pentagrid converter tube is shown in Fig. 10. The oscillator triode section of the tube consists of

the cathode, grid 1 (functioning as a control grid), and grid 2 (functioning as plate for the oscillator triode). (This second grid is called an "anode grid", to indicate its plate action.) Any desired form of oscillator circuit may be connected to these three electrodes; a standard tuned grid, plate coil feed-

the plate current, which is already varying at the oscillator frequency. Thus, the local signal and the incoming signal are mixed in the tube electron stream (this is called electron coupling or electronic mixing).

The C bias produced by  $R_d$  and  $C_d$  acts upon grid 4, thus controlling

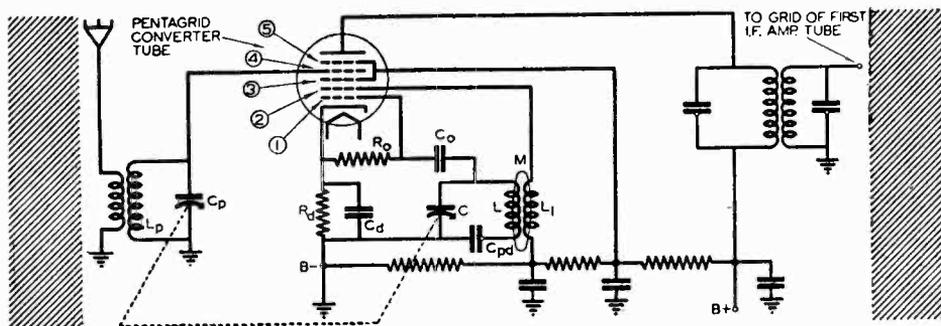


FIG. 10. A practical frequency converter using a pentagrid converter tube.

back oscillator circuit is shown. Coil  $L$ , condenser  $C$  (rotor grounded) and padding condenser  $C_{pd}$  constitute the tank circuit of the oscillator, while coil  $L_1$  in the circuit of grid 2 (the oscillator plate circuit) feeds oscillator r.f. plate current back to the grid tank circuit to maintain oscillation. Condenser  $C_o$  and resistor  $R_o$  together provide automatic C bias for the oscillator. Since this bias is applied directly between grid 1 and cathode, it is independent of the automatic C bias created by  $C_d$  and  $R_d$  for the first detector.

The action of the oscillator sets up, just beyond the second grid, an *electron cloud* which serves as the *virtual cathode* for the other tube elements. Thus, the detector section is furnished an electron stream which is varying at the oscillator frequency. In the detector section of the tube, grid 4 acts as the control grid, grids 3 and 5 as a screen grid, and the plate has its usual function. An incoming signal controls

the flow of electrons from the virtual cathode to the plate. The shielding action of grids 3 and 5 is only partially effective. It is therefore necessary to choose a detector plate load (i.f. transformer primary) which will resonate sharply to the i.f., having at the same time such an  $L/C$  ratio that it will reject other signals. This is necessary in order to prevent i.f. harmonics and other undesired plate circuit signals from feeding back to grid 4 in the input circuit.

With oscillator and detector sections essentially coupled together only by the space cloud, it is possible to design the oscillator and detector sections of a pentagrid converter independently and secure optimum operation of each.

**Disadvantages of Pentagrid Converter Tubes at Ultra-High Frequencies.** Although the pentagrid converter tube gives excellent results at broadcast band frequencies and the lower short-wave frequencies, it is somewhat unsatisfactory at frequencies above

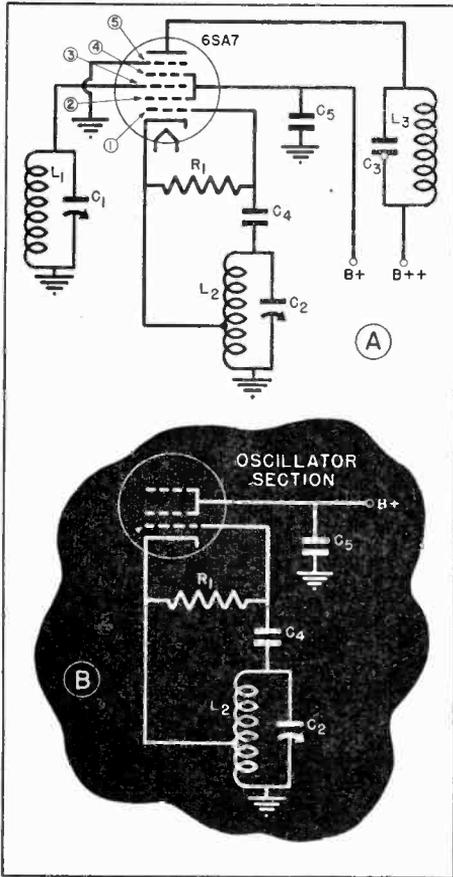


FIG. 11. Another type of pentagrid converter.

about 10 megacycles. One reason for this is that despite the use of a screen grid, the oscillator section of the tube sets up a space charge which at high frequencies affects the input circuit (grid 4 in Fig. 10) directly. At ultra-high frequencies, the oscillator circuit will cause *regeneration* when tank circuit  $L-C-C_{pd}$  is tuned *below* the frequency of the incoming signal, and will cause *degeneration* when this circuit is tuned *above* the frequency of the incoming signal. Since the oscillator of a super is ordinarily tuned *above* the incoming signal frequency, degeneration and consequent loss in signal strength occurs in the high-frequency band and

at the higher frequencies in other bands. To avoid this degeneration, the manufacturers of some all-wave receivers make the oscillator tune below the incoming signal frequency on the highest frequency band.

*Excessive oscillator frequency drift* at frequencies above 10 megacycles is another trouble encountered in a pentagrid converter. The tank coil of the oscillator ( $L$  in Fig. 10) is affected by the a.c. plate resistance of the oscillator section of the tube (the resistance between grid 2 and cathode, which acts on the tank circuit through mutual inductance  $M$ ). This resistance changes the oscillator frequency (as determined by the values of  $C$ ,  $L$  and  $C_{pd}$ ) a certain *fixed percentage* at all times. At low radio frequencies the error in oscillator frequency is so small as to be negligible, but at high radio frequencies the error is larger. The variation is relatively unimportant, however, for if the preselector is a little broad, it is only necessary to change the tuning dial setting slightly to bring in the station satisfactorily. More serious trouble occurs because of variations in carrier intensity (fading) when distant stations are being received. In any oscillator, detector, or combination oscillator-detector circuit, the a.c. plate resistance of the tube will vary with carrier intensity because variations in the carrier cause variations in the average space current of the tube. This varying a.c. plate resistance causes oscillator frequency drift, which can be very severe at the higher frequencies. ► Here is an example: Suppose that a carrier of varying strength changes the a.c. plate resistance enough to cause a maximum error of 0.1% in oscillator frequency. At 1000 kc. this percentage will give only a 1-kc. error in the i.f. beat, and the i.f. amplifier is almost always broad enough to offset such a small error. At 10 megacycles, how-

## PENTAGRID MIXER-FIRST DETECTOR TUBE

To overcome the inherent shortcomings of the pentagrid converter at high frequencies, engineers designed a special pentagrid mixer-first detector tube which has a *mixer* or *injector* grid. When used with a separate oscillator tube, this *pentagrid mixer-first detector tube* is a better frequency converter because it has *negligible frequency drift and negligible degeneration*, even at the very high frequencies encountered in television receivers.

ever, this same percentage gives a 10-kc. error in the i.f. beat, and clearly there will be severe cutting of side bands. Also, if the i.f. amplifier is highly selective, the signal may even fade out entirely unless the set is retuned. ► A special form of a pentagrid converter, which uses a 6SA7 tube, is shown in Fig. 11A. We can consider that the oscillator section of this circuit consists of the elements shown in Fig. 11B. As you can see, this is the kind of Hartley oscillator pictured in Fig. 7C, with the screen grid acting as

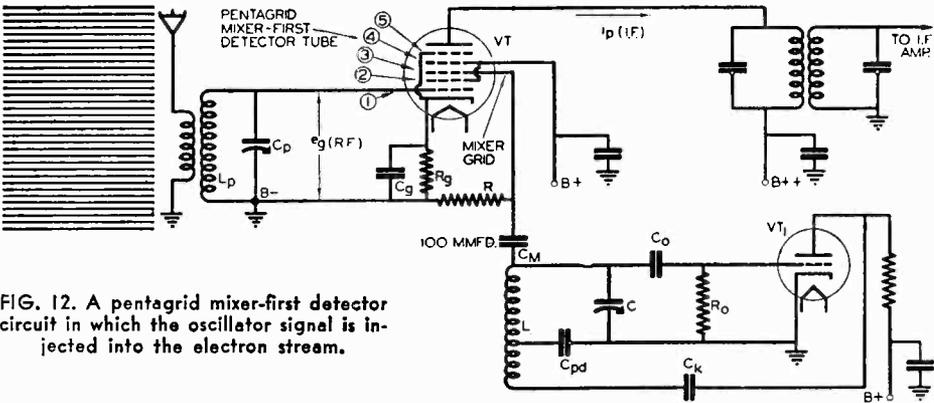


FIG. 12. A pentagrid mixer-first detector circuit in which the oscillator signal is injected into the electron stream.

the plate or anode grid. In the detector section of Fig. 11A, grid 3 acts as the control grid, grids 2 and 4 form a screen for grid 3, and grid 5 is a suppressor. The cathode and plate, of course, also perform their usual functions for the detector section of the tube.

This form of pentagrid converter has somewhat greater sensitivity and less tendency toward oscillator frequency drift than has the circuit shown in Fig. 10. In addition, the 6SA7 tube is single-ended (has no grid cap) and so fits in better than other pentagrid converter tubes with the modern practice of keeping above-chassis wiring to a minimum.

A practical circuit using this tube and capable of giving excellent frequency converter action well up into the very high frequencies is shown in Fig. 12. Tube *VT* is the special pentagrid mixer-first detector tube. Grid 1 is the control grid for the detector; grid 3 is the mixer grid; grids 2 and 4 are shielding grids for the mixer grid; while grid 5 is a suppressor grid.

Tube *VT<sub>1</sub>* is an ordinary triode tube connected into a conventional tuned-grid oscillator whose tank circuit consists of *L*, *C*, and *C<sub>pd</sub>*. Condenser *C<sub>M</sub>* provides capacitive coupling between the oscillator tank circuit and grid 3 of *VT*, which injects the oscillator sig-

nal into the electron stream of the mixer tube. The stability of the oscillator is dependent only upon the design and construction of the oscillator stage and there can be no feedback of oscillator current to the preselector (the  $L_p$ - $C_p$  tank circuit).

► A triode-hexode tube has been designed which combines a triode oscillator and a hexode (6-element) mixer-detector in one envelope. A typical circuit in which this tube is used is shown in Fig. 13. As you can see, the oscillator section of this circuit uses a standard tuned grid oscillator. The grid of this triode section is connected within the tube to the first grid of the hexode section, which then becomes an injector grid and feeds the oscillator signal into the electron stream. Mixing occurs when the incoming signal is fed into grid 3 of the hexode section of the tube. Grids 2 and 4 of the hexode section merely act as shielding screens for grid 3.

This triode-hexode circuit has many of the advantages of the circuit in Fig. 12 for high-frequency operation and, in addition, permits the use of one less tube. Therefore, this circuit is being used more and more in all-wave receivers.

### FREQUENCY CONVERTER RATINGS

When discussing the relative merits of different frequency converter systems, engineers need what might be called a "yardstick of comparison." What they really want to know is how well a frequency converter will change an r.f. signal voltage into an i.f. signal voltage—that is, frequency converter gain.

The important factor in determining

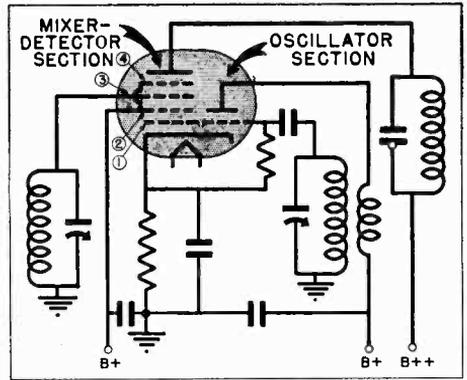


FIG. 13. A triode-hexode tube featuring electron mixing from the oscillator to the first detector.

the gain is the *conversion transconductance* (sometimes called *translation conductance*) of the frequency converter; it is equal to i.f. plate current  $i_p$ , divided by r.f. input voltage  $e_e$  and is designated by the symbol  $S_c$ . Knowing  $S_c$ , the gain of a frequency converter can be found by multiplying the value of  $S_c$  in *mhos* by the a.c. plate load impedance in ohms of the first detector circuit. For example, a 6A7 tube used with a reasonably powerful oscillator circuit will have a conversion transconductance of somewhere between 350 and 520 micromhos at ordinary frequencies, but this value will be reduced greatly at very high frequencies. A 6L7 pentagrid mixer-first detector used with a separate oscillator tube as in Fig. 12 will have a translation conductance of about 475 micromhos, this value changing little even at high frequencies. A commonly used value for the a.c. plate load impedance is 100,000 ohms. This load and a converter with an  $S_c$  of 400 micromhos (.0004 mhos) gives a gain of  $.0004 \times 100,000$ , or 40.

# Oscillator-Preselector Tracking

Ganging of the preselector and the oscillator tuning condensers simplifies tuning of a receiver and eliminates repeat points. However, new adjustment problems are introduced when we attempt to make the oscillator operate above the frequency of the incoming signal by exactly the i.f. value at all times. The oscillator must follow or "track" the preselector, hence we call the alignment of the oscillator and preselector circuits a *tracking adjustment*.

► Let us see first why tracking adjustments are necessary. Turning to Fig. 14A, which shows both the preselector and oscillator tuning circuits, assume that the tuning condensers are set to minimum capacity, that  $L_p$  is exactly like  $L_o$ , and that  $C_p$  is exactly like  $C_o$ . Obviously, both circuits now will have the same resonant frequency. Since the oscillator circuit must tune to a higher frequency, the electrical value of either  $L_o$  or  $C_o$  must be reduced. It is not easy to reduce  $C_o$ , since its minimum capacity is due essentially to stray capacities between rotor and stator plates, so  $L_o$  must be replaced with a coil having enough *less* turns to make the oscillator tune above the preselector by exactly the i.f. value when both condensers are at *minimum-capacity* positions.

But if, after making this adjustment, we set both condensers to their *maximum-capacity* values, we find that the oscillator is no longer exactly the i.f. value higher than the preselector. (The reason is that the numerical difference in frequency between the oscillator and the preselector depends upon the numerical values of both the inductances of the coils and the capacities of the condensers.) Therefore, when we change the numerical values of the capacities of the condensers we also

change the numerical value of the difference in frequency.

Curves 1 and 2 in Fig. 14B show how the resonant frequencies of the oscillator and preselector circuits vary as the tuning dial (to which both condensers are ganged) is rotated from 100 to 0 (100 corresponding to the minimum

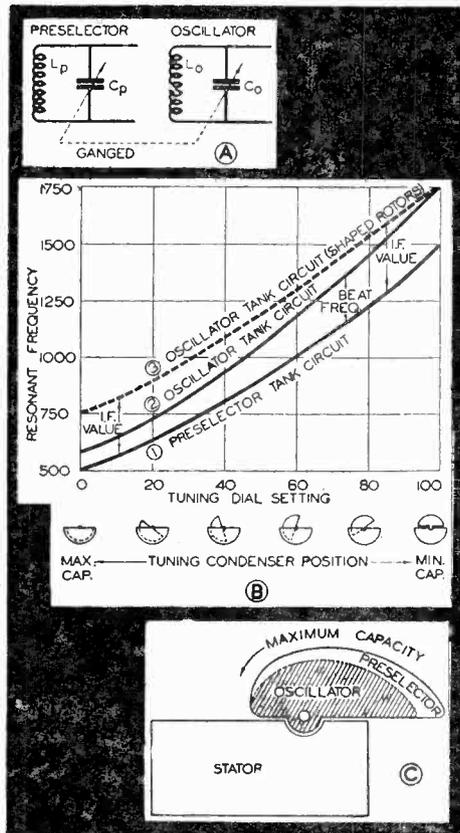


FIG. 14. Preselector-oscillator tracking adjustments are necessary because the resonant frequencies of the two tank circuits (at A) do not differ by the same amount (the i.f. value) at each tuning dial setting. The discrepancy is shown by curves 1 and 2 (at B); curves 1 and 3 show the ideal relationship, which can be secured if the rotor plates of the tuning condensers are specially shaped as at C.

and 0 to the maximum capacity of the tuning condensers). As you can see, the resonant frequency of the oscillator decreases faster than does the resonant frequency of the preselector, so the i.f. beat frequency (the difference between the two) also decreases as the condenser is turned from minimum capacity to maximum capacity. There is need for an adjustment of some sort which will keep the beat frequency constant. We cannot reduce the number of turns on the oscillator coil, for the coil is correct as it now is for the high-frequency setting. Clearly, then, it is necessary to reduce the capacity of  $C_o$  for low-frequency settings.

► A constant i.f. beat frequency over the entire tuning range can be obtained by giving the oscillator rotor plates a different shape from those of the pre-

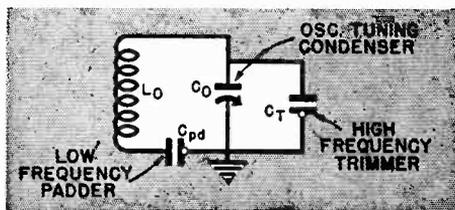


FIG. 15. Here are the adjusters used on the oscillator circuit when the tuning condenser does not have specially-cut plates.

selector. The exact shape required can be figured out by mathematics, but the general shapes will resemble those shown in Fig. 14C. Notice that as the frequency is lowered (by increasing the capacity), the oscillator condenser increases less in capacity than the preselector. Thus we obtain the results represented by curves 1 and 3 in Fig. 14B.

Condensers having specially cut rotor plates are expensive to build and are good only for one particular i.f. value and oscillator coil. Hence, they are clearly out of question for all-wave receivers, because here a different

shape of rotor would be required for each band. For these reasons, all-wave receivers are built with ganged tuning condensers having all sections alike, and two trimmer condensers, known as the low-frequency padder and the high-frequency trimmer, are adjusted to make the preselector and the oscillator track each other.

Cut rotor plates are used extensively in broadcast band receivers, however. Also, they find wide use in auto radio receivers, as they keep to a minimum the number of trimmer condensers which might get out of adjustment because of vibration.

**Low - Frequency Padder Condenser.** When ganged tuning condensers are used, the oscillator condenser will have too high a capacity when its plates are completely meshed. (Remember that the oscillator coil has less inductance than the preselector coil.) This capacity can be lowered by inserting in the tank circuit, in series with  $C_o$ , a trimmer condenser which is called a *low-frequency padder*, a *padder*, or sometimes a *lag condenser*. This padding condenser can be adjusted to lower the tank circuit capacity just enough to give perfect alignment at the 0 (lowest frequency) point on the dial. The padding condenser shown as  $C_{pd}$  in Fig. 15 (and also in Figs. 9, 10, and 12) is considerably higher in capacity than the maximum capacity of  $C_o$ , so that its reactance is almost negligible (it acts as a short circuit) at the 100 or high-frequency setting of the dial.

However, at the low-frequency end, it acts in series with  $C_o$  and, as you know, condensers in series have a lower capacity than either one alone. Hence, the capacity in the circuit is reduced at the low-frequency end of the band, which provides the necessary tracking.

**High - Frequency Trimmer Condenser.** We have assumed thus far

that the exact i.f. value was produced at the 100 or minimum-capacity setting, but this seldom is the case in actual practice because, even with mass production methods, it is difficult to make coils and condensers with exactly the desired electrical values. Manufacturers of superheterodynes compensate for these errors by placing a small trimmer condenser (called the *high-frequency trimmer condenser*) in parallel with the oscillator tuning condenser. This, as well as the padder, must be adjusted to make the receiver function properly. Let us now consider how these tracking adjustments are made in a modern superheterodyne receiver.

► There are two alignment conditions: 1, where the receiver is completely out of alignment; and 2, where the alignment just needs "touching up" or adjusting for maximum response. Considering the first case, the following procedure would be used:

Assuming that the receiver is to be aligned for the 550-1500-kc. broadcast band, feed into the preselector input a 1000-kc. signal (supplied by a signal generator) and set the receiver tuning dial to its 1000-kc. division. Adjust the low-frequency padder  $C_{pd}$  for maximum receiver output; now you are getting exactly the correct i.f. beat for mid-dial settings. Change the signal input frequency to 1400 kc., set the receiver dial at 1400 kc., and now adjust high-frequency trimmer  $C_T$  for maximum receiver output. If there are two positions of  $C_T$  for which maximum output is obtained, choose the one at which  $C_T$  has minimum capacity; the other position is a repeat point. Now change the signal input frequency and the dial setting to 600 kc. and readjust the low-frequency padder for maximum output. It is necessary to repeat the adjustments at 1400 kc. and at 600

kc., since one trimmer has a slight effect upon the other. The final adjustment is made at 1400 kc. If the procedure has been correctly followed, the proper i.f. value will be obtained over the entire tuning range.

These alignment instructions apply to any other frequency band as well. High, medium, and low frequencies in each band are selected; the high-frequency trimmer and the low-frequency padder then must be adjusted to make the preselector and the oscillator track each other. The low-frequency padder is always adjusted for the low and mid-scale frequencies, while the high-frequency trimmer is adjusted at the high-frequency setting. This procedure is called a *three-point track alignment adjustment*. It makes the i.f. value correct for three points, insuring that the i.f. beat will not be off an appreciable amount at other dial settings.

On a few all-wave receivers, the padding condenser for some of the short-wave bands may be a fixed condenser rather than a trimmer. On such bands, there can be only a high-frequency adjustment, so only fair tracking is obtained.

► When the alignment is just to be "touched up" it is unnecessary to make the mid-frequency adjustment, as the high- and low-frequency points are the only ones likely to be off.

## ALL-WAVE SUPERHETERODYNE RECEIVERS

**Number of Bands Required.** An all-wave superheterodyne receiver differs essentially from a broadcast band superheterodyne receiver only in that the all-wave receiver has one or more extra preselector and oscillator tuning circuits, which may be switched in as desired. Since it is difficult to tune a preselector and an oscillator over a range having a frequency ratio greater than about 3.3 to 1, it is necessary to

use a new set of coils in the preselector and oscillator circuits for each band of frequencies. If the broadcast band coils for a receiver cover the range from 540 kc. to 1780 kc. (1780 being 3.3 times 540), the next band will extend from 1780 kc. to 5800 kc. ( $3.3 \times 1780$ ), and band No. 3 will extend from 5800 kc. to 19,100 kc. (these values are approximate). To secure this 3.3 to 1 range in each band, the variable condensers must have a low minimum capacity, stray lead connection capacity must be very low, and the coils must have a very low distributed capacity. When these requirements cannot be met because of chassis layout and design problems, or when it is desired to cover all frequencies from 540 kc. to about 22,000 kc., it is necessary to divide the entire frequency range into four bands and use four sets of coils.

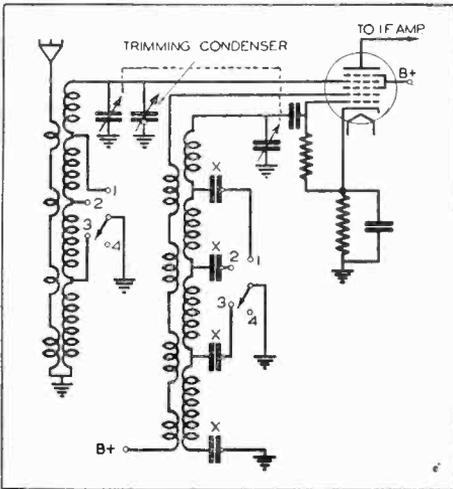


FIG. 16. Preselector and frequency converter circuits of a four-band superheterodyne receiver which employs series coil switching.

**Band Changing by Switching Series Coils.** Changing the number of coils connected together in series is one way of changing the frequency range of a resonant circuit. Fig. 16 illustrates

how this is done in one practical four-band superheterodyne receiver. The four preselector coils are connected together in series, as also are the oscillator coils; when the band-change switches (operated by a single control knob) are set to band 1, only the uppermost coil is in each resonant cir-

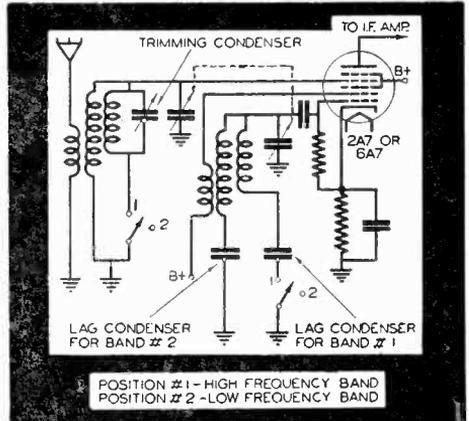


FIG. 17. Preselector and frequency converter circuits of a two-band superheterodyne receiver employing shunt coil switching.

cuit. Switching to band 2 adds another coil to each resonant circuit which lowers the resonant frequency by increasing the circuit inductance. Switching to band 3 adds another coil, and all four coils are used on band 4 (which ordinarily would be the broadcast band). Notice that a different low-frequency padder (x) is used in the oscillator tank circuit for each band. Only one high-frequency trimmer is used, so it is possible to make the high-frequency adjustment on only one band. (Usually this adjustment is made on the more popular broadcast band.) The preselector circuits must therefore be quite broad to compensate for poor tracking in other bands. Another undesirable feature of this series switching arrangement is the fact that the unused coils may absorb energy, making it necessary to use a special band-change

switch which either disconnects or shorts out the unused coils.

**Band Changing by Switching Shunt Coils.** Band changing can be accomplished also by adding coils in parallel; Fig. 17 shows how this is done for a two-band receiver. Each oscillator coil has its own padder condenser to insure good low-frequency tracking. A trimmer condenser is used across one preselector coil to permit a high-frequency adjustment of the higher frequency band (band 1). Unused coils can still absorb energy in this circuit.

Quite often only one oscillator coil is used in two-band receivers, a harmonic of the oscillator being used for the higher frequency band. For example, if the oscillator range for the 550-1500-kc. broadcast band of a receiver having a 460-kc. i.f. value is 1010 kc. to 1960 kc., the second harmonic of the

**Band Changing by Switching Complete Coils.**

Although the band-changing circuits shown in Figs. 16 and 17 have been used in lower-priced receivers, perfect tracking on all bands is not obtainable, and energy absorption by unused coils can prove troublesome. A more practical and more widely used circuit is shown in Fig. 18 as it is applied to a four-band receiver. For each band there is a separate preselector coil with its high-frequency trimmer *C*, and a separate oscillator coil with its high-frequency trimmer *C* and low-frequency padder *X*. The change from one band to another is accomplished by a four-section (four-deck), four-point rotary switch. Since image interference is most objectionable in the broadcast band, an extra tuning circuit is inserted between the antenna and the main pre-selector when the band-change switch

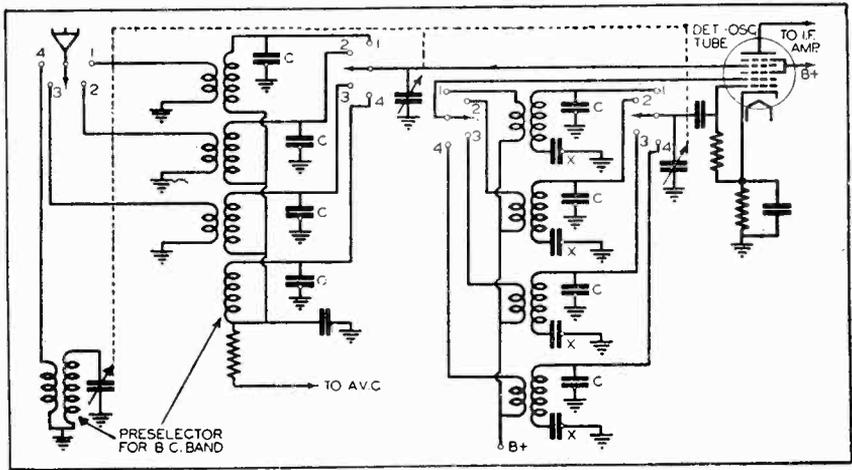


FIG. 18. Preselector and frequency converter circuits of a four-band superheterodyne receiver in which band changing is accomplished by switching complete coils. One extra tuned circuit is used in the broadcast band (band-change switch set to No. 4) to give additional suppression of image interference.

oscillator will vary from 2020 kc. to 3920 kc., the correct variation for the 1560-kc. to 3460-kc. band. In this case band-changing is accomplished simply by tapping the preselector coil.

is at position 4. With complete coils being switched, each band is electrically independent of the others and maximum operating efficiency is attained.

► An all-wave receiver circuit con-

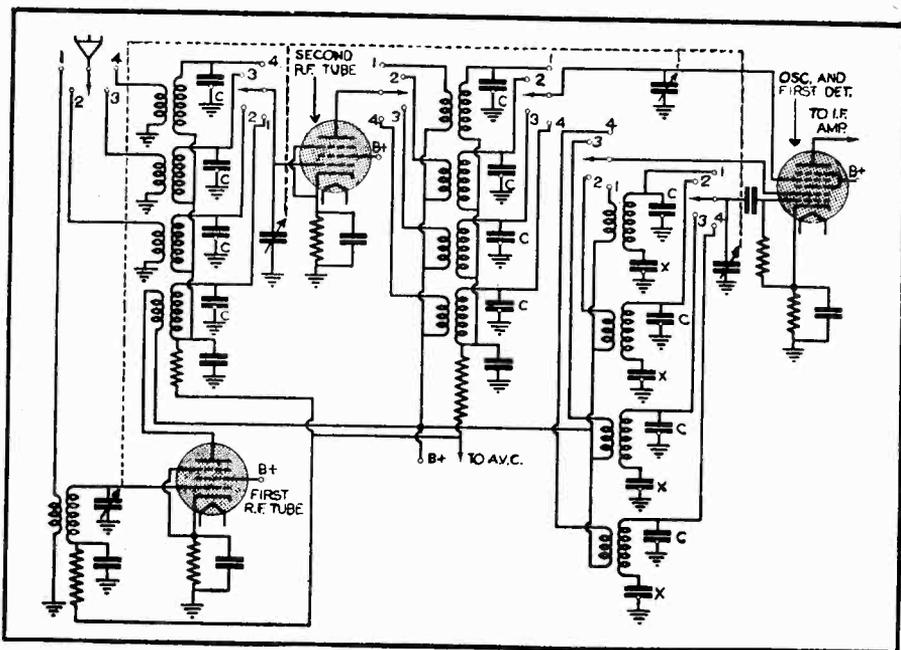


FIG. 19. The preselector in this all-wave superheterodyne circuit uses two stages of r.f. amplification for the highest frequency band (band-change switch set to No. 1) and one stage for all other bands.

taining r.f. amplifier stages in the preselector is shown in Fig. 19. This arrangement gives better image rejection and improves the signal-to-noise ratio, thus lessening the effects of converter noise. Observe that one r.f. amplifier stage is used for the lower frequency bands (2, 3 and 4) but two stages are used for band 1 to offset the reduction in the sensitivity and selectivity of the receiver at high frequencies.

Thus, you can see that all-wave superheterodynes differ from broadcast band superheterodynes only in the sections ahead of the mixer-first detector. All-wave receiver circuits may appear complicated at first glance because of the band-changing switch and the extra parts, but connections can be traced quite easily if only one band is

considered at a time. Various methods for shorting unused coils will be found; these often seem to complicate preselector circuit diagrams.

**Combination Receivers.** When the receiver is a combination AM-FM receiver, the widely different frequency bands require special considerations. It is possible to use a band switch and another set of coils, but many combination sets switch the entire preselector-oscillator-first detector section. This is desirable to keep down losses in the very high frequency FM bands, for this eliminates the use of long leads to band switches, and permits the use of tubes specially designed for each service. These combination receivers will be studied along with FM receivers.

# The I. F. Amplifier

Most of the gain in a superheterodyne receiver is furnished by the i.f. amplifier. In fact, the very purpose of frequency conversion is to permit this section of the super to do its work. The i.f. amplifier also contributes most of the *adjacent channel selectivity*; that is, if the i.f. amplifier is designed to pass all frequencies 5 kilocycles above and below the i.f. value, all stations producing i.f. beats which are outside this range will be tuned out or rejected by the i.f. amplifier.

## CHOOSING THE I.F. VALUE

The receiver designer considers many things when he is choosing the i.f. value for a receiver. He knows that low i.f. values permit high i.f. gain and high selectivity, but require good preselector circuits to eliminate images and repeat spots. Likewise, he realizes that he can use a simple preselector if the i.f. value is high, and can still get reasonable gain by using coils which have pulverized iron cores. A high i.f. value is required if a wide band of side frequencies is to be passed. For a 10-kc. band width, the i.f. should be at least 175 kc.; and for a 20-kc. band width, the i.f. value should be about 460 kc. The 6000-kc. band width used in television may require an i.f. value of at least 15,000 kc.

**Should I.F. Values End in 5?** It is common practice to choose an i.f. value which ends in 5, such as 175 kc., 265 kc., 465 kc., etc. The reason for this is interesting.

Harmonics of the i.f. signal, feeding back to the preselector (through the grid-to-plate capacity of the mixer-first detector tube), produce very annoying squeals when stations are tuned in which have the same frequencies as

these harmonics. For example, with an i.f. value of 175 kc., the fourth harmonic of the i.f. signal will be 700 kc. When a 700-kc. station is tuned in, the fourth harmonic of the i.f. frequency will beat with the 700-kc. carrier to produce an audio squeal frequency which will modulate the 700-kc. signal and, after frequency conversion, ride through the i.f. amplifier.

In the United States, stations in the broadcast band have frequencies ending in 10, such as 960 kc., 1440 kc., etc. By making the i.f. value end in 5, only its *even* harmonics can end in 10 and be equal to a broadcast frequency. For example, the harmonics of a 175-kc. i.f. are 350, 525, 700, 875, 1050, 1225, 1400, etc., of which 700, 1050 and 1400 represent station frequencies in the broadcast band; the harmonics of a 180-kc. i.f. are 360, 540, 720, 900, 1080, 1260 and 1440, of which all but the first value are equal to broadcast station frequencies. An i.f. value ending in 5 thus gives only half as many squeal-producing points on the dial as an i.f. value ending in 10.

The higher the i.f. value, the fewer i.f. harmonics there will be in the broadcast band, which is the band suffering the most interference. This is another factor which makes 455 and 465 kc. the most popular i.f. values.

## TYPICAL I.F. AMPLIFIER CIRCUITS

The most widely used i.f. amplifier is the twin resonant circuit shown in Fig. 20A. When  $C_1$  and  $C_2$ , the trimmer condensers, are tuned for maximum receiver output, a highly selective (sharp resonance curve) amplifier is obtained. When  $C_1$  is tuned below and  $C_2$  above the i.f. value, a rounded or even a double-peak response characteristic

curve is obtained, but the gain is reduced considerably. In the latter case, the mutual inductance  $M$  must be of the correct value for band-pass results; this is a job for the designer. Wider band width can be obtained by shunting one of the coils with a 10,000- to 100,000-ohm resistor, but this gives even further reduction in gain.

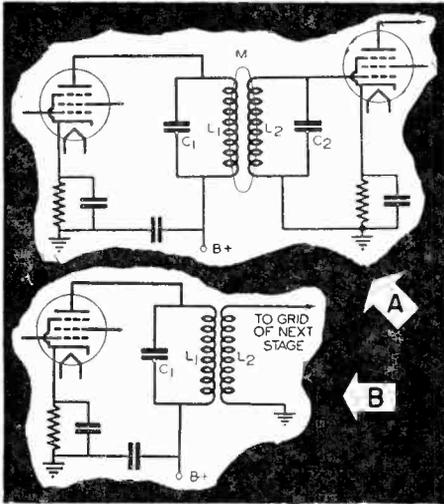


FIG. 20. Typical i.f. circuits.

In any i.f. amplifier, the i.f. transformer in the plate circuit of the mixer-first detector must be extremely selective, and must have a low  $L/C$  ratio in order to short-circuit the harmonics which are produced by detection which might otherwise feed back into the detector input. In addition, the coupling between the transformer coils must be loose; a copper screen is often placed between the two coils to reduce the coupling.

I.F. transformers, after the first one, can have higher  $L/C$  ratios and higher gain. Thus, these transformers may not all be alike—an important point when you are replacing one in service work.

► When high i.f. gain is desired with a minimum number of circuit parts, a

single tuned i.f. circuit like that in Fig. 20B is used. Trimmer condenser  $C_1$  may be placed across either  $L_1$  or  $L_2$ . You will find this circuit most often in midget supers. The gain is about double that of the type shown in Fig. 20A, but of course there is less selectivity.

**High-Fidelity I.F. Amplifiers.** In a high-fidelity receiver the resonance curves of the i.f. amplifier must be practically flat over the entire band of frequencies being transmitted, for this section provides most of the gain of the receiver. Other sections of the receiver must likewise pass all side frequencies without excessive attenuation (unless defects in one section are compensated by a correction in another section). Thus, if all frequencies up to 8500 cycles are desired, the i.f. amplifier must have a band width of 17 kc.

**Variable Selectivity I.F. Amplifiers.** High-fidelity receivers perform beautifully when tuned to strong local stations, but often give squeals and garbled speech (sometimes called “monkey chatter”) when tuned to distant stations. This is because the response of the receiver is so broad that the carrier or side frequencies of an adjacent channel station can interfere with the desired carrier. The remedy obviously is a control which will permit the listener to choose between high selectivity and high fidelity (poor selectivity) characteristics. Three commonly used methods for obtaining variable selectivity in a superheterodyne are given in Fig. 21; each of these provides a continuously variable control, which allows the listener to reduce the fidelity just enough to stop the interference.

In the circuit of Fig. 21A the selectivity is changed by varying the mutual inductance  $M$  of the i.f. transformer (by mechanically varying the spacing between coils  $L_1$  and  $L_2$ ). With

this arrangement, the trimmers are adjusted for peak response when  $M$  is a minimum (weak coupling). Now, when  $M$  is increased, the peak of the response curve flattens out, becoming flat or perhaps even double-humped. As you learned in an earlier lesson, increasing  $M$  has the effect of introducing capacity into one circuit while adding inductance to the other, which is exactly what is needed to get double peak response.

► Another scheme, which has the effect of shunting each resonant circuit with a resistance, is shown in Fig. 21B; this uses a third winding shunted by a variable resistor ( $R$ ). The two resonant circuits originally are adjusted for peak response when  $R$  has a maximum resistance. Reducing  $R$  causes  $L_3$  to absorb energy from both  $L_1$  and  $L_2$ , thus

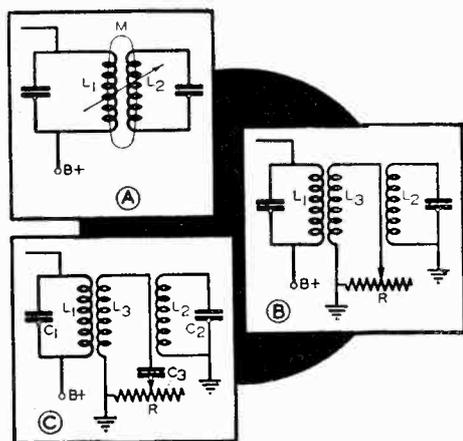


FIG. 21. Three methods of securing continuously variable control of the selectivity of i.f. stages of a superheterodyne receiver are shown here. Although only one i.f. transformer is shown, each i.f. transformer in a receiver may be identically treated and the selectivity controls ganged together.

acting as a load. This widens the response curve, giving higher fidelity.

► A third method, shown in Fig. 21C, uses a third winding  $L_3$  which is con-

nected in series with condenser  $C_3$  and rheostat  $R$ . Resonant circuits  $C_1-L_1$  and  $C_2-L_2$  are tuned for maximum receiver output when  $R$  is set to maximum resistance. Then  $R$  is set to zero resistance and  $C_3$  is adjusted for *minimum* output. Thereafter, the circuit  $L_3-C_3-R$  acts as a loading circuit. As  $L_3$  and  $C_3$  are resonant, the amount of energy absorbed depends on the setting of  $R$ . With  $R$  at a high resistance value, little energy is absorbed and maximum selectivity is obtained. When  $R$  is reduced toward zero, the other circuits are effectively loaded with resistance, reducing the selectivity and gain, and thereby broadening the over-all response. This circuit is just a variation on the one shown in Fig. 21B.

**High-Low Selectivity Switches.** In a variable-selectivity receiver the desired method of controlling selectivity must be incorporated in one or more i.f. circuits. To reduce costs, continuously variable selectivity is sometimes replaced with a high-low selectivity (low-high fidelity) change-over switch, as shown in Fig. 22A. Trimmer condensers  $C_1$  and  $C_2$  are adjusted for single peak response (high selectivity) when the switch is set to point 1. Moving the switch to point 2 makes coil section  $L_M$  common to both resonant circuits; this increases the mutual inductance of the coils. It also adds inductance to circuit  $L_1-C_1$ , and takes inductance away from circuit  $L_2-C_2$ , thereby increasing the band width (giving high fidelity).

► Any scheme which will increase the resonant frequency of one circuit while decreasing the resonant frequency of the other circuit by an equal amount may be used to control selectivity. Another method of doing this is shown in Fig. 22B. When the selectivity switch is at point 1, normal peak adjustments exist. When it is at point 2, condenser

$C_3$  is in series with  $C_1$  and  $L_1$ , increasing the resonant frequency of circuit  $C_1-L_1$ ; condenser  $C_4$  is in shunt with condenser  $C_2$ , lowering the resonant frequency of circuit  $L_2-C_2$ . Condensers  $C_3$  and  $C_4$  are of such capacities as to give the desired band width.

**Comparison of I.F. Stages.** It is interesting to compare the i.f. amplifiers of AM, FM and television receivers. Essentially, the i.f. amplifier delivers most of the amplification and selectivity in each of these receivers, and the basic design is similar. Fundamentally, they differ mostly in the degree of bandpassing and in the i.f. value used.

► Broadcast and so-called all wave AM receivers today use an i.f. value of 455 to 465 kc. (Broadcast receivers have used i.f. values of 130, 175 and 260 kc. in the past.) In AM receivers intended for long-distance reception, the usual band-width is 10 kc. or less, so peak alignment is employed. The high-fidelity types are band-passed somewhat, out to a band width of 15 to 20 kc. It is important that the characteristic be relatively flat-topped so that there will be no discrimination against the modulation frequencies.

► The FM receiver has an i.f. value of from 2 to 4 mc., because the FM process requires a very wide band—about 200 kc. Only a high i.f. value will permit such a wide band, and even here special design is necessary. It is not necessary that the response be flat-topped as the limiter stage cuts off all peaks in amplitude, as you will learn

when you study FM receivers more fully. Usually, the FM receiver will have a greater number of i.f. stages, with the last one or two being designed as limiters.

► In television receivers, the i.f. value is usually 13 mc., and the amplifier must have a band width of about 4.5 mc. to pass even *one* side-band. (Single

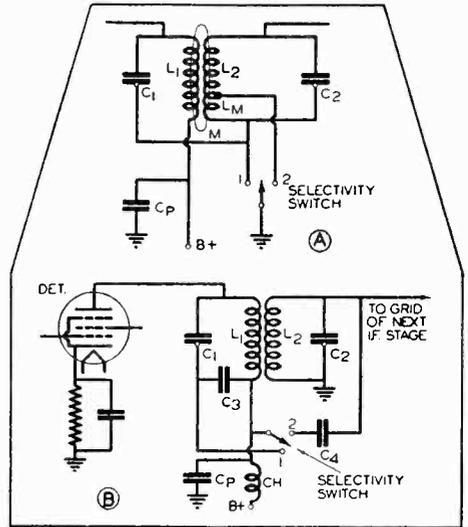


FIG. 22. Two methods for incorporating high-low selectivity switches in a superheterodyne receiver are shown here.

side-band transmission is used in television.) Here, special types of coupling are needed to give the necessary wide band. We will now briefly preview some of the requirements of a television receiver; there will be several lessons on these receivers further along in your Course.

# Television Superheterodyne Receivers

Aside from the use of a cathode ray tube as an image reproducer, the superheterodyne receiver used to pick up sight (video) and sound (audio) television signals is much like an all-wave sound receiver. You have already studied in your course the basic circuits used in a television receiver, for no radically new principles are involved. However, we can present here only a preview of this subject. There are a number of lessons dealing with the special adaptations made on basic circuits to fit television requirements.

The transmission of a 525-line picture which is repeated completely thirty times each second calls for a video frequency range of about 4.5 mc. Partial single side-band transmission is used for television in this country, with the lower side frequencies being partially rejected at the transmitter. Both the sight and sound modulated r.f. carrier signals are radiated by a television transmitter within a frequency channel 6 megacycles wide in the very high frequency band. Eighteen of these channels have been assigned to television; these channels are scattered over the frequency spectrum between 50 mc. and 300 mc.

In each television channel, the carrier center or resting frequency for the audio signal is .25 mc. below the highest channel frequency.\* The carrier for the video signal is 4.5 mc. below the audio carrier, with side frequencies extending above and below this value in

\*The sound portion of a television program is transmitted by means of frequency modulation; that is, the sound intensity variation is transmitted as r.f. frequency variations. The no-modulation frequency (corresponding to the carrier in a.m. systems) is referred to as the "resting frequency." Frequency modulation is discussed in another lesson.

the manner shown in Fig. 23.

► At the receiver, both the sight and sound r.f. signals are picked up by the same antenna, and both are passed through the preselector to a frequency converter stage. A single local oscillator feeding into the frequency converter stage converts both the sight and sound signals and their associated side frequencies to lower i.f. values at which greater gain can be obtained and satis-



Courtesy Stewart-Warner Corp.

A typical television receiver. The lid has a mirror mounted on it. The image on the television cathode ray tube is reflected by this mirror to the audience.

factory frequency response can be secured for all desired signal components.

As the sound and picture signals are on different carrier frequencies, they mix with the single oscillator frequency to produce two different i.f. values. The sight and sound signals thus separate at the output of the frequency converter stage. The sound portion of the television program goes into a sound

i.f. channel having at least a 200-ke. pass band, then to the amplitude limiters and frequency modulation detectors (which are essential in f.m. receivers), from which the sound signal feeds into the audio amplifier and loudspeaker. The sound channel includes provisions for rejecting picture signals.

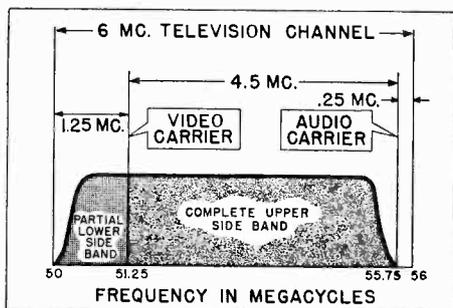


FIG. 23. Relationship between the audio carrier, the video carrier and the side frequencies in a typical 6-megacycle wide television channel. The frequency values specified below the diagram are for the 50-56 megacycle television channel.

► The sight carrier and its side frequencies pass into the video i.f. channel, which has a pass band almost 5.75 mc. wide. There are provisions in this channel for rejecting the audio carrier signal. The output of the video i.f. amplifier feeds through the video detector and video frequency amplifier to the television cathode-ray tube.

► A television picture signal includes impulses which indicate the end of a

line and the end of a picture, along with the variations corresponding to detail along each line of the picture. Although the entire signal including the impulses is fed to the television cathode ray tube, only the video components corresponding to details in the scene vary the brightness of the spot on the screen. At some point ahead of the television cathode ray tube, a stage known as the clipper separates the video signals from the synchronizing impulses. After this, the line impulses are separated from the frame impulses. The separated impulses control oscillators which develop the voltages required to sweep the electron beam across the cathode ray tube screen in the proper manner to reproduce the original scene.

► The requirements for very wide pass bands—6 megacycles in the preselector, 5.75 megacycles in the i.f. amplifier, and up to about 4.5 mc. in the video amplifier stages—have given television receiver design engineers some knotty problems. These have been solved by using plate load resistances of low ohmic value and circuits having wide pass bands, but this results in low gain per stage. To compensate for this, special tubes with high mutual conductance values were developed for television receivers. These tubes also have low interelectrode capacities, to minimize attenuation of higher video frequencies.

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# Lesson Questions

**Be sure to number your Answer Sheet 23FR-3.**

**Place your Student Number on every Answer Sheet.**

***Send in your set of answers for this lesson immediately after you finish them, as instructed in the Study Schedule. This will give you the greatest possible benefit from our speedy personal grading service.***

1. Why must the resonant load in the plate circuit of the first detector be highly selective?
2. What two radio frequency signals in a superheterodyne receiver are combined by the frequency conversion process to produce the i.f. carrier?
3. In addition to giving single-dial tuning, what advantage is secured by ganging the oscillator and preselector tuning condensers?
4. When an interfering signal is heard along with the desired signal, and the frequency of this interfering signal is *above* the frequency of the desired signal *by twice the i.f. value* of the receiver, what type of interference is present?
5. How would you eliminate code interference which is heard at all settings of the tuning dial in a superheterodyne receiver?
6. Why is it desirable to have a stage or two of r.f. amplification ahead of the frequency converter section in a superheterodyne receiver?
7. Give two reasons why the pentagrid mixer-first detector tube, when used with a separate oscillator tube, is a better frequency converter than a single pentagrid converter tube.
8. What two trimmer condensers must be adjusted in order to make the preselector and the oscillator track each other?
9. In what way, essentially, does an all-wave superheterodyne receiver differ from a broadcast band superheterodyne receiver?
10. What section of a superheterodyne receiver provides most of the adjacent channel selectivity?

## SMILE

When you give a smile, you give something which is priceless yet costs nothing. Nobody can buy, beg, borrow or steal your smile, because it is of no value unless you give it away in friendly greeting.

A smile takes but a moment, but its effects sometimes last forever. A smile creates happiness among friends, brings sunshine to the sad, and promotes valuable good will in business.

You don't feel like smiling? Then force yourself to smile. Whistle, hum a tune or sing softly—act as if you were already happy, and the smile will come. A happy smile comes from happy thoughts, not outward conditions.

It isn't what you are or where you are or what you are doing that makes you happy or unhappy—it's *what you think about it*. You'll find just as many happy faces among Chinese coolies sweating in the rice paddies for ten cents a day as you will among any similar-size group of business presidents in this country.

Smile!

J. E. SMITH

**TONE CONTROL,  
VOLUME EXPANSION AND  
NOISE LIMITING CIRCUITS**

24FR-1

**NATIONAL RADIO INSTITUTE**

ESTABLISHED 1914

**WASHINGTON, D. C.**



# STUDY SCHEDULE NO. 24

For each study step, read the assigned pages first at your usual speed, then reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind, then answer the Lesson Questions specified for that step. Study each other step in this same way.

- 1. Satisfying the Human Ear . . . . . Pages 1-5  
Distortion; Noise; Peculiarities of the Human Ear; Characteristic Curves for the Average Human Ear; Tone Controls; Volume Compression; Volume Expansion; Directional Effects. Answer Lesson Questions 1 and 2.
- 2. Simple Tone Controls; Special Bass and Treble Tone Controls . . . . . Pages 5-9  
Four-Position Tone Control; What Is Best Tone Control Setting; Continuously Variable Tone Control; How Tone Controls Reduce Noise; How Tone Controls Affect A.F. Response Curves; Parallel Resonant Bass Control Circuit; Audio Input-Shunting Bass Control Circuit; Treble Controls. Answer Lesson Questions 3 and 4.
- 3. Automatic Tone Controls; Unique Tone Controls . . . Pages 9-15  
Condenser-Type A.B.C. Circuit; Series-Resonant A.B.C. Circuit; Separate Bass and Treble Controls; Motorola Acoustinator; G-E Tone Monitor. Answer Lesson Questions 5 and 6.
- 4. Volume Expansion Circuits . . . . . Pages 15-19  
RCA A.V.E. Circuit; Crosley Auto-Expressionator. Answer Lesson Question 7.
- 5. The Noise Problem; Noise-Reducing Tone Controls; Noise Impulse-Silencing Circuits . . . . . Pages 19-23  
Signal-to-Noise Ratio; Man-Made Interference Noises; Receiver Noise; Lamb Noise Silencer; Diode Noise Limiter. Answer Lesson Question 8.
- 6. Inter-Carrier Noise Suppression Circuits . . . . . Pages 24-28  
Biased Demodulator Circuit; A.F. Amplifier Blocking Circuit; I.F. and R.F. Amplifier Blocking Circuit. Answer Lesson Questions 9 and 10.
- 7. Mail Your Answers for this Lesson to N.R.I. for Grading.
- 8. Start Studying the Next Lesson.

# Tone Control, Volume Expansion and Noise Limiting Circuits

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## Satisfying the Human Ear

THE manufacturers and operators of sound broadcast transmitters have one objective in common with the manufacturers of radio receivers—to make the receiver owner feel that he is listening to the actual studio performance rather than to a loudspeaker. This is by no means a simple task, and as yet has not been completely achieved even in the most expensive radio receivers. Radio engineers are doing everything within their power to improve the faithfulness of reproduction, and each year brings them another step closer towards their goal of perfection.

In this text-book we shall consider the various factors which make it difficult to achieve complete perfection, and study the methods used to satisfy the human ear under different conditions.

*Distortion.* One important requirement for faithful reproduction is freedom from all types of distortion. All of the sound frequencies which are present in the original performance must be present in the reproduced program and must *seem to have* the same loudness relation to each other as in the original performance; furthermore, there must be no frequencies in the reproduced program which did not originally exist (that is, there must be negligible amplitude distortion and negligible noise originating in the transmitting and receiving apparatus).

*Noise.* Broadcast transmitters can be, and in most cases are designed to radiate negligible noise. Radio receivers can likewise be so designed that noise originating in them is negligible.

It is noise which originates outside of the transmitting and receiving apparatus which requires special attention. Man-made interference noise is the chief offender in this respect; for this reason every serviceman should be familiar with the use of noise-reducing antennas for receivers and the application of noise filters to trouble-making electrical devices, all of which are taken up elsewhere in this Course.

Atmospheric noises, which enter the receiver along with the carrier of the desired station, cannot be entirely eliminated, but their annoying effects can be reduced considerably. You will learn that there are three practical types of circuits used for this purpose: 1, *Tone Control circuits*, which cut out high-frequency sound components, thereby eliminating predominating noise signals at a sacrifice of fidelity; 2, *noise impulse-silencing circuits*, which temporarily cut out all signals for the duration of sharp noise pulses which are stronger than the program signals; 3, *inter-carrier noise suppression circuits*, which cut out all signals, including noise, whenever the station signal is so weak that it is drowned out by noise signals. (Increasing the amount of power radiated by a transmitter, in order that the desired signal can better override noise signals, is a fourth solution, but transmitting power is definitely limited by law in order to prevent interference between stations.) Typical examples of the ingenious circuits which have been developed to accomplish these results will be studied in this lesson.

*Peculiarities of the Human Ear.* The characteristics of the human ear have

a lot to do with radio receiver design. Possibly you have noticed that orchestra music which sounds fine at a moderate distance away seems high-pitched (lacking in low frequencies) when you walk a considerable distance away.

The same phenomenon occurs during the fading of a radio program; this can most readily be noticed when a foreign short-wave program is tuned in. At a normal listening level of loudness the music will sound fine, but as the signal fades out, the music seems to become high-pitched or squeaky; as signal strength comes up to normal again, the music gradually becomes more mellow and pleasing.

The reason for this puzzling effect is simply that our ears hear low frequency or bass notes better at medium and high loudness levels than at low loudness levels. When the volume of a radio program drops because of fading, the bass notes seem to be cut far more than the higher frequencies.

Since radio receivers in homes are ordinarily operated at a loudness level considerably lower than that of the original performance, we have the queer situation that even a perfect radio receiver (one having a flat response over the entire audio range) would sound unsatisfactory whenever its output volume differed from that of the original program.

*Characteristic Curves for the Average Human Ear.* The peculiar hearing characteristics of the human ear have been carefully investigated. The results of tests on thousands of persons have been combined in the graph in Fig. 1, which tells how the average ear responds to various frequencies at different loudness levels.

A few words of explanation as to how data for this graph was secured will help you to read its story. First of

all, a 1,000-cycle sound was varied in loudness, while various persons listened, until an average loudness level which could just barely be heard was determined. This was called the 0 (zero) db *loudness* level. The actual intensity of this 1,000-cycle sound was then determined with a sound-measuring instrument, and the meter scale on the instrument was made to indicate 0 db *intensity* level for this condition. Now the same test was repeated for other frequencies, and the intensity level in db at which the sound could just barely be heard was measured with the instrument. Thus, at 500 cycles the average meter reading was +6 db; at 100 cycles it read 38 db, and at 30 cycles the reading was +63 db, etc. Each value was plotted on the graph, and a smooth curve was drawn through all the points to give the 0 db loudness level curve, also known as the threshold of hearing curve. This curve shows clearly that intensity levels considerably greater than 0 db are required below 1,000 cycles in order for sound to be heard, and also shows that the human ear hears best at about 4,000 cycles per second.

The intensity level at 1,000 cycles was then raised 10 db to give a loudness level of 10 db at this frequency, and the intensity level at each other frequency which seemed to give this same loudness was determined. This gave data for the 10 db loudness level curve. The same procedure was repeated at 10 db intervals to secure data for the remaining curves in Fig. 1.

It was found that when the loudness level was made higher than 120 db, the sounds were actually felt by the persons taking the test, the sensation being that of pain in the ears in some instances; the 120 db loudness level curve is therefore known as the *threshold of feeling*.

*How to Use the Ear Characteristics Graph.* Suppose we want to determine the intensity level required for a certain loudness, say 20 db, at a given frequency such as 200 cycles. We simply locate this frequency on the horizontal scale (point A), trace directly upward from it to the point where that frequency line intersects the 20 db loudness curve (point B), and then trace horizontally to the vertical reference scale at the left, where we read +40 db (point C) as the intensity level required in db for that loudness. In other words, for a 20 db loudness level

microphone which picks up all sound frequencies equally well.

We can draw these general conclusions from the curves in Fig. 1: At low loudness levels (from 20 to 40 db), such as those ordinarily produced by a radio receiver loud-speaker in the home, the human ear is most sensitive to sounds in the middle frequency range from 500 cycles to 5,000 cycles, and has difficulty in hearing sounds below and above this frequency range.

As we raise the loudness level of sound, such as by turning up the volume control on the receiver, the sensi-

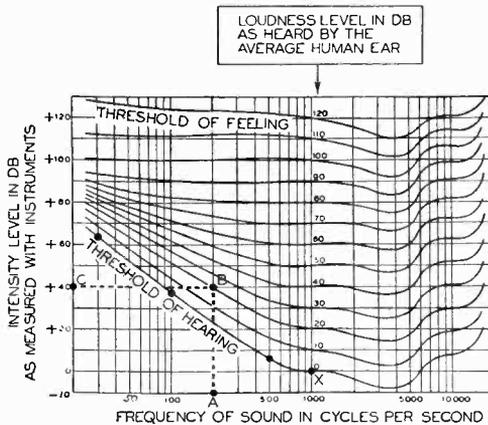


FIG. 1. This standard reference graph gives the average hearing characteristics (frequency response) of the human ear at various loudness levels; in other words, it tells what intensity level will be required for a given loudness level at a given frequency.

our ear requires 20 db more sound power at 200 cycles than at 1,000 cycles. At a 90 db loudness level, however, the curve is practically horizontal between these two frequencies, indicating that our ear can hear both frequencies equally well when they are this loud.

Bear in mind that *loudness level* is a human response to sound, and not something which can be measured with instruments. *Intensity level* is the actual measured level of sound as measured by instruments connected to a

tivity of the ear to low frequencies improves rapidly, so that at a loudness level of 100 db we can readily hear all sound frequencies below 10,000 cycles equally well. The lower the volume control setting on a radio receiver having an ordinary volume control, then, the fewer low or bass notes there will be insofar as our ears are concerned; to a lesser extent this statement also applies to the extreme high frequencies.

To get true high-fidelity reproduction at any desired loudness level, then, it is necessary to compensate for

the shortcomings of the human ear by boosting the response of a receiver at the low and high frequencies the correct amount for each loudness level. Boosting of the lows (raising of the bass notes) is far more important than raising the high-frequency or treble sounds.

*Tone Controls.* Many receivers are provided with separate bass and treble tone controls, which allow the listener to adjust the response of the receiver to secure faithful reproduction or to suit his particular taste at any desired loudness level. Oftentimes, however, the radio engineer makes these tone controls entirely automatic in operation, so that as volume is reduced, the low and the high frequency notes are automatically strengthened with respect to the middle frequencies. The amount of tone compensation provided at each loudness level is governed by the average reaction of the human ear to sound; receiver design engineers continually refer to the standard curves in Fig. 1 for this purpose. Automatic tone control is particularly appreciated by those listeners who do not have the ability to judge the correct adjustments of the bass and treble controls, or who do not want to take the time required to make these adjustments. Both manual and automatic tone controls will be taken up in this lesson.

*Volume Compression.* Of course, no radio program is broadcast at a constant volume level from instant to instant. For example, a symphony orchestra may have a variation in intensity of as much as 100 db from the weakest to the strongest passages. If the transmitter were adjusted to handle the loudest passages without overmodulation, then the weakest passages would be completely drowned out by transmitter circuit noise, and

the average modulation percentage will be too low for economical operation; on the other hand, if the transmitter is set to keep the lowest passages well above the transmitter circuit noise level, then severe overmodulation and distortion would occur on loud passages. A maximum variation of 40 db is about the most a transmitter can handle, although a range of 30 db seems to be preferred in most transmitters. If the signal level is allowed to rise too high, over-modulation and distortion occur; oftentimes this also results in overloading of the transmitter and opening of circuit breakers, throwing the station temporarily off the air. It is the duty of the studio operator who monitors a broadcast to reduce the audio gain of the transmitter for loud passages and boost the gain for weak passages, thereby compressing the sound into the allowable volume range. Automatic volume limiting systems are also employed for this purpose; these compress just before over-modulation occurs, by cutting down gain.

*Volume Expansion.* The compression of the original sound into a limited volume range will reduce the quality of the transmitted program. In a number of radio receivers special volume-expanding circuits are used to counteract this compression at high levels and increase the range of volume somewhat. Full expansion is ordinarily not possible because of the inability of the amplifier in the receiver to furnish the power required for the loudest sounds produced in the studio and because only a part (or none) of the compression occurring at the transmitter is automatic, but there is an appreciable improvement in the quality of reproduction when a volume-expanding circuit is incorporated in a receiver.

Sounds are compressed in much the

same way when phonograph records are made, and many of the better phonograph amplifiers have volume-expanding circuits. Because the high level compression usually is automatic in recording studios, the expansion can be made to correspond almost exactly, and very good fidelity can therefore be secured. Several different automatic volume expansion circuits will be studied.

*Directional Effects.* When we listen to an actual orchestra, our ears are able to determine from which direction each particular sound comes; thus we may note that the drums are to the right of the conductor, while the bass horns are at his left. When all these sounds are reproduced by a single loud-speaker at the receiving point, however, we lose this effect completely, and all of the sounds appear to come from the loudspeaker. With our present system of broadcasting, it is impractical to correct this situation, although it has been done already on an experimental basis.

In one test broadcast, two microphones were located a definite distance apart in the studio, and the sounds picked up by each were independently amplified and broadcast over two separate transmitters. At the receiving location were two separate receivers and loudspeakers, with the loudspeakers located the same distance apart as the microphones in the studio. As a result, the actual directional characteristics of the orchestra were duplicated.

Curiously enough, the lack of directional qualities in a radio program in no way destroys the entertainment value of the program for the average listener. We have become accustomed to radio transmission as it is, and consequently the tremendous expense involved in complete duplicate transmitting and receiving systems for each

broadcast is unwarranted at the present time.

### Simple Tone Controls

Any device which, when introduced into a radio receiving circuit, serves to reduce or remove the higher-frequency sound signals will have the apparent effect of boosting the low and medium frequencies. A condenser is one simple device for doing this, and is therefore widely used in simple tone controls. It is customary to make the insertion of one or more condensers in the circuit for tone control purposes optional, so the listener can select the most satisfactory tone control position for each loudness level.

When a tone control condenser is connected between the grid and the cathode or chassis of an audio amplifier stage or between the plate and the cathode or chassis, the high reactance of this condenser at low frequencies results in negligible by-passing of the signal, but at medium and high frequencies the reactance of the condenser is so low that there is appreciable signal by-passing. A simple condenser tone control gives an apparent boost in bass or low-frequency signals because the signal which it feeds to the next stage has a higher proportion of bass notes (with respect to medium and treble notes) than the signal fed into the tone control.

*Simple Four-Position Tone Control.* One simple and widely used tone control arrangement is that shown in Fig. 2A, where selector switch *SW* serves as the tone control. When this switch is set at point 1, no shunting or by-passing of the signal occurs, and hence all frequencies in the sound signal are fed to  $C_K$  without attenuation. As the switch is set to points 2, 3 and 4, increasingly greater capacities are placed between the plate and ground, and in-

creasing amounts of the middle and high frequencies are shunted to ground without passing through the load resistor  $R_L$  for transfer to the following stage.

In inexpensive receivers there are generally only two positions of the tone control, and only one condenser; in the first position of the tone control switch, which might be labeled "BRIGHT," no condenser is in the circuit and all frequencies are passed uniformly. In the second position of the switch, often labeled "MELLOW," the condenser is in the circuit and the medium and high frequencies are cut

that they will not sound any louder to the human ear than the low and medium frequency signals. It must be remembered that the curves in Fig. 1 were secured by averaging the hearing characteristics of a great many persons. Individuals may therefore deviate considerably from these curves.

*Continuously Variable Tone Control.* In the circuit of Fig. 2A, the tone control must be adjusted in steps, with the result that proper correction of tone will not be possible at certain loudness levels in between these steps. For this reason many people prefer a continuously variable tone control like

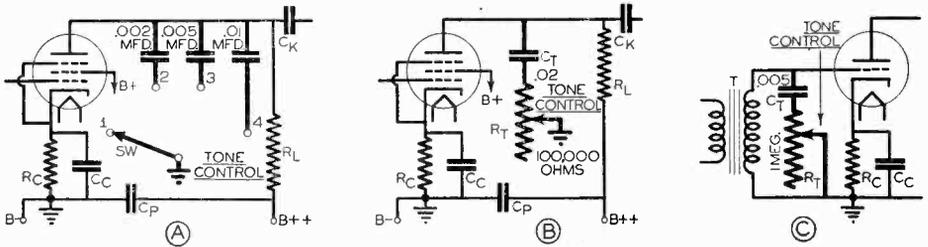


FIG. 2. Simple manual tone control circuits which utilize the signal by-passing action of a condenser.

down, giving a bass-boosting effect. Receivers with three-position tone controls, labeled "BRILLIANT," "BRIGHT" and "MELLOW," would use two condensers.

*What Is the Best Tone Control Setting?* It is incorrect to say that highest fidelity is always obtained at one particular position of the tone control. When the receiver is turned to maximum volume, giving a high level of loudness, that position in which there is no condenser in the circuit to by-pass the higher frequencies will give the best fidelity if the receiver normally has a flat response. On the other hand, when the receiver volume is turned down, it may be necessary to place the highest-capacity tone control condenser in the circuit in order to reduce the highest frequency signals enough so

that shown in Fig. 2B. Here a variable resistor is placed in series with a single tone control condenser. When this resistor is set at zero resistance, only the condenser is connected between the plate of the tube and ground, and we have maximum reduction of the high frequencies (the low and medium frequencies will then come through considerably stronger than the high frequencies). Increasing the resistance increases the net impedance of this tone control path to ground, thus reducing the by-passing effects on the high frequencies. When the maximum tone control resistance of 100,000 ohms is in the circuit, practically no signal frequencies take the path through the tone control circuit and there is no cutting of high frequencies.

A similar continuously variable tone

control arrangement is shown in Fig. 2C as it would be connected between the grid and chassis of an audio amplified stage. The action of this circuit is identical to that in Fig. 2B; a lower value of capacity and a higher value of resistance are required for the grid connection, as you can see by comparing the values specified in Fig. 2C with those in Fig. 2B. It is important to realize that an appreciable amount of attenuation or cutting of high frequencies can be achieved only if the impedance of the tone control circuit is considerably lower than the impedance of the load which it shunts. Since the grid-cathode path of the tube in Fig. 2C is a high-impedance load for the secondary winding,  $C_T$  and  $R_T$  have high ohmic values.

**How Tone Controls Reduce Noise.** Simple tone controls are widely used in all-wave receivers, and are especially valuable when listening to short-wave programs coming from far distant transmitters. The signals from these distant stations are generally accompanied by a great deal of noise; this is because they are received at such low levels that receiver gain must be increased considerably, and noise then becomes more noticeable. It is a known fact that in these noise signals the higher sound frequencies predominate; when the tone control is set to cut down high frequencies, the received signal becomes considerably less noisy. You can easily verify this for yourself by trying the tone control on an all-wave receiver while tuned to a European station; note how the noise is reduced when the tone control is set for MELLOW or BASS reception.

**Tone Controls May Be Connected Anywhere in the A.F. Amplifier.** There appears to be no one preferred position for the tone control in an audio amplifier. Sometimes this control is

connected in the output circuit of a diode detector, right at the input of the audio amplifier, as shown in Fig. 3. Potentiometer  $R_3$  serves as the volume control, with the setting of its movable arm determining the amount of A.F. voltage which is fed through blocking condenser  $C_4$  to the grid of the first audio stage. Tone control potentiometer  $R_2$  is shunted across  $R_3$ , and the movable contact of  $R_2$  is grounded through condenser  $C_3$ . Moving the tone control from position 1 to position 2 gradually increases the shunting effect of  $C_3$ , and therefore increases the cutting or by-passing of high-frequency signals.

A similar tone control arrangement could be used in Fig. 2B by replacing

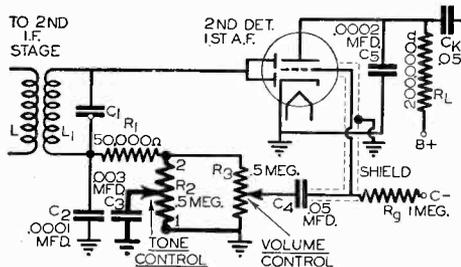


FIG. 3. Simple manual tone control circuit acting on the load of a diode detector; the parts values are those used in the Silvertone Model 1825A superheterodyne receiver.

resistor  $R_L$  with a potentiometer and connecting the movable tap of the potentiometer to the chassis through a .02 mfd. condenser;  $R_T$  and  $C_T$  would then be removed. Likewise in Fig. 2C the secondary of the audio transformer could be shunted with a .5 to 1 megohm potentiometer, with its movable contact grounded through a condenser, in place of the  $R_T$ - $C_T$  tone control arrangement shown.

**How Tone Controls Affect A.F. Response Curves.** The effect of a typical four-position tone control upon the over-all audio frequency response of a

radio receiver is clearly shown by the graph in Fig. 4. Curve 1 is for the tone control setting where there is no shunt capacity to cut down the higher frequencies; this curve therefore represents the normal frequency response of the receiver, indicating the relative output intensity level at each sound frequency. Curves 2, 3 and 4 show how this response is changed as increasingly greater shunt capacity is inserted in the circuit. These curves show clearly how a tone control cuts down the medium and high frequencies in the audio range without affecting the low or bass notes. With variable tone controls like those in Figs. 2B, 2C

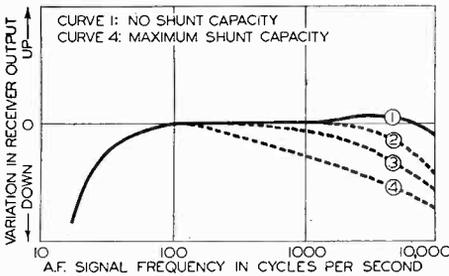


FIG. 4. This graph shows the effect of a four-position tone control upon the over-all frequency response of a typical radio receiver.

and 3, we could have any number of different response curves between 1 and 4, depending upon the control setting.

If the volume level of a receiver is raised to a high value when the tone control setting corresponds to curve 4 in Fig. 4, bass notes will be amplified excessively for the requirements of the human ear, and the over-correction will be readily noticed by the listener as a boomy effect. Some people actually prefer a very strong bass response and will purposely set the tone control for this result, even though they thus destroy the fidelity or quality of reception.

## Special Bass and Treble Tone Controls

In the more expensive radio receivers you will often find two separate tone controls, one for changing the low frequency or bass response and the other for changing the high frequency or treble response. Let us first consider bass controls.

*Parallel Resonant Bass Control Circuit.* Resonant circuits are widely used to control the bass response of a receiver; a typical example is that in Fig. 5A, where condenser  $C$  and iron-core inductance  $L$  form a parallel resonant circuit which has a resonant frequency of about 50 cycles ( $C$  will ordinarily be about .1 mfd., and  $L$  about 100 henrys). At resonance this circuit acts as a high resistance which, being in series with resistor  $R$  (about 10,000 ohms), will result in a high plate load impedance. The amplification of the stage is therefore increased in the low frequency or bass range around 50 cycles.

This parallel resonant circuit acts like a reactance of low ohmic value at frequencies higher than 50 cycles, under which condition resistor  $R$  governs the plate load impedance. The greater the resistance of the resonant circuit at resonance, the greater will be the boost in low frequency response; this resonant resistance can be varied over a wide range by adjusting the setting of potentiometer  $R_B$ . When all of  $R_B$  is in the circuit, the resonant resistance is quite low and there is little or no boosting of bass response. Maximum bass boosting is secured when all of  $R_B$  is shorted out.

*Audio Input-Shunting Bass Control Circuit.* Another widely used bass tone control circuit is shown in Fig. 5B. Inductance  $L$ , potentiometer  $R$  and condenser  $C$  form the bass tone control

circuit, which in this case is not a resonant circuit.

We can neglect the effect of  $C_K$ , as its reactance will be small in comparison to the reactance of  $L$  at any frequency or in comparison to the resistance of  $R$ . Assume for the moment that  $C$  is omitted; clearly we now have a voltage divider consisting of  $R$  and  $L$ , with the reactance between point  $P$  and ground determining the voltage output. Since the reactance of  $L$  at low frequencies is quite small in comparison to the resistance of  $R$ , the bass voltage will be developed essentially across  $R$  (between points 1 and 2). As potentiometer contact  $P$  is moved from point 2 to 1, less and less of this bass voltage is fed to the grid of the next tube. Since the audio system is intentionally made to have a high bass response, position 2 of the movable tap is for high bass response and position 1 is for low bass response.

Although moving contact  $P$  from point 2 to point 1 decreases the amount of bass forwarded to the following tube, it also decreases to some extent the medium and treble frequencies. Condenser  $C$  in Fig. 5B is used to bypass medium and high frequency signal components around the upper section of  $R$ , and consequently the medium and treble notes are passed by the tone control circuit without appreciable attenuation regardless of the setting of  $P$ . This bass control is definitely not a volume control when condenser  $C$  is used; if  $C$  is omitted, however, volume control action will be present along with bass control.

**Treble Controls.** The resonant tone control circuit in Fig. 5A can also be made to serve as a treble control if the values of  $L$  and  $C$  are chosen to provide resonance at a high audio frequency value. With this arrangement, varying the setting of potentiometer

$R_B$  will vary the high frequency response from its normal value to a value considerably above normal.

*Use of a Conventional Tone Control as Treble Control.* A more widely used procedure for securing treble control is that wherein the audio amplifier is designed to have a sharp rise in response at the higher frequencies, and one of the conventional tone control circuits shown in Figs. 2A, 2B, 2C and 3 is used to cut down this high frequency or treble response as desired.

### Automatic Tone Controls

The purpose of a tone control, as

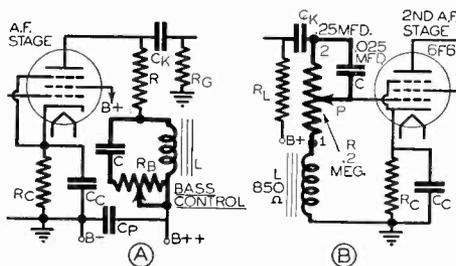


Fig. 5. Typical bass control circuits; that at A can also be made to serve as a treble control, by making L-C resonate at a high audio frequency. That at B is used in a General Electric all-wave receiver.

you already know, is to produce the effect of boosting the extreme low and extreme high-frequency response of the receiver to compensate for the inability of the human ear to hear these frequencies as well as the medium frequencies at low levels of loudness. The same results can of course be accomplished by reducing the level of the middle frequencies more than that of the extreme frequencies when reducing the volume. It is possible to secure this action automatically as the volume control is adjusted by the listener.

Engineers differ as to the extent to which automatic tone control should

correct for the shortcomings of the human ear at low levels of loudness. Some feel that only the bass should be reinforced (automatic bass compensation), while others prefer to reinforce both the treble and the bass (automatic tone compensation) to secure more nearly perfect compensation for the drop in sensitivity of the human ear under the conditions in question.

Automatic bass compensation (often abbreviated as A.B.C.) could be secured with a tone control of the type shown in Fig. 3 if we mounted this tone control potentiometer on the same shaft as the volume control, so that when volume was reduced, the medium and high frequencies would be cut down proportionately in order to satisfy the human ear. While this could be made to work perfectly satisfactorily, there is a much simpler way of accomplishing the same results.

*Condenser-Type A.B.C. Circuit.* A practical automatic bass compensation circuit is shown in Fig. 6A; although quite simple, it is entirely effective. The two terminals of this circuit,  $x$  and  $y$ , are connected either across the diode load resistor or across the output of some audio stage in the receiver. Volume control potentiometer  $R_1$  is tapped at point 2, which need not necessarily be its mid-point. Condenser  $C$  in series with resistor  $R$  between points 2 and 3 serves to attenuate (cut down) the middle and high frequencies at low-volume settings of the control, thus providing automatic bass compensation. Whenever you encounter a circuit like this, where the volume control has a fixed tap which is connected to ground through a condenser and a resistor, you can immediately identify it as an *automatic bass compensation circuit*.

Consider the circuit in Fig. 6A from the viewpoint of a voltage divider. If  $C$  and  $R$  were not present, the voltage  $E$  at any frequency would depend solely upon the position of contact arm  $P$ . With  $C$  and  $R$  connected as shown, and with the reactance of  $C$  decreasing with increases in frequency, the impedance between points 2 and 3 decreases as the frequency increases. The lower this impedance becomes, the more A.F. voltage there will be across section 1-2 of the voltage divider; in fact, at very high audio frequencies almost all of the available signal voltage is developed across points 1 and 2.

When  $P$  is at point 1, all of the A.F. voltage developed across the voltage divider is fed to the grid of the next stage regardless of its frequency. If  $P$  is placed at point 2, only the voltage developed between points 2 and 3 is fed to the grid of the next stage; the reactance between these points decreases at the higher frequencies and consequently there are fewer high-frequency components than normal in the output voltage  $E$  when  $P$  is at 2. We thus have the bass-boosting effect required for automatic bass compensation. At any position of  $P$  between points 1 and 2, the output voltage  $E$  depends upon the ratio of the impedance between  $P$  and 3 to the total impedance between points 1 and 3, and the amount of automatic bass compensation increases from zero (when  $P$  is at 1) to its maximum value when  $P$  is at 2. When  $P$  is moved below 2 (towards 3), the amount of bass compensation remains the same as for point 2 and all frequency components are reduced uniformly as in normal volume control action.

Occasionally you may find an additional condenser and resistor con-

nected in series between point 4 and the chassis, and another condenser and resistor between point 5 and the chassis; this is done to provide increasingly greater bass compensation as contact arm *P* is moved from 2 to 3.

*Series-Resonant A.B.C. Circuit.* Even greater attenuation or cutting of the high and middle frequencies for automatic bass compensation purposes is possible with a series resonant circuit arrangement as shown in Fig. 6B. Coil *L* and condenser *C* in series are adjusted to be broadly resonant to the middle and high frequencies; as a result, the effects of their reactances cancel at these frequencies, leaving only resistance *R* and the coil resistance in shunt with section 2-3 of

distance to give the desired broad tuning over the middle and high frequency range is used. If the *L-C* circuit is broadly resonant only for the medium frequencies, only these medium frequencies will cut down at low volume control settings, and automatic tone control (bass and treble compensation) is secured.

### Unique Tone Control Circuits

*Separate Bass and Treble Channels.* Some manufacturers of radio receivers provide one path in the audio amplifier for low-frequency signals and another entirely separate path for high-frequency signals; this permits amplification of these signals independently

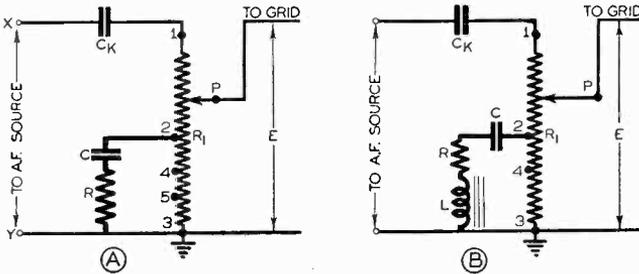


FIG. 6. Two practical automatic bass compensation circuits.

potentiometer *R*<sub>1</sub>; resistance *R* prevents complete cutting of signals at the resonant frequency. At low frequencies, however, the reactance of *C* is so high that the shunting effect of the series resonant circuit can be neglected. As contact arm *P* is moved from point 1 to point 2, lowering the volume, the amount of cutting of medium and high frequencies gradually increases to a maximum value when *P* is at 2. Further reductions in volume (by moving *P* below point 2) give no further increase in bass compensation unless an additional series resonant circuit or condenser-resistor circuit is connected between point 4 and ground. Very often resistor *R* is omitted and a coil which has a sufficiently high re-

and gives a definite control over their relation to each other.

A simple circuit of this nature is shown in Fig. 7. The audio signal output of the first A.F. amplifier tube (*VT*<sub>1</sub>) divides at point 1; the middle and high frequency signals in the audio range take the path through the .001 mfd. condenser *C*<sub>3</sub> to audio amplifier tube *VT*<sub>3</sub>, but the reactance of this condenser is so high at low frequencies that very few low frequency signals get through. The other path for signals from point 1 is through .5 mfd. condenser *C*<sub>2</sub> and through 100,000-ohm resistor *R* to audio amplifier tube *VT*<sub>2</sub>. Both low and high frequencies can pass through this condenser and resistor, but the high frequencies take

the .05 mfd. shunt path ( $C_4$ ) around the .5 megohm potentiometer ( $R_3$ ) to ground and consequently do not act upon the grid of  $VT_2$ . Only the low frequency components and a portion of the medium frequencies are amplified by  $VT_2$ . Resistor  $R$  prevents  $C_2$ - $R$ - $C_4$  from being a complete short for high frequencies, for that would leave none for the treble tube  $VT_3$ . The potentiometers in the grid circuits of  $VT_2$  and  $VT_3$  can be adjusted to provide any desired relationship between the bass, medium and treble frequencies. The outputs of the bass and treble amplifier tubes are combined at point 2

accentuate the bass and treble frequencies, thus giving high-fidelity reproduction of music at normal low sound levels.

The circuit diagram of a typical Motorola tone control system is shown in Fig. 8. The tone control switch is shown at position *V* (VOICE). Under this condition the A.F. signal is developed across load resistor  $R_1$ , all R.F. components accompanying this signal being by-passed to ground by the .0002 mfd. shunting condenser  $C_2$ . This audio signal voltage is then applied to .5 megohm resistor  $R_2$  through blocking condenser  $C_3$  and by-pass conden-

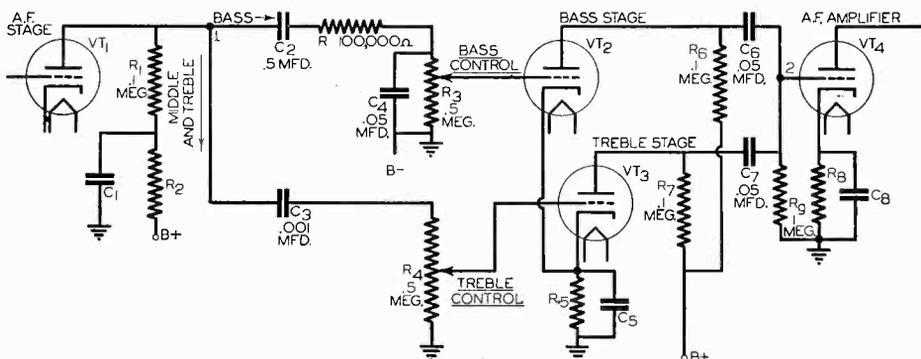


FIG. 7. Special tone control circuit which uses separate channels for amplification of bass and treble frequencies, with independent controls for each channel.

and fed from there to the grid of audio amplifier tube  $VT_4$ .

*The Motorola Acoustinator.* A rather unique bass and treble tone control is employed in some Motorola radio receivers. The audio systems in these receivers are designed to have a peak response at the medium frequencies, this being considered a highly desirable condition for clear voice reproduction. Means are provided for cutting down the medium frequencies and to some extent the treble frequencies when bass compensation is desired, such as at low levels of loudness. Means are also provided for cutting down the medium frequency response so as to

ser  $C_4$ . Since  $R_4$  and  $R_5$  offer no paths to ground, the entire signal voltage across  $R_2$  is applied to the control grid of the second A.F. tube through the  $R_4$ - $R_5$ - $C_5$  combination. The output of this tube feeds into the primary of an audio transformer across which is connected .007 mfd. condenser  $C_6$ . This condenser shunts some of the high-frequency audio signals around the primary, and therefore serves to cut down the high-frequency response. The audio amplifier normally has reduced bass response. As a result, when the tone control is in position *V*, the audio amplifier system has a response which is peaked in the middle frequencies.

Since the majority of voice frequencies are in this middle region, we have a condition which many people consider ideal for the reproduction of voice.

With the tone control in position *M* (MUSIC), the middle frequencies are cut down considerably and bass and treble frequencies therefore predominate. Position *M* is therefore used chiefly for reception of music at a low level of loudness. In this position of the tone control the bass signal currents take the path through resistor  $R_4$  and condenser  $C_7$  to ground, for  $C_5$  is essentially an open circuit at low frequencies. The reactance of  $C_7$  at bass

reactance of  $C_5$  is also quite low in relation to the ohmic values of  $R_4$  and  $R_5$  so at high frequencies we can consider  $R_4$  and  $R_5$  as being in parallel. The tone control input and output voltages will thus be the same at high frequencies, but this voltage will be reduced because  $R_4$  and  $R_5$  load the detector circuit. High and low-frequency voltages are about normal, while medium frequency voltages are cut considerably. The final effect is therefore a boosting of lows and highs.

With the tone control in position *B*, two condensers ( $C_6$  and  $C_8$ ) are connected to ground by the switch.  $C_6$

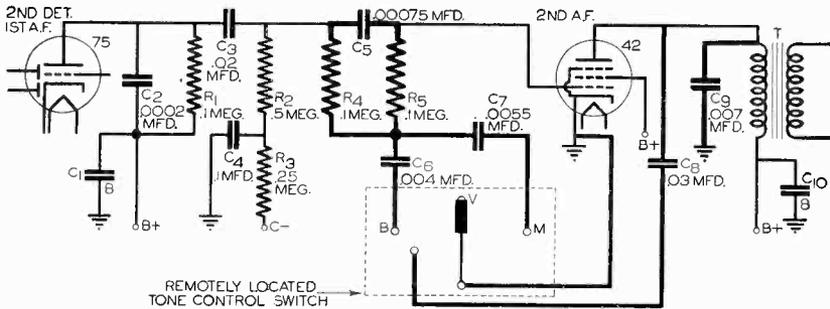


FIG. 8. Heavy lines indicate the circuit of the Acoustinator tone control system as used on the Motorola Model 70 and on many other Motorola receivers.

frequencies (about 50 cycles) is approximately .6 megohm or six times the resistance of  $R_4$ , and consequently nearly all of the bass signal voltage is developed across  $C_7$  and is impressed upon the grid of the type 42 tube through  $R_5$ . The middle frequencies, taking the same path as the bass signals, find a considerably lower reactance at the .0055 mfd. condenser; most of the available medium frequency signal voltage is wasted across  $R_4$ , and only that small portion which is developed across  $C_7$  affects the grid of the type 42 tube. At high frequencies, the reactance of  $C_7$  is so low that it can be considered a short-circuit path. The

acts in much the same way as  $C_7$  in cutting down the response at medium frequencies, while  $C_8$  acts like  $C_6$  in cutting the highs. As a result, position *B* (BASS) gives a high bass response; this reduces static and interference noises to a minimum, and gives a soft mellowness to music.

*The General Electric Tone Monitor.* When an audio signal is fed from the output of an audio amplifier back into the input of an audio stage in such a way that the feed-back voltage is 180 degrees out of phase with the normal input signal for that stage, degeneration (reduction in output) will occur. In the tone control system shown in

Fig. 9, used in General Electric model F-77 and other G.E. receivers, degeneration is purposely introduced in this way to provide a control over tone.

First let us analyze circuit conditions when the tone monitor switch (a type which connects together two adjacent contacts), is in the *S* or speech position. Observe that the switch now shorts  $R_{16}$  and  $C_{15}$ , so that feed-back current from point 1 on the secondary winding of the output transformer flows through the switch, through resistor  $R_6$  to point 2 on the volume control, and then through section 2-G of the volume control to ground, developing across this section an out-of-phase voltage which is applied to the grid of the first A.F. stage through section 2-P of this control. The higher the feedback current, the more degeneration or reduction in signal there will be. Since there is no reactance in the feedback current path (only the 22,000-ohm resistance of  $R_6$ ), the amount of feedback current which causes degeneration will be constant over the entire frequency range. This will lower the gain over the entire frequency range, but will not change the shape of the frequency response curve for the amplifier.

The audio frequency response curve for the amplifier circuit in Fig. 9 is that represented by curve *S* in Fig. 10 (the heavy solid-line curve); this will also be the response curve when the tone monitor is in the *S* position, for the shape of the curve is not affected by uniform reduction of gain over the entire frequency range. Note that there is a peak in the high-frequency region from 3,000 to 6,000 cycles; this occurs because the primary leakage reactance of output transformer primary  $T_5$  acts with Condenser  $C_{20}$  to form a resonant circuit at these frequencies, and maximum power is fed to the input

of  $T_5$ . The peak response in this range, along with uniform response down to 100 cycles, provides a pleasing reproduction of speech at normal room volume.

When the tone monitor switch is placed at position *F*, the foreign-reception position,  $R_{16}$  and  $C_{15}$  still remain shorted. Condenser  $C_{19}$  is now in parallel with resistor  $R_6$ ; at low frequencies the reactance of this condenser is quite high, with the result that feedback current will take the path through  $R_6$ . Normal amplifier response can therefore be expected at low frequencies, for feedback current is

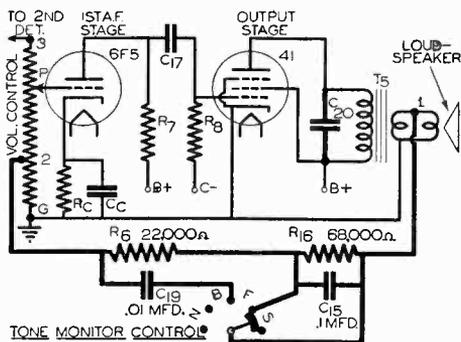


FIG. 9. The G. E. Tone Monitor Control circuit is shown here in heavy lines, as applied to the General Electric Model F-77 receiver (the amplifier circuit is shown here in simplified form).

still at the value determined by  $R_6$ . At higher frequencies the reactance of  $C_{19}$  is naturally less; since this condenser shunts  $R_6$ , more feedback current flows at higher frequencies than would ordinarily be passed by  $R_6$  alone, increasing the degeneration and thus cutting down the high frequency response. The shape of the response curve for this foreign-reception setting is represented by curve *F* in Fig. 10; this differs from the normal response curve only at frequencies above about 1,000 cycles. The attenuation or cutting of the high-frequency components serves to remove objectionable noise signals which ordinarily accompany

programs received from foreign countries.

When the tone monitor switch is in the *N* or normal-reception position, feed-back current flows through the parallel combination of  $R_{16}$  and  $C_{15}$ , and then flows through  $R_6$  to point 2 on the volume control. At low frequencies the reactance of  $C_{15}$  is high, and  $R_{16}$  acts in series with  $R_6$  to reduce the feed-back current practically to zero. As a result, peak bass response is secured. At medium and high frequencies the reactance of  $C_{15}$  becomes so low that it shorts out  $R_{16}$ , and only  $R_6$  is effective in limiting feed-back current. This means that in the *N* position of the tone monitor switch we will have the normal treble peak in the response curve as well as a peak in the bass or low frequency region, as indicated by curve *N* in Fig. 10; this is a desirable condition for listening to music at the low loudness levels usually preferred in the home.

Setting the tone monitor switch at position *B*, the bass position, places condenser  $C_{19}$  across both  $R_6$  and  $R_{16}$ , and inserts the  $R_{16}$ - $C_{15}$  combination in series with  $R_6$ . At low frequencies the reactances of these two condensers are so high that we can neglect them; this leaves  $R_{16}$  and  $R_6$  acting in series and reducing feed-back current almost to zero. Degeneration is thus almost entirely eliminated at low frequencies, and bass response goes up.

At medium frequencies the reactance of  $C_{15}$  is low enough to provide a shunt path around  $R_{16}$ , but  $C_{19}$  is still out of the picture. The 22,000-ohm resistance of  $R_6$  now controls feed-back current, and so we have essentially the same medium frequency response as was secured at positions *S*, *F* and *N*.

At high frequencies  $C_{15}$  shorts out  $R_{16}$ , and  $C_{19}$  provides a low-reactance

shunt path around  $R_6$ . Feed-back current thus increases considerably, cutting down the high-frequency response quite rapidly as frequency increases. At position *B*, then, curve *B* in Fig. 10 will be the response curve of the receiver; it clearly indicates a boosting of bass frequencies.

Naturally each setting of the tone monitor control will sound different to the human ear at each setting of the volume control, because of the changing characteristics of the human ear with loudness. Automatic compensation for the variations in sensitivity of the ear with loudness is secured by

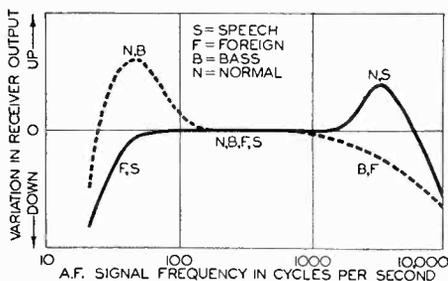


FIG. 10. This graph shows how the G. E. Tone Monitor Control (Fig. 9) changes the audio frequency response of the receiver in which it is used. The graph represents full-volume conditions, when *P* is at point 3; there is then no automatic tone compensation.

making the tone control connection to a tap at point 2 on the volume control potentiometer. Varying the volume control setting thus gives an automatic balance between the normal response of the amplifier and the special response introduced by the tone monitor circuit.

### Volume Expansion Circuits

The compression of sound level is of extreme importance in the practical operation of radio transmitters, for otherwise over-modulation and an excessively low signal-to-noise ratio would exist. Compression of sound level is just as important in the pro-

duction of disc recordings of musical numbers, for excessive volume results in over-cutting of the grooves. Volume compression, of course, destroys the fidelity or faithfulness of a program, for it cuts down the volume of high-level passages. In order to counteract this volume compression, it is necessary to introduce some form of volume expansion in the reproducing device.

Although automatic volume expansion can be applied to both radio receivers and electric phonographs, it is far more satisfactory when the original compression is performed automatically, as is now being done on most recordings. In radio transmitters the compression is done manually by the studio operator, and obviously no automatic circuit in the receiver can exactly compensate for an action which depends for its accuracy upon human alertness and skill.

The most widely used system for automatic volume expansion depends upon the fact that *the gain of a variable mu or super-control pentode tube varies with the negative C bias voltage which is applied.* By applying to a variable mu pentode amplifier tube, in series with its normal negative C bias, a positive D.C. voltage which increases with sound level, the gain of the tube can be made to increase as sound level or volume increases. The necessary D.C. or A.V.E. (automatic volume expansion) control voltage is secured by rectifying the audio voltage and filtering out the A.C. component of the resulting pulsating voltage. This D.C. voltage must of course be time-delayed, for if the A.V.E. control voltage varies too rapidly with fluctuations in sound level, the reproduced music will sound "choppy" and have a "gurgling" effect; likewise if the A.V.E. control voltage is too greatly time-delayed, the benefit of automatic volume expansion

will be lost.

*RCA A.V.E. Circuit.* A typical automatic volume expansion circuit based upon this principle and used in a phonograph amplifier is shown in Fig. 11. The phonograph pick-up at the upper left in the diagram feeds an audio signal into step-up transformer  $T_1$ , across the secondary of which is shunted a 625,000-ohm volume control potentiometer  $R_4$ . Notice that this potentiometer is in an automatic tone compensating circuit, there being two taps on the potentiometer for this purpose. Series resonant circuit  $L_2-C_2$ , connected to the first or uppermost tap, serves to attenuate the middle frequencies and thereby has the effect of raising the bass and treble response at medium and low volume levels. Resistor  $R_2$  broadens the resonant response of  $L_2-C_2$  and limits the amount of attenuation at medium frequencies.  $R_3$  and  $C_3$ , connected between the lower tap on the potentiometer and ground, provide an additional boosting effect for the bass response at still lower volume levels. Series resonant circuit  $L_1-C_1$ , also connected across the secondary of  $T_1$ , resonates at about 700 cycles and serves to cut down the response of the audio amplifier in this region. Resistor  $R_1$  prevents complete cut-off at resonance.

After passing through the volume and tone control circuits, the audio signal is fed through D.C. blocking condenser  $C_4$  into vacuum tube  $VT_1$ , which is ordinarily operated at a high negative C bias and thus normally has low gain. At the same time the audio signal developed across the secondary of transformer  $T_1$  is fed into vacuum tube  $VT_2$ , a triode amplifier; this audio signal has not been acted upon by the automatic tone and volume control circuits. The output of  $VT_2$  is rectified by  $VT_3$  and then filtered to give a D.C.

voltage whose value is dependent only upon the signal level at the secondary of  $T_1$ . This varying D.C. voltage is applied to one of the grids of pentode tube  $VT_1$  in such a way as to increase the gain of the tube in proportion to the level of the original sound. Now let us trace the circuit in greater detail.

Potentiometer  $R_8$  controls the amount of signal voltage applied to triode amplifier  $VT_2$ , and thus provides a control over the amount of volume

Since  $VT_3$  permits electron flow only in the direction from cathode to grid, the A.F. voltage existing across  $R_{10}$  sends through  $R_{11}$  a pulsating current which develops across this .22-megohm resistor a pulsating D.C. voltage, with polarity as indicated. Condenser  $C_{12}$  smooths out the A.F. variations, and the resulting D.C. voltage is applied to the third grid ( $G_3$ ) of  $VT_1$  through filter  $R_{13}$ - $C_5$ , which provides the necessary time delay in the application of

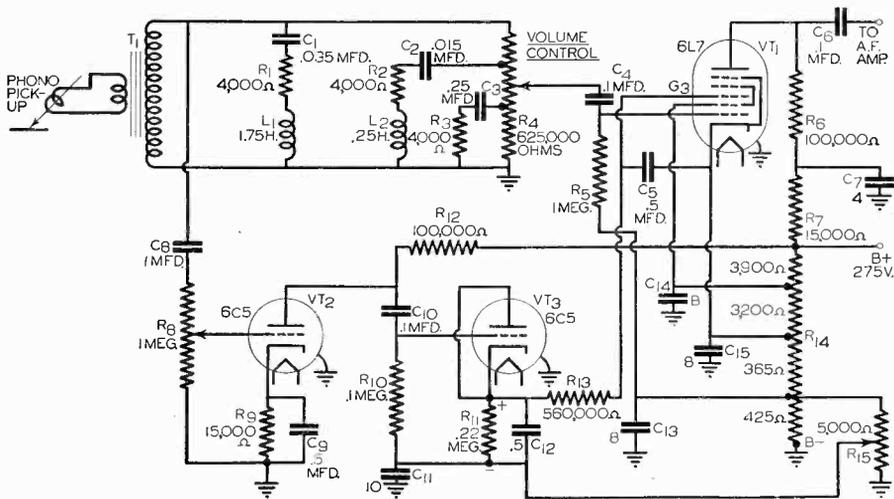


FIG. 11. Automatic volume expander circuit as used in a number of RCA electric phonographs.

expansion.  $R_9$  and  $C_9$  together provide automatic C bias for this tube, and plate voltage is obtained by a connection through resistor  $R_{12}$  to the B+ terminal of the receiver power supply circuit. The A.C. plate current of  $VT_2$  flows through  $C_{10}$ ,  $R_{10}$  and  $C_{11}$  to the chassis, with most of the voltage being dropped across  $R_{10}$ . Observe that the plate and cathode of  $VT_3$  are connected together; this tube therefore acts as a diode rectifier with its grid (acting as anode) connected to one side of  $R_{10}$  and its cathode connected through  $R_{11}$  to the other side of  $R_{10}$ .

the A.V.E. control voltage. The negative terminal of  $R_{11}$  connects to ground through potentiometer  $R_{15}$ , which is connected across a part of voltage divider  $R_{14}$  in such a way as to provide an adjustable negative C bias which will act in series with the positive bias across  $R_{11}$ . Potentiometer  $R_{15}$  need be adjusted only when a new tube is inserted in the volume expander circuit.

At low volume levels practically no D.C. voltage is developed across  $R_{11}$ , and grid  $G_3$  of  $VT_1$  then receives the full negative C bias voltage developed by  $R_{15}$ . As the volume level increases,

the positive voltage drop across  $R_{11}$  increases and counteracts the negative C bias, with the result that the third grid becomes less negative and the gain of tube  $VT_1$  increases.

The action of an automatic volume expander circuit of the type shown in Fig. 11 is illustrated by the graph in Fig. 12. The lower curve tells how the output level varies with the input level when there is no automatic volume expansion, and the upper curve gives the

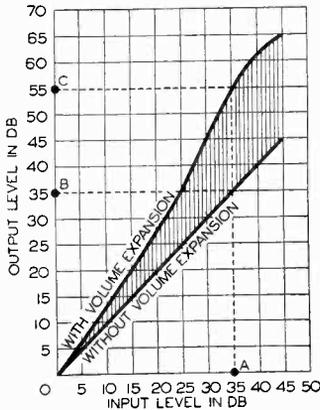


FIG. 12. Automatic volume expansion action of the phonograph amplifier circuit in Fig. 11. In preparing this graph, a definite low input signal level was assumed and called 0 db input level, and the output level under this condition was called the 0 db output level. These curves tell that if the receiver is being fed with a signal of 35 db (point A), the output level without A.V.E. will also be 35 db (point B); with A.V.E., however, the output level will be 20 db higher, or 55 db (point C).

same information for a circuit having automatic volume expansion. Notice that as the input signal level is increased, automatic volume expansion makes the output level increasingly greater than would be obtained without A.V.E. With proper circuit design, an exact reproduction of the original program can be secured even though the volume level is condensed during the recording process in the case of a phonograph record or during the transmitting and receiving process in the

case of a radio system.

*Crosley Auto-Expressionator.* Another interesting automatic volume expansion circuit, which in addition provides automatic bass compensation at low sound levels, is that shown in Fig. 13; as you can see, this is a special form of filter (some engineers call it a differential bridge) connected between the output transformer of the receiver and the loudspeaker. It is used in a number of Crosley receivers, where it is known as an *auto-expressionator*.

All four switches in Fig. 13 are operated by a single control which permits instant change-over from normal operation to A.V.E. and A.B.C. Normal operation is secured when switches  $SW_3$  and  $SW_4$  are open, and switches  $SW_1$  and  $SW_2$  are closed; a study of the circuit will show that under these conditions a direct connection exists between the loudspeaker and the output transformer, with no parts in series or shunted across the loudspeaker.

With the control switch in the A.V.E. position, switches  $SW_1$  and  $SW_2$  will be open and switches  $SW_3$  and  $SW_4$  will be closed, as indicated on the circuit diagram. We will assume for the moment that the loudspeaker is disconnected. Audio frequency current flowing from the output transformer to point A will have two paths to point B; 1, through  $X_1$  (a ballast resistor), resonant circuit  $L_1-C_1$  and  $R_1$ ; 2, through  $R_2$ , resonant circuit  $L_2-C_2$  and ballast resistor  $X_2$ . At point B the currents from the two paths combine again and return to the output transformer. Resistors  $R_1$  and  $R_2$  are equal in value; ballast resistors  $X_1$  and  $X_2$ , whose resistance increase with current, are likewise the same size; resonant circuits  $L_1-C_1$  and  $L_2-C_2$  are also identical in characteristics, and thus the same current would flow over each path when the loudspeaker is disconnected.

When the loudspeaker is connected between points *C* and *D*, the loudspeaker current will depend upon the voltage existing between these points, and this voltage will be the difference between the voltage drops for *A-C* and the voltage drop for *A-D*. The ballast resistors used in this circuit are of special design, increasing in ohmic value as the current through them increases. This means that as the output current of the receiver increases, the resistances of  $X_1$  and  $X_2$  increase, causing the impedance of branch *A-C* to be greater than that of branch *A-D* and causing the impedance of branch *B-D* to be greater than that of *B-C*. Even when cold, the ohmic values of the ballast resistors are slightly greater than the ohmic values of  $R_1$  and  $R_2$ . As a result, the circuit is in an unbalanced condition at all times, with some signal voltage always being applied to the loudspeaker. The unbalance becomes greater as the receiver output current increases and sends more current through the ballast resistors. At low values of power output, when the circuit is most nearly balanced, only a small amount of the total output power is supplied to the loudspeaker.

As the power output of the receiver increases, the circuit becomes farther out of balance and more and more of the receiver output power is applied to the loudspeaker. This increases the intensity of the louder passages, expanding the volume range. The time lag in the heating and cooling of each ballast resistor filament is an important factor in the proper operation of this circuit, for it determines the speed of A.V.E. action.

Now let us see how this circuit works as a bass compensator. Since each resonant circuit resonates at about 40 cycles, we can see immediately that for the medium and high audio fre-

quencies, the reactances of condensers  $C_1$  and  $C_2$  become quite low, and the resonant circuits thus do not create an unbalance at these higher frequencies. At low or bass frequencies, however, the resonant resistance of each resonant circuit becomes quite high, causing a greater-than-normal unbalance and thus sending more current through the loudspeaker. The result is a boosting of the bass notes at low sound levels. At high sound levels, however, the resistance of  $X_1$  increases to the point where it makes the resonant resistance of each resonant circuit prac-

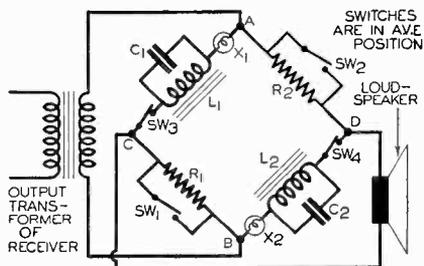


FIG. 13. Crosley Auto-Expressioner circuit, a combination automatic volume expansion and automatic bass compensation arrangement.

tically negligible in so far as unbalance is concerned; this means that bass compensation occurs only at low sound levels.

### The Noise Problem

*Signal-to-Noise Ratio.* One of the most perplexing problems with which the radio engineer has to deal is that of noise. Regardless of whether this noise originates in the receiver or outside the receiver, it is reproduced along with the desired signal and is oftentimes strong enough to be annoying. Recognizing that it is impossible to eliminate noise entirely, the engineer endeavors to use apparatus which will make it negligible with respect to the desired code, sound or picture signal.

This is why he continually refers to the signal-to-noise ratio (the signal voltage or current divided by the noise signal voltage or current).

Even the average radio listener unknowingly recognizes the importance of a high signal-to-noise ratio, for he invariably tunes to a local station when he desires to enjoy a high-quality musical program with a minimum of noise. This local station, being close to the receiving location and having reasonably high power, can produce at the receiving antenna a high enough signal strength to give the desired *high signal-to-noise ratio*.

#### *Man-Made Interference Noises.*

Noise originating outside of the receiver is usually caused by electrical apparatus such as motors, sign flashers and other devices in which sparks occur when electrical circuits are broken. This type of noise is commonly referred to as man-made interference and is most common in industrial cities and towns, where it will predominate over atmospheric noises except perhaps on stormy days. Much of this interference can be reduced by the use of special noise-reducing antennas (considered elsewhere in the Course), but the ideal remedy is elimination of the interference at its source by the use of noise filters.

Radio stations have aided materially in reducing the effects of noise by increasing the power of their transmitters. In extremely noisy locations in cities, owners of radio receivers should be told to tune to the high-power local stations in order to secure a high signal-to-noise ratio at their receivers.

The wave form of a typical noise signal is shown in Fig. 14; notice that there is a general noise level, existing practically all of the time, which is ordinarily not objectionable except when low-power or distant stations are

tuned in; in addition, there are sharp pulses of noise or static at irregular intervals, which usually are objectionable. As a rule these pulses are of very short duration, less than .001 second.

*Receiver Noise.* Inside the radio receiver, thermal agitation and tube shot effect, particularly in the first R.F. stages of a receiver, result in unavoidable circuit noises. Fortunately these circuit noises can be reduced to a level less than that produced by a one-microvolt input signal (by proper selection of circuits and tubes), and the signals of most stations can override this noise level. The mixer-first detector in a superheterodyne receiver can give excessive noise, but a properly

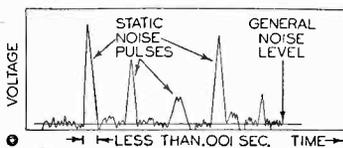


FIG. 14. Wave forms of typical rectified noise signals as they may exist in the detector stage of a radio receiver.

operated stage preceded by R.F. amplification will not have this trouble.

We shall now consider in turn the use of tone control circuits, noise impulse-silencing circuits and inter-carrier noise suppression circuits for noise-reducing purposes.

#### **Noise-Reducing Tone Controls**

An analysis of noise signals will reveal that they contain practically all audio signal frequencies, with the higher frequencies, above 3,000 cycles, predominating. It is for this reason that a tone control which suppresses or attenuates the high audio frequencies is also effective in reducing the amount of noise. Of course, this suppression of the high audio frequencies destroys the faithfulness of reproduction, and consequently tone controls

are far from being a satisfactory solution to the noise problem.

### Noise Impulse-Silencing Circuits

*The Lamb Noise Silencer.* The circuit shown in Fig. 15, developed by J. J. Lamb, serves to eliminate noise pulses by silencing the receiver for the duration of each pulse. This circuit has

generally inserted between the I.F. amplifier and the second detector of a superheterodyne receiver, as indicated in Fig. 15.

The signal voltage developed by the second I.F. transformer between point A and ground is made up of an I.F. carrier modulated by noise signals and by desired intelligence signals. This voltage is applied directly to the con-

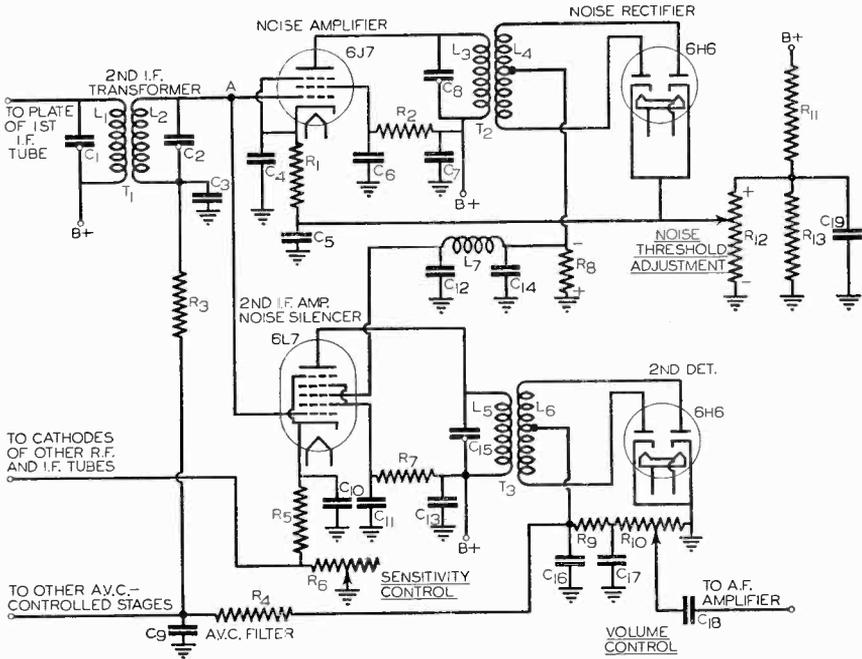


FIG. 15. The Lamb noise silencer, which silences the receiver for the duration of each strong noise pulse, is shown here incorporated in a superheterodyne receiver.

been found quite valuable in commercial and communication receivers which are used for code signal reception. It has not been entirely satisfactory for broadcast receivers, however, for the action of the noise-silencing circuit has the effect of partially destroying the peaks of modulation in the received carrier, thereby creating the more serious problem of amplitude distortion.

The Lamb noise-silencing circuit is

control grid of the 6J7 noise amplifier tube, and also to the control grid of the 6L7 second I.F. amplifier tube. The amplified output of the second I.F. tube feeds through tuned primary transformer  $T_3$  to the 6H6 full-wave diode second detector, which through rectification provides the desired A.F. signal voltage and a D.C. voltage for A.V.C. purposes. The amplified output of the noise amplifier tube, on the other hand, feeds through tuned primary

transformer  $T_2$  (tuned to the I.F. value) to another 6H6 double-diode tube which serves as a full-wave noise rectifier. The cathodes of both the noise amplifier and noise rectifier tubes are connected to the movable contact of noise threshold adjustment potentiometer  $R_{12}$ , and the setting of this control determines the value of the negative bias which is applied to the control grid of the noise amplifier and to the diode sections of the noise rectifier. This bias on the diodes acts much like a delay voltage, in that it determines the minimum signal strength required to swing the diode plates positive and start rectification; the bias is normally set so that rectification occurs only for noise peaks extending above the desired signal level. Note that the noise amplifier tube also receives a bias from the A.V.C. circuit of the receiver through A.V.C. filter  $R_3-C_3$ ; this serves to maintain the proper noise threshold level despite normal variations in desired signal level due to fading.

The rectified noise voltage, which is developed across noise rectifier load resistor  $R_8$  when strong noise peaks are present, is made up of I.F., A.F. and D.C. components. Filter combination  $C_{14}-L_7-C_{12}$  removes the I.F. and A.F. components; the remaining D.C. voltage is applied to the third grid of the second I.F. amplifier-noise silencer tube, driving this grid more negative with respect to ground and thus blocking plate current flow for the duration of each strong noise pulse.

When no noise pulses are present, the noise rectifier plates are negative with respect to their cathodes; no current passes, no D.C. voltage exists across  $R_8$ , and consequently the third grid of the 6L7 second I.F. tube has only the normal negative C bias determined by  $R_5$  and  $R_6$ . Desired signals are therefore amplified and de-

tected in a normal manner by the two lower tubes in Fig. 15.

When a strong noise pulse enters the receiver along with the desired signals, it instantly drives the noise rectifier plates positive, and the D.C. voltage developed across  $R_8$  by the resulting rectified current flow is applied without delay to the third grid of the second I.F. amplifier-noise silencer tube, blocking plate current flow completely. Under this condition no desired or noise signals whatsoever reach the second detector, and consequently the noise is not heard. The instant the noise pulse drops below normal signal level, the noise rectifier stops conducting, the blocking voltage is removed from the third grid of the 6L7 tube, and normal receiver operation is restored.

Naturally there must be no time delay between the arrival of a noise pulse at the noise amplifier and the application of the D.C. silencing voltage to the third grid of the noise-silencer tube if the silencing action is to be instantaneous and is to last only for the duration of each strong noise pulse. Under this condition the receiver is silenced for such a short interval at a time that the action can scarcely be noticed by the human ear.

*Diode Noise Limiter.* If the load of a diode detector is instantly shunted with a large condenser when a noise pulse comes through, the audio voltage developed across this load, as well as the noise pulse, will be greatly reduced for the duration of the pulse. A practical noise-limiting circuit based upon this principle is shown in Fig. 16. This circuit uses a type 6H6 double-diode tube, with section  $D_1$  serving as second detector (resistors  $R_2$  and  $R_3$  form its load) and section  $D_2$  serving as an automatic noise-limiter switch places a large condenser

across  $R_3$ , a part of the detector load, during the time of a noise pulse (the two sections are shown as separate tubes to simplify your study of the circuit, but remember that they are actually in the same vacuum tube envelope). The D.C. voltage developed across the detector load resistors is used for A.V.C. purposes, with  $C_F$  and  $R_F$  serving as the A.V.C. filter. Condenser  $C$  is an R.F. by-pass condenser which keeps I.F. signals out of the detector load. Switch  $SW$  is a panel control, and must be in the closed position when noise-limiting operation is desired.

When no strong noise pulses are present, the normal rectified electron flow is from point 1 to point 3 through the detector load, and point 1 is therefore more negative than points 2 and 3. The D.C. voltage developed between points 1 and 3 charges condenser  $C_1$  through resistor  $R_1$ , which is of high ohmic value. Once this condenser is charged, no additional current flows through  $R_1$ , and points 1 and 4 are then at the same negative potential with respect to ground (point 3). Diode section  $D_2$  now acts as a high resistance, for its anode (point 4) is more negative than its cathode (point 2).

Now suppose that a strong noise pulse is rectified by diode section  $D_1$  along with the desired modulated carrier. The flow of rectified current through the detector load increases instantly, and the D.C. voltage between points 1 and 3 will likewise increase instantly.  $C_1$  will begin drawing current through  $R_1$  in order to become charged to the new voltage value. The time constant of the  $R_1$ - $C_1$  combination is quite high, however (ordinarily it is from 1/10 to 1 second), and consequently  $C_1$  cannot be charged instantly to the new voltage. Since the average noise pulse lasts about 1/1000

of a second, which is considerably less than the time constant, point 4 remains essentially at the previous low negative potential for the duration of the noise pulse, while point 2 goes instantly to the new higher negative potential. This means that the plate of diode section  $D_2$  becomes positive with respect to its cathode for the duration of the noise pulse; current then flows through  $D_2$  and its resistance drops to a quite low value. Since this tube is in series with high-capacity condenser  $C_1$  across  $R_3$ , the net impedance between points 2

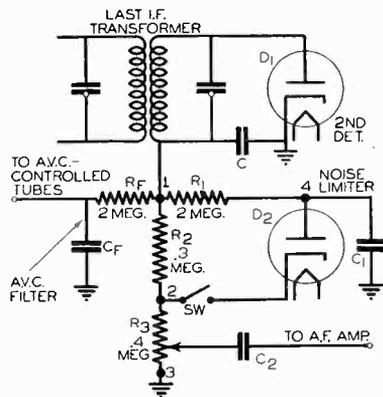


FIG. 16. Diode noise-limiter circuit, which shunts all audio signals to ground through a condenser for the duration of each strong noise pulse when switch  $SW$  is in the closed or noise-reducing position.

and 3 is lowered, and the A.F. voltage across  $R_3$  is therefore lowered. As a result, the A.F. voltage fed to the first audio stage drops considerably for the duration of each noise pulse, and the A.F. output voltage of the receiver is correspondingly lowered. The instant the noise drops below normal signal level again, point 2 returns to its original potential, current flow through  $D_2$  stops, and the  $D_2$ - $C_1$  shunt path across  $R_3$  is no longer effective in limiting the audio amplifier input voltage.

## Inter-Carrier Noise Suppression Circuits

In receivers having A.V.C., the noise which is heard when tuning between stations may be objectionable to some listeners. This inter-station or inter-carrier noise occurs *because automatic volume control raises the gain of the R.F. system to the point where normal external and internal noises become objectionable*. Some receivers have an inter-station noise-suppression circuit which works in conjunction with A.V.C. to silence the receiver when no station is tuned in. This inter-carrier noise suppression system is often known as quiet automatic volume control, abbreviated Q.A.V.C.; since it stops or squelches noise it is also known as a noise-squelching circuit. In some circuits the detector action is delayed for weak signals; in other circuits the I.F. or A.F. amplifier is blocked until the station signal reaches a level which is high enough to override noise which may be present, thus preventing reception of stations having less than the minimum satisfactory signal-to-noise ratio. To be sure, this has the effect of reducing the apparent sensitivity of the receiver; a switch is therefore provided to cut out this noise-suppressing action when maximum sensitivity is required regardless of noise.

The diode noise-limiter circuit shown in Fig. 16 also serves to a certain extent as an inter-carrier noise suppressor. Suppose that the receiver is tuned off a station. Diode section  $D_1$  of course receives no input voltage under this condition, and no current flows through  $R_2$  and  $R_3$ . Points 1 and 2 immediately assume ground potential, and point 4 will likewise be at ground potential after the time-delay interval (less than one second), during which condenser  $C_1$  discharges to ground through  $R_1$ ,  $R_2$  and  $R_3$ , has elapsed.

Noise or static signals which enter the receiver under this condition will, of course, be rectified by diode section  $D_1$ , making point 2 negative with respect to ground. Point 4 remains at ground potential because of time-delay action, making the plate of  $D_2$  positive with respect to its cathode. Noise signals reaching point 1 will therefore take the low-impedance path through  $R_2$ ,  $D_2$  and  $C_1$  to ground. Noise is thus diverted from the input of the A.F. amplifier and will not be reproduced by the loudspeaker when tuning slowly from station to station. Of course, the noise-reducing action of this circuit ceases as soon as the time-delay interval has elapsed.

In the more conventional types of inter-carrier noise suppression circuits, the receiver is made inoperative for weak signals on the assumption that the noise signals heard between stations will be weaker than any of the desired strong carrier signals. Three types of inter-carrier noise suppression circuits are used for this purpose: 1, *biased demodulator circuits*, which place a bias or delay voltage on the diode detector so that only carrier signals above a certain level will be demodulated; 2, *A.F. amplifier blocking circuits*, which block or prevent operation of the A.F. amplifier until the carrier signal exceeds a definite minimum value; 3, *I.F. and R.F. amplifier blocking circuits*, which prevent operation of these amplifiers until the carrier level becomes appreciably higher than the noise level. Let us consider one example of each of these inter-carrier noise suppression methods.

*Biased Demodulator Circuit.* As you know, delayed A.V.C. action can be obtained by inserting in the separate A.V.C. diode rectifier circuit a delay voltage which must be exceeded by the carrier signal before the rectification

required for the production of an A.V.C. voltage can take place. Usually, a separate diode rectifier is then required for demodulation purposes.

If we introduce this delay voltage directly into the diode rectifier circuit used for demodulation, so that only signals above a certain minimum level will be rectified, we have one form of inter-carrier noise suppression. Weak carrier and noise signals cannot overcome this delay voltage, and hence are not reproduced by the loudspeaker as the receiver is tuned from one station to another. With this arrangement, however, the receiver must be tuned to fairly strong carriers in order to secure satisfactory reproduction, for medium-strength signals will undergo a certain amount of amplitude distortion.

A circuit incorporating this biased demodulator type of inter-carrier noise suppression is shown in Fig. 17. In the *D* (distance) position of switch *SW*, the two diode sections of the 6B7 tube together serve for demodulation and A.V.C. purposes, and there is no delay voltage in the diode circuit. Under this condition all signals, including noise, are detected in the usual manner.

When the switch is placed in the *L* (local) position, the voltage drop across cathode bias resistor  $R_C$  acts in series with the diode circuit. This delay voltage must be exceeded by the signal before A.V.C. action and demodulation can take place, and the carrier signal must be reasonably larger than this delay voltage before complete demodulation can occur. If  $R_C$  (which acts also as C bias resistor for the amplifier section of the 6B7 tube) is replaced with a potentiometer and the movable arm is connected to the *L* terminal of the switch, the amount of delay voltage in the rectifier circuit can be varied.

**A.F. Amplifier Blocking Circuit.** In

Fig. 18 is a well-known example of a circuit which employs quiet automatic volume control (Q.A.V.C.) action to block or cut off the input signal to the A.F. amplifier whenever the carrier level becomes so low that noise will be objectionable. As you can see, a special Q.A.V.C. tube is inserted between the detector and the first A.F. stage, with a switch in the cathode lead to make this tube inoperative when ordinary receiver action is desired.

Observe that the plate-cathode resistance of the Q.A.V.C. tube acts in series with a 1-megohm resistor (part

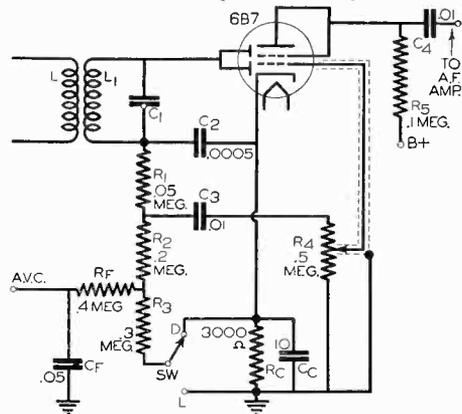


FIG. 17. Biased demodulator type of inter-carrier noise suppression circuit, as used in the Silvertone Model 7149 receiver. This circuit reduces the noise ordinarily heard in an A.V.C.-controlled receiver when tuning between strong stations.

72) between *B+* and ground *G*, with the screen grid of the type 77 A.F. amplifier tube connected to point *X* as indicated. We thus have a voltage divider, with the screen grid voltage of the type 77 tube being determined by the plate-to-cathode resistance of the Q.A.V.C. tube. The lower this resistance, the lower will be the positive voltage developed across the Q.A.V.C. tube and applied to the screen grid of the A.F. tube, and the lower will be the gain of this type 77 tube.

Now let us see how weak signals lower this plate-cathode resistance and

thereby block the first A.F. tube so as to suppress inter-carrier noises. Notice that the control grid of the Q.A.V.C. tube connects to the diode second detector load through a 4-megohm resistor. Under a no-carrier-signal condition, the only D.C. voltage across diode load resistor 55 will be a very low D.C. voltage produced by noise signals; condenser 52 and resistor 53 remove all A.C. component which might otherwise reach the Q.A.V.C. tube. The Q.A.V.C. tube thus gets a small negative C bias, under which condition *its plate current is high and its plate-to-cathode resistance quite low*. This makes the screen

this tube and thereby increasing the screen grid voltage on the 77 tube enough to permit amplification of the desired signals in the usual manner.

In the circuit just described, blocking action was secured by varying the screen grid voltage of the first A.F. amplifier tube; the same results can be secured by applying the blocking voltage to the control grid of this tube, as is done in the inter-carrier noise suppression circuit in Fig. 19. No extra tube is required here, for the triode section of double-diode-triode tube VT<sub>1</sub> is used to produce the required blocking voltage. A desired carrier signal

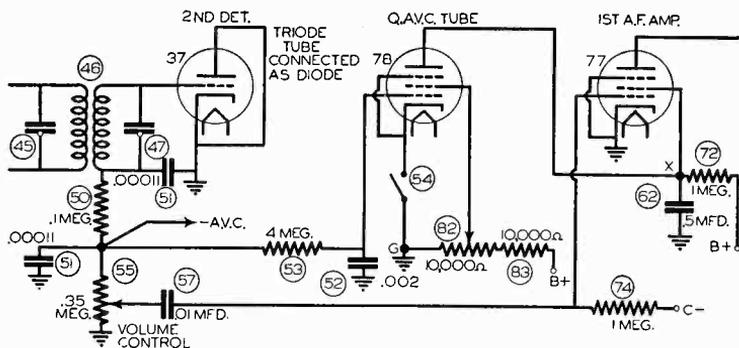


FIG. 18. This quiet A.V.C. circuit, used in the Philco Model 16-122 receiver, acts upon the screen grid of the first A.F. tube, blocking the amplifier whenever the receiver is tuned between strong stations. It thus serves to suppress inter-carrier noise.

grid voltage of the type 77 tube low enough to prevent amplification of any noise signals which are fed to the control grid of this tube through D.C. blocking condenser 57. Potentiometer 82 controls the screen grid voltage on the Q.A.V.C. tube and thus determines the amount of plate current which will flow during this low negative C bias condition. This potentiometer can be adjusted so that signals below any desired level will be blocked.

When a desired strong carrier signal reaches the detector, the resulting D.C. voltage across detector load resistor 55 increases the negative bias on the control grid of the Q.A.V.C. tube, increasing the plate-to-cathode resistance of

is rectified by the combined diode sections of the tube in the usual manner, developing a D.C. voltage across detector load resistor  $R_2$ . The audio voltage which also exists across this resistor is fed through D.C. blocking condenser  $C_5$  to the control grid of the pentode first A.F. amplifier tube  $VT_2$ , while the D.C. voltage is fed through resistor  $R_3$  to the control grid of the triode section of tube  $VT_1$ . Plate current for this triode section flows through resistor  $R_4$ , producing across it a voltage drop which acts with the drop across voltage divider resistor  $R_9$  to provide C bias for  $VT_2$ . You can readily see that when the current through  $R_4$  is small, the C bias on the



of the whisper control tube through  $C_{19}$ . When no carrier signal is present, there will be no rectified current flow, and point  $X$  and the control grid of the whisper control tube will therefore be at cathode potential; as a result a large plate current flows through the triode section of this tube, then through  $SW$ ,  $R_{13}$ ,  $R_{16}$ ,  $R_{17}$ ,  $R_{18}$  up to  $P$  and back to the cathode. This develops a large negative voltage across  $R_{13}$  and since this voltage is applied to the A.V.C.-controlled tubes through the usual A.V.C. feed system, it blocks these tubes.

When weak intelligence and noise signals are present in the output of the I.F. amplifier, such as when tuning between powerful local stations, these pass through  $C_{19}$  to the upper diode section of the whisper control tube, where they undergo rectification and produce D.C. voltage drops across resistors  $R_4$ ,  $R_{20}$  and  $R_7$ . The direction of electron flow is such that point  $X$  on  $R_7$  will be negative with respect to the cathode of the tube; since the control grid of the triode section of the whisper control tube is also connected to this point, it gets a negative C bias. Plate current flow is thereby reduced, and the voltage drop produced by this plate current is correspondingly reduced. This reduces the negative bias applied to the R.F. and I.F. sections through the A.V.C. system, but the negative bias on the second detector diode sections is still high enough to prevent signal rectification.

When a sufficiently strong carrier signal is tuned in, the diode sections of the whisper control tube will pass enough current to block plate current flow through the triode section (by increasing the voltage drop across  $R_7$ ), and practically no current will be drawn through  $R_{13}$  by the whisper control tube. This com-

pletely releases the delay action of the whisper control tube upon the A.V.C. circuit, restoring normal high sensitivity of the receiver. A.V.C. action remains normal with further increases in carrier level. The setting of potentiometer  $R_{18}$  determines the minimum carrier level at which the delay action is released and normal A.V.C. action begins.

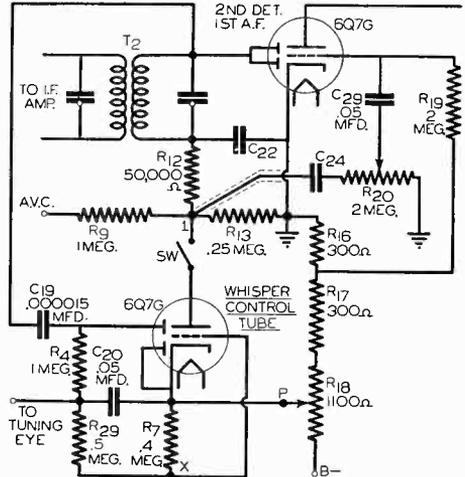


FIG. 20. This inter-carrier noise suppression circuit used in the Silvertone Model 4666 receiver where it is called a "whisper tuning control," serves to block the R.F. and I.F. amplifiers through the A.V.C. system, and at the same time provides delayed A.V.C. action.

The circuit in Fig. 20 thus provides a combination of inter-carrier noise suppression and delayed A.V.C. action. The listener is able to tune from one station to another even at low sound levels or whisper levels without hearing inter-station noise, and thus this circuit has been called a whisper control circuit. Opening switch  $SW$  removes the delay action, allowing reception of weak signals along with noise. Observe that the opening of this switch does not stop the functioning of the diode section of the whisper control tube; this is essential because the diode also provides a D.C. control grid voltage for the electron ray indicator tube or tuning eye.

# Lesson Questions

Be sure to number your Answer Sheet 24FR-1.

Place your Student Number on every Answer Sheet.

Never hold up one set of lesson answers until you have another ready to send in. Send each lesson in by itself before you start on the next lesson. In this way we will be able to work together much more closely, you'll get more out of your Course, and you will receive the best possible lesson service.

1. What three types of circuits are used for reducing the annoying effects of atmospheric noises which enter a radio receiver?
2. At low loudness levels, is the human ear more sensitive to medium-frequency sounds (500 cycles to 5,000 cycles) than to low-frequency sounds (below 500 cycles)?
3. Why does a simple condenser tone control give an apparent boost in bass signals?
4. How can a conventional tone control provide treble control?
5. What type of circuit is present when the volume control has a fixed tap which is connected to ground through a condenser and a resistor?  
*But make for a capacitor*
6. Why is degeneration intentionally introduced in the General Electric Model F-77 receiver circuit shown in Fig. 9.  
*70 control*
7. Upon what fact does the operation of the most widely used automatic volume expansion system depend?  
*Variable biasing with a feedback loop applied*
8. Why should a receiver be tuned to a local station when a minimum of noise is desired?  
*to get a good signal*
9. Why is inter-carrier noise heard in receivers having A.V.C.?  
*the carrier frequency is not suppressed*
10. Name the three conventional types of inter-carrier noise suppression circuits.

## HONESTY

That old proverb, "*Honesty is the best policy,*" is just as true today as ever. Any firm which depends upon repeat business cannot exist for long without following this policy; any man who deals with other people cannot afford to disregard it.

Strict observance of the law will keep a man out of jail, but that does not necessarily make him an honest man. Honesty goes far beyond the law; it involves a careful regard for the rights of others, a truthfulness and sincerity in dealing with others, and a fairness and trustworthiness in matters involving property or business.

It is not enough to act so others will think you are honest; you yourself must know that you are playing the game fair and square if you are to enjoy that real satisfaction associated with absolute honesty.

Be honest, and your reputation will take care of itself. Let your spoken word, your slightest implied action be as good as your signature on a legal contract, and you will enjoy those things which no amount of money can buy—happiness, success, and the respect of your fellow men.

J. E. SMITH

**LIGHT-SENSITIVE CELLS  
FOR CONTROL CIRCUITS**

25FR-3

**NATIONAL RADIO INSTITUTE**

ESTABLISHED 1914

**WASHINGTON, D. C.**



*Review 2-25*

# Study Schedule No. 25

For each study step, read the assigned pages first at your usual speed. Reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind, then answer the Lesson Questions for that step. Study each other step in this same way.

1. The Electric Eye ..... Pages 1-7

Here you learn about the equipment which makes possible almost magical feats in electronics. The apparatus used, the fundamentals of light and illumination, and the types of photoelectric cells are all described. Answer Lesson Questions 1, 2 and 3.

2. Photoemissive Cells ..... Pages 8-14

There are two types of these cells—the vacuum and the gas-filled types. The small amount of gas in the latter type changes the characteristics greatly. Answer Lesson Questions 4 and 5.

3. Photoconductive Cells ..... Pages 14-18

*Reread 3 times*

The photoconductive cell operates as a variable resistor, the resistance changing with light changes. Selenium is used so commonly in these cells that they are called selenium cells, although other cell types, having different characteristics, also use selenium.

4. Photovoltaic Cells ..... Pages 18-24

This cell generates its own voltage, so does not require a power source for its operation. This is very important and leads to many applications of these cells. They are used particularly in exposure meters and light meters. Answer Lesson Questions 6, 7 and 8.

5. Electron-Multiplier Cells ..... Pages 24-28

The extremely high sensitivity of these cells makes them very useful where small light values are encountered. They do require a high-voltage power supply, but this is outweighed in many applications. Answer Lesson Questions 9 and 10.

6. Mail your Answers for this Lesson to N.R.I. for Grading.

7. Photoelectric Control Circuits with Relays .. Reference Text 25X

Here's a fascinating subject which is becoming of increasingly greater importance as the industrial world turns to electronics for its automatic controls. Read it once, to familiarize yourself with the contents of the book. You will then know where to find this information when you actually need it. If you're doing electronic work now or are deeply interested in it, however, you'll want to study this book just as thoroughly as a regular lesson.

8. Start Studying the Next Lesson.

# LIGHT-SENSITIVE CELLS FOR CONTROL CIRCUITS

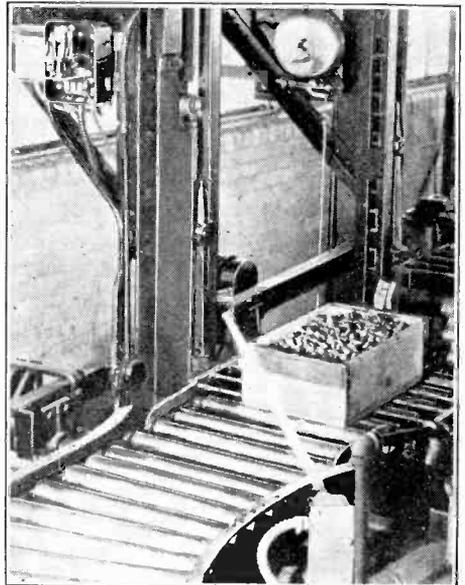
## The Electric Eye

**U**NDoubtedly you have read about the magic *electric eye*, a seemingly mysterious device which causes doors to open as you walk toward them, turns on roadside signs in the country as an automobile approaches, sounds alarms when anyone walks over a forbidden area, prevents dangerous machines such as punch presses from operating while the hands of the worker are in the way, and does thousands of other equally amazing and practical feats. In this lesson you will learn about the many different types of electric eyes, which are called *photoelectric* or *light-sensitive cells* by technical men. You will learn how each type of cell is constructed, how it functions, and how it can be made to replace man's eyes.

► Thousands of light-sensitive cells are in use today in every corner of the world, responding to beams of light which may be perfectly invisible to the human eye, detecting every change in illumination from the sun or from other sources of light; these cells change their *electrical characteristics* with variations in the light which they "see". These light-sensitive cells start and stop heavy machinery, count objects moving past at a mile-a-minute speed, guard against fire, smoke, water, and burglars, and even "read" books for blind persons. Cigars, beans, eggs, fruit, and other products are being graded as to color or shade by light-sensitive cells, faster and more

accurately than by the human eye.

Although the field of photoelectricity is not new, its development into commercial practicality has taken place within the last few years. Scientists have known for more than one hundred years that certain electrical effects could be obtained by exposing chemical elements and com-



Courtesy General Electric Co.

A typical industrial use of photoelectric equipment. When the light beam is interrupted by a box, a number of actions are possible. The unit may actuate a counter; may sound an alarm; or may operate automatic machinery such as an elevator. It may also separate cases according to size or shape by operating an assorting apparatus. Thus, the same equipment may be used for many purposes, depending on what it is used to control.

pounds to light, but the lack of suitable apparatus to make use of this electrical effect, and the poor sensitivity of the light-sensitive cells then available prevented the commercial use of this photoelectric action.

Recent developments in the field of television and electronics resulted in a great demand for photoelectric devices, and today the electric eye is looked upon as a dependable and invaluable device for industrial and commercial applications. As men in industry and business realize the value of electronic control, more of these devices will find their way into everyday use. Only the imagination of man

stallation, so you can understand better the important part played by the light-sensitive devices which you will study. The six important basic parts of a complete photoelectric control installation are:

**1. The Light Source.** The light which is directed upon the electric eye may be the natural light from the sun or artificial light from an incandescent lamp, gas flame, arc light, etc.

**2. Light Beam Apparatus.** On some photoelectric installations, it is necessary to concentrate the light into a narrow beam in order to make it travel over a definite path before

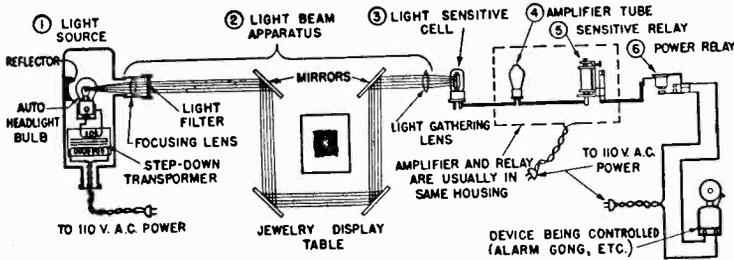


FIG. 1. Simplified diagram of a complete photoelectric installation such as might be used to protect valuable jewelry on a display table. The infra-red light filter on the light source makes the light beam practically invisible.

stands in the way of accomplishing deeds which are best called *magic*.

► As a sideline for the radio man, the field of photoelectric control offers great opportunities, for, in this branch of electronics, there are many simple, basic applications requiring only standard equipment, a knowledge of the fundamentals of photoelectricity and radio circuits, plus mechanical ingenuity and common sense.

## A COMPLETE PHOTOELECTRIC INSTALLATION

Before taking up the different types of light-sensitive cells, we will describe briefly a complete photoelectric in-

stallation, so you can understand better the important part played by the light-sensitive devices which you will study. The six important basic parts of a complete photoelectric control installation are:

reaching the light-sensitive cell; lenses and curved mirrors are used at the light source to accomplish this. Again, it may be necessary to change the direction of the beam of light with a mirror, or to make the light beam invisible to the human eye by using filters which absorb the visible light rays. Where insufficient light reaches the light-sensitive cell, it may be necessary to use a collecting lens which gathers light and concentrates it upon the relatively small area of the light-sensitive cell.

**3. The Photoelectric Cell.** The electric eye or light-sensitive cell changes its electrical characteristics in

response to changes in illumination.

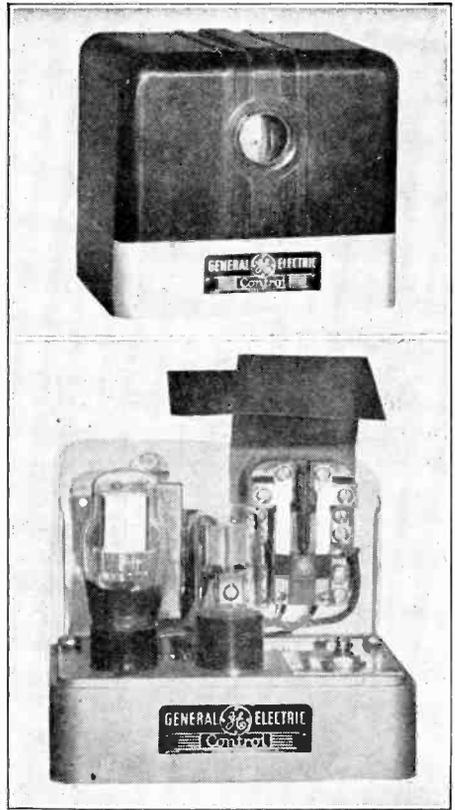
#### 4. The Photoelectric Amplifier.

With certain types of light-sensitive cells, it is necessary to build up the strength of the variations in current or voltage from the light-sensitive cell by means of a vacuum tube amplifier, which may contain one or more ordinary radio amplifier tubes or gaseous tubes.

**5. Sensitive Relays.** The photoelectric cell or amplifier output is used to control some circuit which may be power operated, so a relay is used to close (or open) the power circuits. When the relay is connected directly to the output of the light-sensitive cell, a super-sensitive relay is needed; when connected into the plate circuit of the amplifier tube, an ordinary sensitive relay is satisfactory. Relays which operate on currents of less than 250 microamperes are classed as *super-sensitive*; those which require from .5 to 100 milliamperes (at low power) are classed as *sensitive* relays.

**6. Heavy-Duty Relays.** An additional relay, used after the sensitive relay, is necessary in installations where the preceding relay is not capable of handling the current required by the device being controlled. In some very large installations two or even more power relays, one operating the other, are required. Relays which require more than 100 milliamperes are considered the *heavy-duty* type.

► The simplified diagram in Fig. 1 gives you the relations between the various parts in a typical photoelectric installation (the alarm gong sounds if the light beam is intercepted at any point along its path). Fig. 2 shows a typical commercial photoelectric unit having several of the basic parts mounted in one housing.



Courtesy General Electric Co.

FIG. 2. A typical commercial photoelectric assembly. The upper picture is one of the housing, while the lower gives a view of the interior, showing an amplifier tube, the photocell, and a relay. This unit may be used in many ways, requiring only a light source to form a complete installation.

By properly choosing circuits and relays, you can make a change in cell illumination produce any desired control operation, choose the degree of light intensity at which the relays will operate, and speed up or slow down the action of your controls as much as you desire. The only actual limitations to a photoelectric control system are the sensitivity of the light-sensitive cell with its associated apparatus and the ingenuity of the control engineer.

► An example of how this photoelec-

tric equipment operates is the installation in which a photoelectric eye is used to open the doors of a garage when a car enters the driveway. The light source can be mounted on a post on one side of the driveway; this source throws a beam of light across the driveway at such a height that the beam will be intercepted by a car coming in or going out. The beam of light is directed on a light-sensitive cell mounted on the opposite side of the driveway. The apparatus is so connected that nothing happens while the beam of light illuminates the light-sensitive cell. When an automobile approaches, the light beam is interrupted; the light-sensitive cell detects this immediately and causes the value of the current in the plate circuit of the amplifier tube to change. This operates the sensitive relay; its contacts close and send current through the heavy-duty or power relay. When the contacts of the power relay close, current flows through the electric motor which operates the door-opening mechanism. All this happens so quickly that the garage doors are completely open by the time the car reaches them.

### WHAT IS LIGHT?

The importance of *light* in any

photoelectric installation should be quite obvious from what has been said up to this time. Let us, therefore, start your study of light-sensitive cells by first learning just what light is and how it is measured.

The greatest source of light is the sun; it sends out waves which are identical with those which we use in radio communication, except that they are a great deal shorter in wavelength. Light waves which can be seen by the human eye vary in length from 40- to 70-millionths of a centimeter. The wavelength of light can, for convenience, be expressed in millimicrons, units of length equal to one-thousandth of one-millionth of a meter or one ten-millionth of a centimeter. The human eye, therefore, responds to a light between 400 and 700 millimicrons. Radio waves, which range from .01 to 25,000 meters in length, are more than one million times as long as visible light waves.

The wavelength determines the color of the light; a combination of wavelength (colors) produces the sensation of white light. The human eye does not respond to all colors equally; it is most sensitive to yellow and green. Study Fig. 3 carefully, noting how the human eye responds to the different colors in the visible spectrum.

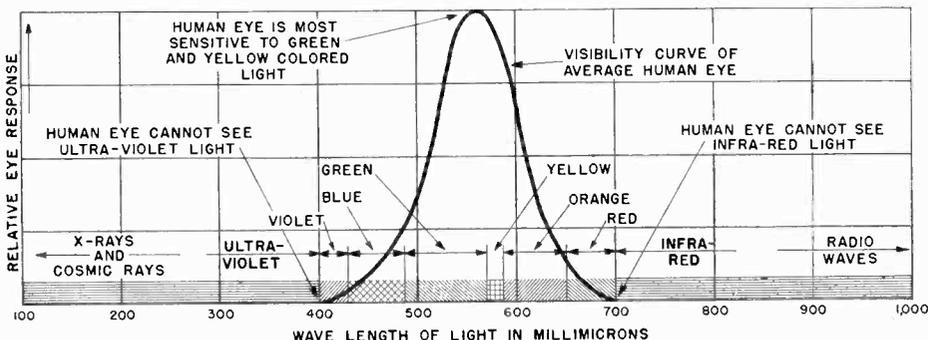


FIG. 3. The relative sensitivity of the average human eye to light of various colors (various wavelengths) is given by this curve.

The electric eye, in addition to "seeing" those frequencies of light which can be detected by the human eye, will respond also to ultra-violet light and infra-red light, both of which are invisible to the human eye. It is this characteristic of the electric eye which makes it possible to use invisible light beams to control machinery or to operate burglar alarms.

Light-sensitive cells respond to various types of artificial light as well as to natural light. The ordinary incandescent electric lamp is the most common artificial source of light for photoelectric equipment. Its filament, a very fine wire of high resistance, is heated by a current of electricity until it becomes incandescent and gives off light. Electric lamps designed for automobile headlights are ideal for photoelectric work because the source of light is concentrated into a very small space, approximating a point source of light. The smaller the source of light, the easier it is to focus that light into a beam.

Other artificial sources of light include natural gas lights, coal gas lights, the carbon arc lamp, the mercury vapor lamp, gaseous conduction tubes (better known as neon tubes), and fluorescent lamps.

**How Light Is Measured.** As you know, the wax or tallow candle was one of the first artificial sources of light. When new light sources began to take the place of the candle, it was natural that their power should be expressed in terms of the old and familiar candle. Eventually a candle made according to certain specifications and burned under certain conditions was selected as the unit of light intensity, and the light given off by this candle was said to have an intensity of *one candlepower*. If an electric bulb was found to be 40 times as

strong as this candle, it was said to have a candlepower rating of 40. The average electric lamp used in the home has an intensity of about one candlepower per watt of power used.

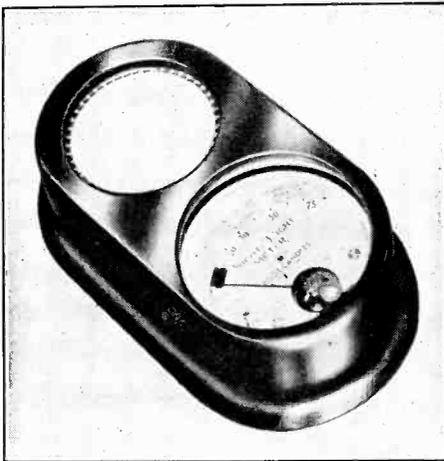
You may have noticed that the strength of the light on a certain object, such as a book, decreases very rapidly as the book is moved away from the source of light. Actually the intensity of the illumination produced by a source of light varies *inversely as the square of the distance* from the source of light. In other words, if the illumination at a point two feet from the source is a certain value, the illumination at a point four feet from the source (twice as far away) will be one-fourth of that at the first point.

The practical unit of illumination is the *foot-candle*. This is the *intensity of the light on a surface* which is directly facing, and one foot away from, a light source which has an intensity of one candlepower. For example, the correct illumination for general reading purposes is about 15 foot-candles; this means that the illumination on the printed page should be equivalent to that which would be produced by 15 standard candles located one foot above the page.

Another unit of illumination, the *lumen*, represents the *amount* of light (or light flux) falling on a surface one square foot in area, every point of which is one foot from a point source of light having a strength of one candlepower. Practically, however, to determine the amount of light (number of lumens) falling on a uniformly-illuminated surface of limited area, you multiply the area of the surface (in square feet) by the intensity of the illumination at the surface (in foot-candles). The answer will be in lumens. To determine the

number of lumens of light emitted by a lamp, multiply the candlepower rating of the lamp by the number 12.6.

The foot-candle is a measure of the *intensity* of light at a *certain point*, such as at the face of a light-sensitive cell; the lumen is a measure of the *amount* of light falling on a *given area*, such as the sensitive surface of a light-sensitive cell. *Light intensity at the cell (in foot-candles) multiplied by effective area of the cell (in square feet) gives the number of lumens of light on the cell.*



Courtesy Weston Electrical Instrument Corp.

This type of light meter is used to measure the illumination in a home or an office. If the amount of light is too low, corrective measures can be taken and the results checked by the light meter.

*Brightness* is another factor which must be considered by the photoelectric engineer, as he will often direct a light on a certain surface and use an electric eye to pick up the light reflected from that surface. Brightness can be expressed in *candles per unit area*; this means that brightness is a measure of the ability of an illuminated surface to act as a source of light. For example, a surface has a brightness of 10 candles per square foot when each square foot of this

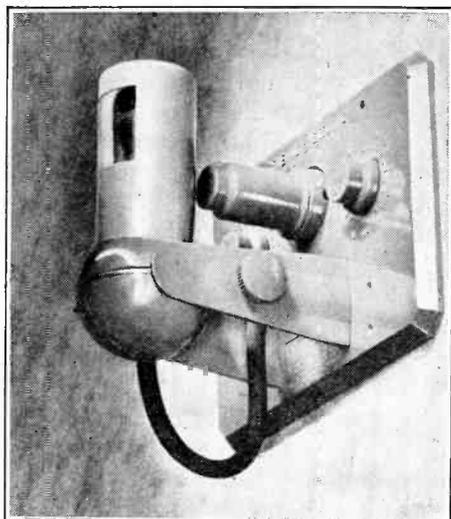
surface reflects as much light in a given direction as would 10 candles.

**Light Sources.** In most photoelectric applications, it is very desirable to have a definite beam of light directed on the electric eye. Naturally, it is desirable to use as small a light source as possible, in order to economize on power and make a compact unit. Most commercial light sources for photoelectric work use the small but powerful 32-candlepower automobile headlight bulb (the required low voltage is obtained from a step-down transformer which is connected to a 110-volt a.c. line), concentrating the light into a beam with a lens. Since only that light which falls on the lens is useful in producing the beam, reflectors are generally used back of the bulb to reflect light to the lens. Some light is absorbed at each reflection and at each passage through a lens, so it is usually necessary to make adjustments of the light source and the relay apparatus until satisfactory operation is secured.

## TYPES OF CELLS

Although photoelectric or light-sensitive cells of various types are being manufactured today by many different firms, all of these can be divided into four basic classes. These four classes of light-sensitive cells are:

**1. Photoemissive Cells.** In these cells, electrons are emitted from the cathode of the cell by the action of light, and are collected by an anode which is at a positive potential. Photoemissive cells are better known as *photocells*; some technicians prefer to call them phototubes, since they are always built into glass envelopes as are glass type radio tubes. In many technical books and articles you will find all types of light-sensitive cells referred to as photocells, but you can



Courtesy General Electric Co.

A wall-mounted photocell assembly. This type is particularly designed to turn on lights in a room when the illumination falls to an unsatisfactory level, and to turn off lights when the level rises above the preset value. The two controls are used to set these upper and lower limits.

generally determine which type of cell is meant from the nature of the discussion. Remember that when we speak of photocells in this book we are referring specifically to *photoemissive cells*.

**2. Photoconductive Cells.** In these cells the resistance (or conductivity) of a material changes under the action of light. Selenium is the resistance material most commonly used in these cells. You will find that photoconductive cells which use selenium are often referred to as *selenium cells*.

**3. Photovoltaic Cells.** These cells develop their own voltage under the action of light. They are referred to often as photoelectric cells, but photovoltaic cells or self-generating cells are

the terms which will be used in this Course. The term photoelectric cell, as used in this Course, will refer to light-sensitive cells in general, regardless of their type.

#### 4. Electron - Multiplier Cells.

These are really a kind of photoemissive cell, but are so different from the usual photocell that we should treat them as a separate type. An electron-multiplier cell has one cathode and a number of anodes, each of which is maintained at a different positive potential. Through a series of actions, which we will describe later in this lesson, this cell is able to produce many more electrons than are driven out of the cathode by photoelectric action. In this Course, we shall refer to electron-multiplier cells simply as *multiplier cells*.

► Each light-sensitive cell has its own characteristics, and naturally transfers these characteristics to the associated photoelectric apparatus. Some cells are more dependable than others; some require external sources of voltage which may fail; some cells have a comparatively short life—all these factors must be taken into consideration where failure of the photoelectric control would in any way endanger human life or damage valuable equipment. We are going to devote the remainder of this lesson to detailed descriptions of each type of cell, telling how they are constructed, how they operate, and basically, how they are used. When you have finished, you will have all the basic information you need to understand the operation of modern photoelectric equipment.

# Photoemissive Cells

The photoemissive cell can be compared to a diode or two-element tube, for both contain two electrodes mounted in a glass envelope. In the radio tube, the electrons which make up the current through the tube are produced by heating the cathode (filament); but, in the photoemissive cell, the light rays activate the sensitized surface of the cathode and cause electrons to be emitted. The other electrode, called the anode, attracts the emitted electrons, as it is positively charged with respect to the cathode. *The emission varies with the amount of light; the more light there is shining on the cell, the greater is the emission.* Current will flow through the tube; the amount depending on the emission and the applied voltage. The tube acts like a variable resistance, the resistance value following the light variations.

The construction of a typical photocell is shown in Fig. 4. Here the cathode is a semicircular cylinder of metal (usually oxidized silver) supported inside the glass envelope by stiff wire leads made of nickel. The anode is simply a nickel rod or wire placed in the axis of this cathode cylinder. The anode is untreated, but on that surface of the cathode which faces the anode is a thin film made up of caesium oxide, sodium, potassium, or lithium. For practical purposes, just consider the cathode film as simply a layer of light-sensitive chemical compound.

Soda-lime glass is used as the envelope for most of the inexpensive photocells made today, since it is cheap and easy to form into the desired shape. Higher-priced cells use either pyrex glass or fused quartz, as

these have lower light losses and allow more ultra-violet rays to pass. Lead glass is never used for photocells. It combines chemically with the materials used on the cathode, discoloring the envelope, and lead glass is a poor transmitter of light (it absorbs a high percentage of the rays).

► The modern photocell is made by automatic machinery in much the same way as radio tubes are made. The proper chemical is sprayed on the cathode, the two electrodes are mounted on the glass stem, and then

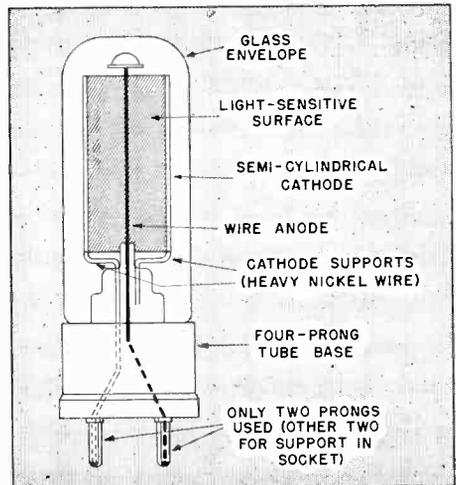


FIG. 4. The connections and parts of a photoemissive cell.

this stem is sealed to the glass envelope. The gas is evacuated by vacuum pumps, and the tube is given a special treatment in a high-frequency electrical furnace to complete the processing of the coating on the cathode.

Notice that the anode in a photocell is the *smallest* electrode; this is necessary so that the anode will not cast too large a shadow on the cathode and reduce the efficiency of the cell. In

regular radio tubes the opposite holds true; here the plate or anode is the largest electrode.

► Two types of photocells are in common use, differing only in that one contains a gas while the other has a "hard" or high vacuum. Gas cells give an increased current output for a given amount of light. The sensitivity of a photoemissive cell is controlled by the cathode materials; the introduction of a gas reduces the effect of the space charge and permits greater sensitivity to be attained.

► Since the current passed by the cells is too small to operate even a super-sensitive relay, photoemissive cells are always used with one or more amplifier tubes. A typical circuit which can be used either with gas or vacuum type photocells is given in Fig. 5. The photocell is connected so that the voltages of the *B* and *C* batteries will be applied to it through *R*. (Its current will not operate the relay.) An ordinary radio tube such as the 1H4G (or any triode, provided the proper voltages are applied) serves as amplifier tube; the relay is a sensitive type, with its contacts connected into the power supply leads of the device being controlled.

In this basic circuit the grid of the amplifier tube has a negative bias, and therefore draws no current. The relay resistance (usually somewhere between 1000 and 10,000 ohms) is comparatively low with respect to the circuit resistance, so we can consider the photocell and resistor *R* to be in series across points *A* and *B* of the batteries. When the photocell is dark, (no light on it) there is no emission, so practically no current flows through it and *R*. Hence, the voltage across *R* is very low, and the grid of the tube can be considered to have almost the same negative potential as point *A*.

With a highly negative grid, little or no electron current passes from cathode to plate in the amplifier tube, and the relay armature is not attracted to the relay core.

When light falls on the photocell, its emission rises and more current can flow through *R* and the photocell, so the voltage across *R* becomes greater. Electron flow is from *A* through *R*,

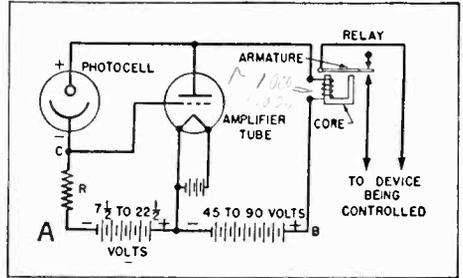


FIG. 5. A basic circuit which can be used with the photoemissive cell.

through the photocell and through the relay to *B*; point *C* is therefore more positive than point *A*. The grid thus becomes more positive than point *A* when light falls on the photocell. With a lowered negative bias on the grid, plate current flows through the relay, causing the armature to be attracted to the relay core. In a practical circuit the grid bias would be controlled by a potentiometer connected across the grid bias battery, which would allow the circuit to be adjusted for maximum response to the desired changes in light.

It is only necessary to interchange the positions of *R* and the photocell in this circuit in order to make the relay operate when the cell is darkened (as when some object interrupts a light beam directed on the cell). In this reversed circuit, light on the cell produces a high current, but the voltage across the low cell resistance is small and is "cancelled" by the high grid

bias. Darkening the cell raises its resistance so much that most of the voltage drop across the combination of  $R$  and the cell is developed across the cell. This makes the anode of the cell more positive than it is when the cell is illuminated; thus, the negative bias of the grid is reduced, and the increased flow of plate current closes the relay.

### VACUUM TYPE PHOTOCELLS

In the vacuum photocell the total tube current is made up of electrons that are emitted by the cathode (see Fig. 6). This cell will respond to light variations of almost any frequency, which means that it is ideal for use in the fastest of counting jobs. The capacity between the anode and cathode is the only important limitation on the highest light variation frequency to which this cell will respond.

Just as curves are used to illustrate the characteristics of the vacuum tubes which you have studied, so can curves be used to illustrate photocell characteristics. The stronger the light the greater will be the current flow, but does this relation hold at all times? Is the current affected by the voltage applied to the cell? These are important questions which can be answered easily by making some simple tests.

A standard incandescent electric lamp is used in securing characteristic curves for all types of light-emissive cells, and the voltage applied to it is adjusted to operate the filament at a constant temperature. The illumination on the cell is varied by changing the distance between lamp and cell rather than by varying the light source.

**Current-Illumination Curve.** This important photocell characteristic curve is obtained by applying a fixed

voltage (say 40 volts) to a cell and measuring (with a microammeter) the current passed by the cell at different intensities of illumination. With a vacuum type photocell, the results obtained will give a curve like that in Fig. 7A, which shows that the current output is directly proportional to the amount of light falling on the cell. That is, the cell current doubles when the light flux is doubled.

**Current-Voltage Curve.** If a vacuum type photocell, illuminated by a constant light intensity of about .5 lumen, is connected to a variable source of d.c. voltage, and measurements are made of the current passed

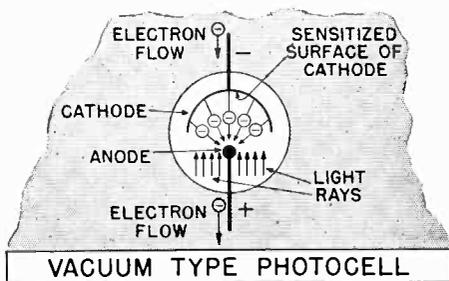


FIG. 6. Here are the details of operation of the vacuum type photocell.

by the cell for each value of voltage, the curve obtained will be like that shown in Fig. 7B. Notice that once the voltage reaches a certain value (above the knee of the curve) the cell current increases very little as the voltage is further increased. At this point, called the *saturation value* of current, practically all of the electrons emitted by the cathode under the action of the fixed light are drawn to the anode. Clearly, then, there is little to be gained by increasing the voltage above this value. A small size vacuum cell can safely withstand up to 500 volts; often such a high value must be applied to the cell because of circuit conditions.

The curve in Fig. 7B can be made to show the relation between the sensitivity of the cell and the cell voltage simply by dividing the current values by the number of lumens of light on the cell. The vertical scale at the right in Fig. 7B is obtained in this way, by dividing each value of current in the scale at the left by .5, which is the number of lumens of light on the cell. You can easily see that a vacuum type cell should be operated at a voltage corresponding to a point *above the knee* of the curve if greatest sensitivity is to be obtained.

**Color Response Curve.** Another very important photocell characteristic is its response to light of different colors (wavelengths). The curve in Fig. 7C gives this information for a typical vacuum type photocell. In general, the color response of a photocell differs considerably from that of the human eye (shown by the dash-dash curve).

These color response curves are important when the job is one of assorting objects by color; filters must frequently be used to correct the response so that the proper selection will occur, or another cell with a more favorable response must be used. Various light-sensitive materials are used on the cathode to get certain desired color characteristics. Quartz must be used for the envelope where a tube is to be highly sensitive to ultra-violet light, since ordinary glass does not transmit ultra-violet rays.

The characteristics shown in Fig. 7C show a peak in the response in the infra-red light region, so this cell is an excellent one to use with invisible light beams in burglar alarms, etc.

### GAS TYPE PHOTOCELLS

The gas type photocell differs from the vacuum type only in that, after

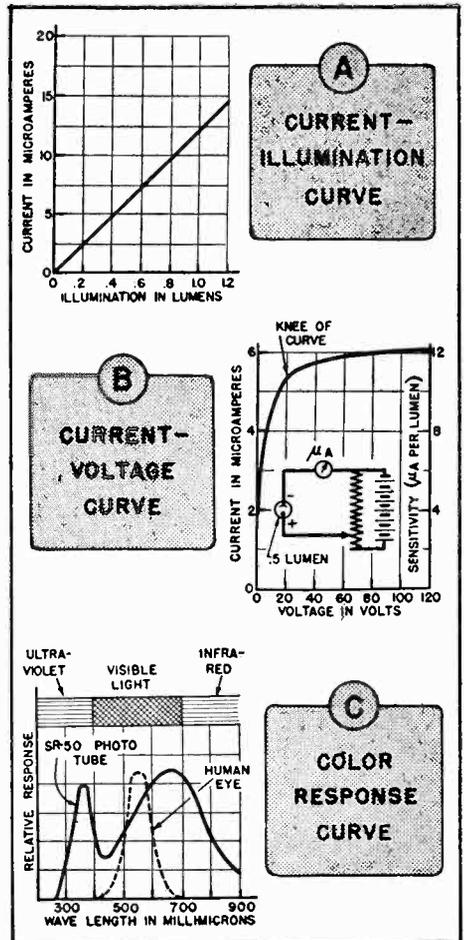


FIG. 7. Typical characteristic curves for vacuum type photoemissive cells. (These curves are for the Westinghouse SR-50 photocell.)

the cell is evacuated, a small quantity of an inert gas (a chemically inactive gas which will not combine with the tube elements) such as argon, helium, or neon is admitted before sealing off the tube. In this gas type cell the electrons emitted by the cathode, traveling at high speed towards the anode, have sufficient force to knock out electrons from gas atoms with which they collide. Thus, they split up the atoms into positive ions and free electrons; this process is known as *ionization*. The electrons resulting

from these collisions are attracted to the anode along with the emitted electrons, and serve to build up the photocell current. The positive ions (atoms from which electrons have been knocked out) move to the space cloud near the cathode, neutralizing the electrons there and allowing more of the electrons from the cathode to find a free path to the anode. If the ionization is made too strong by excessive anode voltage, a glow discharge similar to that in neon sign tubing will be produced. Then the gas ions will bombard the cathode and destroy the tube.

This "ionization by collision" process is shown in Fig. 8. The original tube current can be increased as much as ten times by the ionization of the gas. However, the amount of ionization increases rapidly as the voltage is increased, so the voltage must be kept below a critical value in order to prevent a glow discharge from destroying the tube. With too-high voltages the gas type tube will pass current even when no light is falling on the cathode. The operating voltages vary between 25 and 100 volts for the average gas photoemissive tubes, depending upon their construction. Under no conditions should this voltage be exceeded.

Characteristic curves for a typical gas type photocell are given in Figs.

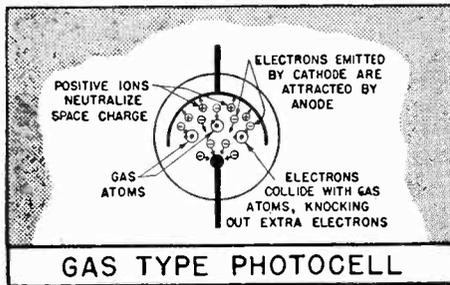


FIG. 8. The gas cell depends upon gas ionization for increased sensitivity.

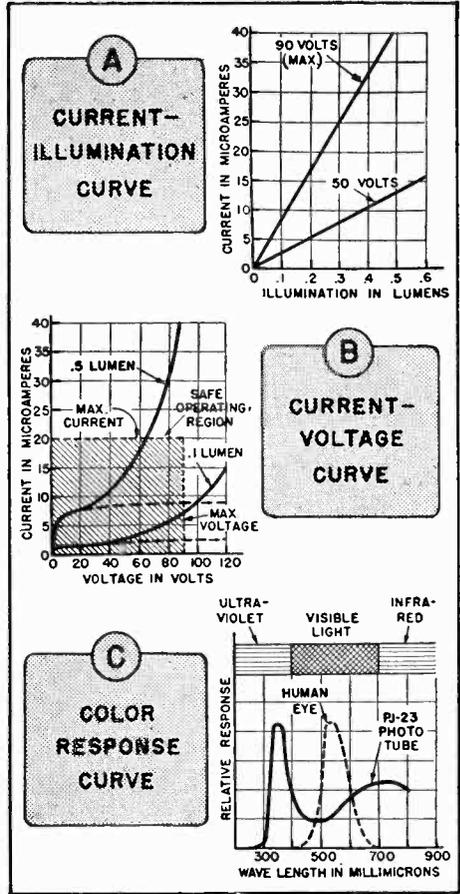


FIG. 9. Compare these gas cell curves with those of the vacuum type in Fig. 7. (These curves are for the General Electric PJ-23, which is typical.)

9A, 9B, and 9C. The current passed by the gas type cell increases practically in direct proportion to the illumination, just as in the vacuum type cell (Fig. 7A).

The curves in Fig. 9B give you a very good idea of the characteristics of a gas type photocell. At voltages below 20 volts this cell behaves almost exactly like a vacuum type cell (Fig. 7B) because, at these low voltages, the emitted electrons do not reach a sufficiently high speed to knock electrons

out of the atoms of gas inside the photocell. At some voltage above about 20 volts (above the knee of the curve) ionization starts, and further increases in voltage give much greater increases in current than are obtained with a vacuum type cell. The dotted lines (Fig. 9B) show how the current would vary at higher voltages if there

the tube to limit current to a safe value in case the illumination is accidentally increased.

**Gas Ratio.** In the gas photocell, the ratio of the current passed at its maximum safe operating voltage, to the current passed just before ionization (and gas amplification) begins, is known as the *gas ratio* of the tube.

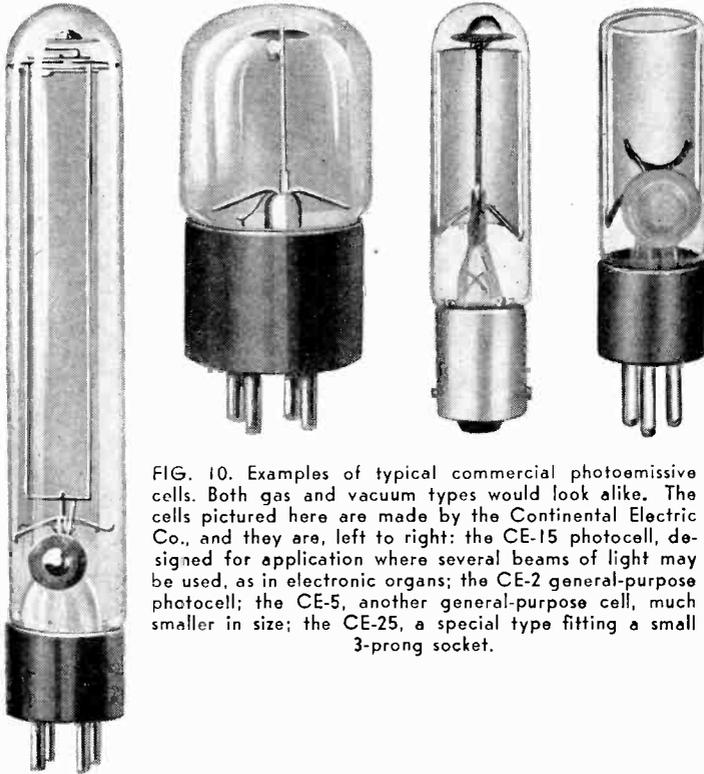


FIG. 10. Examples of typical commercial photoemissive cells. Both gas and vacuum types would look alike. The cells pictured here are made by the Continental Electric Co., and they are, left to right: the CE-15 photocell, designed for application where several beams of light may be used, as in electronic organs; the CE-2 general-purpose photocell; the CE-5, another general-purpose cell, much smaller in size; the CE-25, a special type fitting a small 3-prong socket.

were no gas in the tube.

The voltage and current ratings specified by manufacturers for gas type tubes must be carefully followed if the tube is to be in operation for long periods of time. Voltages slightly higher than rated values shorten tube life considerably. In general, the maximum safe operating voltage of the average gas type photoemissive cell is about 100 volts. It is a good idea to place a resistor in series with

For example, the gas ratio in Fig. 9B is about 7; this value is obtained by dividing the current at 90 volts by the current at 25 volts, the illumination being held constant at a value which limits the current at 90 volts to a safe value. The values of illumination used are generally specified for gas ratios, since the ratio depends somewhat on the illumination (it becomes less as illumination on the cell is decreased).

**Color Sensitivity.** The color

response of a gas type photocell depends upon the kind of glass used for the envelope and upon the nature of the light-sensitive material used on the cathode. The curve in Fig. 9C is therefore not a general response, but is instead the color characteristic associated with the cell used in making this curve and others like it. As you can see, this particular tube has high sensitivity to ultra-violet and infrared, but comparatively low sensitivity to visible light.

### GAS AND VACUUM PHOTO-EMISSIVE CELL RATINGS

Just as radio tubes have certain definite ratings which must not be exceeded, so do photocells have maximum voltage and current values which cannot be exceeded in ordinary practice.

The *maximum anode current* is the maximum value of current which can be safely passed by the tube. For a.c. this is the peak value of current.

The *maximum anode voltage* is the maximum value of voltage which can

safely be applied to the tube. For a.c. this is the peak value of voltage.

The *maximum illumination* is more important in connection with gas type cells than with vacuum type cells. At high values of illumination, the voltage applied to a gas type photocell must be considerably lower than the rated value in order to keep the current down to a safe value (in the safe operating range shown in Fig. 9B).

The *sensitivity* of a photocell is generally expressed as the current passed in microamperes per lumen of light flux. This is usually measured at a light intensity of either .1 or .5 lumen, so that sensitivity ratings of various tubes can be compared. The sensitivity of vacuum type cells varies from about 5 to 35 microamperes per lumen, while for gas cells, rated sensitivity values may be as high as 300 microamperes per lumen.

Fig. 10 shows a number of typical photoemissive cells. Gas and vacuum type photocells look the same, since the gas used is invisible.

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## Photoconductive Cells

The photoconductive cell is essentially a resistance whose ohmic value varies with the light falling upon the cell. The stronger the light falling on the cell, the lower becomes the resistance. There is no "emission" of electrons here; the cell is just a resistance which varies with light. When placed in a circuit with a voltage source, the current flow will depend on the voltage and the resistance value at the moment. When dark, the cell resistance is high but it still will pass some current. Hence, there is no "cut-off" of current; it is just varied from a low value to a higher value by the cell

resistance variation.

The photoconductive effect was noticed first by an engineer named Willoughby Smith, who, while stationed in the Azores Islands in 1871, observed that the selenium resistors he was using changed their resistance when exposed to sunlight. Reporting the discovery to a group of scientists, he announced: "By the aid of the telephone I heard a ray of light fall on a bar of metal!"

Photoconductive cells are made almost exclusively from selenium, although certain other compounds, one of which is thallium oxysulfide, show

appreciable photoconductive effects. Many different forms of selenium cells are on the market today.

► The term "cell" is somewhat misleading when used in connection with photoconductive cells and photoemissive cells, as these do not generate their own voltage and are not therefore to be compared to a battery. The word *cell* is, nevertheless, commonly used among technical men for all classes of light-sensitive devices, and is used as such in this lesson. You will be able to tell what is meant by referring to the circuit diagram, because the symbols for light-sensitive cells are quite different from the symbols for batteries.

### SELENIUM CELLS

A selenium cell consists essentially of two electrodes between which is deposited a thin film of selenium. The electrodes, which can be copper, iron, nickel, aluminum, silver, gold, platinum, metal alloys, lead, graphite, or carbon, are mounted on some insulating block such as quartz, glass, clay

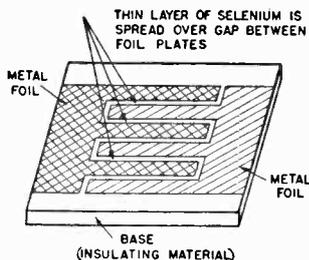


FIG. 11. This sketch shows the basic construction of the selenium cell.

compounds, porcelain, slate, mica, or bakelite.

The construction of a simple selenium cell is pictured in Fig. 11. Two pieces of metal foil are cut out as shown, and cemented to the flat base, after which a thin layer of molten selenium is spread evenly over the foil plates and all gaps between the foil.

Sometimes the selenium in powder form is sprinkled over the plates, then heated and spread out, or selenium is heated to its boiling point and the vapors allowed to condense on the plates. In any event, the cell must be annealed by heating carefully until the selenium changes from its pitch black form to the gray crystalline form which is highly sensitive to light, then the cell is cooled slowly.

► Commercial selenium cells are made in a number of different ways. One type of cell has two wires wound over an insulating slab, with a thin layer of selenium deposited between the wires. Each wire serves as a terminal of the finished cell, the resistance between the two wires being determined by the resistance of the selenium layer. A flat piece of slate is sometimes coated with a film of graphite which is polished with a chamois skin, then divided into two interlacing sections with a sharp tool and selenium deposited over the entire surface.

In other types of selenium cells a thin film of platinum, gold, or silver is fused into the surface of a block of glass or quartz. Then, this thin metal film is divided into two sections by a zigzag or comb-like line made with a precision instrument known as a "dividing engine." There may be as many as one hundred lines or scratches per inch on the cell. When the metal film has been divided into two separate electrodes in this manner, a thin layer of selenium is placed over the entire unit, this selenium serving to bridge the gaps between the two electrodes. This method gives a very small separation between the two plates and a long gap covered with selenium.

► The cell whose characteristics are given in Fig. 12 belongs in this last class; it is highly sensitive to changes

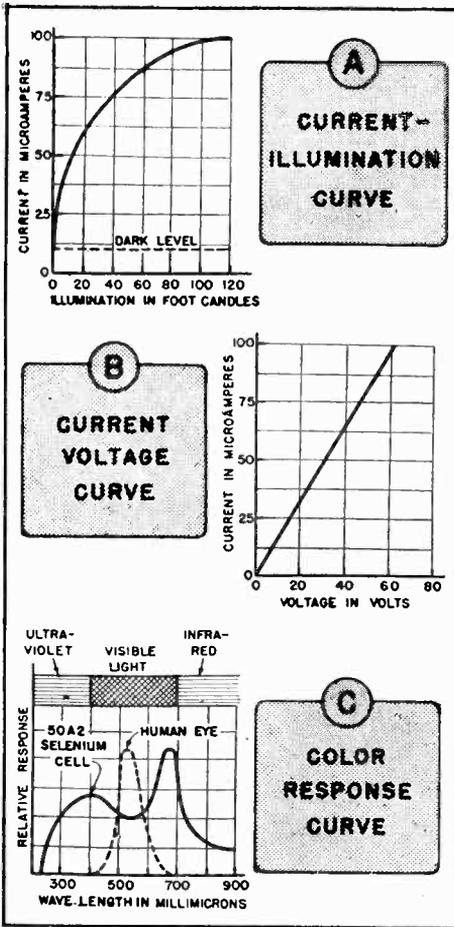


FIG. 12. The Acousto-Lite Laboratories 50A2 selenium cell was used in making these curves.

in illumination but is not recommended for operating voltages higher than 80 volts or light intensities of more than 20 foot-candles. As a precaution against breakdown from excess voltage or excess current, a 1-megohm resistor is usually connected in series with the cell.

By looking first at the Current-Illumination Curve (Fig. 12A), you can see that the current passed increases with illumination, but the effect of the light (the increase in current) is less at the higher values of illumination. Notice that the current

does not drop to zero when the cell is dark (the applied voltage being held constant). Selenium cells have a definite "dark" resistance, so current can drop only to the dark level when light is cut off from the cell. The resistance of this particular cell is about 10 megohms in the dark and about 2 megohms at an illumination of 10 foot-candles, hence it has what is called a dark-to-light resistance ratio of 5.

When illumination is held constant on a selenium cell and the voltage is varied, the current varies exactly as it would for an ordinary fixed resistance. You know that the current passed by a resistor varies directly with the voltage applied; that is why the curve in Fig. 12B is a straight line.

You will find that most selenium cells have a maximum sensitivity to red or infra-red light, with a lower peak (in some instances) in the ultra-violet region (Fig. 12C).

Photoconductive cells are generally used with vacuum tube amplifiers, just as are photoemissive cells, but it is possible to make them with a resistance sufficiently low to permit direct operation of a sensitive relay. A battery must be connected in series with the cell and relay, as photoconductive cells do not generate a voltage. The current passed by the cell under constant illumination depends only on the voltage applied to the cell; the maximum current is determined by the maximum voltage which can be applied without causing breakdown of the selenium in the gap between the plates.

The amplifying circuits required for selenium cells are very similar to those used with photoemissive cells. A fundamental selenium cell circuit is given in Fig. 13. Light falling on the

selenium cell changes its resistance, thereby changing the bias voltage on the grid of the tube. In the circuit shown, increases in light make the grid of the tube less negative (more positive), because the cell resistance decreases, allowing a larger current and hence a larger voltage drop across  $R$ . This voltage is opposite in polarity to the bias, so it reduces the bias and allows the plate current to increase and operate the relay. With this circuit, therefore, the device being controlled by the relay operates whenever a strong light falls on the cell. (Notice—the response of this circuit to light falling on the cell is similar to that of the circuit in Fig. 5, in which a photoemissive cell is used.) If the positions of the selenium cell and the resistor  $R$  are reversed, the relay will be closed when the cell is dark but will open just as soon as sufficient light falls on the cell.

Selenium cells generally are sealed in moisture-proof cases, because they

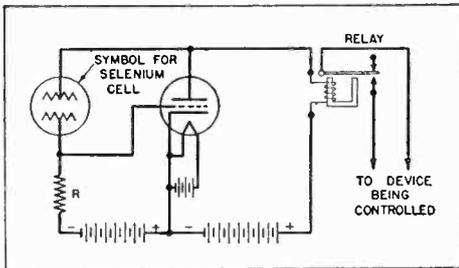


FIG. 13. Notice the symbol used to represent the selenium cell in this basic circuit.

are quite sensitive to changes in humidity. The stability of the cells is in general rather poor; those cells which have a very high sensitivity to changes in light usually have a short life, and their sensitivity changes with use. Mounting in moisture-proof cases also improves cell stability, but with many cells the current passed varies

slightly even with constant illumination. Their power output is limited by the maximum voltage which can be impressed without causing breakdown across the electrodes, and by the amount of heat developed in the cell (excess heat changes the selenium to an insensitive form).

► Selenium cells ordinarily have a time lag and are rather slow to respond to changes in light. It takes several minutes after an increase in

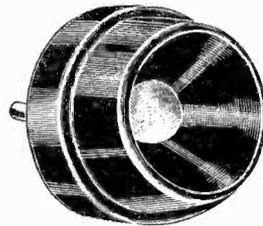


FIG. 14. This H. H. Eby Co. selenium cell is representative of these cells, noted for their small size and for their wide use by experimenters.

light before an ordinary selenium cell will allow maximum current to flow. However, the current reaches 95% of its final value in about .03 second, so the cell is satisfactory for use as a make-break control.

It is possible to change the construction so this cell will respond to light variations (modulations) up to 6000 c.p.s., but the selenium cell is rarely used for audio frequency work.

#### Advantages of Selenium Cells.

Some selenium cells are very sensitive to infra-red light, and so are valuable where control is to be secured with an invisible light beam. Selenium cells can be made to have a low sensitivity to changes in light and a high current-handling ability, or a high sensitivity with low current-handling ability, whichever is desired. Their output is in general considerably greater than that of photoemissive cells. A typical selenium cell is shown in Fig. 14.

## PRECAUTIONS IN USING SELENIUM CELLS

**1. Keep the cell cool.** The heating effect resulting from: 1, too much current through the cell; 2, from exposure to an intense light source; or 3, from using the cell in locations where temperature is excessive, will cause the selenium in the cell to become soft and possibly melt, which makes it unfit for photoelectric use.

**2. Keep the cell dry.** When it is not in use, place it in a dark box containing a few pieces of calcium chloride. The chloride will absorb any moisture which may have collected near the cell.

**3. Avoid high potentials.** Use the

lowest possible voltage which will give the desired results. Choose high-resistance relays or place a limiting resistance in series with the cell to protect it. In general, 4 milliamperes are more than ample for relay work.

Exposure to intense lights for long periods of time should be avoided, for this causes "fatigue" which makes the cell temporarily (and in many cases, permanently) insensitive to light. Cells not in use should be kept in the dark, but should be exposed to light regularly, for short periods of time, to aid in retaining their sensitivity. If the resistance of a cell drops greatly, it can be raised, at least temporarily, by applying pulsating or alternating currents.

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## Photovoltaic Cells

Photovoltaic cells are really small batteries or sources of direct current, since they generate a current which varies with the intensity of the light falling on the cell; more correctly, they transform the radiant energy of light directly into electrical energy. Although the voltage output of these cells is quite low, the current delivered is sufficient to operate an indicating meter or a *super-sensitive* relay without the use of any batteries or auxiliary apparatus.

**Types of Photovoltaic Cells.** There are two general types of photovoltaic cells: the *dry* or electronic type, which is today the most common, and the *wet* or electrolytic type. These differ only in the methods of construction and in general characteristics, for each generates its own voltage.

### DRY PHOTOVOLTAIC CELLS

This type of cell consists essen-

tially of a metal disc, perhaps 1/16-inch thick, on one side of which is a film of light-sensitive material; this basic construction is illustrated at A in Fig. 15. The metal disc forms the positive terminal of the cell, and a thin metal film deposited on part or all of the sensitized surface forms the negative terminal. (This film is so thin it is translucent—that is, it will allow light to pass through it.) The action of light forces electrons to the surface of the sensitized layer, where they are collected by the thin metal film which serves as the negative terminal of the cell. The metal disc must make up for the electrons drawn out of the sensitized layer by the action of light by furnishing electrons to the sensitive layer. This loss of electrons makes the disc become the positive cell terminal. A voltage therefore exists across these terminals when the cell is illuminated, and an electron flow will

be in the direction indicated at *B* in Fig. 15.

Photovoltaic cell circuits are quite simple. The cell is connected directly to the coil terminals of some type of super-sensitive (meter type) relay, as in Fig. 15C. Since the contacts of this relay ordinarily cannot handle the current required by the device to be controlled, they are generally used to control the current to the coil of a sensitive relay. In some cases, the sensitive relay may be used to control a heavy-duty relay.

► While a dry photovoltaic cell can be made in a number of different ways, the following is a typical manufacturing procedure. An iron disc about 2 inches in diameter and 1/16-inch thick is first cleaned thoroughly, then covered on one side with ordinary solder. A thin layer of selenium is deposited over the layer of solder, then the selenium is annealed or heated under



Courtesy Selenium Corp. of America.

A typical photovoltaic cell.

pressure. When the cell has cooled, a thin film of gold, silver, or platinum is deposited on the selenium surface, this film being thin enough to allow light to pass through. This film can be applied as a very thin sheet of the metal (called "gold beater's metal") or can be sprayed on the selenium in molten form from a special spray gun. In some cases the translucent metal film is deposited on the selenium by a process called "cathode sputtering." The cell is completed by making contact to the iron disc and to the translucent metal film with thin metal washers of the same diameter as the iron disc. Naturally the cells must be handled very carefully, because the translucent metal film will rub off very readily. A glass window is customarily set into the cell housing to protect the light-sensitive layer.

Although selenium is used as the light-sensitive element, do not confuse these cells with the selenium photoconductive type. The photovoltaic cell *generates* a voltage, while the

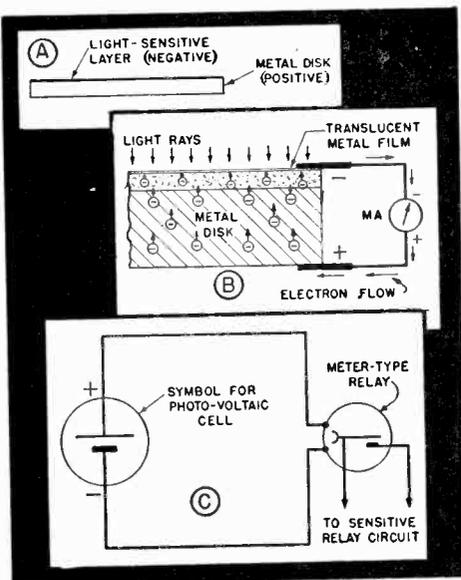


FIG. 15. The construction and operation of the photovoltaic cell differs from that of other cells, particularly in that it generates a voltage without any external battery.

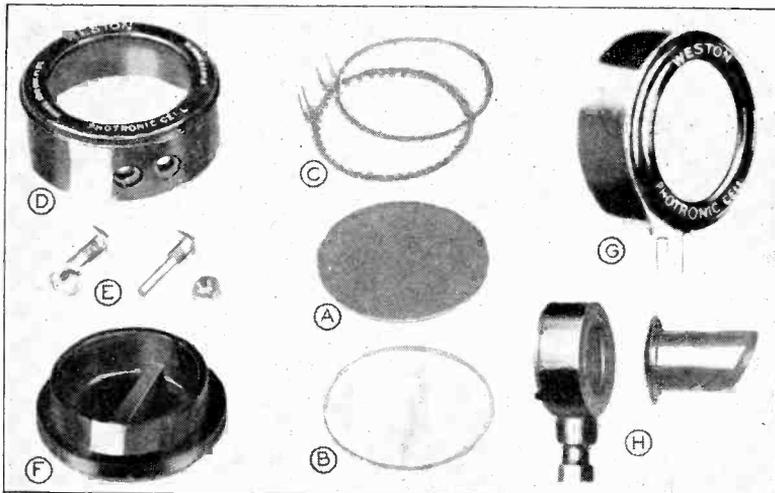
photoconductive cell acts only as a variable resistance.

**Photronic Cell.** The Weston Photronic cell, one of the first commercial selenium-iron type cells to be developed in the United States, is constructed in much the manner described above; today this is one of the best known of all photovoltaic cells in this country.

The component parts of a Photronic cell are shown in Fig. 16; at A is the

housing. The projecting ends of the bolts are of the same diameter as radio tube prongs, and fit into an ordinary four-prong tube socket.

Characteristic curves for the Weston Photronic cell are given in Fig. 17. The current output varies with the external resistance connected to the cell as well as with the illumination; the output is linear (proportional to illumination) for low values of external resistance. This linear relation for



Courtesy Weston Electrical Instrument Corp.

FIG. 16. View of component parts of the Weston Photronic cell, a dry type photovoltaic cell. A—sensitized iron disc; B—glass window; C—metal contact rings; D—bakelite housing; E—terminal bolts; F—bakelite cap; G—assembled cell; H—Photronic cell in weatherproof housing, with visor to keep out unwanted light.

thin disc made of iron, on one side of which is the layer of light-sensitive selenium. The cell is assembled as follows: The glass window is placed in the bakelite housing; the iron disc, with one of the metal contact rings on each side, is placed in the housing next, with the sensitive side against the glass; the contact rings are turned so each terminal lug is over one of the holes in the side of the housing, and the terminal bolts are set into place, heads inside the housing; finally the bakelite cap is screwed into the

low resistances holds true even when the cell is in direct sunlight (about 175 lumens of illumination). It is necessary, therefore, to consider the resistance of meters or relays used with Photronic cells. Low-resistance meters are used generally where it is necessary that the current produced be directly proportional to the light intensity. However, for low values of illumination (below .1 lumen or below 10 foot-candles), a sufficiently linear response can be obtained with instruments having 1000-ohm or higher

resistances, as the curves are practically straight for illumination of such low intensity (see Fig. 17A).

The voltage output of the Photronic cell for varying intensities of illumination, measured by a method which involved no flow of current through the cell, is shown in Fig. 17B. The curve is not linear, and the values of voltage are quite low. This cell, together with all other photovoltaic cells, is not used generally with vacuum tube amplifiers which require large changes in grid voltage to get useful changes in plate current.

The relative color response curve for the Photronic cell, given in Fig. 17C, shows that the cell and the human eye have a maximum response to about the same color, yellow-green. The manner in which a glass window absorbs ultra-violet light can be seen; a much higher response to ultra-violet light is obtained with a quartz window in the cell. Filters (panes of colored glass) which give the Photronic cell almost exactly the same color response as the human eye can be obtained from the manufacturer, and these filters are necessary whenever the cell is to replace the human eye in making measurements of light. The Weston illumination meter, where the cell is connected directly to a microammeter calibrated to read in foot-candles, is an example of this use.

Photronic cells should be used as current sources rather than voltage sources, because the current output of the cell varies directly with the light falling on the cell. In order to obtain a constant voltage from this type of cell, it is necessary to connect a resistance across the cell and take off the voltage across the resistor. This voltage then will be proportional to the light falling on the cell.

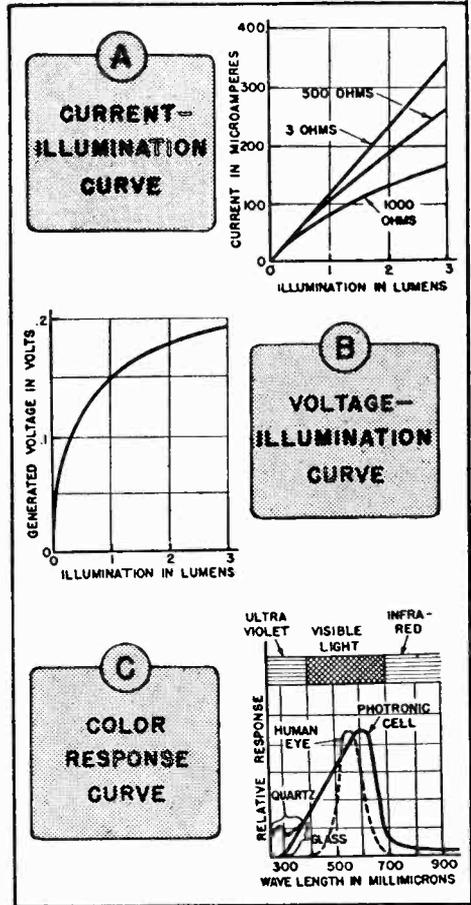


FIG. 17. These characteristic curves are for the Weston Photronic cell, a representative photovoltaic type.

It is possible to connect two or more Photronic cells *in parallel* to obtain a greater current output for a given intensity of illumination. Connections are exactly the same as for dry cells, minus to minus and plus to plus. The dark resistance of these units is quite high, which means that a number of cells at different points can be connected together in parallel without the possibility of one cell feeding current into the others. The total current delivered by a system made up of a number of dry photovoltaic cells will



Courtesy DeJur-Amaco Corp.

A photovoltaic cell is used in this exposure meter. When pointed at the scene to be photographed, the cell operates a meter, which gives information needed for proper exposure of film. This is an important photoelectric application.

be proportional to the total light falling on all of the cells; for instance, if three cells in parallel furnish 100 microamperes when illuminated by light sources of equal intensity, the combination of cells would give exactly 67 microamperes if one cell were completely darkened. Individual cells have this same characteristic; practically the same current response is obtained when light is concentrated on one part of the cell as when the *same amount* of light is spread uniformly over the entire active surface.

**The Electrocell.** The Electrocell, a dry disc type of self-generating cell imported from Europe, consists of a light-sensitive layer of selenium deposited on the surface of an iron disc. Contact is made to the layer with a transparent conducting film of silver. Transparent lacquer sprayed over the greater part of the cell protects it from rough handling. The sensitivity of

this cell is 480 microamperes per lumen; in direct sunlight the  $1\frac{3}{4}$ -inch diameter cell will deliver as much as 20 ma. It is claimed that the Electrocell, in the larger sizes, delivers enough current to operate a sensitive relay directly, thus eliminating the need for an expensive super-sensitive relay. The smallest Electrocells,  $\frac{3}{8}$  inch in diameter, have a sufficiently low capacity to be used for modulations up to 8000 cycles.

**Other Types.** Another type of dry photovoltaic cell which is now on the market, the Westinghouse Photox cell, consists of a disc of copper on which has been formed a thin film of cuprous oxide. Contact is made to the copper and to the oxide layer in much the same way as in the selenium-iron type cell. Gold is the material used as the translucent film on this cell.

► The Lange cell, of foreign manufacture, is quite similar in construction to the above-mentioned photovoltaic cells. According to data furnished by the manufacturer, it has a very good current output.

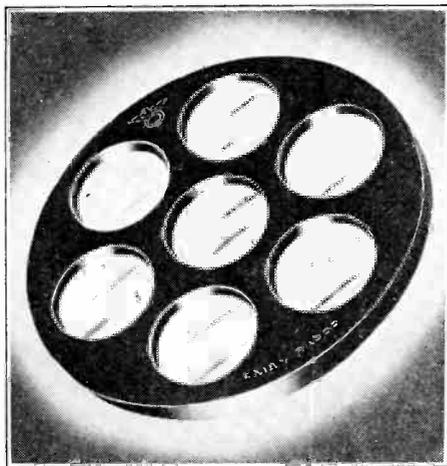
► The General Electric Company's selenium-on-iron type photovoltaic cell is mounted in a bakelite case and provided with prongs which permit its being mounted in an ordinary four-prong radio socket.

**General Characteristics of Dry Photovoltaic Cells.** Dry photovoltaic cells usually are connected directly to super-sensitive relays. These are built much like a milliammeter or microammeter, but have contacts on the moving pointer and fixed contacts on the meter scale. Such relays respond to currents of the order of microamperes, and are therefore quite costly.

For all practical purposes, the response of a dry photovoltaic cell to

changes in light can be considered to be practically instantaneous. Actually these cells are fast enough to detect the passage of a rifle bullet through a beam of light. Because of the high parallel capacity of the cell, however, (about .5 mfd.) dry photovoltaic cells are not suitable for such a use as responding to a light beam which has been modulated at audio frequencies. Photoemissive cells are used more generally for this purpose.

A typical assembly of voltaic cells in parallel is shown in Fig. 18. This connection gives a higher output for a



*Courtesy Selenium Corp. of America.*

FIG. 18. A number of photovoltaic cells can be connected together as shown here, giving a far greater output.

given light value, as was explained for the Photronic cell.

The output of a photovoltaic cell can be increased considerably by connecting a small potential, not over 6 volts, in series with the cell. Too high voltages may permanently change the characteristics of the cell. In fact, the manufacturers of the Photronic cell recommend that no external voltages be used if maximum cell life is to be obtained. A photovoltaic cell behaves much the same as a photo-

conductive cell, when an external potential is used in series with the cell.

Dry photovoltaic cells are believed to have an unlimited life; that is, they will retain their characteristics for long periods of time if kept at temperatures below about 120° Fahrenheit. The cells must be tightly sealed in their cases, because they are critically affected by chemical vapors and by dampness.

## WET PHOTOVOLTAIC CELLS

The photovoltaic cell in wet form is now over 100 years old, as its principle, known as the Becquerel effect, was discovered by Edmund Becquerel in 1839. While experimenting with the ordinary type of voltaic cell known in that day, he noticed that his cell gave a much greater output in direct sunlight than in the subdued light of his laboratory.

In its simplest form, the wet photovoltaic cell consists essentially of two metals which are immersed in an electrolyte, with one of these metals or electrodes exposed to a source of light. Research workers have developed two types of wet photovoltaic cells, one in which the electrodes themselves are light-sensitive, and another in which the electrolyte is light-sensitive, but neither type is believed to be of great commercial importance at the present time.

Wet photovoltaic cells can be constructed with a number of different electrode materials and electrolytes. One form has two copper electrodes, on one of which is a film of cuprous oxide. Another type uses one copper electrode on which is a film of cuprous oxide, and one lead plate; the electrolyte is a lead nitrate solution.

The wet photovoltaic cell has a number of drawbacks. In some types, destructive gases are formed while the

cells are standing idle. Industry in general hesitates to adopt equipment like this, in which there is a possibility of leakage of liquid.

Naturally, polarity must be con-

sidered when connecting photovoltaic cells of all types into a circuit, as the cells are really small batteries. In the wet cell, the electrode which has the oxide layer is always *positive*.

---

## Electron-Multiplier Photocells

A multiplier photocell is a multi-electrode photoemissive cell which uses the principle of "secondary emission" to amplify tremendously the electron stream given off from its cathode when the cathode is exposed to light. The RCA 931-A, a typical multiplier cell, has eleven electrodes—the cathode, nine special anodes known as "dynodes," and the regular anode from which the electron stream passes to the outside circuit.

The dynodes are curved metal plates which are coated on one side with a mixture of chemicals. This coating has the property of emitting electrons when a fast-moving electron strikes it. As many as eight or ten electrons may be emitted from the coating when one electron hits it (the number depends to some extent on the velocity of the electron striking the surface). The electrons given off from the coating are called "secondary" electrons, and the emission itself is known as "secondary emission."

### HOW THE CELL OPERATES

The elements of the RCA 931-A are arranged as shown in Fig. 19. The dynodes—numbered 1 through 9—are maintained at successively higher positive voltages with respect to the cathode, usually in 100-volt steps. That is, dynode 1 is 100 volts more positive than the cathode, dynode 2 is 100 volts more positive than dynode 1, and so on. The dynodes are so

shaped and arranged that electrons from one of them tend to go to the dynode with the next higher number—for example, electrons emitted from dynode 2 will go to dynode 3, those from 3 will go to 4, and so on.

As you can see, dynode 9 practically surrounds the anode. This construction is used because the potential between the anode and the cathode will fluctuate with the load voltage drop and such a fluctuating potential would disturb the movement of electrons from one dynode to the next. Dynode 9, however, acts as a shield and prevents this undesirable effect from occurring.

The anode is a grid rather than a solid plate. This construction of the anode allows electrons from dynode 8 to pass through it to dynode 9; then, because of the geometry of the space around the anode, and because the anode is usually maintained at a potential around 250 volts higher than 9, electrons emitted from 9 are collected by the anode and passed on to the external circuit.

► Let's trace the electron path through this cell to see why high current amplification occurs. Light falling on the cathode causes emission of electrons, just as in the regular photoemissive cell. These electrons are attracted to dynode 1. Each of these electrons, when it strikes dynode 1, may cause the emission of up to eight or ten secondary electrons from the

dynode. These secondary electrons are attracted to dynode 2, where each of them causes the emission of several more secondary electrons. This process is repeated all along the electron path through the tube, with each secondary electron from a dynode causing the emission of many others from the next dynode. As you readily can see, the number of electrons flowing in the tube increases enormously as we go from one dynode to the next. In fact, if the tube worked perfectly, with every electron from one dynode going on to the next and causing the emis-

all odds the most sensitive cell in use today.

### CELL SHIELDING

The mica shield shown in Fig. 19 is used to prevent possible feedback of positive ions from the anode to the cathode. Although the glass envelope in which the tube elements are housed is as well evacuated as is commercially practicable, a small amount of air is left in it, and some of this air may be ionized by the large electron current in the space between the anode and dynode 9. If such ions were able to flow to the cathode or to one of the first dynodes, they would cause secondary emission from these elements which would, of course, be amplified by the other dynodes of the cell. This would make the electron flow through the cell undesirably erratic, and might change the amplifying characteristics of the cell considerably. The mica shield, which allows no ions to pass through it and emits no secondary electrons when struck by ions, prevents the ions from reaching the cathode or any but the highest-numbered dynodes, and so reduces their effects to a minimum.

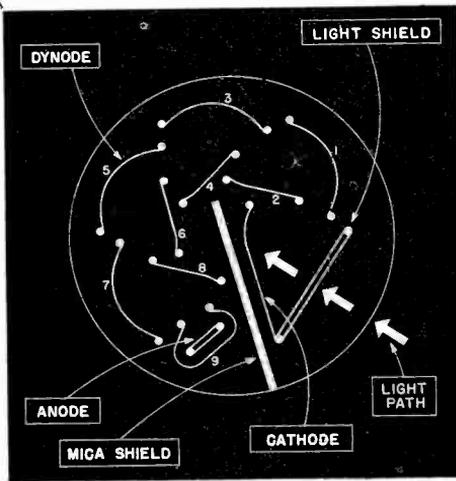


FIG. 19. The electron-multiplier cell contains a number of elements, so shaped and placed that secondary emission tremendously increases the sensitivity.

sion of ten electrons from it, then just *one* electron emitted from the cathode would cause *one billion* electrons to arrive at the anode!

Of course, such perfection of operation does not occur, but the tube can be made to produce current amplifications of as much as 200,000 times—which means that 200,000 times as many electrons can be drawn from the anode as are emitted from the cathode. This makes the multiplier photocell by

► The light shield (see Fig. 19) is a small grill through which light must pass to reach the cathode. It is used principally to prevent undesired light from reaching the cathode. Also, the grill acts as an electrostatic shield, since it is electrically connected to the cathode.

### CHARACTERISTICS OF MULTIPLIER CELLS

Since the 931-A multiplier cell is an 11-electrode tube, rather than a diode like other photoemissive cells, we must take ten operating potentials into account when we discuss the performance of the cell. This is not really

as complicated as it looks, since the voltage between the cathode and each successively-numbered dynode *usually* increases by equal steps. (This is not always true: sometimes the voltage step to one dynode is made unequal to the steps to the other dynodes, in order to control the amplification of the tube.) For ease in discussing the cell characteristics, engineers often

*amperes per lumen*—that is, an illumination of 1 lumen on its cathode would produce a current output of 2 amperes. The current output of the multiplier is thus 160,000 times as great as that of the photocell for the same amount of illumination!

This, however, does not mean we can draw 2 amperes out of multiplier cell; such a high current would ruin the cell at once. The maximum rated current output of the 931-A cell is 2.5 milliamperes. The high sensitivity

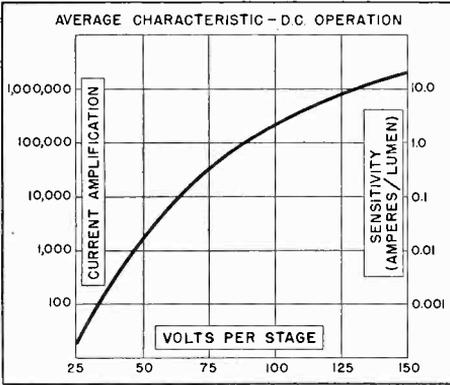
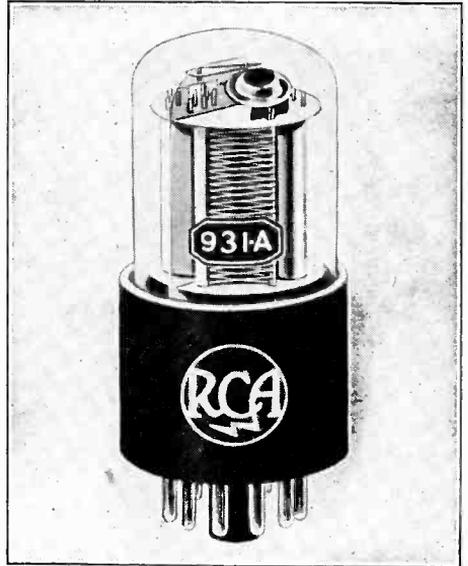


FIG. 20. Sensitivity and gain curves for the electron-multiplier cell.

consider the cell to be an amplifier, with each dynode representing one stage; this allows them to draw characteristic curves in terms of the volts-per-stage used on the cell.

Such a curve is shown in Fig. 20. This shows both the current amplification of the 931-A and its sensitivity in amperes per lumen of light falling on the cathode surface. A comparison of this curve with the current illumination curve for an ordinary vacuum photocell (Fig. 7A) shows you at once how much greater the current output of the multiplier cell is. At an illumination of 1 lumen, for example, the vacuum photocell has a current output of about 12.5 microamperes. The 931-A multiplier cell, operated at 100 volts per stage (a usual operating potential), has a sensitivity of 2



Courtesy RCA Mfg. Co.

A typical electron-multiplier cell is shown here.

means we must be sure never to expose the cell to too much light—it might be ruined.

Under normal operating conditions, the output current of the multiplier cell is directly proportional to the illumination. The response of the cell when illuminated by a modulated light beam is perfectly flat up to extremely high frequencies of modulation. The amplifying action of the tube has no adverse effect on its fre-

quency response, because secondary emission from the dynodes occurs instantaneously when they are struck by electrons. At very high frequencies, however, the fact that the electrons take a definite time to move from one element to the next limits the response of the cell. As Fig. 21 shows, the

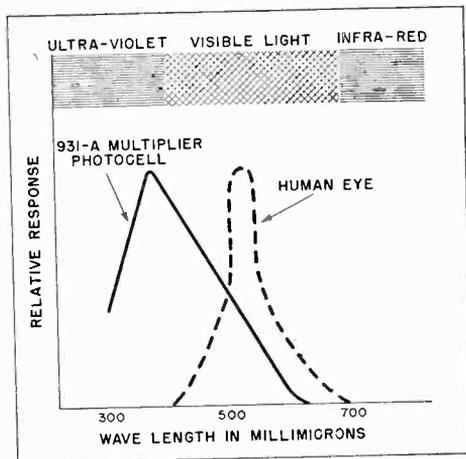


FIG. 21. Color response curves for the electron-multiplier cell.

color response curve of the 931-A cell has a peak in the ultra-violet region and drops off rather rapidly on either side of this peak.

### Power Supply Requirements.

Multiplier cells have one disadvantage not shared by other kinds of light-sensitive cells — high voltages are necessary to operate them. The anode may be as much as 1250 volts above the potential of the cathode. Naturally, such high voltages are dangerous, and you should take all possible safety precautions if you work with such a cell. Apparatus using these cells is usually designed so that it is hard to reach the high-voltage electrodes, and so that the primary circuit of the high-voltage power supply must be broken before you can get at the cell. Whenever possible, the

positive high-voltage terminal of the voltage supply circuit (connected to the anode of the cell through the load) is grounded; the cathode, connected directly to the negative terminal of the supply, then operates below ground (highly negative with respect to ground). This allows the high-voltage section of the circuit to be safely enclosed, while the output section (the part of the circuit you are most apt to touch) will be fairly close to ground potential. Of course, if the circuit happens to be defective, high voltages may appear anywhere in it—so you should always shut off the

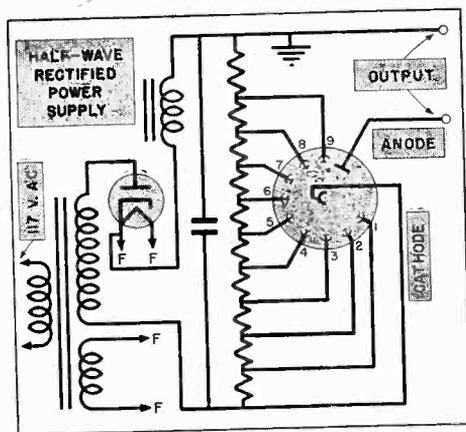


FIG. 22. Where d.c. is wanted for operation of an electron-multiplier, a circuit like this one can be used.

power and discharge any charged condensers before touching any part of the circuit.

Ordinarily, the operating voltages for the various stages of a multiplier cell are obtained from a tapped voltage divider which is placed across either a full-wave or a half-wave rectified source of current. A simple half-wave power supply circuit for the cell is shown in Fig. 22. A cell also can be run directly from an a.c. source, using a circuit like that in Fig. 23.

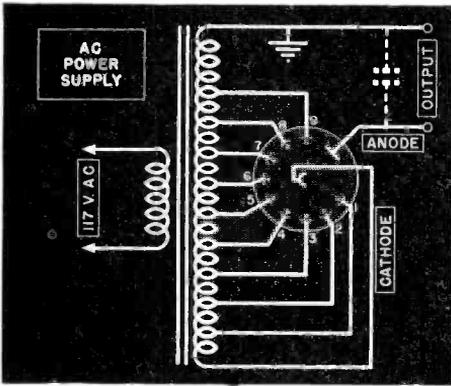


FIG. 23. A.C. voltages can be used as the operating voltage for many applications of the electron-multiplier cell.

This circuit is particularly suitable for relay operation.

► A magnetic field, as you know, can

change the direction of an electron stream. For this reason, it is sometimes necessary to shield a multiplier cell magnetically when it is operated near a strong magnetic field, so that the field will not interfere with the desired flow of electrons from dynode to dynode.

► In addition to its enormous sensitivity, the 931-A multiplier cell has a low noise level, low dark current, and freedom from distortion. Practical applications of the cell include the operation of relays, sound reproduction from film, facsimile transmission, and scientific research involving low light levels. This cell is a recent development and is now coming into more widespread use.

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# Lesson Questions

Be sure to number your Answer Sheet 25FR-3.

Place your Student Number on every Answer Sheet.

Send in your set of answers for this lesson immediately after you finish them, as instructed in the Study Schedule. This will give you the greatest possible benefit from our speedy personal grading service.

- ✓ 1. Name the six basic parts of a complete photoelectric installation.  
*source, path, object, cell, relay, alarm*
- ✓ 2. What two kinds of light are invisible to the human eye, yet can be "seen" by a photoelectric cell? *ultra violet, infra red*
- ✓ 3. Name the four classes of light-sensitive cells. *photoresistor, photovoltaic, photoconductive, electron multiplier*
4. If the photocell and resistor  $R$  are interchanged in Fig. 5, will the relay close when: 1, the photocell is *lighted*; or 2, when the photocell is *darkened*?
5. What is the maximum safe operating voltage of the average gas type photo-emissive cell? *100V*
6. Which of the light-sensitive cells will operate a super-sensitive relay *without* the use of a battery or an amplifier? *photovoltaic*
7. What is used with a Photronic cell to make it have exactly the same color response as the human eye? *filters of colored glass*
8. How would you connect two or more Photronic cells to secure a greater current output for a given intensity of illumination? *Parallel*
9. What basic principle is used in the electron-multiplier cell to obtain a high light-sensitivity? *secondary emission*
10. How may the amplification of an electron-multiplier cell be varied?  
*By varying the illumination*

## ACTION SPEAKS FOR ITSELF

*Be sure you are right, then go ahead*—this has been the motto of many of the world's great men.

In most cases you know instinctively what is right, your decision being based upon your past training, your experience, your common sense and your conscience. In these cases, *act!* Waste no valuable time arguing with others who know less than you; waste no time trying to “pound” your ideas into a cynical world—take the initiative yourself.

It is a thousand times better to *do things* and let your deeds speak for themselves than to spend time explaining why your proposed course of action is right. Friends can hinder your success if you take time to justify your actions to each one of them.

If you need advice—if you are not exactly certain you are right, then go to men who are capable of giving authoritative answers to your questions. You'll find that successful men are glad to answer serious, well-planned questions. Analyze their advice in connection with your own experiences, make your decision, then act!

Give this plan a tryout. You will accomplish a great deal more work, and you will be a lot happier.

J. E. SMITH

**HOW SOUND REPRODUCERS  
OPERATE**

25FR-1

**NATIONAL RADIO INSTITUTE**

ESTABLISHED 1914

**WASHINGTON, D. C.**



# Study Schedule No. 26

For each study step, read the assigned pages first at your usual speed, then reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind, then answer the Lesson Questions specified for that step. Study each other step in this same way.

1. Introduction; Types of Loudspeakers; Bi-Polar Magnetic

**Driving Units - - - - - Pages 1-6**

A Complete Loudspeaker has Three Sections; The "Weakest Link" in Radio; What You Will Study; four general groups of loudspeakers; Magnetic Loudspeakers; Dynamic Loudspeakers; Condenser Loudspeakers; Crystal Loudspeakers; Headphone Construction; How a Headphone Works; Why Strong Permanent Magnets are Necessary; Headphone Ratings. Answer Lesson Questions 1, 2 and 3.

2. Analysis of Vibrating Systems; Mechanical Resonance

**in Headphones - - - - - Pages 6-11**

Electrical equivalent of mass and compliance; Natural Frequency of Vibration; Velocity of a Diaphragm; Maximum Displacement; Velocity of a Vibrating Mass; Facts to Remember About Vibrating Systems; Why headphone diaphragms must have a high natural resonant frequency. Answer Lesson Questions 4 and 5.

3. Balanced Armature Type of Magnetic Driving Unit; Dynamic Driving

**Units; Condenser Driving Units; Crystal Driving Units - Pages 12-19**

Construction of balanced-armature unit; how it works; comparison with other driving units; piston action in a dynamic driving unit; diaphragm construction; how a dynamic driving unit works; voice coil design; factors affecting force upon voice coil; construction of condenser driving units; drawbacks of condenser unit; basic action of a crystal loudspeaker; construction of crystal units; drawbacks of crystal units. Answer Lesson Questions 6 and 7.

4. Loudspeakers Can Also Serve as Microphones; Action of a Horn as a Loudspeaker Coupling Unit; Design of Dynamic Driving

**Units for Horn Loudspeakers - - - - - Pages 19-28**

Inter-communicating systems use loudspeakers as microphones; equivalent mechanical circuit of a vibrating diaphragm; Measuring Loudspeaker Efficiency; Loading the Diaphragm; Exponential Horns; Mouth Area; Throat Area; Construction of Horn; Horn Ratings; diaphragm mounting methods; efficiency of horn loudspeakers. Answer Lesson Questions 8, 9 and 10.

5. Mail your Answers for this Lesson to N. R. I. for Grading.

6. Start Studying the Next Lesson.

# HOW SOUND REPRODUCERS OPERATE

## Diaphragm-Type Units

### Introduction

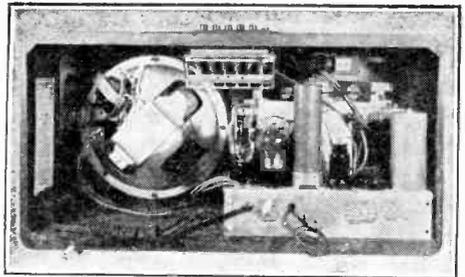
**I**N RADIO receivers and in public address systems the loudspeaker serves to convert audio frequency currents into sounds which can be heard by the human ear. The A.F. currents in the last stage of the audio amplifier are made to set in motion an electro-mechanical system, and this in turn causes the air in the vicinity of the loudspeaker to vibrate in accordance with the variations in the audio signal, thus producing sound having all the characteristics of the original sound.

*A Complete Loudspeaker has Three Sections.* From what you have already studied in this Course and from your own actual experience, you can readily see that a complete sound reproducer or loudspeaker consists of three essential sections: 1, an electro-mechanical *driving unit*, often called a *motor*, which converts A.F. currents into vibratory motion; 2, the *diaphragm* or *cone*, which when driven by the motor sets the surrounding air into motion; 3, the *air coupling system*, consisting essentially of a horn or baffle, which improves the efficiency of the loudspeaker by setting a larger amount of air into vibration and by preventing the reproduced sound components from cancelling each other.

*The "Weakest Link" in Radio.* Perfection in radio transmitter and receiver design is of little value unless the loudspeaker, the final link in

the radio system, is capable of reproducing the entire range of audio signals with true fidelity and is capable of operating at the required maximum loudness level without distorting the reproduced sound. Contrary to general impressions, *the loudspeaker is the weakest link in the entire radio system.* The design of high-fidelity loudspeakers is an intricate and involved task even for the engineer who specializes in this particular branch of radio engineering. Even today, after years and years of research on the subject, engineers are still striving for the perfect loudspeaker.

*What You Will Study.* Although as a Radiotrician and Teletrician you are not particularly interested in the exact details of loudspeaker design technique, a general discussion of some of the problems encountered in building loudspeakers and a study of the various solutions which have been developed for these problems will



The loudspeaker is the mechanical means of converting electrical signals into sound waves.

serve to make you realize the limitations of loudspeakers. Furthermore, a familiarity with the constructional features of various types of loudspeakers and with their performance under various conditions will be of value to you in any branch of radio. Only the more widely used loudspeakers, which you will be likely to encounter in your practical work, will be considered. Basic principles of loudspeaker construction will be stressed, but enough actual examples of commercial loudspeakers will be taken up to illustrate how these principles apply to all loudspeakers.

### Types of Loudspeakers

Although some loudspeaker manufacturers have assigned trade names to their various types of loudspeakers, you will find that technical men in general prefer to describe loudspeakers according to their construction or operation.

Loudspeakers can be divided into four general groups according to the type of motor or driving unit which they employ: 1, *magnetic loudspeakers*, in which the moving element is made of iron; 2, *dynamic or moving coil loudspeakers*, in which the moving element is a coil of wire; 3, *condenser loudspeakers*, in which one plate of a large two-plate condenser is the moving element; 4, *crystal loudspeakers*, in which a crystal is the moving element.

*Magnetic Loudspeakers.* That type of magnetic loudspeaker which depends for its operation upon the attraction of a bi-polar driving unit for an iron diaphragm has long since been superseded by the other types to be discussed. The bi-polar driving unit is still used almost universally in

headphones, however, and will therefore be taken up in this lesson. Magnetic loudspeakers in use today are of the *balanced armature type*; when a balanced armature unit is made to drive a diaphragm, we have a *diaphragm type magnetic loudspeaker*. On the other hand, if a balanced armature unit drives a cone, we have a *magnetic cone loudspeaker*.

*Dynamic Loudspeakers.* When the fixed magnetic field for a dynamic or moving coil loudspeaker is furnished by an electromagnet, we have what is known as an *electrodynamic loudspeaker*. When this fixed magnetic field is provided by a permanent magnet, we have a *permanent magnet dynamic loudspeaker*, often abbreviated as *P.M. dynamic loudspeaker*. Either of these dynamic or moving coil units can be used to drive a small cone, a large cone or a diaphragm. With a small cone or diaphragm, the air coupling unit is usually a horn; with large cones, of the size commonly found in radio receivers employing dynamic loudspeakers, a baffle ordinarily serves as the air coupling system.

*Condenser Loudspeakers.* These are also known as *electrostatic loudspeakers*, for they depend for their operation upon the forces of attraction and repulsion existing between two charged plates. The moving plate is ordinarily about one foot square, so no cone or diaphragm is required. Condenser speakers are used so seldom today that only the fundamental operating principles will be taken up in this lesson.

*Crystal Loudspeakers.* The driving units of crystal loudspeakers are all essentially the same, but they may be made to drive either a diaphragm

or a cone, and may be used with either a baffle or horn.

Once you become familiar with loudspeaker construction and operation, you will find that practically all of the loudspeaker names or designations which you encounter will be self-explanatory.

### Bi-polar Magnetic Driving Units

No discussion of sound-reproducing units would be complete without a consideration of the headphone unit. Aside from the fact that this bi-polar magnetic driving unit was the essential part of all early loudspeakers, headphones are still widely used today by commercial radio operators, by radio amateurs, by experimenters, and by those who are hard-of-hearing.

**Headphone Construction.** The constructional details of a typical headphone unit are shown in Fig. 1, the important parts being clearly identified. The operating principle of this bi-polar magnetic driving unit can be more clearly understood by referring to the simplified diagram shown in Fig. 2A. The permanent magnet *PM*, semicircular in shape and made of hard steel, produces a fixed magnetic flux which flows through the two soft iron pole pieces *P*, and then through two air gaps to the thin iron diaphragm *D* (about .003 inch thick), which is firmly clamped around its edges to the housing and which is accurately spaced away from the pole pieces. The diaphragm is attracted toward the pole pieces; one explanation is that the magnetic circuit always adjusts itself to have the least reluctance (opposition to the flow of magnetic flux), while another is that the north (*N*) pole piece makes

the section of the diaphragm directly above it a south (*S*) pole, and the south pole piece makes the diaphragm section above it a north pole. Since opposite poles always attract, the

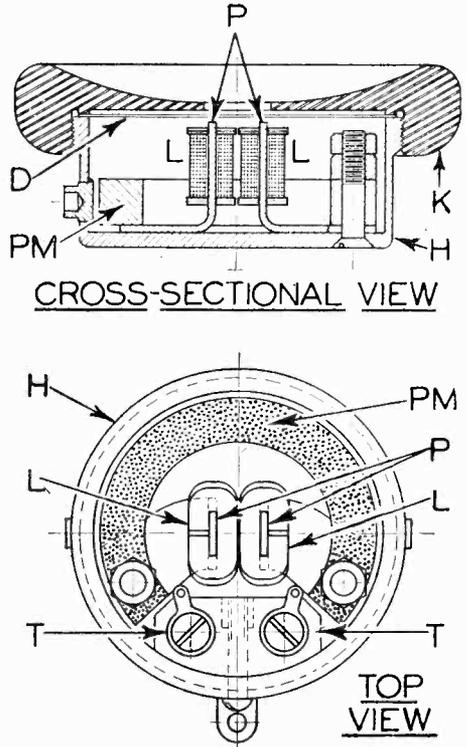


FIG. 1. These two diagrams show the important constructional details of a typical bi-polar magnetic driving unit as used in a headphone. The parts are: *PM*—hard steel permanent magnet; *L*—coils; *P*—soft iron pole pieces; *D*—soft iron diaphragm; *H*—metal housing; *K*—ear cap; *T*—headphones terminals, connected directly to the coils.

diaphragm is bent toward the pole pieces.

The diaphragm is made stiff enough and is originally separated far enough from the pole pieces so that it cannot actually touch the pole pieces under any normal conditions of operation.

*How a Headphone Works.* Observe that around each of the pole pieces in Fig. 2A is a coil of wire, with the two coils connected together in series in such a way that the magnetic flux due to each will be in the same direction through the permanent magnet. When a current is sent through these coils, it will produce an additional magnetomotive force which either aids or opposes the magnetomotive force of the permanent magnet (depending upon the direction of current flow) and therefore either in-

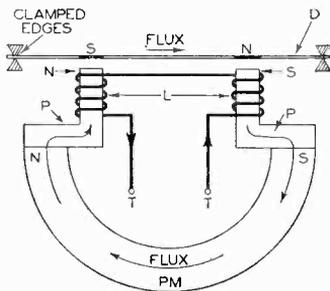


FIG. 2A. Simplified diagram of a bi-polar magnetic headphone unit, showing coil connections and flux paths.

creases or decreases the amount of magnetic flux flowing through the magnetic circuit of the headphone. Any change in this total magnetic flux naturally changes the attraction of the pole pieces for the diaphragm; when the current flowing through the coil is varying at an audio frequency, the diaphragm will move toward and away from the pole pieces at the same frequency. With proper design of the entire headphone unit, the movement or displacement of this diaphragm will correspond exactly to the variations in the audio frequency currents,

and this diaphragm will produce sound waves which correspond in wave form to that of the A.F. signals.

*Why Strong Permanent Magnets are Necessary.* One of the most important factors affecting the performance of a magnetic loudspeaker is the permanent magnetic field. Let us see what would happen if this field were not used. Assume that we are sending a pure sine wave alternating current through the coil of the bi-polar driving unit in Fig. 2A, with permanent magnet *PM* temporarily replaced by a soft iron piece of similar shape. When this alternating current is zero there will be no magnetic field and no pull on the diaphragm. As the current swings positive, the two poles will become magnetized with opposite polarity, and the diaphragm will be pulled toward the pole pieces. This attraction will continue until the current again reduces to zero. As the current reverses for the negative half of the cycle, the polarity of each pole will reverse, but the poles will again attract the diaphragm. As a result, the diaphragm will vibrate at twice the frequency of the current passing through the coil. Clearly this is an undesirable condition for sound reproduction.

The use of a strong permanent magnetic field eliminates this undesirable frequency-doubling effect in a bi-polar headphone unit, for the permanent field provides a continuous attraction for the diaphragm which is merely varied by the alternating current. The change in the attraction of the pole pieces for the diaphragm depends upon the original magnetization and upon the change in flux caused by the signal current flowing through the

coils (the change in attraction actually depends upon the product of these two factors). Increasing the strength of the permanent magnet and increasing the strength of the signal currents are thus possible ways of increasing the sensitivity of a headphone unit.

Modern headphone units have permanent magnets made of chrome or tungsten steel, which produce a strong permanent flux and thereby give a highly sensitive unit with a minimum of frequency doubling. Permanent magnets made of Alnico produce even greater magnetomotive forces, and this recently developed alloy may therefore be found in new high-quality headphones. Placing the diaphragm as close as possible to the pole pieces increases the amount of permanent flux and thus increases the sensitivity of the unit, but at the same time this procedure reduces the maximum available volume; the practical headphone unit is therefore based upon a compromise between sensitivity and volume.

If the coil current in a bi-polar magnetic driving unit is to produce a large A.C. magnetic flux, many turns of wire are needed on the coils, and the A.C. magnetomotive force produced by these turns must develop a large A.C. flux. You know that in a magnetic circuit the amount of flux produced for a given magnetomotive force depends upon the reluctance of the flux path, just as the current for a given voltage depends upon the resistance of the path for current in an electrical circuit. The first essential of a low-reluctance flux path is a small air gap between each pole piece and the diaphragm, but on the other hand, the air gap must be long enough

to permit the diaphragm movements which are necessary for maximum required volume. With the arrangement shown in Fig. 2A, however, the A.C. flux produced by the coils must flow through the permanent magnet; unfortunately this has a high reluctance because it is made from a hard, tempered steel or steel alloy rather than from soft iron.

A simple solution to this difficulty is shown in Fig. 2B; it involves placing a short-cut path for A.C. magnetic flux across the bottoms of the

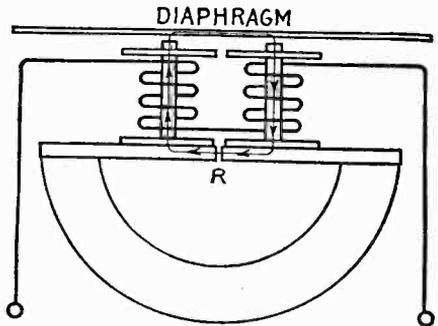


FIG. 2B. Simplified diagram of a special type of bi-polar magnetic headphone unit in which a leakage reactance path  $R$  for magnetic flux is purposely introduced to improve performance.

pole pieces. The air gap  $R$  which is placed in this path has a low enough reluctance to make the *alternating* magnetic flux prefer it to the path through the permanent magnet, but at the same time this air gap offers considerably more reluctance to the *permanent* magnetic flux than does the desired path through the pole pieces and the diaphragm.

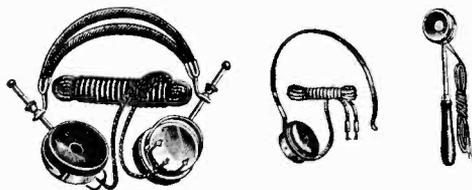
*Headphone Ratings.* Headphones are commonly rated according to their D.C. resistance, even though the impedance value gives a far better

indication of operating characteristics. For a given type of headphone construction, having a given size of bobbin for each coil, we can use a large number of turns of very small wire, thereby securing a high D.C. resistance and a high impedance, or we can use a small number of turns of larger wire to secure a low D.C. resistance and low impedance. Since headphones ordinarily serve as plate loads in amplifier stages, it is the impedance value (generally measured at the aver-

manufacturers supply this information.

### Analysis of Vibrating Systems

A complete study of the mechanically vibrating parts used in loudspeakers involves a number of new fundamental ideas which you should understand before proceeding farther with your study of loudspeakers. A mechanical vibrating system has certain properties which compare with those of an electrical vibrating or



Radio operators for commercial or amateur communication systems ordinarily use a pair of headphones like those shown here, in which two units are mounted on the headband and connected together in series. Single headphones with a headband are widely used by experimenters, by hospitals having radio facilities for each patient, and by the hard-of-hearing. A single headphone unit with a handle is used chiefly by those who have only slightly impaired hearing and who therefore need use the sound-amplifying system only at intervals.

age voice frequency of 1,000 cycles) which is of importance; the designer will choose a headphone impedance value which best matches the required plate load of the amplifier tube. In general, the impedance of a headphone unit at 1,000 cycles will be from five to ten times its ohmic value for direct current.

When headphones are being used primarily to convert low intensity signals into audible sounds, they should be rated on the basis of how much A.F. power in milliwatts is required to produce a just audible sound; as yet, however, very few

oscillating system; for instance, you will learn that an inductance (coil) in an electrical oscillating circuit can be compared to *mass* in a mechanical vibrating system, capacitance in an oscillating circuit can be compared to *compliance* in a mechanical system, and resistance in an oscillating circuit can be compared to *friction* or to *mechanical resistance*. A few examples will help you to understand what these new terms mean.

Consider any large, heavy body on wheels; an automobile will do as an example. When pushing the car on a smooth, level road, you know that

considerable *force* is required to set it into motion, but once the auto is moving at a definite speed, it can be kept at that speed with little additional effort. In order to set such a heavy object into motion, we must overcome the inertia of its *mass*; the heavier the object, the greater is its mass and inertia, and the more we must push to set it into motion. We have the same situation in an electrical circuit; an inductance has an electrical inertia which serves to limit the initial flow of current. *Mass in a mechanical vibrating system therefore corresponds to the inductance of a coil in an electrical circuit.*

Now let us consider an ordinary steel coil spring. Experience has taught us that force is required to stretch the spring, and now we learn that this ability of a spring to stretch under the application of force is referred to by engineers as its *compliance*. The greater the compliance of a spring, the more it will stretch when a given force is applied. This compares very closely to a condenser, where greater capacity means that we can store more electricity in the condenser.

As you well know, motion of any object involves friction. It is far more difficult to move an automobile from which all four wheels are removed than a car having wheels, for without wheels there is a great deal more friction between the chassis and the pavement. You know also that friction can be reduced by making the sliding surfaces perfectly smooth and by lubricating the sliding surfaces with grease. Friction of the type which exists when one object rubs against another is quite familiar to you, but in loudspeakers it is the friction pro-

duced by an object moving in air which is of importance. An example will illustrate the nature of this friction.

Suppose we take a mass, as shown in Fig. 3A, and a spring, as shown in Fig. 3B, and connect them in series as shown in Fig. 3C. When both the spring and the mass are stationary, pull down on the mass and then release it. The spring stretches when you pull down, but since it is under tension it immediately starts to pull the mass up again after you have re-

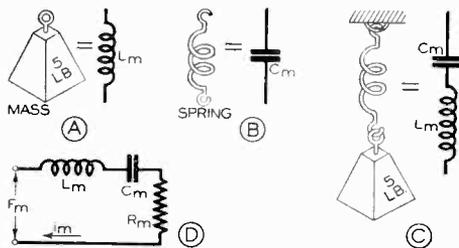


FIG. 3. These diagrams will help you to remember that mass can be considered as a mechanical inductance, that springiness or compliance of any kind (a spring) can be considered as mechanical capacitance, and that a complete mechanical vibrating system acts in much the same way as a series-resonant electrical circuit.

duced it. Having reached its original position, the mass continues moving upward because it has acquired momentum. The mass stops moving upward when it has lost this momentum; the mass then drops downward, moving past its normal stationary position again because of momentum, and continues to bob up and down in what we call mechanical vibration. This vibration continues until all of the original energy imparted to the spring and mass by pulling it down is wasted in friction. This friction occurs in two different ways here, between the mass and the air particles, and internal friction produced

by bending of the spring. If this entire system were in water or in a thick oil, there would be even greater friction between the water or oil particles and the mass than in the case of air, and the vibration would stop sooner.

We need not necessarily have five-pound weights and large, stiff springs in order to secure a mechanical vibration; the weight or mass need be only a fraction of an ounce, and the spring can have far more compliance than that shown in the illustration. No matter how small the mechanical structure happens to be, vibration can be secured if there is mass and compliance in the system.

Now you can readily see that the diaphragm in a bi-polar magnetic driving unit has mass and compliance. It will therefore vibrate. Radio engineers have found that its exact behavior during vibration is far easier to understand if the mechanical system is replaced by an equivalent mechanical circuit which has all the appearances of an electrical circuit. Figure 3D shows such a circuit for the diaphragm of the headphone unit; it will make the mechanical action of a headphone considerably easier for you to understand. Remember, however, that this is not the actual electrical circuit in the headphone; it is simply an imaginary series-resonant circuit which acts in the same way as the mechanical system.

*Natural Frequency of Vibration.* Notice that we have mechanical force  $F_m$  acting upon mechanical inductance  $L_m$ , upon mechanical capacitance  $C_m$  and upon mechanical resistance  $R_m$ . When this force is applied for a short instant of time and then removed, the system goes into vibration and me-

chanical current  $i_m$ , which you can think of as the *velocity* at which the diaphragm moves back and forth, is the result. The natural frequency of vibration of the diaphragm (or of any other physical object) will be determined by the mechanical inductance (mass) and the mechanical capacitance (compliance) of the diaphragm or other object. When a continuously vibrating force is applied, the vibration of the assembled parts will follow this force, and the amount of mechanical current will depend upon the mechanical reactance of  $L_m$  and  $C_m$

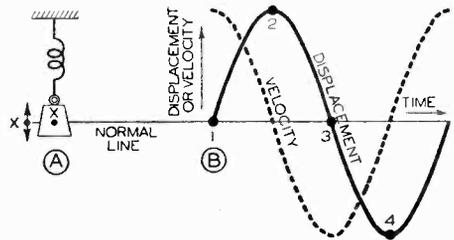


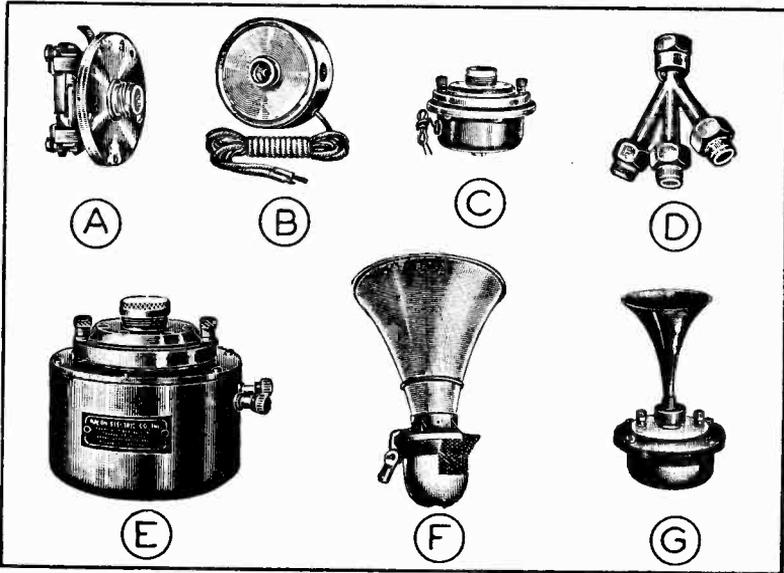
FIG. 4. A careful study of this simple diagram will help you to understand the meaning of *velocity* or *mechanical current* in connection with a loudspeaker.

and the mechanical resistance of  $R_m$ , just as in any series resonant circuit. By adjusting the frequency of the vibrating force or by changing  $L_m$  and  $C_m$  mechanically, you can make the mechanical circuit act either as a mechanical resistance, a mechanical inductance, or a mechanical capacitance.

*Velocity of a Diaphragm.* Let us get a clearer picture of this mechanical current or velocity, for it is extremely important in connection with our study of loudspeakers. Suppose we have a mass and compliance arranged as shown in Fig. 4A; we set this mechanical system into vibration. Under this condition, any particle on

this mass, such as point *X*, will be moving continually above and below its normal position or normal line. If we plotted from instant to instant the position of particle *X* with respect to this normal line, we would secure the

When particle *X* is moving across the normal line at point 1, it has a greater mile-per-hour speed than at any other point along its path; in other words, the particle has maximum velocity when it crosses the



These diagrams illustrate a number of interesting features of horn loudspeakers. *A*—Low-power magnetic driving unit of the balanced armature type, with diaphragm feeding into a standard  $\frac{3}{8}$ -inch throat for use with horn loudspeakers; *B*—balanced armature magnetic driving unit in its housing, with connecting leads clearly visible. Units like this can handle up to 2 watts of input power; *C*—electrodynamic driving unit for horn loudspeakers. Note the two field coil leads coming out of the side of the housing or pot; *D*—special three-way connecting throat which permits the attachment of three separate driving units to a single trumpet or horn; *E*—another electrodynamic driving unit for horn loudspeakers. Size and weight are general indications of power-handling ability in units of this type. A 20-pound unit will handle approximately 25 watts of input power continuously and 50 watts on peaks, while a 6-pound unit will handle about 5 watts continuously and 10 watts on peaks. If more than the rated continuous power is fed at any instant of time, the unit may rattle because of excessive diaphragm displacement. The knurled protective cap on the throat is removed when the unit is installed; *F*—dynamic driving unit employing a large diaphragm which is coupled to a trumpet horn; *G*—horn loudspeaker designed specifically for high frequencies, from 3,000 cycles to 15,000 cycles. The trumpet horn used on this unit is spun from a single piece of aluminum.

solid line curve shown in Fig. 4*B*. Yes, this plotting of displacement against time for a vibrating mechanical system results in a pure sine wave; this is why mechanical vibration of a simple mass and compliance has a simple sine wave motion.

normal line. A quarter of a cycle later, when particle *X* is at point 2, its highest position above the normal line, it actually comes to a standstill for an instant, and its velocity is zero. As particle *X* drops down toward the normal line again, its velocity in-

creases again to a maximum as it goes through point 3, and then gradually decreases to zero again at point 4. If we plot this velocity of particle  $X$  from instant to instant, we secure the dotted line curve in Fig. 4.

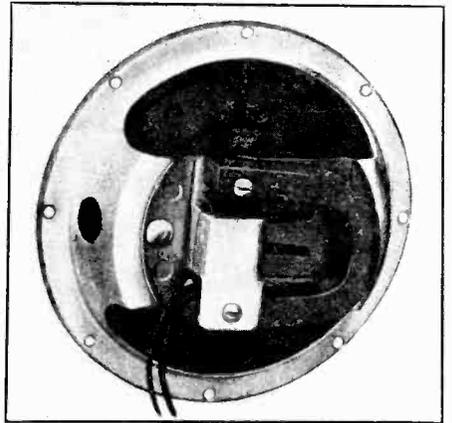
*Maximum Displacement.* When we deal with the vibrating systems used in loudspeakers, we are particularly interested in the peak or maximum displacement of the mass or of particles on it. In the bi-polar magnetic driving unit, for example, a strong coil current will give a greater displacement than a weak current, and an alternating coil current will cause the diaphragm to move in and out in an alternating manner similar to that shown in Fig. 4B. If the alternating current is of high frequency, then the diaphragm will likewise vibrate at this same high frequency.

Suppose that the diaphragm is moving the same distance (has the same peak displacement) first for a low frequency and then for a high frequency; clearly it will take less time to go through one complete cycle at the higher frequency. The speed or velocity of the diaphragm is therefore *considerably greater* at the higher frequencies of vibration.

*Velocity of a Vibrating Mass.* At a given frequency of vibration, any increase in displacement likewise means an increase in the velocity of the mass or particle. Your own experience will prove this to be true, for you know that the farther you pull the weight in Fig. 3C down from its "at rest" position, the faster will it move upward past its "at rest" position when released. The velocity of a vibrating mass therefore depends upon *the frequency of vibration* and upon *the maximum displacement of*

*the mass* from its normal position.

When the diaphragm of a headphone or loudspeaker is vibrating, it causes particles of air in its vicinity to vibrate in a similar manner. These air particles will vibrate at the same frequency as the diaphragm, and will also have the same velocity as the diaphragm at any instant. The vibration is transmitted from one air particle to another, giving rise to sound waves. It is the frequency at which



Courtesy Arlavor Mfg. Co.

This magnetic loudspeaker, available with cone diameters ranging from 3 inches to 8 inches, will be found in many small Universal A.C.-D.C. receivers. It uses a balanced armature type of driving unit. Note the U-shaped permanent magnet.

these air particles vibrate which determines the pitch or tone of the sound, and it is the *velocity* of the *air particles* which determines the power\* (loudness or volume) of the sound.

\* You know that in an electrical circuit like that in Fig. 3D, the power absorbed by the resistor is equal to the square of the current multiplied by the ohmic value of the resistor ( $P = I^2R$ ). Likewise, in a mechanical vibrating system, the square of the air velocity multiplied by the mechanical resistance gives the *mechanical power* used by the system; this represents acoustical power.

*Facts to Remember about Vibrating Systems.* It will be some time before you will be able to appreciate fully the facts just brought out in connection with the vibrating systems used in loudspeakers. For your present study of loudspeakers, it will be sufficient for you to keep in mind the following important facts:

1. Any physical object has *mass*, which can be considered as *mechanical inductance*.
2. A physical object can have *compliance* (you can call this springiness if you prefer), and this characteristic can be considered as *mechanical capacitance*.
3. The motion of any physical object is accompanied by friction, which represents power used or wasted.
4. Any physical object which has mass and compliance can be shocked into vibration by a sudden application of force; it will then vibrate at its natural frequency of vibration, which is determined by its *mechanical inductance* (mass) and its *mechanical capacitance* (compliance). Increasing either of these two values lowers the natural frequency of vibration.
5. The application of a steady alternating force to a vibrating system will cause a corresponding alternating or vibrating motion of the system at the same frequency. We can consider this mechanical vibrating system as equivalent to an electrical series resonant circuit. When the natural frequency of the system corresponds to the frequency of the applied alternating force, resonance occurs and the system acts as a mechanical resistance. At frequencies above resonance, the vibrating system acts as a mechanical inductance (a mass). At frequencies below the natural resonant frequency, the system acts as a mechanical capacitance.
6. Since mass can be considered as an inductance, and compliance as a capacitance, these parts of a mechanical

vibrating system have the additional property of mechanical reactance, which takes frequency into account.

7. The application of an alternating force to a vibrating system causes the system to vibrate with a velocity dependent upon its mechanical reactance and mechanical resistance.

## Mechanical Resonance in Headphones

Referring again to Fig. 2A, we recognize that the diaphragm is the mechanical vibrating system in this bi-polar magnetic driving unit. Even though quite small, the diaphragm has mass (weight) and compliance (springiness). When the diaphragm is set into vibration by an alternating current flowing through the coils, it causes air particles in the vicinity to vibrate in a corresponding manner. The resulting sound produced naturally represents radiated energy, and therefore the diaphragm must overcome mechanical resistance in radiating this energy. The diaphragm has a small mass and therefore its mechanical inductance is low; being quite stiff, it has a low compliance and its mechanical capacitance is likewise small. The natural frequency of vibration of the diaphragm is therefore quite high; in fact, diaphragms in high-fidelity headphone units are designed to have a natural frequency of vibration which is considerably above the highest audio frequency being reproduced. This prevents resonant peaks in the response curve of the headphone. Clearly the question of mechanical resonance is of great importance when high fidelity is required from a bi-polar magnetic driving unit such as is used in a headphone.

## Balanced Armature Type of Magnetic Driving Unit

Although the bi-polar magnetic driving unit is remarkably simple in design and can be made to have high sensitivity, it is not entirely suitable for loudspeaker applications where high sound output (high volume) is required. Application of excessive input signal voltage to this type of unit results in distortion, due either to the diaphragm striking the pole tips or to a frequency-doubling effect which occurs when the air gap is increased to permit greater volume. Furthermore, normal movements of the diaphragm cause the air gap to vary, with the result that the permanent magnetic flux also varies and a certain amount of distortion occurs. When the diaphragm is made stiff enough to prevent it from hitting the pole pieces, it then becomes quite thick, and its increased mass lowers the natural frequency of vibration. A stiff, light-weight disc would be desirable, but this cannot be easily obtained in the bi-polar type of construction.

The balanced armature type of magnetic driving unit, illustrated in Fig. 5A, overcomes most of these disadvantages. A light-weight strip of soft iron, called the armature, is pivoted in the center of a coil through which flows the A.F. signal current. This coil may have only a few turns when large A.F. currents are available, but may be built with several thousand turns for low A.F. currents. The coil does not interfere in any way with the movement of the armature. The ends of the armature move between two U-shaped pole pieces which are clamped to a U-shaped permanent magnet which is generally larger

and more powerful than the permanent magnet used in bi-polar units. The permanent magnet in this case makes one set of pole pieces permanently of north polarity and the other of south polarity. One end of the armature is rigidly linked to a diaphragm which may be made of mica, duralumin, or any other light material; if the diaphragm is not sufficiently stiff a spring steel metal strip may be attached to the other end of the armature as indicated, to increase the stiffness (reduce the compliance) of the system.

When no current is flowing through the coil, the armature is not magnetized. All four pole pieces then attract the armature equally, and we have a balanced condition in which the armature stays midway between the pole pieces. It is from this characteristic that the balanced armature magnetic driving unit gets its name. When a current  $i$  flows through the coil in the direction shown (arrow indicates electron flow), this current will magnetize the armature and give it the polarity indicated in Fig. 5A. The  $N$  pole of the armature will be attracted to the nearest  $S$  permanent pole and will be repelled by the permanent  $N$  pole. A similar action occurs at the other end of the armature, which is magnetized with south polarity. The result is that the armature twists on its pivot in a counter-clockwise direction, as indicated by the curved arrow line, pushing the diaphragm upward. When the current through the coil reverses its direction, the armature is twisted in the other direction and the diaphragm is pulled downward.

Figure 5B shows the two paths taken by the magnetic flux produced

by the armature when currents flow through the coil. These paths are entirely through soft iron, which has a low reluctance and therefore permits a large amount of flux to circulate. Notice that movement of the armature in either direction will cause one air gap in one of the paths to decrease and the other air gap in that path to increase, with the result that the total air gap for any one path remains essentially the same regardless of the position of the armature. This is a desirable condition for distortionless operation.

Now let us see what effect the position of the armature has upon the path taken by the permanent magnetic flux. Referring to Fig. 5B, let us assume that the armature is rotated counter-clockwise, so that air gaps  $g_1$  and  $g_2$  are considerably smaller than the other two. Naturally the flux coming out of the north pole of the permanent magnet will choose to take the path through pole  $N_1$  because  $g_2$  offers less reluctance than  $g_3$ . This flux will pass through the entire armature and then through gap  $g_1$  to pole  $S_2$ . The result is that poles  $N_1$  and  $S_2$  are strengthened, while poles  $N_2$  and  $S_1$  are weakened. But remember that the motion of the armature depends upon repulsion as well as attraction; an increase in the attraction on any one end of the armature is offset by a decrease in the repulsion on the same end, with the result that the effect of the permanent magnetic flux upon the armature is essentially independent of the position of the armature.

A balanced magnetic driving unit can swing through considerably greater distances than a bi-polar magnetic unit without serious distortion.

The use of large and powerful permanent magnets made of new magnetic alloys makes possible a large air gap, with resulting increase in output volume. The moving system can be made light enough to prevent mechanical resonance in the range of audio frequencies being handled.

With the balanced armature type of construction there is generally ample

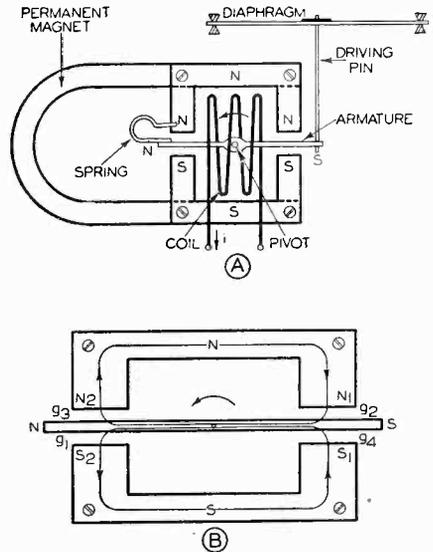


FIG. 5. These diagrams illustrate the construction and principle of operation of a balanced armature type magnetic driving unit.

room for the coil, permitting the use of larger-sized wire with fewer turns in order to reduce the coil inductance. This allows higher-frequency audio currents to flow, increasing the frequency range of the loudspeaker. In general, then, a balanced armature type magnetic driving unit gives higher output and a wider frequency range than a bi-polar magnetic driving unit. With proper design the magnetic type of driving unit can be

made to have reasonably uniform frequency response over the middle range of frequencies, from 150 cycles to 3,000 cycles.

### Dynamic Driving Units

The amount of audio sound power radiated by a given loudspeaker at a definite frequency depends upon two things: the area of the diaphragm, and the displacement or maximum movement of the diaphragm. These two factors together determine the

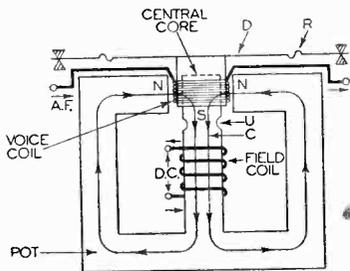


FIG. 6. Simplified diagram illustrating the operating principles of an electro dynamic driving unit.

volume of air moved and thereby determine the amount of sound produced at a particular frequency. (The in-and-out motion of a loudspeaker diaphragm is often referred to as *piston action*, for the diaphragm acts much like the piston in a water or air pump; the larger the piston area and the farther it moves, the more water or air will be moved.) The greater the volume of air which is moved by a given diaphragm, the greater is the load on the diaphragm and the greater is its mechanical resistance.

With the maximum displacement of the diaphragm limited by the air gap length in magnetic types of loud-

speaker driving units, engineers turned to an entirely new type of electro-mechanical driving system, known as the *dynamic loudspeaker*. A study of the simplified diagram in Fig. 6 will show the operating principles of this dynamic unit, which consists simply of a small coil, carrying signal current, which is attached to a large diaphragm and which is located in a strong magnetic field.

Bear in mind that the diagram in Fig. 6 represents a cross section of a cylindrical unit. The permanent magnetic field is provided by sending direct current through a field coil wound around the center cylindrical soft iron core. An iron shell or *pot* surrounds this core and coil to complete the magnetic path up to the air gap in which is located the voice coil; this coil consists of a few turns of fine wire wound on a bakelite-impregnated paper or mica coil form. It is essential that the air gap be kept as small as possible without having the coil or its form rubbing against the cylindrical pole piece or the central core. To reduce the thickness of this voice coil, a spiral groove is often cut in the form for the wires. Mica is often used for the coil form because it will not warp when subjected to excessive heat due to resistance losses in the voice coil. The leads to the voice coil must, of course, be exceptionally flexible and arranged in such a way that they do not interfere with the movement of either the coil or the diaphragm.

Diaphragm *D*, to which the voice coil is attached, is usually made of stiff duralumin, in order that the coil will return to its normal no-current position when the voice coil current drops to zero. The diaphragm is cor-

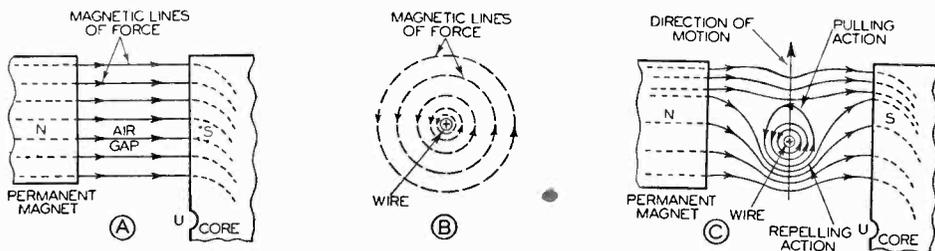
rugated around its edges, such as at *R*, in order that the inner portion of the diaphragm can move with the voice coil even though the outer edges are rigidly clamped in position.

Now let us see how this dynamic driving unit operates. First of all, I want to point out that the operation is the same regardless of whether the permanent magnetic flux through the air gap is produced by a coil carrying direct current or by a powerful permanent magnet having essentially the same shape as the soft iron core shown

is present in the vicinity of the voice coil at all times.

As you know, any wire which is carrying current will have around it a circular magnetic field like that shown in Fig. 7*B*. If the electron flow through this wire is into the paper, the magnetic lines of force will have the direction shown.

When this single current-carrying wire is in the air gap, we have the conditions shown in Fig. 7*C*. The two magnetic fields react with each other. Notice that the magnetic lines



**FIG. 7.** What makes the voice coil move in a dynamic loudspeaker? These diagrams will tell you. The fixed magnetic field is shown at *A*, while the magnetic field around a voice coil wire is shown at *B*. The interaction of these two fields, as at *C*, results in motion of the voice coil. The + symbol in the wire indicates that the electron current is flowing into the paper (away from you) through the wire.

in Fig. 6. It is the interaction between the permanent magnetic flux and the flux produced by the voice coil current which produces motion; the principle is the same as that applying to electric motors.

To understand just why the voice coil should move when current passes through it, consider only one part of the air gap, as shown in Fig. 7*A*. The magnetic lines of force here come out of the ring-shaped *N* pole piece and go through the air gap to the cylindrical center core which is made an *S* pole by the permanent magnet or by the field coil. This is the field which

of force underneath the wire are in the same direction and reinforce each other, while those above the wire are in opposite directions and tend to cancel out. There is a crowding of flux lines below the wire and a less-than-normal number above the wire; the motion of the wire will be such as to redistribute the flux more uniformly, and the wire will therefore move upward as indicated by the long arrow. If the current through the wire reverses, the magnetic field of the wire will be reversed in direction and the wire will be forced downward. With a voice coil in the air gap, this same

action takes place at all points in the air gap and on all turns of the voice coil, with the result that the entire voice coil is either moved upward or downward depending upon the direction of current flow.

The strength or magnitude of the force acting upon the voice coil depends upon three things: 1, *the strength of the fixed magnetic field existing in the air gap*; 2, *the length of the wire used for the voice coil*; 3, *the voice coil current*. Increasing any one or all of these three factors increases the force acting upon the voice coil.

If the force acting on the voice coil is to be proportional to the voice coil current at all times, the magnetic flux must be of uniform strength throughout the air gap. You will generally find a groove cut into the central core at *U* to prevent the flux from thinning out at the lower edge of the air gap; furthermore, the length of the voice coil will generally be less than the height of the air gap.

The weight of the diaphragm and voice coil assembly should be low, and the diaphragm should have sufficient stiffness so that mechanical resonance will occur above the audible frequency range.

Since the voice coil in a dynamic unit has only a few turns of wire (from 1 to 50 turns), its reactance is quite low even for the highest audio frequencies; the high-frequency response of a dynamic loudspeaker is therefore not limited by the voice coil inductance, but rather by the mass of the moving system. Diaphragm movements of one-half inch are not uncommon in dynamic loudspeaker units which are capable of handling large amounts of audio power.

## Condenser Driving Units

Although condenser loudspeakers are relatively little used today, they have certain special advantages which make it advisable for you to be familiar with their construction and operating principles. They depend for their operation upon the attraction of a positive charge for a negative charge, and might therefore be called electrostatic units.

In a condenser driving unit a heavy, stationary plate is charged with one polarity and a light, movable plate, mounted a short distance away, is charged with opposite polarity. The separation between plates is uniform and is made as short as possible without causing the charges to equalize by jumping across the air gap. As you know, these two plates constitute a condenser; increasing the areas of the plates and reducing the distance between them increases the capacity of the condenser and thereby increases the charge which can be stored on the plates. For a given electrical charge on the plates, reducing the air gap between them increases the force of attraction between the plates.

Application of a high-voltage A.F. signal to the plates is not enough to give the desired operation, for the plates will always be of opposite polarity and will therefore always attract each other (except when the voltage is zero, which occurs twice during each cycle). As a result, we have a frequency-doubling effect just as would occur in a bi-polar magnetic unit if there were no permanent magnet. The remedy is quite simple and comparable to that used in the head-phone unit; a D.C. or polarizing voltage is applied to the plates, so that

there is a constant attraction between the plates. The A.F. voltage acts in series with this fixed or polarizing voltage, increasing or reducing the attraction. In this way the displacement of the light, movable plate corresponds to the wave form of the A.F. signal.

In practical condenser loudspeakers the movable plate usually consists of metal foil which is cemented over a sheet of thin, stretched rubber which is supported at its edges. A D.C. polarizing voltage of about 500 volts is applied to the plates. The essen-

limited life, becoming hard or deteriorating with time. The low-frequency response of a condenser unit is rather poor, and some provision for correcting this must ordinarily be made in the audio amplifier. The high-frequency response is quite good, however, for the reactance of the loudspeaker unit decreases with frequency and the mass of the moving system is not great enough to prevent high-frequency movement. High-frequency response is further improved by punching holes in the stationary plate to permit free movement of air

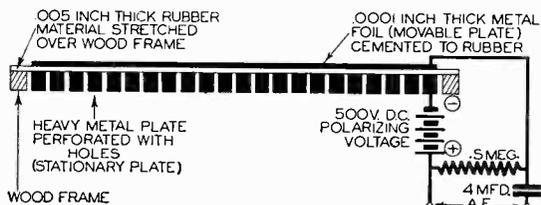


FIG. 8. Cross-section view of a typical condenser loudspeaker, showing essential constructional features. The plates were usually made about 8" x 12" in size.

tial features of this construction are shown in Fig. 8.

Although condenser loudspeakers are simple in construction and comparatively inexpensive, their power output is limited because of the extremely limited displacement of the moving plate (the maximum obtainable displacement is always much less than the thickness of the stretched rubber sheet). When increased output is desired, it is necessary to use several condenser loudspeaker units arranged side by side and connected in parallel; each unit ordinarily is about one foot square. Another drawback is the fact that the rubber used as an insulating material has a

through this plate. Sounds travel out from a condenser loudspeaker at right angles to its moving plate, and the units therefore have desirable directional characteristics.

### Crystal Driving Units

The basic action of a crystal loudspeaker is as follows: The application of a voltage of given polarity to the faces of specially-cut crystal slabs causes a change in shape which is used to drive a diaphragm or cone.

Rochelle salt crystals, which can be grown artificially from chemical solutions in quite large sizes and in a relatively short time, are most often

used in crystal loudspeakers. A typical Rochelle salt crystal slab might be cut to a size of about 2.5 inches square and  $\frac{1}{8}$  inch thick, as shown in Fig. 9A. The slab must be cut from the crystal in a definite manner in order to secure a maximum change in shape when voltage is applied. When this crystal element is rigidly fastened at three of its corners, *A*, *B* and *C*, the application of an electric charge to its faces by means of tinfoil sheets cemented on each side will cause length *A-D* to increase or decrease, depending upon the manner in which the slab was originally cut from the complete crystal and the polarity of the applied charge. This change in length is of course quite small, but engineers have discovered how to utilize it to greatest advantage.

In the practical crystal loudspeaker, two slabs are cemented together. One increases in length when a voltage with a given polarity is applied, while the other decreases in length; the result is that the combined slab bends at its free corner through a distance far greater than the original change in length of either crystal. The principle is much like that of the bi-metallic strip so widely used in some thermometers (two different metals are welded together in the form of a thin rectangular strip; one expands more than the other when temperature increases, with the result that the strip bends or curls with changes in temperature).

When two crystal elements are cemented together, giving a two-crystal unit, we have what is known as a *bi-morph cell*. The construction of the usual bi-morph cell is as shown in Fig. 9B, where tinfoil sheets are cemented to each face of each crystal

before the crystals are cemented together; in this way each crystal can be so charged that one will expand while the other contracts.

The construction of a typical crystal driving unit is shown in Fig. 9C. Three corners of the bi-morph crystal cell are rigidly supported between rubber pads, while a metal cap and a driving link or lever are cemented to the fourth corner. The entire cell is

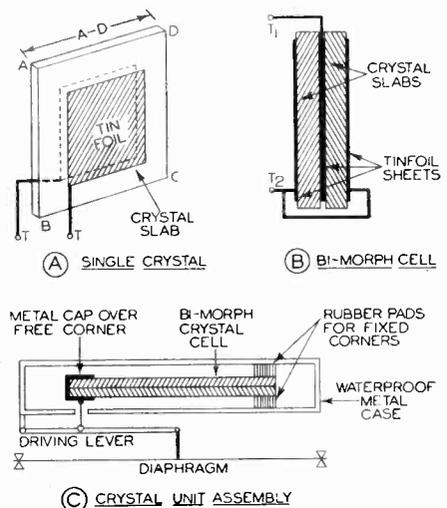


FIG. 9. How does a crystal loudspeaker work? These three diagrams give you the complete story.

mounted in a water-proof housing, for Rochelle salt crystals dissolve in water and deteriorate in the presence of moisture. They are also affected by high temperatures and must not be used in locations hotter than  $125^{\circ}$  F. The motion of the free edge of the crystal unit is transferred to the diaphragm by the mechanical link or lever system.

Crystal driving units are essentially condensers; in fact, a 2.5 inch square bi-morph crystal cell has a capacity

of about .03 mfd. This means that its reactance will decrease at the higher frequencies, and it will therefore have a very good high-frequency response. Although crystal driving units can be designed to reproduce the entire audio frequency range, they are primarily used for the reproduction of the higher audio frequencies.

### Loudspeakers Can Also Serve as Microphones

Before considering further details of sound-reproducing units, it is worth noting that any device for converting A.F. signals to sound can also be operated in reverse, so that sounds produced in the vicinity of the diaphragm or moving system will develop an electrical signal of corresponding wave form; in other words, a loudspeaker will also operate as a microphone. The better the fidelity of the loudspeaker as a sound reproducer, the better will be its performance as a microphone. One basic difference must be kept in mind; sound reproducers are designed to furnish large sound output powers, with the diaphragm or moving element pushing a large volume of air, while microphones are usually designed to respond to movements of small volumes of air or to weak sound inputs. In modern intercommunicating systems, small dynamic loudspeakers are widely used as microphones.

### Action of a Horn as a Loudspeaker Coupling Unit

We have now considered the various methods ordinarily used for setting the diaphragm of a loudspeaker into vibration. Naturally we want this vibration of the diaphragm to pro-

duce a large sound output. Since the horn is a common air coupling system used for this purpose, it will be taken up next.

As you already know, the acoustical characteristics of a vibrating diaphragm in a driving unit may be represented by an electrical circuit like that shown in Fig. 10. The mass of the diaphragm is represented by  $L_m$  (mechanical inductance), the compliance or springiness of the diaphragm is represented by  $C_m$  (mechanical capacitance), and the mechanical resistance of the diaphragm is represented by  $R_m$ . The mechanical force applied to the diaphragm by

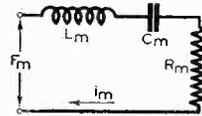


FIG. 10. Equivalent mechanical circuit of a vibrating diaphragm in a loudspeaker.

the driving unit is represented by  $F_m$ ; for simplicity we can assume that it is an alternating force having a simple sine wave characteristic. We can then vary the frequency of this source and determine the effects of these variations on the performance of our loudspeaker.

The simple series circuit in Fig. 10 will serve as our guide for analyzing the action of a vibrating diaphragm. The applied force  $F_m$  causes the diaphragm to vibrate, and at any frequency the velocity of vibration will be governed by the mechanical reactances of  $L_m$  and  $C_m$  and by the mechanical resistance of  $R_m$ . When the two mechanical reactances are exactly equal, their effects cancel and

we have mechanical resonance, with only  $R_m$  in the circuit to limit mechanical current flow. This mechanical current ( $i_m$ ) represents the velocity of the diaphragm, and consequently also represents the velocity of air particles in the vicinity of the diaphragm. It is this air velocity which contributes toward the desired sound output. At input frequencies below the resonant frequency of the diaphragm, the mechanical reactance of  $C_m$  will be greater than that of  $L_m$ , and the diaphragm will act as a mechanical capacitance in series with a mechanical resistance. At above-resonance frequencies, the diaphragm will act as a mechanical inductance in series with a mechanical resistance but neither the mass (inductance) nor the compliance (capacity) involves loss of power; they merely offer opposition to changes in diaphragm position. It is the mechanical power absorbed by  $R_m$  which determines the loudspeaker sound output.

From Fig. 10 it is obvious that increasing the value of  $R_m$  increases the amount of mechanical power absorbed and therefore increases the sound output of a loudspeaker. Likewise increasing the driving force  $F_m$  will cause greater mechanical current flow, increasing the diaphragm displacement and velocity and thereby increasing the air velocity and the sound output power.

In the case of crystal or condenser driving units, the mechanical force  $F_m$  can be increased directly by increasing the A.F. voltage applied to the driving unit, while in other types of driving units an increase in the applied A.F. voltage will cause greater A.F. current to flow and this will increase the mechanical force  $F_m$ . In-

creased mechanical force  $F_m$  obviously will cause greater diaphragm displacement and greater air velocity, with more power being absorbed by  $R_m$ .

If a driving unit of a loudspeaker were placed in a vacuum,  $R_m$  would merely represent losses due to friction in the diaphragm itself; there would be no useful sound output since no air would be moved. The only limitations to the velocity of motion of the diaphragm would be this very low value of  $R_m$  and the difference between the mechanical reactances of  $L_m$  and  $C_m$ ; at resonance these mechanical reactances would balance out, and we could expect extremely large velocities and displacements of the diaphragm. Considering these facts from a practical standpoint, we can easily see that with a loudspeaker operating outside the gondola of a balloon high in the stratosphere (where atmospheric conditions approach a vacuum), the displacement of the diaphragm might be large enough to ruin the driving unit.

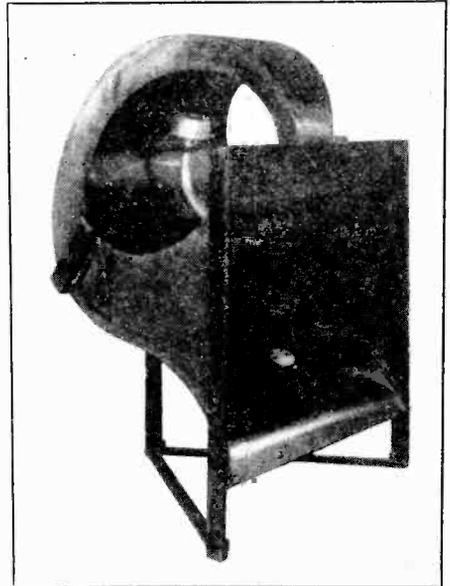
When the diaphragm is allowed to act upon air, a load is applied or coupled to the diaphragm and this is equivalent to increasing the value of  $R_m$ . This is a desirable condition, for it allows us to secure a greater amount of useful work from the loudspeaker.

*Measuring Loudspeaker Efficiency.*  
When loudspeaker engineers desire to check the efficiency of a particular loudspeaker as a sound reproducer, they measure the electrical resistance of the loudspeaker input terminals under the following two conditions: 1, with the driving unit blocked, so that the diaphragm and the other moving elements cannot move; this gives the

resistance due to electrical losses in the driving unit only, and is known as the *blocked resistance* of the loudspeaker; 2, with the moving elements free to move.\* These resistance measurements are repeated for a number of frequencies over the entire audio range. The difference between the two measured resistance values at each audio frequency then represents the additional resistance due to the production of sound; this is called *motional resistance*. The ratio of the motional resistance to the total resistance when the diaphragm is free determines the loudspeaker efficiency; multiplying this ratio value by 100 gives the percentage efficiency. If, for example, these tests show that a loudspeaker has an efficiency of 10% at 1,000 cycles, the engineer knows that he will only get 2 watts of acoustical power out of the loudspeaker when he feeds 20 watts of electrical power into the loudspeaker.

*Loading the Diaphragm.* Loudspeaker engineers quickly learned that small diaphragms operating in free air without horns or other air coupling systems gave little sound output regardless of their velocity or displacement; measurements of loudspeaker efficiency verified this by indicating a very low motional resistance. The reason for this is easy to see; air directly in front of the diaphragm is alternately compressed and thinned out, and the same effect occurs at the back of the diaphragm. Whenever there is compression in front, there is thinning out or rare-

faction at the back. Since compressed air tries to spread out or relay its effects to adjacent air particles in the easiest way possible, it merely moves around the diaphragm to the rear to equalize the thin air there, and only a very small volume of air is thus actually set into vibration by the diaphragm.



Curled exponential horn with circular cross-section near throat; the  $\frac{3}{8}$ " diameter throat fits into a standard driving unit. Note the square cross-section at the mouth and the rectangular cross-section in the middle region. The shape of cross-section is unimportant as long as the cross-sectional area of the horn increases in an exponential manner from throat to mouth.

One way of overcoming this trouble is to use a large diaphragm or cone which will set a large volume of air into motion. Another way is to prevent the air in front of a small diaphragm from reacting on the air in back of the diaphragm. Either of these procedures increases the load on the diaphragm, thereby increasing the

\* In order to secure a true A.C. resistance measurement at the chosen frequency, the reactance of the voice coil is tuned out by means of a condenser in both cases.

motional resistance, producing more acoustical output and increasing the loudspeaker efficiency.

The mounting of a megaphone or conical-shaped horn like that shown in Fig. 11A around the diaphragm of a driving unit was an early attempt to load the diaphragm by preventing air from escaping behind it, and also was intended to direct the moving air toward the listener. Results were not satisfactory, however, for the use of a conical-shaped horn did not take into account the fact that air in motion travels in definite natural paths. We have a small amount of air vi-

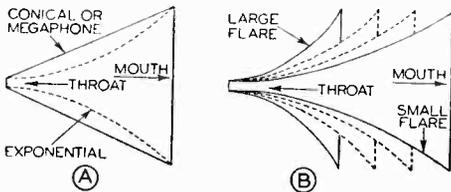


FIG. 11. Comparison of shapes of a conical horn with various types of exponential horns used with loudspeaker driving units.

brating at high velocity near the diaphragm; what we really want is a horn which will cause a greater amount of air a farther distance out to vibrate at a somewhat lower velocity, this process continuing until at the opening of the horn a very large volume of air will be vibrating at a low velocity. There must be no dead spots in the horn (regions unaffected by the vibration of the diaphragm); there must be no cancellation of effects along the length of the horn, no turbulence of air and no wasted energy. If all this is realized, the entire amount of air in the horn will be properly coupled to the driving unit, increasing the motional resist-

ance and giving the desired increase in efficiency. Securing this desirable condition is known as matching the impedance of the air with the impedance of the driving unit; it serves to load the unit effectively and give a maximum of useful sound power output.

*Exponential Horns.* No doubt you are already familiar with loudspeaker horns. At the small end or *throat* of a horn the cross-sectional area is small, while at the large end or *mouth* the cross-sectional area is large and is a maximum. It makes little difference whether the cross-sectional shape of a horn is square, rectangular or circular; it is the manner in which the cross-sectional area increases from the throat to the mouth which determines the effectiveness of the coupling between the driving unit and the air in space. When this area increases according to what mathematicians call an *exponential* formula, giving a graceful curve like that shown by the dotted lines in Fig. 11A, the coupling will be most effective.

For a given throat area and a given mouth area, there can be any number of exponential horn shapes, but each would have different length and a different flare, as indicated in Fig. 11B. The *flare* of a horn refers to the amount of spreading outward in a given horn length; the shortest horn shown in Fig. 11B thus has the greatest flare. The amount of flare has a definite and important effect upon the frequency range of a horn loudspeaker, for it determines the efficiency of coupling at various frequencies in the audio range.

The formula used by loudspeaker design engineers in computing the shapes of various forms of **exponential**

horns involves considerable mathematical knowledge, and will not be taken up here since it is seldom if ever needed by the practical man. There are certain facts to be derived from this formula which are of practical interest, however. The smaller

the horns having smaller flares. Remember—a horn which is to provide effective coupling between the air and a loudspeaker driving unit at low frequencies should have a small flare.

*Mouth Area.* Although the area of the mouth of a horn does not have an



*Courtesy Utah Radio Products Co.*

**Assembly line in the Utah loudspeaker factory. Finished electrodynamic loudspeakers are placed on the conveyor belt which moves down the center of the long table.**

the flare (amount of spreading out) of the horn, the lower will be the frequency which can effectively be coupled to the driving unit and the longer will be the horn for a definite throat and mouth area. That horn in Fig. 11B having the largest flare consequently cannot be expected to reproduce low frequencies as well as

appreciable effect upon the cut-off frequency of the horn (the lowest audio frequency which it will radiate effectively), the mouth area does affect the uniformity of the loudspeaker response curve. A small mouth results in a sudden change in the velocity of air particles as they leave the horn, causing some of the sound

waves to be reflected back into the horn. This gives rise to peaks and dips in the loading of the driving unit, causing a ragged sound output. In general, for a horn with a circular cross-section, the diameter of the mouth should be equal to at least one-fourth the wavelength of the lowest frequency which is to be reproduced (the wavelength of sound is equal to its velocity of travel, 1,089 feet per

*Throat Area.* The throat area of a horn is controlled essentially by the size of the diaphragm. It is common practice to design driving units for use with a circular throat having a diameter of  $\frac{5}{8}$  inch, in order that standard horns and driving units can be used interchangeably. With a considerably larger throat area, a short horn with a small flare would provide a fairly low-frequency cut-off, but ex-



*Courtesy Vac-O-Grip Co.*

Four dynamic cone loudspeakers with single-piece spun aluminum exponential horns are here mounted on an ingenious carrying frame which can be used on any passenger car without damaging the roof. Rubber vacuum cups beneath each support prevent the assembly from sliding off while the car is in motion.

second, divided by its frequency in cycles; at 50 cycles, then, the wavelength would be  $1,089 \div 50$ , or about 22 feet). Thus a horn which is to handle frequencies down to 50 cycles should have a mouth diameter equal to at least  $22 \div 4$ , or 5.5 feet. The length of this horn would be quite long, for the small flare required to secure efficient coupling at this low audio frequency would make necessary the long length in order to secure the required mouth area.

perience has shown that such a horn would not give efficient coupling at higher frequencies. With a small diaphragm and the conventional small throat, good high-frequency output is possible but the movement or displacement of the diaphragm must be increased in order to secure good bass output. Yes, there are plenty of tough problems confronting the loudspeaker designer.

*Construction of Horn.* Any material is satisfactory for horn con-

struction as long as it is dead as far as sound is concerned; in other words, the material must not vibrate under the influence of sound waves. Aluminum is an excellent material, but when used for the longer horns, it should be reinforced with ribs on the outside to prevent the large surfaces from vibrating in unison with the diaphragm. Plywood is widely used for horns of rectangular cross sections. Some horns are made of papier mache (molded paper fiber), while others are made of layers of cloth impregnated

section which screws onto the driving unit is generally spun or cast from aluminum, and has the same form of exponential curve as the remaining sections of that particular horn.

Where space prevents the use of a long, straight horn having the desired characteristics (a horn length of 18 feet is by no means unusual where frequencies down to 50 cycles must be reproduced), the horn may be curled in the manner shown in Fig. 12B. The sharpest bend in the horn must not cancel any of the high-frequency

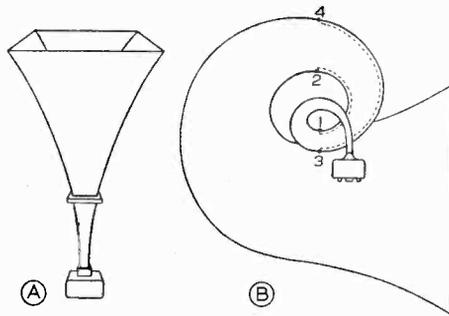


FIG. 12. Straight and curled exponential horns having standard  $\frac{3}{8}$ " throats.

with a special glue or binder and formed to the desired shape. Horns which must withstand outdoor weather conditions generally are of sturdy metal construction; when plywood or some other non-metallic material is used, it should be treated to withstand moisture.

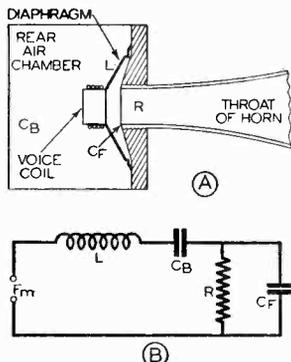
As a rule, a straight horn like that shown in Fig. 12A will give better results than a curled horn like that in Fig. 12B. It is common practice to build up a large horn in sections, different constructional procedures being followed for each type and size of horn and each type of material. That

sounds. If the difference in length between sound paths 3-4 and 1-2 in Fig. 12B, around a sharp bend, is less than one-half the wavelength of the highest frequency to be reproduced, the cancellation of sound will not be serious. Cancellation occurs whenever two sound waves become 180° out of phase, and this can occur when one wave travels one-half wavelength more than the other. Curled horns can generally be designed to meet any reasonable space requirements while still observing the flare, mouth area and curvature specifications required for a desired performance.

**Horn Ratings.** If a horn is able to resist heavy sound pressures without self-vibration, it can handle unlimited sound output power. Good horns are therefore rated according to frequency range and *not* according to power-handling ability. Either high- or low-power driving units may be attached to any particular horn. Oftentimes several driving units are used on a single horn by redesigning the throat end of the horn or by inserting the proper type of connecting unit. In general, a long horn with a small flare and a large mouth will have a wide frequency range and will have high efficiency.

### Design of Dynamic Driving Units for Horn Loudspeakers

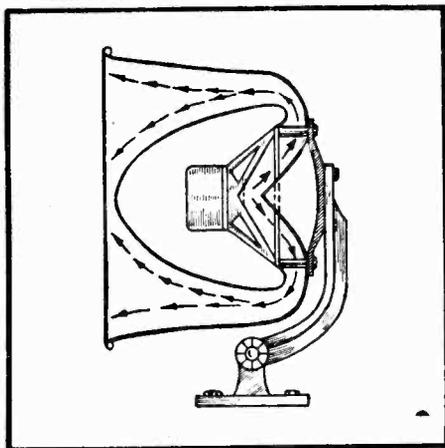
In the high-power dynamic driving units used with horn loudspeakers, the



**FIG. 13.** Method of mounting horn with  $\frac{3}{8}$ " throat on driving unit having a diaphragm considerably greater in diameter than  $\frac{3}{8}$ ", and equivalent mechanical circuit for such a combination.

diaphragm is generally made considerably larger than the  $\frac{3}{8}$  inch throat diameter of the horn, in order to move a larger volume of air and secure a higher initial air velocity. The ar-

rangement of a dynamic driving unit might be as shown in Fig. 13A. The diaphragm is made in the shape of a cone, to prevent buckling when a



Marine-type mounting of dynamic cone loudspeaker. This weather-proof unit, made by Atlas Sound Corp., gives excellent distribution of sound and at the same time is capable of withstanding the extremely severe weather conditions encountered on board ships, on sound trucks and even on fire trucks. Tests have shown that a stream of water can be played directly into the mouth of the loudspeaker without affecting loudspeaker operation. The horn is made from heavy-gauge aluminum and steel spinings. Arrow lines show paths of sound waves.

strong force is applied at its center by the voice coil.

The mounting of the diaphragm is such that it has a certain amount of natural springiness, which returns it to a normal position when no current flows through the voice coil, or the cone and the chamber in back of it are made air-tight so that the air space in back of the diaphragm will act as a cushion or spring which returns the diaphragm to its normal position. The diaphragm therefore has mechanical capacitance. Furthermore, since the diaphragm and voice coil together have a certain amount of mass, they also have mechanical inductance. The experience of loud-

speaker engineers has shown that these act in series, as shown in Fig. 13B. The load which is placed upon the cone-shaped diaphragm by the exponential horn can be represented as a mechanical resistance acting in series with the mechanical inductance and mechanical capacitance. We must not overlook the air chamber between the cone and the surface directly in front of it which supports the throat end of the horn. This air

the compliance of the rear chamber ( $C_B$ ) form a series resonant circuit. Resonance will occur at some particular frequency, and at resonance the input force  $F_m$  will depend upon the length of the wire in the voice coil, upon the voice coil current and upon the flux density in the air gap surrounding the voice coil. Furthermore, at resonance the input force will be acting solely upon mechanical resistance  $R$ , and practically all of the in-



*Courtesy Racon Electric Co., Inc.*

Battery of trumpet horn loudspeakers set up in front of the Hotel Astor at Times Square, New York City for a special event. This arrangement for radiating sound equally well in all directions requires a number of horns, with separate driving units for each.

space can be considered an extra mechanical capacitance  $C_F$  acting in parallel with the mechanical resistance.

We know how an electrical circuit resembling that in Fig. 13B will behave under various conditions, since it is a common radio circuit, and consequently we can predict the behavior of our loudspeaker by studying this circuit.

Assuming that the value of  $C_F$  is small, as it ordinarily is, we can see that the mass of the cone ( $L$ ) and

put force will serve to produce useful sounds (assuming a perfect exponential horn). The loudspeaker designer can select the values of  $L$  and  $C_B$  so that mechanical resonance occurs at a low audio frequency, thus providing reinforcement of bass notes, or at some intermediate frequency. This is one way of securing a desired frequency response for a loudspeaker.

At high frequencies the mechanical reactance of  $C_F$  becomes important. Referring to our equivalent mechanical circuit in Fig. 13B,  $C_F$  by-passes

high-frequency signals around the load, and consequently in the loudspeaker this air chamber ahead of the cone definitely suppresses the high audio frequencies. The volume of air in this chamber determines to a considerable extent its mechanical capacity, and therefore the effects of this chamber can be reduced by making it as small as possible while still allowing ample room for cone movement. The shape of this air chamber

account when designing the driving unit for a horn loudspeaker.

A number of typical mounts for the throat and diaphragm of a dynamic horn loudspeaker are shown in Fig. 14. In each case the air chamber ahead of the diaphragm or cone is designed to give desired performance while reducing undesirable effects. When a wide range of sound frequencies is to be reproduced, it is generally best to use two loudspeak-

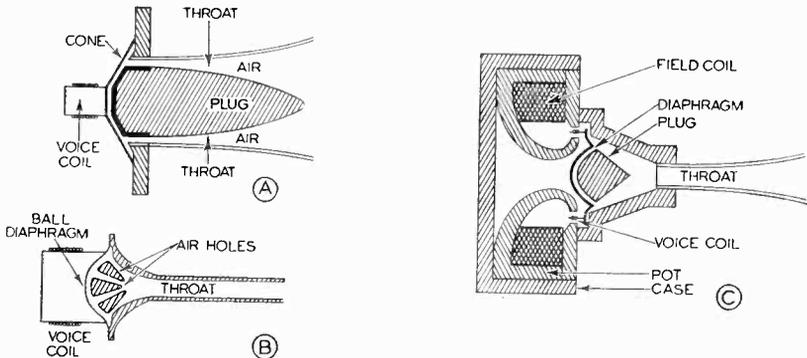


FIG. 14. Constructional features of three typical dynamic driving units are shown here. *A*: Unit employing a ring-shaped air chamber between the cone and the throat (secured by mounting a plug in the throat). The chamber is carefully shaped to reduce cancellation of sound waves at the higher frequencies. *B*: Unit using multiple outlet paths from diaphragm to throat (secured by forming air holes in a plug) to release the back pressure. *C*: Complete dynamic driving unit energizing a modification of the ring-shaped air chamber shown at *A*.

alongside the throat is important for another reason; if the shape is such that air in the chamber can take several paths, of varying length, to the throat of the horn, it is possible that some of the high-frequency sounds will reach the throat out of phase and will cancel. In high-power horns the air pressure in this chamber may be tremendous; air cannot be compressed uniformly under these conditions, and distorted wave forms are the result. All these factors must be taken into

ers; one loudspeaker for frequencies from 50 cycles to 6,000 cycles, called a "woofer," and another loudspeaker for frequencies from 6,000 cycles to 15,000 cycles, called a "tweeter."

The science of horn loudspeaker design has now advanced to the point where efficiencies of up to 50% can be expected, with reasonably flat response. This means that the best horn loudspeakers will deliver 5 watts of sound power for each 10 watts of electrical input to the voice coil.

## TEST QUESTIONS

Be sure to number your Answer Sheet 26FR-1.

Place your Student Number on every Answer Sheet.

Never hold up one set of lesson answers until you have another ready to send in. Send each lesson in by itself before you start on the next lesson. In this way we will be able to work together much more closely, you'll get more out of your Course, and you will receive the best possible lesson service.

1. Name the three sections of a complete loudspeaker.
2. Into what four general groups can loudspeakers be divided?
3. How is the undesirable frequency-doubling effect eliminated in a bi-polar headphone unit?
4. What does mass in a mechanical vibrating system correspond to in an electrical oscillating circuit?
5. What determines the natural frequency of vibration of any physical object?
6. What three things determine the force acting upon the voice coil of a dynamic loudspeaker?
7. What is the basic action of a crystal loudspeaker?
8. What is meant by the flare of a horn?
9. If a horn is to provide effective coupling between the air and a loudspeaker driving unit at low frequencies, should a large or a small flare be used?
10. Are horns rated according to power-handling ability?

## AT THE END OF THE RAINBOW

The only pot of gold you'll find at the end of the rainbow is the one which you put there yourself.

Now, when your best earning years are still ahead, is the time for you to fill that pot of gold. You're an N.R.I. student—you're carrying the ball down the field right now for a touchdown—and everything favors you to make the goal you have in mind.

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J. E. SMITH

**HOW CONE-TYPE  
LOUDSPEAKERS WORK**

27FR-2



**NATIONAL RADIO INSTITUTE**

**WASHINGTON, D. C.**

**ESTABLISHED 1914**

# STUDY SCHEDULE No. 27

For each study step, read the assigned pages first at your usual speed, then reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind, then answer the Lesson Questions specified for that step. Study each other step in this same way.

- 1. Cone Loudspeakers; Dynamic Loudspeaker Design**  
Problems - - - - - **Pages 1-7**  
Shape and Size of Cones; Cone Mountings; Cones of Dynamic Loudspeakers; "Tweeter" Cones; The Spider; Mechanical Resonance in Dynamic Loudspeakers; Piston Action; Double Voice Coils. Answer Lesson Questions 1, 2, 3 and 4.
- 2. Baffles; Box Baffles** - - - - - **Pages 7-16**  
Purpose of a Baffle; Irregular-Shaped Baffles; Mechanical Characteristics of Box Baffles; Equivalent Electrical Circuit of a Loudspeaker-Baffle Assembly; Natural Resonant Frequency; Infinite Baffles; Standing Waves; Bass-Reinforcing Box Baffles; Stromberg-Carlson Acoustical Labyrinth; Special RCA Box Baffles; Acoustic Clarifiers. Answer Lesson Questions 5, 6, 7 and 8.
- 3. Sound Diffusers; Horn-Shaped Baffles** - - - - - **Pages 16-19**  
Polar Radiation Patterns; Deflecting Vanes; Tweeter and Woofer Loudspeakers; Types of Horn-Shaped Baffles; Loudspeaker Efficiencies; Exponential Horns. Answer Lesson Question 9.
- 4. Loudspeaker Impedance** - - - - - **Pages 19-24**  
Mechanical Inductance, Capacitance and Resistance of a Loudspeaker; Electrical Impedance of Loudspeaker; Determining Loudspeaker Impedance; Matching, Typical Loudspeaker Matching Circuits; Replacement of Output Transformers. Give special attention to the last three subjects here, because they are important practical applications of the theoretical information in the first part of this study step.
- 5. Field Coil Connections; Acoustical Problems of High-Fidelity Reproduction** - - - - - **Pages 24-28**  
Use of Field Coil as Filter Choke; Types of Field Coil Connections; Separate Field Supplies; Elimination of Hum; Shading Ring; Hum-Bucking Coil; High-Fidelity Two-Loudspeaker Systems and their Response Curves; Room Acoustics; Loudspeaker Replacement Hints. Answer Lesson Question 10.
- 6. Mail Your Answers for this Lesson to N. R. I. for Grading.**
- 7. Start Studying the Next Lesson.**

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1949 Edition

# HOW CONE-TYPE LOUSPEAKERS WORK

## Cone Loudspeakers

**L** OUSPEAKERS of the horn type, having the required wide frequency range for high-fidelity reproduction, are too bulky for use with the average home radio receiver even when curled to occupy a minimum of space. Engineers recognized this fact soon after loudspeakers first came into use, and eliminated the horn entirely by increasing the size of the diaphragm and setting a large amount of air directly into motion.

Since a large, flat diaphragm will buckle or bend readily even when driven from its center, loudspeaker engineers shaped this diaphragm in the form of a cone, attaching the driving unit to its apex or point. The outer edges of the cone are free to move, being held in alignment by a soft leather ring or washer cemented to the edges of the cone and supported by a circular steel frame. The entire cone moves in and out in unison with the driving unit, at least for low frequencies.

In the early days of radio, a cone as large as 36 inches in diameter was quite common, being used to secure good low-frequency output, but the high-frequency response of these cones was irregular. Today cones are generally from 6 to 14 inches in diameter; because loudspeaker engineers now have a better understanding of the exact operation of a cone loudspeaker, remarkably flat response characteristics are being secured.

The cone arrangement of a typical cone loudspeaker is shown in Fig. 1. Observe that the frame of the cone is held against a flat surface known as a baffle, which may be either of wood,

plywood, fiber board or other sound-absorbing material in which a circular hole the same diameter as the cone has been cut. This baffle board serves to prevent sound waves produced at the rear of the cone from interfering with sounds produced at the front of the cone. For the present we will assume that the baffle is so large that no interference or can-

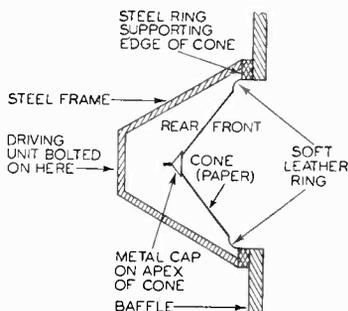


FIG. 1. Cross-section view of a cone loudspeaker and its supporting structure, as mounted on a wood baffle. Driving unit is not shown.

cellation of sound can take place; later in this text-book we will consider baffle design in detail.

In general, cone loudspeakers will be found with balanced armature magnetic driving units, with electrodynamic or permanent magnet dynamic driving units, and with crystal driving units. A number of typical cone loudspeakers are shown in Fig. 2.

The unit in Fig. 2A is typical of cone loudspeakers using a balanced armature magnetic driving unit. The apex or point of the cone is driven by a metal extension arm or lever attached to the vibrating armature. In many cone loudspeakers, the mechanical lever system is so arranged that it provides mechanical amplification of the cone movement;

in other words, the apex of the cone may move twice or three times as far as the point on the armature to which the lever is attached. Even with this mechanical lever arrangement, the movement of the cone is definitely limited by the movement of the balanced armature between the pole pieces. Magnetic loudspeaker units of this type ordinarily will not handle much more than 2 watts of electrical input power.

sume this to be an electrodynamic unit, with the permanent magnetic flux being produced by direct current flowing through the field coil. If there are no additional leads, the unit is of the permanent magnet dynamic type.

The cone unit alone of a dynamic loudspeaker is shown in Fig. 2D. Observe that the voice coil is attached directly to the cone. The size of the voice coil leads has been exaggerated

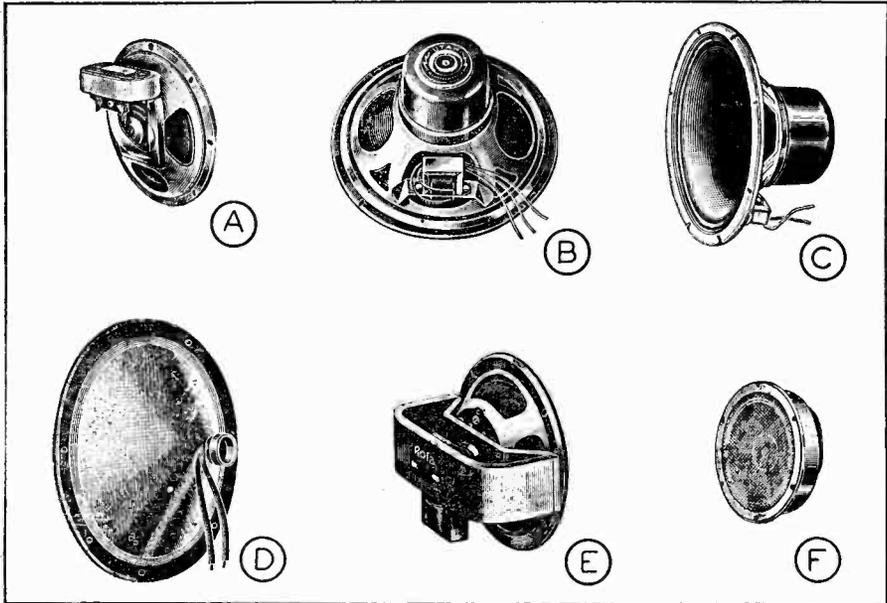


FIG. 2. Typical cone loudspeaker units.

In Figs. 2B and 2C are two views of a representative dynamic cone loudspeaker. The transformer which matches the voice coil impedance to the output impedance of the power stage of the receiver can be seen mounted directly on the loudspeaker frame, as is the usual practice. If, in addition to the leads coming from the matching transformer, there are two or three more leads coming from the pot or cylindrical housing at the back of the loudspeaker, we can as-

sume this to be an electrodynamic unit, with the permanent magnetic flux being produced by direct current flowing through the field coil. If there are no additional leads, the unit is of the permanent magnet dynamic type. The cone unit alone of a dynamic loudspeaker is shown in Fig. 2D. Observe that the voice coil is attached directly to the cone. The size of the voice coil leads has been exaggerated in the drawing in order to make them visible; actually they are made of very fine flexible wire. The outside edges of the cone are cemented to a cardboard ring, with corrugations or a soft leather ring just inside the cardboard to permit free movement of the entire cone. The cardboard ring is in turn either cemented or bolted to the steel ring which is a part of the loudspeaker frame. This type of cone construction is widely used in order to simplify replacement

of cones which have become damaged or have deteriorated through continuous use.

A permanent magnet dynamic cone loudspeaker is shown in Fig. 2E. Here the large U-shaped steel bar serves as the permanent magnet. Both tips of this bar are magnetized with the same polarity, the center being of opposite polarity. The soft iron central core for the voice coil is attached to the center of the permanent magnet, and is therefore opposite in polarity to the soft iron pole piece which is in contact with the ends of the U-shaped permanent magnet.

A crystal cone loudspeaker unit is shown in Fig. 2F. Ordinarily this will be found with a small, light cone about 5 inches in diameter. Crystal loudspeakers are often called "tweeters," and are used to reproduce the higher audio frequencies, in conjunction with an ordinary loudspeaker designed to reproduce the bass notes.

### Dynamic Loudspeaker Design Problems

*The Spider.* One important part of a dynamic cone loudspeaker has not yet been taken up, simply because it does not appear in ordinary loudspeaker illustrations; this part is the *spider*, a springy sheet of fiber or bakelite material cut as shown in Figs. 3A and 3B. The spider serves two purposes, that of *centering the voice coil* with respect to the soft iron core and pole pieces, and that of *returning the voice coil and cone to a normal position* when the driving force drops to zero.

In horn loudspeakers the back of the driving unit is completely enclosed and the air in the enclosed rear chamber provides the springiness required to return the diaphragm to its normal position; in dynamic cone

loudspeakers, however, this back air chamber is usually absent. The ordinary cone has little natural springiness because of the nature of its mounting, and therefore a spring must be used to provide the restoring action which is essential to correct loudspeaker operation. This spring is called a *spider*.

The *internal spider* unit shown in Fig. 3A is cemented inside the voice coil at the point where it joins the cone. The center part of the spider is fastened to the cylindrical iron core

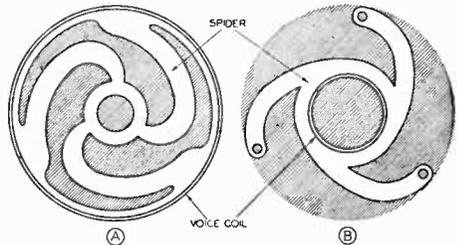


FIG. 3. Typical spiders used in cone dynamic loudspeakers to keep the cone and voice coil assembly properly centered with respect to the central iron core. That at A is for the front of the cone, and the form shown at B is for mounting on the back of the cone, around the voice coil. Spiders also provide a restoring spring action for the cone. Spiders of the type shown at B, which must surround the voice coil, will be considerably larger than those at A, which fit inside the voice coil.

inside the voice coil but is held away from the core by a bushing. A machine screw passing through the center hole of the spider and the bushing firmly anchors the spider. When this screw is loosened, the spider, voice coil and cone assembly can be moved a small amount in any direction to permit exact centering of the voice coil.

Sometimes the spider is of the shape shown in Fig. 3B, and is cemented to the outside of the voice coil at the point where it joins the cone. Three machine screws, one for each leg of the spider, hold it firmly against the outside housing or pot of the loudspeaker. Occasionally you will find an *external spider* of this

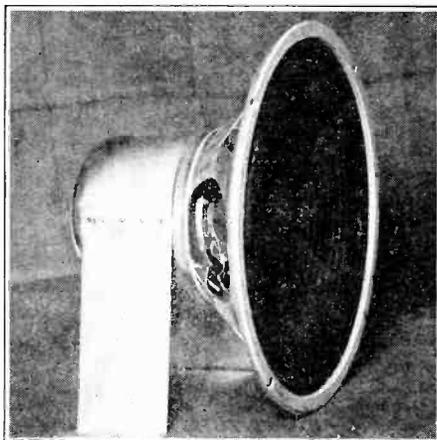
type made from webbed cloth which has been treated with bakelite varnish to give it springiness.

Since dynamic loudspeakers are by far the most common of any types in use today, it will be worth while to consider in detail a number of their peculiar design features which are not apparent at first sight. A great many design problems must be solved in connection with the driving unit, the cone and the spider in order to secure good frequency response and high efficiency.

You already know that the force acting upon the cone of a dynamic loudspeaker is determined by the current flowing through the voice coil, by the length of the wire used in the voice coil, and by the flux density in the vicinity of the voice coil. Before we can determine the effectiveness of this force in producing air velocity or sound, however, we must analyze the characteristics of our moving or vibrating system. Referring to Fig. 4A, we can see that the voice coil has a certain amount of mass, which we can consider as mechanical inductance  $L_V$ . The cone has a mass which can be considered as mechanical inductance  $L_C$ , and the air which is moved directly by the cone also has a certain amount of mass, which can be represented as mechanical inductance  $L_A$ . The spider provides most of the springiness or compliance, and this can be represented as mechanical capacitance  $C_S$ . The opposition offered by air particles to the movement of the diaphragm can likewise be represented as mechanical resistance  $R_A$ . Experiments have shown that all these parts can be considered as acting in series, as shown in Fig. 4B. Note the absence of a mechanical capacitance shunting  $R_A$ ; there being no air pocket in front of the cone, the dynamic loudspeaker does

not have this capacitance which in a horn unit reduces the high audio frequency output. The grille cloth usually found in front of a dynamic loudspeaker is loosely woven, so that sound waves can travel through it readily at all audio frequencies.

*Mechanical Resonance in Dynamic Loudspeakers.* From our knowledge of electrical circuits like that shown in Fig. 4B, we know that mechanical resonance will occur in our vibrating system when the combined mechani-



Courtesy Utah Radio Products Co.

The external spider, surrounding the cone end of the voice coil, can be seen on this photograph of a Utah dynamic loudspeaker.

cal reactances of  $L_V$ ,  $L_C$ , and  $L_A$  are exactly equal to the reactance of  $C_S$ . In a dynamic loudspeaker the spider (or whatever mechanical restoring spring action is used) has a high compliance, and hence the mechanical capacity is high. High values of capacity acting with high values of mechanical inductance make mechanical resonance occur at low frequencies in dynamic cone loudspeakers. In the average dynamic cone loudspeaker unit, resonance occurs below 100 cycles. This resonant action is usually utilized to reinforce the response of the loudspeaker at bass frequencies.

At resonance the entire input power is utilized in producing useful sounds; unless the cone is properly loaded by means of a suitable baffle, the movements of the cone will be excessive at resonance, resulting in distortion or even in damage to the voice coil, and particularly to its spider.

**Piston Action.** At frequencies above resonance, the circuit acts as a mechanical inductance in series with the mechanical resistance, which means that the driving force has only

the unit decreases with increasing frequency. Although this would tend to give greater output at high frequencies, unfortunately the cone will vibrate as indicated in Fig. 4C only for certain frequencies. This means that the high-frequency response of the loudspeaker will have many irregular peaks rather than the desired uniform response.

In a 12-inch cone (the diameter of the free edge), this change from piston action to vibrating cone action

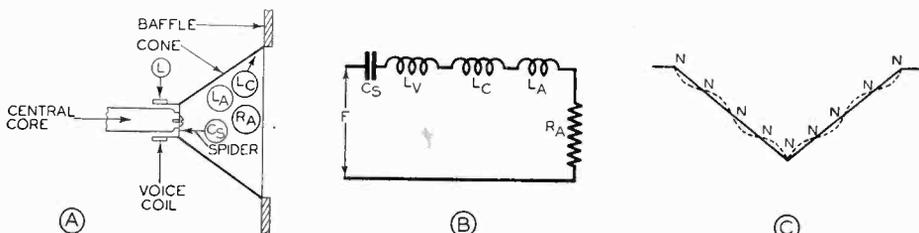


FIG. 4. A cone loudspeaker like that at A can be studied by means of an equivalent acoustical circuit like that at B. The diagram at C illustrates how a cone itself can vibrate at high audio frequencies; the points marked N represent nodes, which are points at which there is no vibration. The dotted line curve represents the maximum amplitude of the vibration at any other point along the cone.

mass and resistance to oppose it. You know that inductive reactance increases with frequency, and consequently our mass reactance in this mechanical system would likewise increase if it were not for the fact that the mechanical resistance  $R_A$  increases also and compensates for the increased mass to a certain extent. (Actually, tests have shown that the air resistance  $R_A$  increases uniformly with frequency up to the point where the diameter of the cone becomes one-half wavelength.) This essentially flat or uniform output response is obtained up to the frequency where the cone begins to vibrate with nodes and peaks as shown in Fig 4C. Under this condition the cone is no longer acting upon the air in the manner of a piston, but is itself vibrating. Once this condition is reached, the mechanical resistance  $R_A$  stays essentially constant, while the mass reactance of

takes place at about 750 cycles. For an 8-inch cone the change occurs at about 1,000 cycles, and for smaller-diameter cones it occurs at correspondingly higher frequencies.

Since pure piston action is desirable in the cone of a dynamic loudspeaker in order to secure a flatter frequency response, the loudspeaker designer uses a scheme which automatically reduces the diameter of the cone for the higher frequencies. One such scheme involves placing a number of concentric corrugations in the cone, as shown in Fig. 5A; the result is that at low frequencies the entire cone will be effective, with edge 1 serving as the free edge. At slightly higher frequencies the free edge of the cone will move inward to the corrugation at point 2; under this condition only that part of the cone between corrugation 2 and the voice coil will be in motion. At increasingly higher

frequencies, the effective cone diameter becomes increasingly less; at the highest frequency, corrugation 3 becomes the free edge.

A typical dynamic cone loudspeaker having corrugations in the

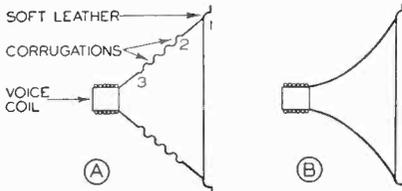


FIG. 5. Two methods of improving the high frequency response of a cone loudspeaker are illustrated here.

cone to reduce self-vibration and thereby improve the high-frequency response is shown in Fig. 6.

Another scheme for increasing efficiency by securing pure piston action involves using a special shape of cone, the free edge of which will automatically shift towards the voice coil as the audio frequency increases. A suitable shape for accomplishing this result is shown in Fig. 5B; since the curve of this cone corresponds to that of a parabola (a special geometric curve), it is often called a *para-curve* or a *curvilinear* diaphragm.

A third scheme involves gradual thinning out of the cone material from the voice coil toward the free edge. Each of these schemes gives a reasonably flat frequency response up to about 3,000 cycles, with decreased and somewhat non-uniform output at higher frequencies. This does not mean, however, that no high-frequency output is obtained; actually the high-frequency output can be quite satisfactory for ordinary radio receiver requirements, but for more nearly perfect reproduction, a second loudspeaker, designed specifically for uniform high-frequency reproduction, should be considered.

*Double Voice Coils.* Even if we could limit cone movement to that region immediately in the vicinity of the voice coil at high frequencies, in order to secure the desired piston action, the voice coil itself would still have too much mass for perfect results. This mass can be reduced by careful design of the voice coil in order to keep its weight at a minimum, and this in turn extends the upper frequency range of the loud-

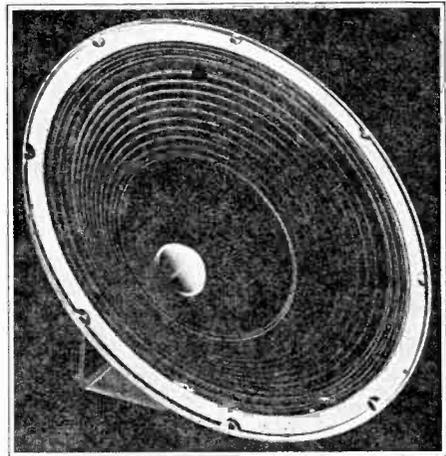


FIG. 6. The corrugations which serve to reduce the effective diameter of the cone at higher frequencies are clearly visible in this photograph of an RCA Victor dynamic cone loudspeaker unit.

speaker a certain amount. To further extend this range, two voice coils are sometimes used together, as illustrated in Fig. 7. The coils are connected by an elastic coupling band which can be represented as  $C_1$ . Coil  $L_1$  has considerably greater mass than  $L_2$ . At low frequencies the mechanical capacitance  $C_1$  is ineffective (the elastic coupling does not bend), and the driving forces produced by both coils act upon the cone. At high frequencies only the smaller coil is capable of moving at the high rate of vibration involved, and the elastic coupling at  $C_1$  allows this coil alone to drive the cone, while  $L_1$  remains

essentially fixed. In this way the cone can be made to produce uniform outputs up to about 6,000 cycles, with lowered output at higher frequencies than this. Coils  $L_1$  and  $L_2$  each have their own leads, being externally con-

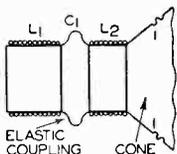


FIG. 7. The double voice coil construction shown here is sometimes used to reduce the effective mass of the vibrating system at high frequencies and thus improve the high-frequency response.

nected together in series.  $L_1$ , the heavier coil, is ordinarily shunted with a condenser whose value is such that high-frequency currents will be by-passed around it and will flow only through  $L_2$ , where they will produce a useful mechanical force.

### Baffles

When the voice coil pushes the cone of a dynamic loudspeaker forward or away from the pot, the air in front of the cone is compressed or made heavy, while the air at the rear of the cone is simultaneously thinned out or made rare, as indicated in Fig. 8A. Those particles of air which are compressed will exert force on adjacent air particles, transferring this compressed condition to nearby air particles at the speed of sound, which is about 1,089 feet per second. It is a natural tendency for these compressed air particles to move outward in all directions, with many of them moving around the edges of the cone to the rear. If the air at the rear of the cone is rarefied at the time when compressed air particles arrive, the particles rush to the rear to equalize the air pressure, and this rush of particles in turn brings more compressed air particles from the front to the rear. The result is cancellation of useful sound, since sound is produced the

instant that air is compressed at the front of the cone.

To limit this natural tendency for air to avoid doing useful work, we can place around the loudspeaker a baffle made up of a square or circular board having cut in it a hole equal to the diameter of the cone. A baffle such as this is indicated in Fig. 8B; we can immediately see that air particles which are compressed in front of the cone must travel completely around the baffle before they can reach the back of the cone. It takes a certain amount of time for this compressed air condition to be transferred from

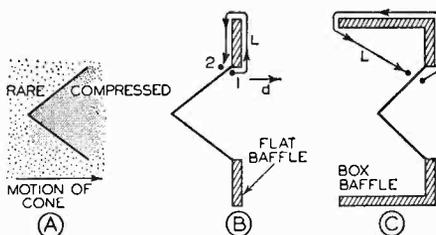


FIG. 8. The shortest sound wave path from the front of a cone around the baffle to the rear must be at least one-half the wavelength of the lowest sound frequency to be reproduced if cancellation of sound waves is to be avoided.

point 1 around the baffle to point 2, and this time should be made long enough to allow the cone to begin compressing the air at the rear on its backward movement. Under this condition there is practically no cancellation of compressed air by rarefied air, and a considerably greater amount of useful sound is radiated by the front of the loudspeaker.

Let us assume that the cone is moving in and out in a sine wave manner. When the cone is farthest forward, compression of air at point 1 will be a maximum. This compressed air will take the shortest path to point 2, where a rarefied condition exists at this moment; naturally this will be path  $L$  around the baffle. The time

taken for air movement along this path will be equal to the length of the path divided by the speed of sound; if this time is such that the cone has gone through a complete half cycle by the time the compressed air reaches point 2, then the cone will be compressing air at point 2, the compressed air from that point will meet air which is equally compressed, and no cancellation will take place. As a result, the compressed air at the front will be *forced* to move in the desired

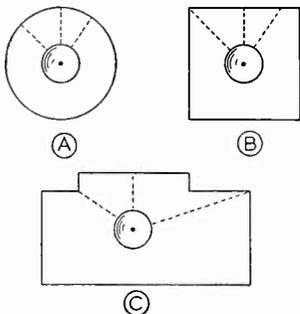


FIG. 9. The irregular-shaped flat baffle at C gives more uniform frequency response than either the circular baffle at A or the square baffle at B. Dotted lines indicate possible sound wave paths.

forward direction. The path length required to prevent front-to-rear cancellation can readily be figured out by dividing 1,089 by twice the frequency of the lowest sound signal to be reproduced. For example, if the lowest frequency to be reproduced is 50 cycles, the minimum path length  $L$  would be equal to 1,089 divided by 2 times 50, which is 10.89 feet.

The longer the path  $L$  around the baffle, the lower will be the cut-off frequency of the loudspeaker. Where space is limited, as it is in the conventional radio receiver cabinet, the loudspeaker is usually mounted in a box like that shown in Fig. 8C. Here path  $L$  would be measured as indicated.

Although each baffle has a definite cut-off frequency, with sounds below

this frequency being eliminated or greatly reduced in strength, this is no assurance that frequencies above the cut-off value will be reproduced without attenuation. Suppose that the diameter of the circular baffle shown in Fig. 9A is such that cut-off occurs at 100 cycles. When a 200-cycle sound is fed to the loudspeaker mounted on this baffle, the cone will go through one complete cycle during the time required for the sound to travel from the front to the rear, and consequently the compressed air traveling around the baffle will encounter rarefied air at the rear. Cancellation takes place and loudspeaker output is low at 200 cycles. This same effect will occur at each higher multiple of the cut-off frequency (3 times, 4 times, 5 times, etc.), but in the case of these higher multiples the cone will have gone through so many cycles during the time required for the compressed air to get around the baffle that most of the sound produced at the front of the cone will have traveled too far out to be affected. It is at the frequency equal to twice the cut-off frequency that cancellation is the most serious.

*Irregular-Shaped Baffles.* With a square baffle like that shown in Fig. 9B, the paths around the baffle are of many different lengths, so that even though cancellation occurs on some paths there will always be other paths at which a particular frequency is not entirely cancelled out. The result is that the frequency response of the loudspeaker system is far more uniform in the low frequency region down to the minimum cut-off frequency (determined by the longest baffle path length). The most uniform response at low frequencies is obtained with an extremely large baffle or with a special form of completely enclosed box baffle, sometimes

known as an *infinite baffle*. An irregular shape of baffle like that shown in Fig. 9C is better than a square or circular baffle, for this provides a number of widely different path lengths.

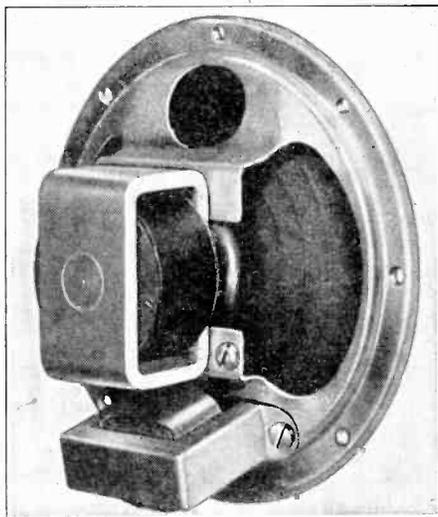
### Box Baffles

The cabinet in which the cone loudspeaker of the average home radio receiver is mounted is essentially a box. Although this box does give the desired long path for sound around the baffle, the presence of the box or cavity behind the loudspeaker is not always desirable. This cavity can in itself be a sound resonator, acting much in the manner of a horn and setting air into vibration at some frequency. In the average radio receiver cabinet this cavity can be shocked into vibration at a low or bass frequency in many cases, producing an undesirable high bass output. The cavity is in addition a load on the cone, increasing its mechanical resistance and altering the frequency response of the loudspeaker; on the other hand, if this load produced by the cavity is properly proportioned, it can be a real aid to loudspeaker operation. Since many loudspeaker installations take into account the design of the back resonating chamber or box baffle, we will analyze this problem in greater detail in order that we can better understand the unique constructions which are sometimes employed.

With a cone mounted on a large flat baffle, the back of the cone must move equally as large a mass of air as the front of the cone. This mass has an inertia which opposes any force which tries to set it into motion, and consequently we can consider this mass as a mechanical inductance.

With a flat baffle, the mass of air in back of the cone has little stiffness,

for it can expand equally well in all directions behind the baffle. As we bend back the edges of the baffle to form a box-like chamber around the rear of the cone, we confine the mass of air to a limited region and reduce the mass upon which the back of the cone is compelled to act. Confining this air to a definite volume increases its stiffness, with the result that



Typical electrodynamic loudspeaker, available from Jensen Radio Mfg. Co., Chicago, Ill., in 5, 6 and 8-inch sizes (cone diameters) with various output transformers and various field coil resistance values for replacement purposes.

greater force must be applied to the cone in order to move this air. This reduces the compliance of the rear chamber, thereby reducing its mechanical capacitance. (In the case of a flat baffle, this mechanical capacitance was so high that its effect could be neglected.) The mechanical force must therefore act simultaneously upon a mechanical capacitance and a mechanical inductance in parallel.

We know also that a certain amount of sound always escapes from the rear of the chamber; this radiated sound must be represented by me-

chanical resistance acting in series with the air mass. A completely closed box has no sound radiation at the rear and therefore has zero mechanical resistance. A flat baffle has a large mechanical resistance, its value being equal to the mechanical resistance existing at the front of the cone. An open box has a medium value of mechanical resistance.

With these facts in mind, we can set up the equivalent mechanical circuit of a loudspeaker mounted in a box baffle, and use this as our guide in analyzing the good and bad fea-

tures of the back chamber air; the mechanical resistance introduced by sound radiated from the rear of the baffle is represented as  $R_B$ , acting in series with  $L_B$ .

We can now see that  $C_B$  and  $L_B$  in Fig. 10B together form a parallel resonant circuit. Furthermore, this resonant circuit can be shocked into self-oscillation by the driving force  $F$  acting through the reactances of  $L_C$  and  $C_C$ , under which condition the air in the cavity will vibrate and radiate sound through the cone. This self-produced sound may set the cone into vibration, producing frequencies which were not present in the original program and giving the so-called boomy response which some loudspeakers have. Furthermore, if the driving force  $F$  has the same frequency as the resonant frequency of the cone loudspeaker unit (resonant circuit  $L_C-C_C$ ), the cone will be set into excessive vibration, giving a whooping sound. This occurs because  $L_C-C_C$  acts as a low mechanical resistance at resonance, allowing more of the driving force  $F$  to be applied to the cavity and increasing the sound output of the back chamber at that particular frequency.

*Natural Resonant Frequency.* Increasing the size of a box baffle increases the volume of the cavity and thereby increases the values of mechanical capacitance  $C_B$  and mechanical inductance  $L_B$ ; this lowers the natural resonant frequency of the cavity (the resonant frequency of the parallel resonant circuit). If the box is made sufficiently large or if an infinite flat baffle (where no sound waves whatsoever can escape to the rear of the cone around the baffle) is used, the natural frequency of the region behind the cone can be made so low that the effects of cavity resonance will not be heard at all.

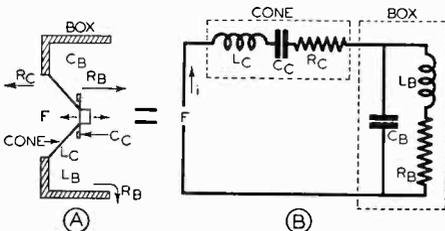


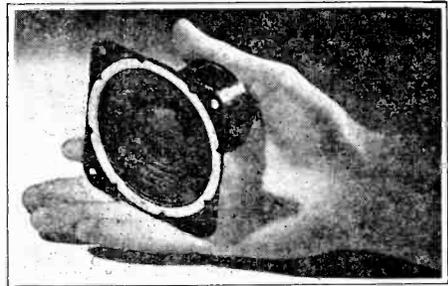
FIG. 10. Rear chamber of a box baffle, and equivalent circuit of the entire system, including the chamber.

tures of this particular baffle. Figure 10A shows a cone and voice coil mounted in a box baffle, and Fig. 10B represents the equivalent circuit including the back chamber. The mechanical force acting upon the entire system varies in an alternating manner corresponding to the variation in voice coil current. This force acts first upon the mass of the cone and voice coil assembly, represented in Fig. 10B as mechanical inductance  $L_C$ , upon the spider which provides springiness or mechanical capacitance  $C_C$ , and upon the mechanical resistance due to sound radiation from the front of the cone, represented as  $R_C$ . The additional load placed upon the cone by the back chamber can be represented by mechanical capacitance  $C_B$  in parallel with mechanical inductance  $L_B$ , representing the mass

A radio receiver which has an open back is generally designed so that when the cabinet is kept a few inches away from the wall or is placed in a corner so there is free movement of air behind the cabinet,  $C_B$  will be high enough in value to prevent audible cavity resonance effects. Placing the radio receiver back up against the wall tends to close up the back chamber and reduce  $C_B$ ; under this condition cavity resonance may occur at an audible bass frequency, in the form of boominess and the whooping sounds mentioned before. Remember: radio receivers with open backs will sound best when kept a few inches away from the wall or when located across a corner of a room.

*Infinite Baffles.* Suppose that we closed up the back of the cabinet com-

will be no vibration of the cone at resonance, no whooping sounds, no sound waves will be radiated by the back of the cabinet, and we will have what is known as an *infinite baffle*. Figure 11 shows more clearly what happens under this condition; curve 1 shows how cone velocity, which determines sound output to a great



Courtesy Oxford-Tartak Radio Corp.

Truly a midget is this Oxford permanent-magnet dynamic loudspeaker unit with 3" diameter cone, designed for use in midget table model radios and in intercommunication systems.

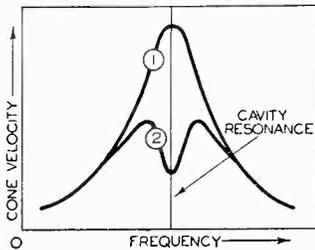


FIG. 11. These curves show how cone velocity (which determines sound output) varies with frequency for an ordinary box baffle (curve 1) and for a box baffle which is tuned to the natural resonant frequency of the moving system (curve 2).

pletely, thereby making the values of  $C_B$  and  $L_B$  constant. This, of course, does not in itself eliminate cavity resonance, but if we make the resonant frequency of  $C_B-L_B$  equal to the resonant frequency of  $L_G-C_C$ , an entirely new effect will take place. When the driving force  $F$  has this same frequency, the series resonant circuit will act as a very small resistance, but the parallel resonant circuit will act as a very high resistance, greatly limiting the flow of mechanical current  $i$ . This means that there

extent, increases gradually with frequency up to a maximum value at a frequency equal to the natural resonant frequency of the cavity. At frequencies above cavity resonance, the cone velocity drops again. Curve 2 is for the condition where the cavity is tuned to the natural resonant frequency of the cone; you can see that the peak has been reduced considerably at resonance, while cone velocity is still high enough to give the system desired bass reinforcement. *Adjusting the cavity resonant frequency to the natural frequency of the cone system* will greatly reduce the undesirable effects of cavity resonance in a box type baffle.

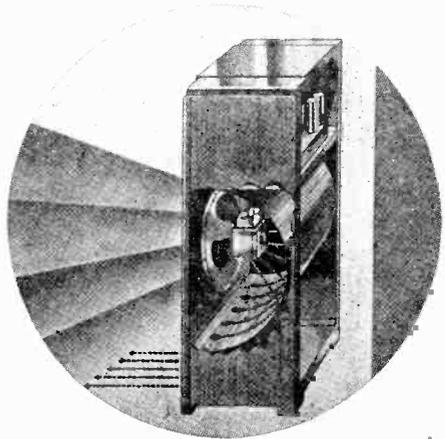
*Standing Waves.* At frequencies higher than the cavity-resonance frequency, the cavity will act as a low-reactance condenser which does not seriously affect the response of the loudspeaker. At the higher sound

frequencies, however, where the wavelengths involved become shorter than the dimensions of the box, there will be considerable reflection of sound from one side of the box to the other. If these reflections occur continually over the same path, the compressions and rarefactions of air will be reinforced at definite points, and *standing sound waves* will occur; these give resonant effects much like those produced in organ pipes and in musical wind instruments. In other words, sounds at certain frequencies will be prolonged somewhat, due to the production of harmonics, and the reproduced sound will not be natural.

Using a shallow cavity or making it irregular in shape so that reflected sounds cannot build up along the same path will cure this trouble. Other remedies involve lining the box with a sound-absorbing material, hanging heavy sound-absorbing materials down from the top inside the box, or lining the sides of the box with rectangular pieces of wood, thereby preventing sounds from being reflected back and forth over the same path. Whenever sound waves are reflected repeatedly over a path which is some multiple of a half wave in length, compression will occur at the same points along this path for each reflection. Reinforcement of sound occurs under this condition due to the building up of pressure, and the resulting standing waves produce annoying sounds. The box itself should be made of heavy wood, so it will not vibrate at the lower bass frequencies.

So far we have considered box baffle arrangements which are intended to prevent cavity resonance at low frequencies and standing waves at high frequencies. In each case the sounds produced by the rear of the cone were either suppressed or absorbed, and were therefore wasted.

*Bass-Reinforcing Box Baffles.* Increased bass output is practically always desirable in a loudspeaker if it can be secured without at the same time producing undesirable cavity resonance effects. We know that the sounds coming from the rear of the cone are out of phase with those coming from the front; if, however, we can reverse the phase of the bass notes coming from the rear and allow them to emerge at the front of the loud-



RCA Sonic-Arc Baffle, also known as the Magic Voice, which provides bass reinforcement and at the same time prevents standing waves.

speaker, we can secure desirable reinforcement of bass notes without increasing cavity resonance problems. The loudspeaker arrangement which accomplishes this result is known as a *low-pass, phase-reversing acoustical filter*.

The back of the cabinet or box is ordinarily closed completely when bass reinforcement is desired, and outlets for air are provided either underneath the cabinet or at the front, below the main outlet at the front of the cone. Our equivalent mechanical circuit for this condition is therefore the same as that shown in Fig. 10B. Again we can assume that undesirable effects of cavity res-

onance are greatly reduced by making the resonant frequencies of the cavity and the cone identical.

Above the resonant frequency of the cone or the cavity, the series resonant circuit  $L_C-C_C$  acts as a mechanical inductance, while the parallel-resonant circuit  $L_B-C_B$  acts as a mechanical capacitance. To simplify further our analysis, let us assume an above-resonance condition (in which the frequency of mechanical force  $F$  is higher than the natural resonant frequencies of the cone and box). In

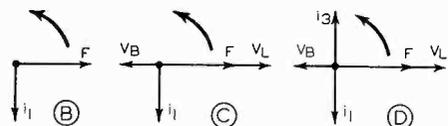
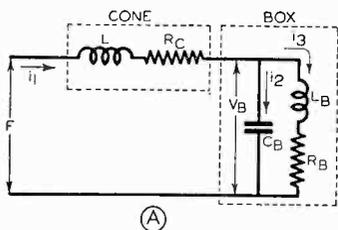


FIG. 12. Equivalent acoustical circuit of a closed box baffle which is acting as a low-pass, phase-reversing acoustical filter at frequencies above the resonant frequency of the cone, and vector diagrams showing how it acts.

this case, series resonant circuit  $L_C-C_C$ , representing the cone assembly, will act as a mechanical inductance (which we can designate as  $L$ ) in series with mechanical resistance  $R_C$ ; the equivalent mechanical circuit will now appear as shown in Fig. 12A.

If we can prove that mechanical current  $i_3$  flowing through mechanical resistance  $R_B$  is  $180^\circ$  out of phase with mechanical current  $i_1$  flowing through  $R_C$ , we will have shown that radiation of sound from the cavity will be  $180^\circ$  out of phase with the sound radiated by the back of the cone, and will therefore be *in phase* with sound radiated from the front of the cone.

Above resonance, parallel resonant circuit  $L_B-C_B$  will act as a mechanical capacitance, and if by design its mechanical reactance is *less than* the mechanical reactance of  $L$ , the mechanical force  $F$  will feel an inductive load. The mechanical current  $i_1$  delivered by  $F$  will therefore lag  $F$  by  $90^\circ$ .

A vector diagram will show at a glance the exact phase relationships between the various mechanical currents and voltages in our circuit. Let us use  $F$  as our reference vector, drawing it as shown in Fig. 12B. Since we have just found that  $i_1$  lags  $F$  by  $90^\circ$ , we draw  $i_1$  downward,  $90^\circ$  clockwise from  $F$ .

The mechanical force acting upon  $C_B$  and  $L_B$  will be equal to the applied force  $F$  minus the mechanical force dropped across inductance  $L$ . (We are now neglecting  $R_C$  and  $R_B$ , as they have little effect.) This mechanical force (mechanical voltage)  $V_L$  which is dropped across  $L$  will lead the current  $i_1$  through  $L$  by  $90^\circ$  (coil voltage always leads coil current by  $90^\circ$ ), so we draw vector  $V_L$   $90^\circ$  counter-clockwise from vector  $i_1$ , as in Fig. 12C. Now we can see that  $V_L$  is in phase with  $F$ .

We also know that the voltage across a condenser lags its current by  $90^\circ$ . Since parallel resonant circuit  $L_B-C_B$  acts as a condenser at frequencies above resonance, mechanical voltage  $V_B$  across this circuit will lag  $i_1$  by  $90^\circ$ ; we therefore draw in vector  $V_B$   $90^\circ$  clockwise from  $i_1$ , as in Fig. 12C. Since  $V_B$  acts upon  $L_B$ , we can say immediately that mechanical current  $i_3$  through  $L_B$  will lag  $V_B$  by  $90^\circ$ , and can draw in vector  $i_3$   $90^\circ$  clockwise from  $V_B$ , as in Fig. 12D. Our vector diagram now shows clearly that  $i_3$  is  $180^\circ$  out of phase with  $i_1$ , which is exactly what we desired to prove.

By proper selection of the mass ( $L_B$ ) and compliance ( $C_B$ ) of the box cavity in relation to the mass ( $L_C$ ) and compliance ( $C_C$ ) of the loudspeaker cone, an engineer can secure a loudspeaker system which will behave like the circuit in Fig. 12A. Sound waves escaping from the box will then be in phase with the sound radiated from the cone front, and the desired bass reinforcement will be secured at frequencies above the cavity resonant frequency.

At high sound frequencies, mechanical capacitance  $C_B$  serves as a shunt path for mechanical current, with the result that practically no mechanical current flows through  $R_B$ , and no high-frequency sounds emerge from the box cavity.

*Stromberg-Carlson Acoustical Labyrinth.* An excellent example of the baffle design features just discussed is the acoustical labyrinth or winding passageway which is built into the cabinets of some Stromberg-Carlson receivers for four distinct purposes:

- 1, to prevent cavity resonance;
- 2, to prevent standing high-frequency sound waves;
- 3, to give a low cut-off frequency for the baffle without resorting to an excessively large cabinet;
- 4, to give reinforcement of bass response by allowing low frequency sounds radiated from the rear of the cone to travel through the labyrinth and emerge at the front in phase with the sounds normally radiated from the front of the cone. The two views in Fig. 13 illustrate the nature of this special loudspeaker baffle construction.

Observe that a felt hood covers the entire rear face of the cone, with only the pot exposed to air for cooling purposes. Sounds radiated by the rear of the cone are therefore directed into the rectangular cross-section passageway which winds back and forth down

to the bottom of the cabinet. The insides of this passageway or labyrinth are lined with a porous sound-absorbing material which is held together by coarse metal screening. High-frequency sounds are totally absorbed by this material, while lower frequency sounds are only partially absorbed. The long passageway with its outlet at the far end has the essential features of mass, compliance and resistance which are necessary for a low-pass phase-reversing acoustical filter at bass frequencies, and the result is that bass sounds emitted at

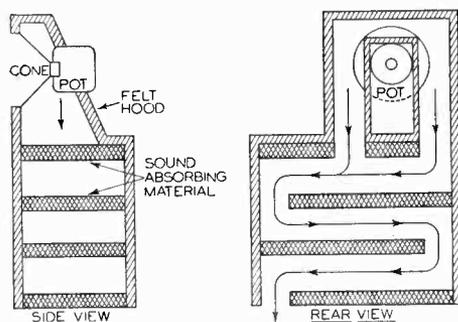


FIG. 13. Essential constructional features of the Stromberg-Carlson Acoustical Labyrinth.

the lower end of the labyrinth reinforce the bass sounds radiated from the front of the cone.

*Special RCA Box Baffles.* To secure the same effects as are provided by the acoustical labyrinth, RCA engineers use the construction shown in Fig. 14A for some of their receivers. The back chamber is completely closed, but there are holes at the bottom of the chamber, each surrounded by a length of pipe, through which air can escape. The back chamber thus has mass and compliance, and that air which escapes through the pipe provides mechanical resistance. The pipes are intended to adjust the mass in order to make the chamber resonate at the same fre-

quency as the cone. The entire cabinet of this RCA receiver is made of heavy wood, to prevent it from vibrating and radiating sounds from all sides when powerful bass notes are being reproduced. Only sounds emerging from the pipes are capable of reinforcing the normal output.

Pipes are by no means essential in the design of an acoustical low-pass filter; simple outlets are sufficient if the cabinet is properly designed to secure the correct resonant frequency for the cavity. Many RCA receivers have this simplified construction, illustrated in Fig. 14B. The back of the cabinet is closed by a curved sound reflector which serves two purposes, that of reflecting high-frequency sounds in all directions to prevent standing sound waves, and that of stiffening the back so it will not vibrate as readily as would a flat board. The air enclosed by this chamber provides compliance and mass, and the air escaping through the outlet at the bottom provides mechanical resistance; we thus have the conditions necessary for bass reinforcement.

Another example of this simplified construction is illustrated in Fig. 15. This is a special high-fidelity, high-power loudspeaker unit made by Jensen and known as the Peridynamic or bass reflex loudspeaker. The cabinet is made of thick, solid material so its sides will not vibrate. The box is made shallow so standing waves will be negligible. The cavity is properly proportioned to give the essential requirements of an acoustical filter.

Some manufacturers of radio receivers and cabinets follow the practice of closing up the back of the cabinet with a solid board which is reinforced with crossed ribs to prevent it from vibrating, allowing sounds to escape only from the corners of the

back through holes which are left at these points. Again, proper design gives desirable bass reinforcement and eliminates undesirable features of a

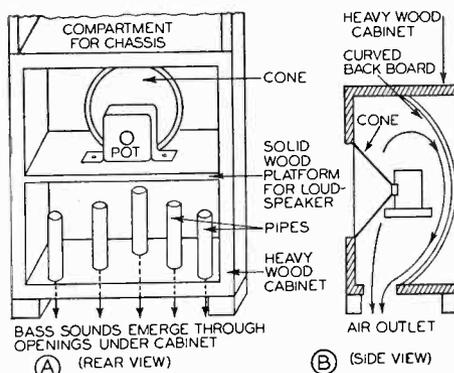


FIG. 14. Constructional features of two types of low-pass, phase-reversing acoustical filters used in RCA receivers.

cavity. Do not think that because a loudspeaker cabinet appears simple in construction, it is easy to design and build; careful engineering is required in order to prevent cavity resonance, to prevent standing sound waves at high frequencies, and to secure the proper proportioning of acoustical elements in the cone and cavity in order to obtain the desired bass reinforcing action.

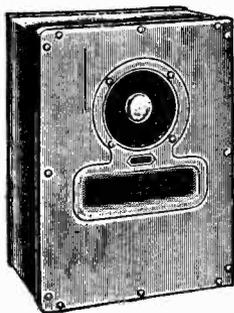


FIG. 15. Jensen Peridynamic loudspeaker unit. Reinforcing bass notes emerge from the rectangular opening below the cone.

*Acoustic Clarifiers.* When no attempt is made to utilize bass sounds emerging from the rear of a loudspeaker system, the chief problem is

that of removing cavity resonance effects or making them inaudible. It is possible to do this by inserting in the cavity an acoustic clarifier, a device which itself vibrates readily at all times and serves to cancel the air vibrations in the cavity. Some Philco receivers utilize this principle, in the form of a small cone which is suspended on an elastic spider mounted in the cabinet cavity or chamber. Any sudden loud sounds or sustained sounds which would otherwise set the cavity into vibration are transferred to this suspended cone, causing it to vibrate and absorb the energy present in the chamber. As a result there is very little vibration of the air in the chamber, and no sounds are emitted from the rear. Several of these acoustic clarifiers are generally used in a single cabinet, so as to increase the coupling between the cavity and the clarifiers. The clarifiers are sufficiently broad in frequency response to prevent cavity resonance effects when the back of the cabinet is open and the cabinet is placed flat against a wall. Remember, however, that acoustic clarifiers are not loudspeakers.

The principle of an acoustic clarifier can best be explained by referring to the electrical circuit shown in Fig. 16. Here we have a parallel resonant circuit made up of  $L_1$  and  $C_1$ , driven by an A. C. voltage source  $V$ ; this, as you know, is equivalent to the parallel resonant circuit representing the back chamber of a loudspeaker cabinet. At resonance, large currents flow through  $L_1$  and  $C_1$ , making this circuit act like a high resistance. If we place across this parallel resonant circuit a series resonant circuit made up of  $C_2$  and  $L_2$ , and make both resonant circuits tune to the same frequency, the series resonant circuit will at resonance act as a short across the

parallel resonant circuit. As a result, the source  $V$  will be supplying current to the series resonant circuit, while the parallel resonant circuit will receive practically no current. This series resonant circuit is the equivalent of the acoustic clarifier; it cancels the sound waves which otherwise would result in oscillation or vibration of the air in this back chamber.

### Sound Diffusers

When sound is produced by a small source, all audio frequencies travel equally well in all directions away

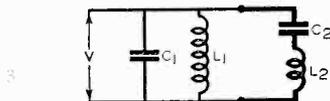


FIG. 16. An electrical circuit explaining the operation of the Philco Acoustic Clarifier.

from the source. When the size of the sound-producing source is quite large, however, as is the case with loudspeakers using large diaphragms and cones, only the low audio frequencies will travel equally well in all directions. The higher frequencies tend to travel best straight ahead of the loudspeaker, concentrating into a beam which becomes smaller and smaller in size as frequency is increased.

Loudspeaker engineers utilize what is known as a polar radiation pattern to show how a particular loudspeaker radiates various audio frequencies in different directions; one such pattern, representing the manner in which a cone loudspeaker with a baffle radiates one particular sound frequency in various directions, is shown in Fig. 17A. Let us first see how a pattern such as this is secured, for then we will be better able to appreciate the interesting information which it can give about a loudspeaker.

With our loudspeaker and baffle set up in a large room having sound-absorbing walls, floor and ceiling, and a definite audio frequency fed through the voice coil of the loudspeaker, we measure the loudness of the sounds produced by the loudspeaker at various points such as at  $P_1$ ,  $P_2$ ,  $P_3$  and  $P_4$ , which are all the same distance away from the center of the loudspeaker and make various angles, such as  $\theta$  (Greek letter *theta*), with the center line  $O-P_3$ . We will designate as  $I_3$  the loudness level measured at point  $P_3$ . We now draw on paper a simple sketch of our loudspeaker, much like that in Fig. 17A, and mark point  $O$  as the approximate center of the sound-producing source. Point  $P_3$  is placed on the diagram next, directly in front of the loudspeaker, and at any convenient distance away. A line is drawn from  $O$  to  $P_3$ , and along this line we plot the measured value  $I_3$  to any convenient scale, starting from  $O$ . We next move our measuring apparatus to position  $P_2$ , which is off to one side of line  $O-P_3$  by the angle  $\theta$  but is still the same distance away from  $O$ . We measure the angle  $\theta$  made by lines drawn from  $O$  to  $P_2$  and  $P_3$ , and use this angle as our guide in locating  $P_2$  on the diagram. The loudness level measured at  $P_2$  is now plotted along line  $O-P_2$ , starting from  $O$ ; this gives point  $I_2$  in Fig. 17A. The same procedure is repeated for points  $P_1$ ,  $P_4$  and various other points which are all the same distance away from  $O$  and at various angles off to each side. A smooth curved line drawn through the resulting points  $I_1$ ,  $I_2$ ,  $I_3$  and  $I_4$  now gives the polar radiation pattern. In the case of Fig. 17A, this pattern tells us that at a definite distance away from the loudspeaker, sounds will be loudest directly in front of the cone and will gradually become

weaker as we go off to one side or the other of the center line.

Radiation patterns for three different frequencies, as secured with a large dynamic loudspeaker mounted in a box baffle, are shown in Fig. 17B. All three curves are for measurements made at the same distance away from the loudspeaker. These curves tell that a listener at position  $P_1$  would hear middle frequencies best, with bass frequencies slightly less loud and treble frequencies about half as loud. On the other hand, a listener at position  $P_2$  would hear bass frequencies the best, with middle frequencies slightly weaker and treble sounds

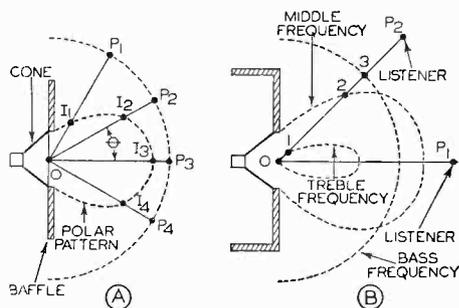


FIG. 17. Polar sound radiation patterns for cone loudspeakers.

scarcely distinguishable at all. A line drawn from any other listening position to point  $O$  will give this same information for that position; simply observe where the line intersects each of the three curves. Now you can readily understand why the output from a loudspeaker sounds best when you are located directly in front of it.

Uniform distribution of sound from a loudspeaker throughout a room is highly desirable, for the non-uniform distribution shown in Fig. 17B tends to cancel out, for certain listening positions, all the advantages gained by careful design of the radio receiver and loudspeaker. Since high frequencies give the greatest trouble in this respect, the loudspeaker engi-

neer often builds vanes in front of the loudspeaker to deflect these frequencies and spread out the beams. If the vanes are of the proper size, at least as wide as the wavelength of the high-frequency sounds to be deflected, the vanes will have no effect upon low and medium frequencies.

A wall-type loudspeaker, with deflecting vanes mounted in front of the cone for this purpose, is shown in Fig. 18A. Sometimes these vanes are mounted behind the grille cloth or are made a part of the cabinet itself.

A well designed deflector for high-frequency sounds is shown in Fig. 18B; this is a curved cone anchored rigidly inside the regular sound-producing cone of the loudspeaker. High-frequency sounds are deflected by this cone and spread out on all sides, while low- and medium-frequency sounds travel around the cone just as if it were not present.

Horn loudspeakers concentrate all sounds into relatively narrow beams, with the highest frequencies being concentrated the most. When a horn loudspeaker is used as a tweeter for the reproduction of high frequencies only, it is often sectionalized in the manner shown in Fig. 19 in order to spread out the beam and make the distribution of high-frequency sounds similar to that of the low- and medium-frequency sounds as produced by the woofer (low- and medium-frequency) loudspeaker. The presence of vanes in one or more units of a reproducing system is usually a sign of high-fidelity reproduction over a wide range of frequencies; this particularly holds true for the loudspeakers used in public address systems.

### Horn-Shaped Baffles

Large cone loudspeakers are widely used in public address systems requir-

ing high sound outputs. Cone loudspeakers mounted in a large flat baffle or in a box baffle radiate sound over a wide angle; this may not be

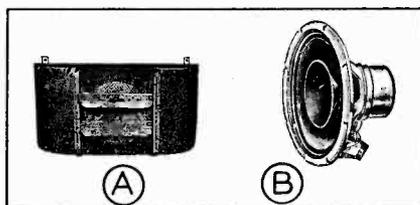


FIG. 18. Examples of cone loudspeakers which utilize deflecting vanes to spread out high-frequency sound waves.

desirable in some installations, for it is often necessary to concentrate sound in a definite direction and over a definite area. Horn-shaped baffles are used for this purpose; when properly designed, these make the loudspeaker more directional and at the same time improve its efficiency considerably.

*Loudspeaker Efficiencies.* The ordinary large cone loudspeaker, when mounted in a box baffle or on a flat baffle, rarely has an efficiency of greater than 3 per cent, whereas a driving unit used with an exponential horn can have an efficiency as high as 50 per cent. This means that if 10 watts of electrical power are fed into

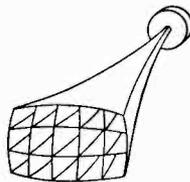


FIG. 19. Horn loudspeakers designed for use as tweeters are often divided into sections as shown above in order to spread out high-frequency sounds.

a cone loudspeaker, only about .3 watt of acoustical power will be delivered. This low cone loudspeaker efficiency is satisfactory for the aver-

age home radio receiver, but in loudspeakers designed for high-power public address systems it is necessary to improve this efficiency by improving the coupling between the cone and the surrounding air. Loudspeaker engineers therefore turn naturally to the exponential horn when the efficiency of an ordinary baffle for a cone loudspeaker proves unsatisfactory.

An early attempt to improve the directional characteristics of a cone loudspeaker is shown in Fig. 20A; the performance of this flat-sided horn was no better than that of the early megaphone used with smaller driving units. Modern exponential horn

sounds back and forth inside the horn. The area of the mouth of the horn approaches the area of a circle whose diameter equals  $1/4$  wavelength at the lowest frequency to be reproduced.

### Loudspeaker Impedance

Any loudspeaker has a definite electrical impedance. The value of this impedance will depend upon the frequency of the source voltage applied to the loudspeaker, upon the electrical characteristics of the loudspeaker (inductance and resistance for magnetic and dynamic driving units, and capacitance with resistance

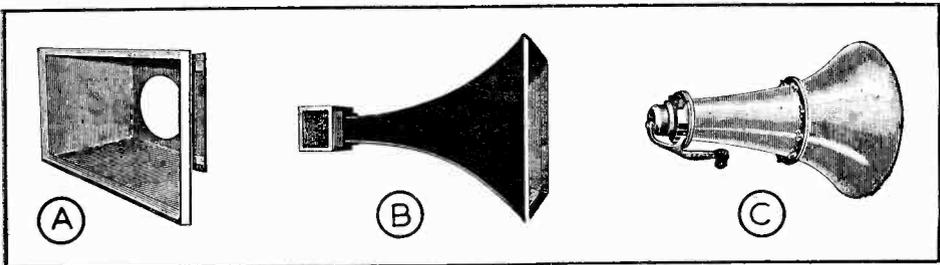


FIG. 20. Examples of horn baffles for cone loudspeakers. A—simple flat-sided, megaphone-like horn; B—complete cone loudspeaker unit with true exponential horn baffle having a square cross-section; C—another complete cone loudspeaker unit with an exponential horn of the trumpet type, spun from aluminum.

baffles or cone loudspeakers are shown in Figs. 20B and 20C. Since these start with a large throat, they need not be extremely long in order to secure good response over a wide range of frequencies. If the loudspeaker and cone are designed with a closed back, to give high air velocity at the front of the cone, and the usual precautions are taken to secure piston action over the entire audio range of frequencies, the addition of an exponential horn of this type will greatly improve the efficiency of operation.

In Fig. 21 is shown a large exponential horn which is fed by six large cone loudspeakers. Careful design is essential to prevent reflection of

for crystal and condenser driving units), and upon the load which is placed upon the loudspeaker by the surrounding air.

We have already seen that the driving unit of a loudspeaker, when

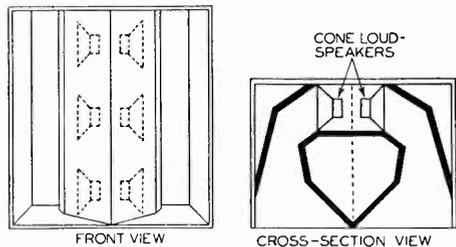


FIG. 21. Two views of an unusually large exponential horn baffle which is fed by six large cone loudspeakers. This assembly is capable of delivering high sound output power at high efficiency. It is made by Lansing Manufacturing Company, Los Angeles, Calif.

in operation, has mechanical inductance, mechanical capacitance and mechanical resistance. Let us consider a simple loudspeaker in which these three mechanical components act in series upon a dynamic cone loudspeaker having electrical inductance and resistance. The complete circuit for this loudspeaker is best represented as in Fig. 22. Here  $e_P$  represents the A.C. source voltage applied to the loudspeaker (this will be the A.C. output voltage of the audio amplifier),  $r_P$  represents the resistance of the source (the A.C. plate resistance of the output tube),  $R_V$  represents the resistance of the voice coil,  $L_V$  represents the inductance of the voice coil,  $M$  represents the electromechanical coupling,  $L_M$  the mechanical inductance (the mass of the moving element),  $C_M$  the mechanical capacitance (the compliance or springiness of the spider), and  $R_M$  the mechanical resistance. Clearly these mechanical components exist only when the driving unit is in motion; when the voice coil is wedged in position so it cannot move, the source would feel only the electrical components,  $R_V$  and  $L_V$ . By measuring the electrical characteristics of the loudspeaker under this condition, we can determine what these electrical values are.

When the voice coil and cone of the loudspeaker are free to move, and the loudspeaker is accomplishing useful work in converting electrical power into acoustical power or sound, the source feels all of the mechanical components just as if they were in an electrical circuit like that shown in Fig. 22. For example, if the source frequency is such that the mechanical reactance of  $L_M$  equals the mechanical reactance of  $C_M$ , mechanical resonance occurs and only mechanical resistance  $R_M$  is reflected through  $M$

into the primary circuit. When the source frequency is above the resonant frequency of the secondary, the mechanical reactance of  $L_M$  exceeds that of  $C_M$ , with the result that a mechanical inductance and mechanical resistance are reflected into the primary as a mechanical capacitance and mechanical resistance.

These facts are mentioned merely to show that when the electrical impedance of a loudspeaker is measured across its terminals (between terminals 1 and 2 in Fig. 22), a different value of impedance will be secured at each frequency; furthermore, this impedance will not be made up of voice coil reactance and resistance alone, but will also include those effects of the mechanical components which are felt by the source. The circuit in Fig. 22 tells us that the power input to the loudspeaker and consequently the useful sound power output will be a maximum when the source impedance (resistance)  $r_P$  is exactly matched to the load impedance (the loudspeaker input impedance) as measured between terminals 1 and 2.

Clearly the matching of the loudspeaker impedance with that of its voltage source is quite important if greatest sound power output is to be obtained. The circuit in Fig. 22 shows that the D.C. resistance  $R_V$  of the loudspeaker, which can be measured with an ohmmeter, cannot be considered alone when securing a proper match; it is the loudspeaker impedance as measured while the loudspeaker is actually in operation which must be matched to the source impedance.

*Determining Loudspeaker Impedance.* Unfortunately the impedance of a loudspeaker is not constant for all frequencies, nor is the mechanical load of a loudspeaker as simple as is represented electrically in Fig. 22.

For these reasons it is necessary to measure the loudspeaker impedance at the frequency which we wish to reproduce at greatest volume, and secure a correct impedance match for this frequency.

For the average loudspeaker an impedance match at about 1,000 cycles is usually quite satisfactory. With tweeter loudspeakers, however, a match at 4,000 cycles would be better. Loudspeaker manufacturers recognize these problems, and specify a value of loudspeaker impedance which, if matched, will give the best over-all results over the frequency range to be handled. For example, when the impedance of a loudspeaker voice coil is specified as 8 ohms, this will take into account mechanical as well as electrical factors and will probably be an impedance measurement made while the loudspeaker was reproducing a 1,000 cycle audio signal.

When the impedance of a dynamic loudspeaker is unknown, some servicemen measure the voice coil resistance with an ohmmeter and assume an impedance value equal to 1.5 times this measured D.C. resistance. We know that this value can be considerably in error, but it at least serves as a guide when a more accurate impedance value is not available.

**Matching.** Knowing the impedance of the loudspeaker at the average or predominant frequency in the range of audio sounds being handled, the next step is to match this impedance to the impedance of the electrical signal source, in order to secure maximum useful sound output. When vacuum tube amplifiers, which are the usual signal sources for loudspeakers, feel anything different from the correct, properly matched load, the wave form of the signal fed to the loudspeaker will be distorted. The reason for this is simply that with an improper im-

pedance match the output tube of the amplifier no longer operates over the linear portion of its dynamic  $E_g-I_p$  characteristic. Regardless of whether the output stage uses class A, push-pull or push-push operation, best results are obtained when the load placed on the stage has the correct impedance value.

With the loudspeaker impedance and the required load impedance of the amplifier known, there remains only the connecting together of source and load by means of a suitable matching or output transformer. Since the re-

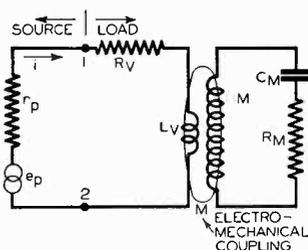
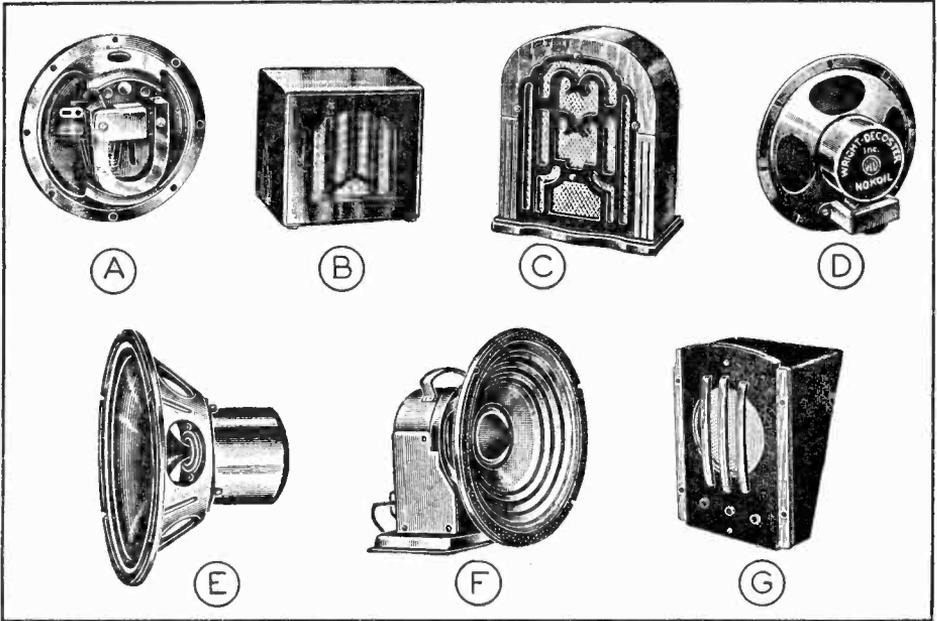


FIG. 22. Equivalent circuit of a complete loudspeaker unit connected to a source of audio frequency power.

quired plate load impedance is usually much greater than the loudspeaker impedance, a step-down transformer is needed; its turns ratio is determined simply by dividing the required plate load impedance by the loudspeaker impedance and then taking the square root of the resulting number. (Taking the square root of a number means finding a smaller number which, when multiplied by itself, will give the original number.) The primary winding, which connects into the plate circuit of the output tube or to the plates of the tubes in a push-pull or push-push output stage, will have the greatest number of turns and will therefore usually have the greatest D.C. resistance. The secondary winding has the least turns and connects to the loudspeaker voice coil leads. Here is an example: If a push-pull output stage

requires a 2,000-ohm load for greatest undistorted output power, and the loudspeaker has a specified impedance of 20 ohms, our impedance ratio is 2,000 divided by 20, which is 100. Taking the square root of 100, we get 10 as the turns ratio of the output transformer. The primary winding

*Typical Loudspeaker Matching Circuits.* A typical connection of a loudspeaker and output transformer to a single tube output stage is shown in Fig. 23A. The recommended plate load impedance of the pentode output tube and the average voice coil impedance of the loudspeaker determine



These diagrams illustrate a number of the interesting features of cone loudspeakers which are taken up in this lesson.

**A**—Balanced armature driving unit which is coupled mechanically to a medium-sized free-edge cone. This unit is commonly known as a magnetic loudspeaker.

**B and C**—Typical small cabinets used for magnetic loudspeakers.

**D**—Permanent-magnet dynamic cone loudspeaker; this is similar in appearance to many electrodynamic loudspeakers, but the trade name NOKOIL indicates a permanent magnet construction.

**E**—Special form of dynamic loudspeaker employing a para-curve or curvilinear cone in order to provide uniform response over a wide range of frequencies.

**F**—Dynamic loudspeaker employing a corrugated cone construction for uniform response over medium frequencies; the conical plug over the moving coil in the center of this cone serves to diffuse high frequency sounds in all directions and to increase the frequency range over which piston action is secured.

**G**—Special dynamic loudspeaker cabinet designed for wall mounting. Magnetic loudspeakers are also available in cabinets like this.

will therefore have 10 times as many turns as the secondary winding. With the loudspeaker connected to the secondary winding and an audio signal of average frequency being fed into the primary winding, the measured impedance of the primary will be 2,000 ohms, and we therefore have a correct impedance match.

the turns ratio required for the output transformer.

A typical loudspeaker connection for a two-tube output stage is shown in Fig. 23B. With class AB or A' push-pull operation, a C bias resistor must be inserted in the circuit at point X to provide a negative C bias for each tube, and this resistor must be

shunted with a high-capacity condenser. Since only one tube works at any instant of time, and feeds into only one-half of the output transformer primary winding, we must multiply the recommended plate load impedance for one tube by 2 in order to determine the turns ratio for the entire output transformer.

With high-impedance loudspeakers, such as those employing magnetic, condenser or crystal driving units, the loudspeaker impedance may be very close to that required by the output tube. The loudspeaker may in this case be connected as in Fig. 23A, using

placement transformer in every case in order to insure a correct match and duplicate the original performance of the receiver. This is particularly important in the case of receivers or public address amplifiers designed for high-fidelity operation.

When exact replacement transformers are not available, you can make a satisfactory replacement with little apparent change in performance by selecting a transformer having the correct turns ratio and a sufficiently large power-handling rating. Few radio men are able to order transformers on this basis, however; for

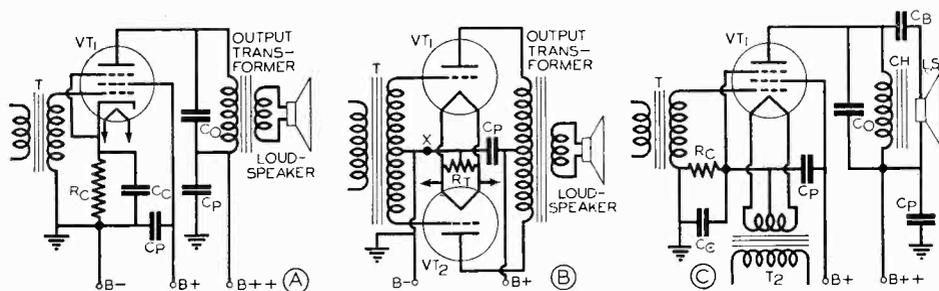


FIG. 23. Typical loudspeaker connections to output stages. Condenser  $C_0$  is often included to prevent self-oscillation of the output stage, to alter the frequency response of the amplifier, or for both reasons.

a 1-to-1 turns ratio output transformer (having the same number of turns on the primary as on the secondary), or a direct loudspeaker connection to the plate circuit may be made in the manner shown in Fig. 23C. In this latter case the D.C. plate current of the output tube flows through iron-core choke coil  $CH$  to the power supply, while signal currents flow through D.C. blocking condenser  $C_B$  and the loudspeaker. The inductance of choke  $CH$  must be sufficiently high to prevent excessive bypassing of audio frequency current around the loudspeaker.

**Replacement of Output Transformers.** When output transformers become defective and must be replaced, it is best to use an exact re-

this reason radio supply houses list replacement transformers for use with particular output tubes and for either single- or two-tube output stages. These transformers are designed on the assumption that most dynamic loudspeakers have a normal voice coil impedance of about 10 ohms.

Radiotronics oftentimes prefer to use universal output transformers, a few of which they can keep on hand at all times. These transformers are available in various power ratings, and have a number of taps on the secondary to provide various turns ratios. The primary winding has a center tap which is used with two-tube output stages but is ignored in the case of a single-tube output stage. The taps on the secondary usually

provide for voice coil impedances of from 1 to 30 ohms. Use that tap which gives the greatest volume along with clearest tone when an actual radio program is being reproduced. For a more accurate determination of the correct tap, connect a cathode ray oscilloscope to the voice coil terminals, feed into the audio amplifier a pure sine wave signal first at 100 cycles, then at 1,000 cycles and 4,000 cycles, and select that tap which gives the greatest output voltage along with sine wave output for all three of these frequencies.

### Field Coil Connections for Electrodynamic Loudspeakers

Most of the dynamic or moving coil loudspeakers used in radio receivers are of the electrodynamic type, employing a field coil for the production of a fixed magnetic flux. The average power used is from 7 to 15 watts. In many cases these coils are made to serve also as chokes for the power pack system, for at power line frequency the field coil has an appreciable inductive reactance. When used in this manner in place of a filter choke, the D.C. resistance of the field coil becomes important, for this resistance reduces the net D.C. supply voltage which is available for the tubes in the receiver.

The loudspeaker field coil connection shown in Fig. 24A is without doubt the most widely used connection today in radio receivers. The D.C. resistance of a field coil used in this manner will usually be somewhere between 500 and 2,000 ohms. Since the D.C. current drawn from the power pack by the various receiver circuits must pass through this field coil, the coil must be capable of handling this current without excessive overheating. When the receiver has a push-pull output stage, com-

plete filtering of the plate and screen grid voltages for the stage is not necessary, and a somewhat higher D.C. voltage is obtained by connecting to point *X* rather than to *B+*.

Since the power pack output voltage in a universal A.C.-D.C. receiver is considerably lower than that in an A.C. receiver, any loudspeaker field coil connection which would reduce this output voltage even more is obviously not desirable. For this reason the field coil of a loudspeaker in a universal receiver is usually independently supplied with rectified current in a manner similar to that shown in Fig. 24B. In this circuit, one half of the 25Z5 double diode rectifier tube supplies current for the field coil, with filter condenser *C*<sub>1</sub> removing the pulsations in the half-wave rectified output current. The other diode section of this tube likewise delivers half-wave rectified current which is smoothed by filter condenser *C*<sub>2</sub> and choke coil *CH*. This choke coil is sometimes replaced by a resistor or is omitted entirely, and a higher-capacity filter condenser is used to provide the necessary filtering. The D.C. resistance of the loudspeaker field coil is usually larger than 2,000 ohms in a circuit of this type, in order to limit field coil current and prevent overloading of the rectifier.

In addition to providing a fixed magnetic flux and filtering the power pack output current, a loudspeaker field coil is sometimes made to serve a third purpose, that of providing a negative C bias voltage for the receiver stages. A tap is provided on the field coil for this purpose, as indicated in Fig. 24C; the value of the bias voltage produced depends upon the resistance which is present between the *B*-- terminal of the power pack and the tap, and upon the amount of D.C. current flowing

through this resistance. The entire D.C. voltage drop across the loudspeaker field coil can also be utilized for C bias purposes; the C -- terminal in Fig. 24C thus provides a greater negative bias voltage than the C -- terminal.

*Separate Field Supplies.* Loudspeakers used with public address amplifier systems must often be located a considerable distance away from the

Fig. 24E. The vacuum tube rectifier normally supplies an output voltage of over 250 volts, and the field coil used with this supply must therefore have a high resistance in order to limit the current to a safe value. With the dry disc type of rectifier, the output voltage will usually be less than 100 volts, and a lower-resistance field coil can be used. The present trend is toward use of permanent magnet

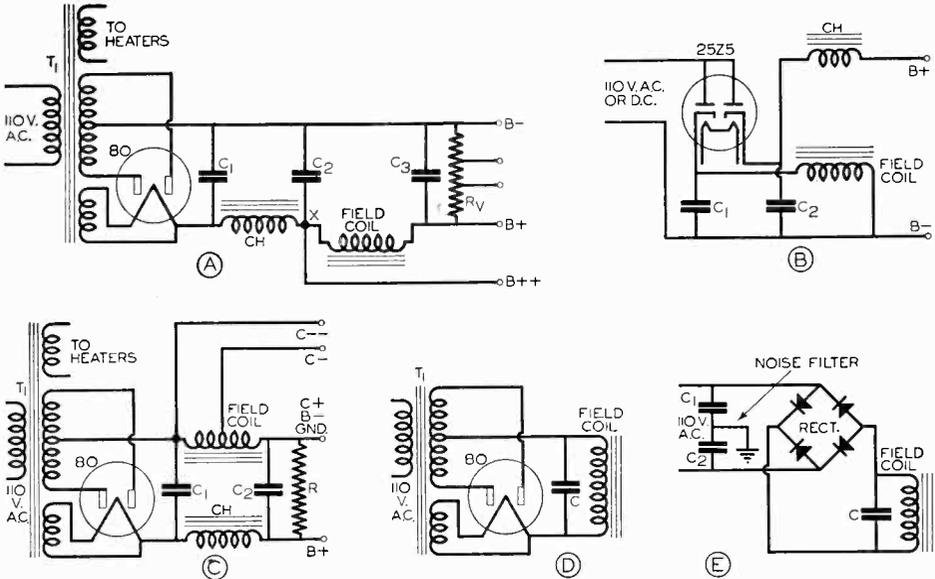


FIG. 24. Typical electrodynamic loudspeaker field supply connections.

amplifiers. If field coil leads were run from the loudspeaker location to the amplifier, the resistance in these leads would be excessive. The same holds true for extra loudspeakers, which are often connected to radio receivers but located some distance away from the receiver. In cases like these, it is necessary to use a separate field supply located at the loudspeaker if it is of the electrodynamic type. This field supply may consist of a full-wave vacuum tube rectifier like that shown in Fig. 24D, or may be a full-wave bridge type rectifier using copper-oxide discs, as indicated in

dynamic loudspeakers, however, thus eliminating entirely the need for a separate power supply at the loudspeaker location.

With battery receivers the extra current drain due to the field coil of an electrodynamic loudspeaker would obviously be undesirable; for this reason the loudspeakers used in battery receivers will generally have either a balanced armature type magnetic driving unit or a permanent magnet dynamic driving unit. Auto radio receivers use electrodynamic loudspeakers extensively, with the field coil so designed that it can be con-

nected directly to the 6-volt storage battery; permanent magnet dynamic loudspeakers are also used with auto radios.

*Elimination of Hum.* We have seen how the rectified current supplied to the field coil of the loudspeaker in an A.C. receiver is usually incompletely filtered. Up to a few years ago this ripple current was not objectionable because the average loudspeaker was not able to reproduce sounds as low as the hum frequency of 120 cycles. Recent improvements in loudspeaker design have extended the low frequency response of the loudspeaker well down to the lowest audible sound frequencies; the result is that any ripple currents induced in the voice coil are heard as objectionable hum. This condition arises because magnetic flux produced by the field coil passes through or links with the voice coil, giving the effect of a transformer in which A.C. or ripple current going through the primary induces corresponding A.C. voltages in the secondary or voice coil. The two basic methods used for eliminating hum due to A.C. in the field of a dynamic loudspeaker are: 1, use of a shading ring; 2, use of a hum-bucking coil.

*Shading Ring.* A large copper ring called a *shading ring* or shading coil is placed around the central core, as shown in Fig. 25A. This shading coil acts as a short-circuited turn which, when A.C. currents flow through the field coil, sets up an out-of-phase A.C. flux in the core to cancel out the effects of the original A.C. flux. The result is that no A.C. voltages are induced in the voice coil.

*Hum-Bucking Coil.* A more widely used scheme, shown in Fig. 25B, involves winding on the central core a number of turns of wire (usually a few less than are in the voice coil). This extra coil, known as a *hum-*

*bucking coil*, is connected in series with the voice coil but is wound in the opposite direction. Any A.C. flux which is present in the core induces an A.C. voltage in the voice coil and also in the extra coil. Since these voltages are out of phase and acting in series, they balance or buck each other, and no A.C. current flows through the voice coil to react with the permanent magnetic flux. Circuit diagrams usually indicate when a

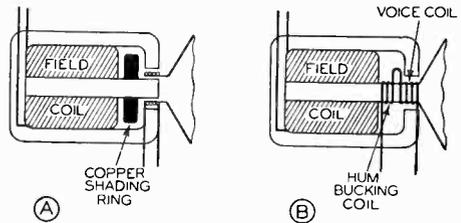


FIG. 25. Here are two methods used by loudspeaker design engineers for overcoming hum which is due to the presence of A.C. components in the field coil current of an electrodynamic loudspeaker.

hum-bucking coil is present in the loudspeaker; typical schematic circuit diagrams illustrating this hum-elimination feature are shown in Fig. 26.

### Acoustical Problems of High-Fidelity Reproduction

The output of the average loudspeaker begins to drop rapidly at frequencies above 4,000 cycles. This drop in high-frequency output can be offset to a certain extent by designing the audio amplifier to have a peak response in the high-frequency region, but there is a limit to the amount of treble-boosting which can be incorporated in the audio system without introducing new circuit problems. This is one of the reasons why modern high-fidelity loudspeaker systems usually employ two separate loudspeakers, a woofer, which is capable of giving flat response up to about

3,000 cycles, and a tweeter, which provides satisfactory reproduction of the higher frequencies. Typical response curves for a woofer and a tweeter are shown in Fig. 27. The

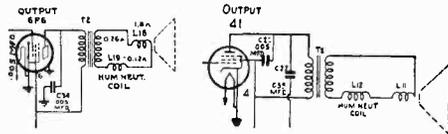


FIG. 26. Schematic circuit diagrams of radio receivers usually indicate, in a manner similar to that shown above, whether a hum-neutralizing (hum-bucking) coil is used in the loudspeaker.

dotted line gives their combined response over the region in which they overlap, and shows that the combined effect is a greatly increased range over which flat response is secured. A low-pass filter (often simply a shunting condenser) is placed in the woofer circuit and a high-pass filter is placed in the tweeter circuit (a series condenser will often suffice) to prevent useless currents from overloading the loudspeakers.

In public address systems, both woofer and tweeter may be of the horn type, or the woofer may be a dynamic cone loudspeaker using an exponential horn baffle, with a horn type tweeter. Deflecting vanes are often built into the tweeter to broaden its directional characteristics.

In home radio receivers the woofer

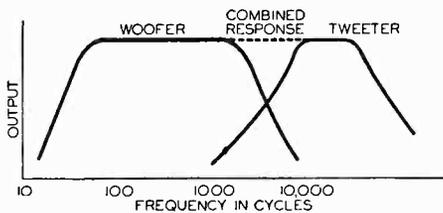


FIG. 27. Response curves for a high-fidelity, two-loudspeaker arrangement.

is usually a dynamic cone loudspeaker mounted in a box baffle having acoustic low-pass phase-reversing construc-

tion, with a tweeter loudspeaker of either the cone type using a crystal driving unit or the trumpet horn type also mounted on the baffle. A typical woofer-tweeter combination in a special box baffle is shown in Fig. 28.

*Room Acoustics.* Although most people recognize the fact that a theatre or auditorium must be properly treated with sound-absorbing material in order to give natural reproduction of sounds, the average radio receiver owner gives very little attention to the acoustic treatment of the room in which his radio receiver

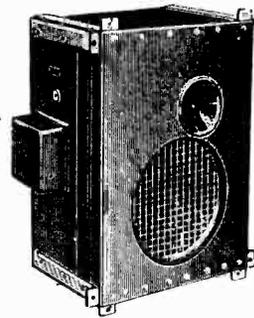


FIG. 28. A large dynamic cone loudspeaker (woofer) and a horn type tweeter are here mounted on a single box baffle to give uniform response over a wide range of audio frequencies. The inside of the box is lined with sound-absorbing material to prevent standing waves.

is located. It is the duty of the Radiotrician to point out that a room having overstuffed furniture, heavy draperies, and a large, heavy rug is desirable. There will then be sufficient sound absorption to prevent sounds from being reflected back and forth between the walls of the room and various objects in it, and there will be no echo or reverberation effects. The Radiotrician can also recommend that the loudspeaker be pointed or directed in such a way that sound can travel a maximum distance before being reflected. If the radio is in one corner of a room, the receiver should be facing another corner

diagonally opposite, giving a long path through air for the radiated sounds.

High-frequency sounds are so directional in nature and so easily absorbed by rugs and furniture that they should be heard directly rather than after reflection. Some receivers utilize inclined sounding boards or baffles which cause these high-frequency sounds to be directed at an upward angle, away from the floor, for this purpose.

By this time it should be quite clear to you that the securing of high-fidelity reproduction in a radio receiver system or public address system can be best obtained only by proper design of each and every link in the long chain between the microphone in the broadcast studio and the listening location in the home.

*Loudspeaker Replacement Hints.*  
Tone controls can compensate to a

certain extent for deficiencies in loudspeaker response, but manufacturers of high-quality receivers will sometimes introduce equalizer circuits in an audio amplifier in order to make the amplifier response peaked in a region where the loudspeaker response is low. The combination response of the system can in this way be made almost uniform for high sound levels. Tone controls then correct for the peculiarities of the human ear at low sound levels. Changing of loudspeakers may therefore alter the performance of a high-fidelity receiver, unless an exact duplicate replacement loudspeaker is used. When the loudspeaker cone in a high-fidelity receiver must be replaced, always use the exact duplicate replacement cone which is supplied by the receiver manufacturer; cone design has considerable influence on loudspeaker response.

# Lesson Questions

Be sure to number your Answer Sheet 27FR-2.

Place your Student Number on every Answer Sheet.

Send in your set of answers for this lesson immediately after you finish them, as instructed in the Study Schedule. This will give you the greatest possible benefit from our speedy personal grading service.

1. Why are large flat diaphragms unsatisfactory for loudspeakers?  
*They are not springy and also buckle*
2. What two purposes does a spider serve in a dynamic loudspeaker?  
*To center the voice coil and acts restoring spring*
3. Does mechanical resonance occur at high, medium or low frequencies in dynamic cone loudspeakers?  
*low*
4. Why is pure piston action desirable in the cone of a dynamic loudspeaker?  
*for uniform response to all freq.*
5. Is an irregular shape of baffle better than a square or round baffle?  
*yes*
6. Can the natural resonant frequency of a box baffle be lowered by increasing the size of the box?  
*yes*
7. How can the undesirable effects of cavity resonance be reduced in a box type baffle?  
*increase path of air and use sound absorber mat.*
8. What are the four distinct purposes of the acoustical labyrinth?  
*prevent cav. res., damp, control, cut off*
9. What are the approximate maximum obtainable efficiencies for a large cone loudspeaker in a box baffle and for a driving unit used with an exponential horn?  
*.03% 50%*
10. What are the two basic methods of eliminating hum due to A.C. in the field of an electrodynamic loudspeaker?  
*shading ring  
Run backing coil*



## DELIVER THE GOODS

“There are 57 rules for success. The first is to deliver the goods. Never mind the rest.”

Like many striking assertions, the quotation above is not altogether true, because there are other rules which cannot be ignored. But there is a lot of truth in this statement.

If you want to be a success in life, deliver the goods. Employers want men who *earn* their salary each and every day—men who have the training required for their particular jobs and actually do apply this training to their work.

You can always excuse yourself if you fail—but nobody else will ever excuse you. Customers may be polite to you and may feel sorry for you, but they will go elsewhere the next time they need radio service work. Employers are equally indifferent to excuses, for they must have good men if they themselves are to deliver the goods.

Make it your business to be where you are needed, when you are needed, with the service or help that is needed. Have all the knowledge which may be required. Be the man who delivers the goods and gets the money.

*J. C. Smith*

**CURRENT, VOLTAGE AND  
RESISTANCE MEASUREMENTS**

28FR-3

**NATIONAL RADIO INSTITUTE**

ESTABLISHED 1914

**WASHINGTON, D. C.**



# STUDY SCHEDULE NO. 28

For each study step, read the assigned pages first at your usual speed. Reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind, then answer the Lesson Questions for that step. Study each other step in this same way.

1. Direct Current Meters ..... Pages 1-10

The most widely used measuring device is the D'Arsonval d.c. meter. Learn its principles and all other meters are easy to understand. Here you learn how the meter range is extended and how meters are calibrated. This section is full of practical information on using current meters—information which may prevent your damaging your equipment. Answer Lesson Questions 1, 2, 3 and 4.

2. Alternating Current Meters ..... Pages 10-19

There are many ways of measuring alternating current. These methods have advantages and disadvantages which you must keep in mind; otherwise, you can get misleading readings. Here the principles of these methods are fully developed. Pay particular attention to the information on the copper-oxide rectifier, as this is the most widely used a.c. rectifier. Answer Lesson Question 5

3. Voltmeters ..... Pages 19-22

Voltages are measured by the same kind of meter used for current. The important difference is the arrangement of resistors used with the meter. A voltmeter will not always indicate the true operating voltage, depending on its sensitivity. The reason for this, as well as the practical means of allowing for this condition, is clearly brought out by practical examples. You won't be confused by your voltmeter readings after reading this section. Answer Lesson Questions 6 and 7.

4. Ohmmeters ..... Pages 22-25

The ohmmeter is probably the most used service instrument. With it you determine resistor values, check condensers for shorts or leakage and check coils and circuits for continuity. Practical data on using and safeguarding this valuable device is given. Answer Lesson Questions 8 and 9.

5. Multimeters ..... Pages 25-28

The multimeter is a combination current meter, voltmeter and ohmmeter, built into a single unit for convenience in using. The two basic types are covered, so you will know how to use either. Of course, there are many different scale ranges available, as each manufacturer has his own ideas as to the most desirable ones. Practically all the standard makes are good, however, once you learn to read their scales. This section closes with an important review of meter tolerances and their practical importance. Answer Question 10.

6. Mail your Answers for this Lesson to N.R.I. for Grading.

7. Start Studying the Next Lesson.

# CURRENT, VOLTAGE AND RESISTANCE MEASUREMENTS

## Direct Current Meters

**M**ETERS are used constantly in radio work of all kinds. As a radio serviceman, you'll make at least a few meter measurements on practically every set you handle.

Early in your N.R.I. Course you learned the elementary facts about what these instruments are and how they are used to measure current, voltage and resistance. In this lesson you'll learn the practical details of actual commercial meters—how they're made, how they work, what to expect from them, and—most important of all—the correct ways to use them.

### HOW IS CURRENT MEASURED?

We cannot see an electric current, but we can detect a number of effects produced by its flow. We might use any one of these effects to measure the amount of current flowing.

For example, we might use a thermometer to measure the heat produced by a current flowing through a wire. From this heat measurement, we could calculate the amount of current.

Or, we might send the current through a resistance wire that will become hot enough to give off light—a common light bulb, for example. By measuring the amount of light produced, we could determine the amount of current flow.

Again, we might make the current flow through a certain silver salt solution, causing a silver electroplating

action. By weighing the amount of silver thus "plated," we would have another means of measuring current. (This last, under carefully controlled conditions, is an international standard method of determining exact current flow.)

But these methods are generally too slow and cumbersome for practical uses. We need a device which will react at once to changes in current and which can be used anywhere, not just under laboratory conditions.

One of the first electrical discoveries was that the magnetic field which always accompanies a current flow would make a compass needle move. Experiments soon proved that the distance the needle moves depends on the amount of current flowing. This was a highly important discovery, for it pointed out the basic principle used in most modern meters. Let's now see how these meters use this magnetic effect to measure current.

### THE D'ARSONVAL METER

Early experimenters soon found that the compass needle method of measuring current was not accurate enough, since it depended on the amount of magnetism in the needle and the effect of the earth's magnetic field. A meter (known as the D'Arsonval type, after its inventor) was invented which overcame these objections.

Instead of placing a compass within

a coil, a small coil is suspended so it can move, and is placed in a strong magnetic field. When the current to be measured flows through the coil, a magnetic field is set up which interacts with the fixed field, causing the coil to move. The basic principle of this operation is shown in Fig. 1. You remember that like magnetic poles will repel. If a small pivoted magnet is placed as shown in Fig. 1A, it will move

coil. Therefore, one particular value of current will make the coil rotate to one particular place. A larger current will rotate the coil farther; a smaller one will rotate it less. By attaching a pointer to the coil, the amount of movement can be indicated on a scale. This scale, when properly marked, will show us the amount of current flowing through the meter.

This is the basic principle of the D'Arsonval type meter movement. It has been developed into the modern, practical meter shown in Fig. 2. Let's look at its features.

The small, powerful magnet is made from special steel or metal alloys, chosen for strong magnetic qualities and long magnetic life. The stronger the fixed field, the more the coil will move for a particular current, so the permanent magnetic strength is made as high as possible. The magnet is specially treated and aged until the field strength remains constant.

The pole pieces are soft iron, carefully shaped to give the desired magnetic field distribution. If the meter scale is to be linear (that is, so adjusted that equal increases in current will produce equal increases in meter movement), the magnetic field must be uniform through the gap in which the coil turns. A soft iron core is inside the coil form. The coil moves through a small gap between this core and the pole pieces. By making this gap small to reduce the reluctance of the magnetic path, we obtain a strong field.

To make the coil easy to move, it is wound on a very light, thin metal form, and the coil and form are suspended between almost frictionless pivots with jewel bearings. The number of turns and size of wire used to make the coil depend on the current range and sensitivity desired for the meter.

The coil must start to move from

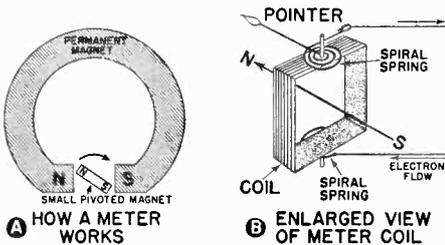


FIG. 1. The coil has a magnetic field and will move when placed in a fixed field.

in the direction of the arrow. A current-carrying coil, such as the one shown in Fig. 1B, has a north and a south pole. If this coil were put in place of the small magnet in Fig. 1A, it too would move.

Unless some means are taken to stop it, once the coil starts to rotate, it will continue to do so. That is why the two fine spiral springs are attached at the ends of the coil. When the coil moves, the springs are twisted so that they oppose the coil movement. The more the coil rotates, the more opposition the springs give. Thus, the coil moves only to the point where the magnetic force causing rotation is balanced by the retarding action of the springs. When the current stops, the magnetic field of the coil disappears, and the springs move the coil back to the starting position.

The magnetic force causing rotation of the coil is proportional to the amount of current flowing through the

the same position each time, in order for the pointer to come to rest at the proper point on the meter scale. This starting, or zero, position is determined by the springs. They are wound in a loose spiral so that they oppose any rotation which would either wind them tighter or unwind them. When the coil moves, one is wound while the other is

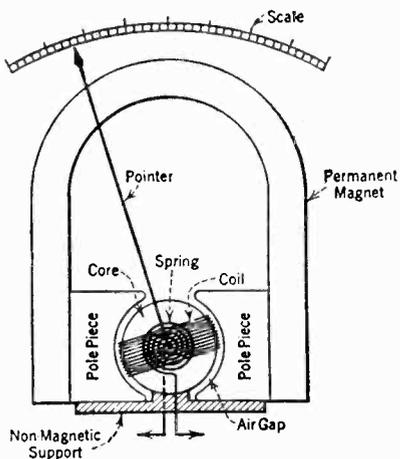


FIG. 2. The names of the meter parts.

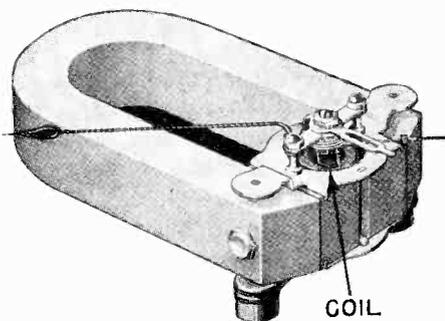
unwound. The springs thus oppose a coil movement in either direction away from the starting position, which is the position the pointer takes when no current is flowing in the coil.

Naturally, these springs won't always remain perfectly balanced. That is why practically all good meters have a zero adjustment. This is a small screw which usually protrudes through the case of the meter just below the meter coil. By turning this screw, the upper coil spring can be moved enough to balance the springs and bring the meter pointer back to the zero position.

The springs are also used to make the electrical connections to the coil, as shown in Fig. 1B. The springs are metal, so they make ideal leads. Of course, this means they must be insulated from each other and from the meter frame.

**Meter Damping.** When a current flows through it, the coil (because of its inertia) tends to overshoot or move past the correct point a small distance. Then the springs pull the coil back on the other side of the correct point. In other words, there is a wavering back and forth for a time before the meter pointer settles down to the proper readings. To keep this wavering as small as possible, the meter coil frame is made of metal. When it moves through the permanent magnet field, eddy currents are induced in the frame. These eddy currents set up an opposing field which slows down the rate of coil movement, so that the coil comes to the correct point and stops. This steadying action is called damping.

Another common method of damping is to place a resistor between the meter terminals. A voltage is induced in the coil as it moves through the fixed field. This induced voltage causes a current flow through the resistor and coil, which sets up an opposing field with a result like that caused by the eddy currents induced in the coil form. In both cases, the damping action ceases as soon as the coil stops moving.



What the inside of a meter looks like.

The resistor value which will permit the most rapid coil movement, without noticeable wavering at full-scale meter reading, is called the *critical damping value*. This value varies widely with different meters—some may require

10,000 ohms, while others may need as low as 100 ohms. A resistor below the critical value causes overdamping and a slow meter movement, while too large a resistor does not damp enough.

The eddy current method of damping does not affect the meter range at all. The resistor method may or may not affect the current range of the meter, depending on the value of the resistance needed. You will learn more about this later on in this lesson.

## CALIBRATION

The meter we have described will indicate when current is flowing. However, we are usually most interested in the actual amount of current flowing, so we want the meter to have a scale marked off in units of current flow.

By passing known currents through a meter, we can mark the scale according to the deflection obtained and thus calibrate the meter. "Calibrating" is the technical term for finding the accurate values of points on the scale of an instrument. For example, suppose you had a stick of wood with scratches on it that you wanted to use as a ruler. You'd "calibrate" it by finding how many inches each scratch was from one end of the stick. Meters are calibrated the same way—except, of course, each point on the scale is marked in terms of the current, voltage or other quantities needed to make the meter needle swing to that point.

Large laboratories, such as the Bureau of Standards, determine currents very accurately by chemical or other means and hand-calibrate a standard meter. Then the standard meter is placed in series with a meter to be calibrated, as shown in Fig. 3. The amount of current passing through the two meters is varied in steps, and the meter readings are noted for each step. Since the standard meter is indicating the exact current flowing through both

meters, the other meter can be directly calibrated from the standard. This method is used in the manufacture of the more accurate and expensive meters.

Meters of reasonable accuracy and much lower in price can be made by duplicating the hand-calibrated meter as closely as possible in all parts. These production meters usually have about the same response as the meter on

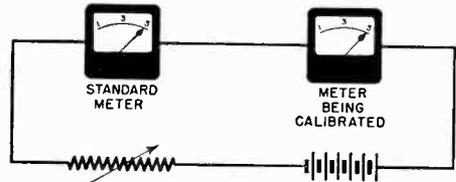


FIG. 3. Calibrating a meter by comparison.

which they were modeled, so the same scale can be used for all of them with fair accuracy. To improve the accuracy of their mass-produced meters, manufacturers adjust the spring tension and magnetic field so that the full-scale reading of each meter is correct.

**Percentage Accuracy.** Highly accurate, hand-calibrated meters are expensive, and not necessary for service work or ordinary testing. Servicemen generally use production meters with printed scales. The percentage of accuracy of such meters is expressed in terms of full-scale deflection. For example, suppose a milliammeter has a full-scale reading of 100 ma. If it is off 5 ma. when the needle gives a full-scale deflection, the meter is said to have a **5% accuracy**. ( $5/100 = .05$  or 5%). In other words, when the needle points to 100 ma., the actual current being measured may be 95 ma. or 105 ma., or any value in between.

Now this 5% accurate meter may have much poorer accuracy for readings at the low end of the scale. When the meter points to 10 ma., the meter may still be wrong by 5 ma., which

would mean a 50% error. ( $5/10 = 0.5$  or 50%). Of course, most of the better meters are held within 1% or 2% of full scale, and the inaccuracy at the low end of the scale is not as great as that given in our example. But here's a practical tip—if several meter ranges are available, use the one giving a deflection of more than half-scale. Then your reading will have the least error.

**Parallax.** Inaccurate readings may result from looking at a meter scale from an angle rather than from right above it. Figure 4 shows you why.

In order to swing freely, the pointer has to be at least a little distance above the scale. Now when you look right down on the pointer, you get one reading. Say it's 10. If you are off to one side, the pointer will seem to give a different reading. It might be 9 or 11, depending on which side you are. The farther you are to either side, the farther off the apparent reading will be from the true one. (You can prove this for yourself by trying to read your watch at an angle.) This shift in the apparent position of the pointer, due

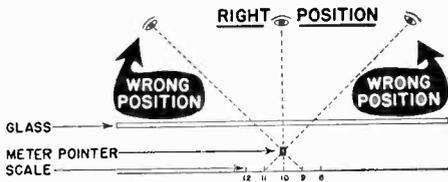


FIG. 4. Always look straight at a meter to avoid parallax.

to the position of the observer, is called parallax.

Most laboratory meters have a small mirror on the scale under the pointer, so unless your eye is directly above the pointer, you will see a reflection of the pointer in the mirror. With such meters, you can avoid parallax by always moving to where the reflection disappears behind the pointer before making the reading.

In service and communications

work, highly accurate readings are usually unnecessary. Even so, good service meters have pointers shaped like knife blades. These pointers look thinnest when viewed from the correct position.

## EXTENDING CURRENT RANGES

In practical service and laboratory work, both large and small currents have to be read. We could use a number of different meters having the re-

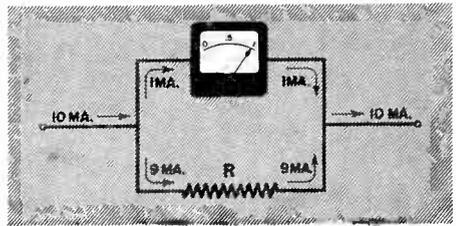


FIG. 5. The current flow divides between the meter and shunt.

quired ranges, but meters are costly and bulky. It is far better to have one meter and a means of changing its current range.

**Shunts.** Suppose we have a 1-ma. meter and want to measure 10 ma. We can do so by putting a resistor across the terminals of the meter of such a value that 9/10 of the current coming into the resistor-meter combination will flow through the resistor and only 1/10 will flow through the meter. In other words, we use the resistor to "bypass" 9/10 of the current. Figure 5 shows how this is done. When we read the meter, we simply remember that there is 10 times as much current flowing as the meter shows—so we multiply the meter reading by 10 to get the actual current flow. If our meter reads 0.5 ma., we know the real current is 5 ma. ( $0.5 \times 10 = 5$ ); if it reads 1 ma., the actual current is 10 ma. ( $1 \times 10 = 10$ ).

Since the "by-pass" resistor  $R$  makes

a parallel (or shunt) path around the meter, it is called a shunt. The ohmic value of  $R$  is calculated so that it will pass a current equal to the difference between the total current to be measured and the amount the meter needs for full-scale deflection. For example, suppose we want to measure currents up to 25 ma., but all we have is a 1-ma. meter. Then we'll have to find a shunt  $R$  which will pass 24 ma.

To do this, we use Ohm's Law,  $R = E/I_s$ . The voltage  $E$  is the voltage across the meter terminals. This voltage is called the meter millivolt rating and is equal to the basic meter current range multiplied by the meter resistance, so we can substitute for  $E$  in our Ohm's Law equation the quantity  $I_M \times R_M$ . ( $I_M$  is the basic meter current;  $R_M$  is the meter resistance.) The current  $I_s$  is the current which passes through the shunt. This current equals the total current minus the basic meter current, so we can substitute for  $I_s$  the quantity  $I_T - I_M$  ( $I_T$  is the total current flowing;  $I_M$ , as before, is the basic meter current.)

Now our Ohm's Law equation reads:

$$R = \frac{R_M \times I_M}{I_T - I_M}$$

We can find  $R$  by substituting values for the other terms. (All currents must be figured in the same units—that is, all in milliamperes or all in amperes. Then  $R$  will come out in ohms.)

Suppose our 1-ma. meter has a resistance of 100 ohms. The current  $I_T$  we want to measure is 25 ma. Then,

$$R = \frac{100 \times .001}{.025 - .001} = \frac{.1}{.024} = 4.2 \text{ ohms,}$$

and we've found the size shunt we need.

To find the actual total current flowing, multiply the reading on a shunted meter by the ratio of the current range of the meter with shunt to the current range without shunt. In our example, a 1-ma. meter was made a 25-ma. meter by the use of a shunt, so meter

readings must be multiplied by 25 to learn the actual current flow ( $25/1 = 25$ ). Remember—the meter itself is not passing 25 ma. Only 1 ma. goes through the meter, and 24 ma. go through the shunt.

You remember that a shunt resistor helps damp a meter. Most shunts have a resistance less than the critical damping value, so the meter movement is slowed down somewhat. This overdamping is not usually objectionable, however.

### MULTI-RANGE CURRENT METERS

Very likely the meter you'll use in your service work will have a number of ranges, with several built-in shunts and a switch arrangement. Figure 6 shows one way such a meter might be made. When the selector switch is in position 1, the meter is not shunted. In other switch positions, different shunts are thrown in, giving various ranges. Typical ranges might be: 1 ma.; 10 ma.; 100 ma.; 10 amperes. Whatever

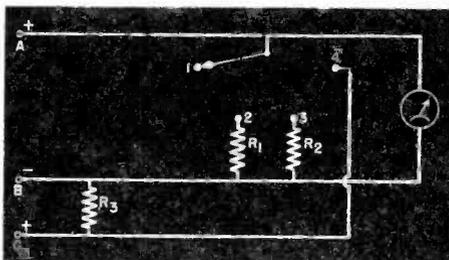


FIG. 6. Multi-range current meter.

the ranges, the manufacturer will mark the scale for each range so you won't have to do any figuring. The only usual exception is when one range is 10 times another, as when the ranges are 10 ma. and 100 ma. There, you may have to multiply by 10 when using the 100-ma. position, getting your readings from the 10-ma. scale. This is easy, however.

The greater the current range extension, the smaller the shunt resistor

must be. With meters of low resistance, the shunt necessary to extend the range above 1 ampere is usually only a fraction of an ohm—so even a small amount of switch-contact resistance in series with the shunt might cause a large current to flow through the meter. To avoid this contact-resistance trouble on the high current range, many service instruments use the extra terminal *C* shown in Fig. 6. Terminals *A* and *B* are used for the basic range and for the low extensions. For the high-current range, terminals *B* and *C* are used. Shunt  $R_3$  is connected right between *B* and *C*. The selector switch is now in the meter circuit. In measuring high currents, switch-contact resistance will not matter greatly, as it adds to the meter resistance instead of to the shunt resistance. Also, the high current does not flow through the selector switch contacts. However, Fig. 6 does not represent the best arrangement, as it is possible to damage the meter if the switch develops too much contact resistance in position 2 or 3 or should open-circuit when switching.

This damage is avoided in the circuit shown in Fig. 7, which is known as a "series or ring shunt." When the selector switch is in position 1, the lowest range is obtained. Note that this is not the basic meter range, as the resistors  $R_1$ ,  $R_2$  and  $R_3$  are acting as a shunt across the meter.

When the switch moves to position 2, resistors  $R_2$  and  $R_3$  are the shunt, while  $R_1$  is added to the meter resistance, since it is now in series with the meter. Similarly, in position 3,  $R_3$  is the shunt while  $R_1$  and  $R_2$  are in series with the meter.

There are two advantages to this arrangement; the meter is protected against damage at all times, and the resistors are relatively independent of the meter resistance, so they can be chosen to be any reasonable values.

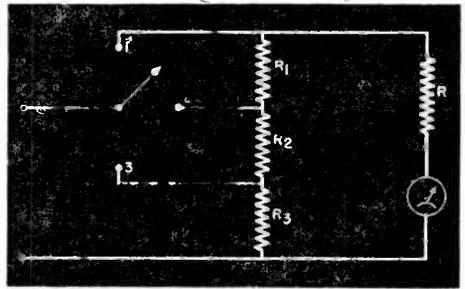


FIG. 7. The use of a ring shunt makes this a better multi-range current meter.

Switch-contact resistance causes no trouble, since it adds no resistance to either the meter or shunt circuit. Should the switch become defective, the entire meter circuit is opened and protected instead of damaged.

## USING A DIRECT CURRENT METER

There are five things you must keep in mind when using a current meter. Let's see what they are and why they're important. Then we'll summarize our findings in five easy-to-remember rules.

**1. Series Connection.** The most important thing to remember is that the current meter must be *in series* with the circuit. If necessary, unsolder a lead or cut a wire so the meter can be placed in the circuit. *Remember, the circuit current you want to measure must pass through the meter itself. If a milliammeter is ever connected across a circuit part (in parallel), it is almost certain to be burned out.*

**2. Correct Point.** In the simple circuit shown in Fig. 8, the meter may be placed anywhere in the circuit. The meter readings at *A*, *B* and *C* will be the same, because the same current flows through the entire circuit. However, when checking screen-grid or pentode tubes another problem arises. The screen-grid current also flows in the cathode lead in returning to the B supply. Hence, the plate current alone

can be measured only at positions *B* or *C*. Position *A* will give both screen and plate currents together.

When more than one tube is used, as shown in Fig. 9, notice there are now two paths for current, one for each tube. If you want to measure the plate current for tube  $VT_1$  only, be careful to locate a point where *only* the plate current of this tube is flowing. The cathode lead of tube  $VT_2$  may connect

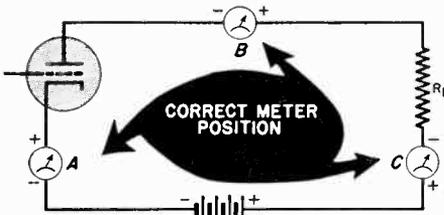


FIG. 8. Be sure to place a current meter in series and use the proper polarity.

directly to the cathode of  $VT_1$  and the  $B+$  ends of resistors  $R_1$  and  $R_3$  may be directly connected together, so positions such as *A* and *C* of Fig. 8 may not be usable. The shaded areas of Fig. 9 show the proper places to measure the plate current of  $VT_1$ . You must be sure to insert your meter where only the *desired* current is flowing. Otherwise, false readings will result. The meter may even be damaged if the sum of the currents exceeds the meter range you have chosen.

**3. Current Range.** The current range of the meter must be higher than the expected current; otherwise, the meter may be badly damaged or even ruined. The circuit current is determined by Ohm's Law,  $I = E/R$ , where  $R$  is the total of all the resistance in the circuit. In a circuit like Fig. 8, with a fixed supply voltage, the tube plate resistance plus resistor  $R_1$  determines the current flow. The current will be about 5 to 10 ma. if this is an ordinary amplifier tube. However, if the tube or resistor  $R_1$  is short-circuited in some

way, the resistance of that part is removed and the current will be many times higher—perhaps enough to burn out the meter.

This danger is the reason that current measurements are not made very often in service work, except where voltage and resistance measurements indicate that it is safe to do so. When using a multi-range current meter, *always start with the highest current range* and switch to lower ranges if the readings permit. Voltage and resistance measurements properly made will usually make it possible to calculate the current, saving you the trouble of opening the circuit except in rare instances. (In transmitters it is necessary to know certain current values at all times for adjustment and maintenance purposes, so current meters are permanently connected in the proper plate and grid circuits.)

**4. Meter Resistance.** In the usual plate circuit, the tube and other parts

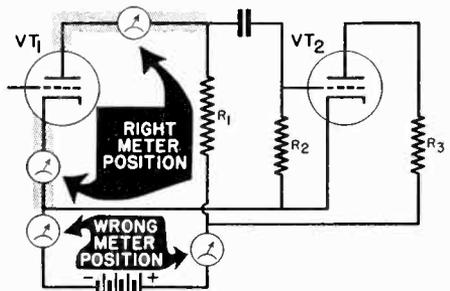


FIG. 9. The right meter position is important.

have resistances many times the meter resistance, so inserting the meter does not affect the amount of current flow much. For example, a meter of 100 ohms added into a circuit of 10,000 ohms obviously will not greatly change the circuit resistance. Suppose we were to apply 200 volts to a 10,000-ohm resistor. From Ohm's Law ( $I = V \div R$ ), the current would be .02 ampere or 20 ma. Now if the circuit resistance is in-

creased 100 ohms by the insertion of a 100-ohm meter, the total resistance will be 10,100 ohms. The current then will be .0198 ampere or 19.8 ma. You would not be able to read the average meter close enough to tell the difference between 19.8 ma. and 20 ma., so for all practical purposes, the meter has not changed the current flowing in the circuit.

However, if the meter is inserted in a low-resistance circuit, it will frequently change the current or may even determine the amount of current flow. For example, a tube filament may have a resistance of 40 ohms. If a 100-ohm meter is connected in series, the original current will be cut considerably. If the voltage is 2 volts, the original filament current for a 40-ohm filament resistance would be 50 ma. With the meter resistance added in series, the total is now 140 ohms, so the current would be cut to about 14 ma., which is about  $\frac{1}{4}$  of the original value. Here you would have to depend on voltage and resistance measurements. In laboratory work particularly, the effect of the meter must be carefully considered, and the lowest resistance meter available, with the required range, must be used. If a suitable meter is not available, other types of measurements must be made.

**5. Polarity.** The direction of movement of the meter coil and the attached needle depends on the direction of the magnetic force, which in turn depends on the direction of current flow through the coil. If the meter is connected backwards, it will indicate backwards or down-scale.

The meter terminals are marked so that proper connections can be made. Just remember to connect the positive terminal of the meter toward the positive terminal of the source, so electrons enter the *negative* meter terminal.

To help fix this in your mind, refer

back to Fig. 8 where the correct meter polarity is indicated. When a tube is in the circuit, it acts as a signpost, showing the direction of electron flow through the entire circuit. Electrons, of course, flow only from the cathode of a tube to its plate and continue to flow in the same direction through the rest of the circuit. To make the meter needle read up-scale, the electrons must *enter* the *negative* meter terminal. Should the meter connections be reversed so the electrons enter the meter terminal marked +, the meter needle will read down-scale.

If the circuit does not contain a tube, you have to determine the polarity of the source and follow the rule of connecting the positive meter terminal toward the positive of the source.

Now let's summarize what we've learned about using current meters.

1. Always connect the meter *in series* with the circuit.

2. In multiple circuits, connect the meter where *only* the desired current flows.

3. Always use a meter range *higher* than the expected current.

4. The meter (and shunt, if used) resistance must be *much smaller* than the circuit resistance.

5. Connect the meter so electrons enter the *negative* terminal.

## PULSATING CURRENT

Suppose we tried to measure a.c. with our d.c. meter. What would happen?

If it were very slowly changing a.c., the coil would turn first in one direction and then in the other, as indicated by the fact that we must watch polarity. This means that half of the time the meter needle would be off-scale. This would not happen with 60-cycle a.c., however, because the meter movement is too slow to follow a.c. variations occurring more than 5 or 10 times per second. For alternating currents

of higher frequency than 5 cycles, the inertia of the meter movement makes the D'Arsonval meter indicate the *average* current over a period of time.

When you analyze an a.c. wave, you will notice that in each cycle the cur-

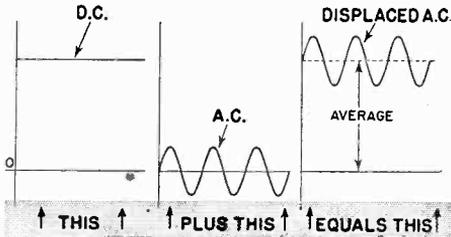


FIG. 10. The average value of a pulsating d.c. wave is equal to the d.c. component.

rent is alternately positive and negative, so the *average* current is zero. That is, the positive swings are equal to the negative swings, so their effect on a slow-moving object is zero. This is important; it means that a D'Arsonval meter alone will not read on a.c.—it can only indicate the a.c. average (zero), regardless of the amount of a.c. current passing through it.

Now a pulsating current has both a d.c. and an a.c. component, as shown

in Fig. 10. Our meter won't read the a.c., but it will read the d.c. So, on pulsating d.c., a meter reading shows only the amount of d.c. present.

In the case shown in Fig. 11A, an a.c. cycle has been rectified by a half-wave rectifier. Now the pulses exist in the

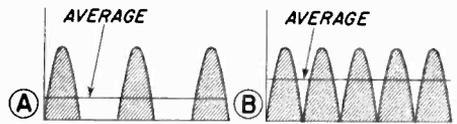


FIG. 11. Notice that full-wave rectification (B) produces a d.c. average twice that of half-wave rectification.

same direction all the time, so this is another form of pulsating current. Even though there are gaps when no current flows, there is still an average value. (Naturally, this average is low; it's only about .3 of the peak value.) The D'Arsonval meter will indicate this average value. When a full-wave rectifier is used, the average value is higher, as shown in Fig. 11B. The average is now about .64 of the peak value.

Just remember this—any pulsating current has a d.c. average which can be read on a D'Arsonval meter.

## Alternating Current Meters

In discussing alternating current meters, we will consider power line, a.f. and r.f. currents all as a.c. The only difference is one of frequency; however, this is important because some a.c. meters cannot measure the higher audio or r.f. frequencies accurately.

In the preceding section, you saw that a D'Arsonval meter could not by itself measure a.c., because the average value of a sine wave current over a period of time is zero. But our meter can be used to measure a.c. if we find

some way to make this average value different from zero. Actually, there are three general classes of a.c. measuring devices, and D'Arsonval meters are used in two of them. The three classes are:

1. The D'Arsonval d.c. meter with a vacuum tube or copper-oxide rectifier.

2. The d.c. meter with a conversion device, such as the thermocouple or photoelectric cell.

3. True a.c. meters, such as the hot-wire, magnetic-vane and the electro-

dynamometer types.

Let's examine each type to see how it works.

## ALTERNATING CURRENT RECTIFIERS

**Vacuum Tubes.** Any diode tube will rectify a.c., as current can pass through the tube only when the plate is positive with respect to the cathode of the tube. In the circuit shown in Fig. 12, the diode tube *VT* passes current only when terminal 1 is positive with respect to terminal 2. The resulting half-wave pulses have an average value which is about .3 of the a.c. peak. This average value will produce a deflection on a D'Arsonval meter.

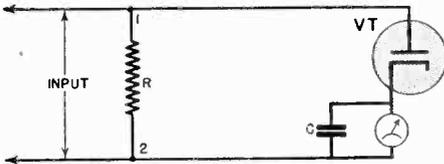


FIG. 12. A diode can be used to rectify a.c.

The condenser *C* in Fig. 12 is used to by-pass the high-frequency pulsations around the meter. The meter movement is a coil and so has inductance. If not by-passed, the inductive reactance would be added to the meter resistance and might both upset the circuit and make it non-linear with respect to frequency. The condenser also serves to smooth out the pulsating current, bringing the peak value down and raising the d.c. average. This not only gives a greater sensitivity, but also reduces the danger of high peak values damaging the meter.

The rectifier tube and meter can't be placed in series with the circuit whose current we wish to measure, because the tube will change the a.c. in the circuit to a pulsating current. We want rectification for the meter, not for the complete circuit. We get it by inserting resistor *R* in series with the circuit. The a.c. voltage developed across the resistor then is applied to the tube-

meter combination. Thus, we are actually measuring the current indirectly by the voltage drop produced in this resistor. The meter is calibrated so all these conditions can be accounted for. Usually, it indicates the effective or r.m.s. value of the a.c.

Figure 13 shows how a triode tube may be used for rectification. The C bias is adjusted so that the plate current is nearly zero. When an alternating current flows through resistor *R*, an a.c. voltage is developed across it. Negative alternations make the grid even more negative and so have practically no effect on plate current. However, the positive alternations swing the grid in the other direction so that plate current pulses flow. The meter then responds to the average plate current.

This plate current is proportional to the a.c. grid voltage, which is determined by the alternating current in the circuit and the value of resistor *R*. Thus, the plate current is an indirect measure of the circuit current.

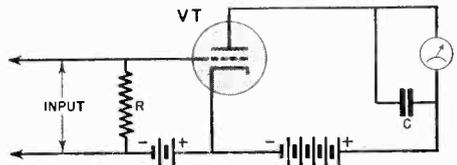


FIG. 13. How a triode rectifier is connected.

Resistor *R* acts just like a shunt in this circuit. For example, we may need 1 volt on the grid to produce a full-scale meter deflection. By using a 1000-ohm resistor, a circuit current of 1 ma. would give 1 volt. If we use a 100-ohm resistor, 10 ma. would be needed to produce 1 volt.

**Copper - Oxide Rectifiers.** The D'Arsonval meter and copper-oxide rectifier\* combination is the most

\*Selenium on iron is also used in place of copper oxide, and the same rectifier circuits are used.

widely used a.c. measuring device, and is found in the equipment used by most servicemen. The copper-oxide rectifier is stable, relatively reasonable in cost, and can be designed to work on small currents.

Each rectifier element consists of a copper disc or washer, which has one surface oxidized by a special heat treatment. A contacting washer, usually made of lead, is pressed on the oxide layer. This oxide layer rectifies because it permits electron flow through the layer in one direction much more readily than in the other. In other words, electrons can move easily from the copper through the oxide to the lead contacting disc but they encounter a fairly high opposition when trying to move in the other direction.

There are several different ways of connecting these elements, but the most common is the full-wave bridge circuit shown in Fig. 14. Four separate elements are used in the bridge. In practice, these elements would be mounted on an insulated single bolt, with insulating washers between the elements.

The arrows alongside the elements in Fig. 14 indicate the directions in which electron movement most easily occurs for each half cycle. When terminal 1 is negative (2 is positive), the electrons move in the direction shown by the solid arrows. At point A the electrons choose the path toward point B, because rectifier W conducts better than rectifier X for this half of the a.c. cycle. At point B, the electrons go through the meter instead of rectifier Z, since Z offers a high impedance to electrons moving in this direction. Reaching C, the electrons pass through rectifier Y to point D, and from there go to terminal 2.

When the cycle reverses, terminal 2 becomes the negative terminal. Now electrons follow the dotted arrows from terminal 2 to D, then through rectifier

Z to B. From here, the path is through the meter to C, then through rectifier X to A and so to terminal 1. Note that regardless of the polarity at 1 and 2, electrons *always* pass through the meter in the same direction, which is exactly the action of a direct current. Hence, a direct current meter can be used at M.

We use full-wave rectification in this circuit because it gives a higher

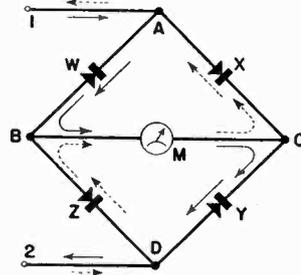


FIG. 14. The copper-oxide rectifier in a bridge is the most common rectifier for service work

average current, about twice that for half-wave rectification. This practically doubles the meter sensitivity by giving a higher meter deflection for the same amount of alternating current, so that our meter has an a.c. sensitivity almost as good as that on d.c.

If one element of the bridge burns out, half-wave rectification will be obtained instead of full-wave. Then, since the average current is only about half that for full-wave, the meter will only read about half as much as it should. You can easily spot a damaged rectifier element in a meter of this type—check a known a.c. current; if the reading is half what it should be, one of the elements is damaged.

The back-to-back circuit is another form of full-wave bridge, with the advantage of using only two rectifier elements instead of four. The circuit gets its name from the rectifier connections, where the “backs” are connected together as shown in Fig. 15. The resis-

tors  $R_1$  and  $R_2$  replace two of the rectifier elements of the usual bridge circuit, and are chosen to have a resistance about halfway between the conductive and non-conductive values of a copper-oxide element. (An element may have only 100 ohms opposition to electron flow one way and several thousand ohms opposition to electron flow in the other direction.)

There is a continuous a.c. path through the resistors, but this current does not pass through the meter. When electrons move from source terminal 1, rectifier  $W$  becomes conductive, so there will be an electron flow through this rectifier to point  $B$ . Rectifier  $Z$  opposes current flow by having more resistance than the path through meter  $M$  and resistor  $R_2$ , so the major electron movement is from 1 to  $A$ , through  $W$  to  $B$ , through  $M$  to  $C$ , through  $R_2$  to  $D$  and thus to 2. On the next half cycle, the major electron movement is from terminal 2, through rectifier  $Z$ , then through the meter and resistor  $R_1$  back to terminal 1. Thus, when the rectifier is conducting, it offers less opposition than the resistor directly across from it, but when the cycle reverses the resistor is more conductive than the rectifier.

To extend the a.c. range with either form of bridge, shunts are connected between terminals 1 and 2 instead of across the meter. If shunts were used across the meter, higher currents would have to flow through the rectifier elements and would burn them out.

Copper-oxide rectifiers have two faults: 1, they are easily damaged; and 2, they have a high shunting capacitance between terminals.

Reasonable care to prevent overloading will protect them from burn-out. They must not be exposed to high temperatures for very long. Heat tends to change the chemical characteristics of the oxide layer, causing a change in

the rectifying ability of the unit. Too much heat, such as from a soldering iron, can ruin a rectifier.

The high self-capacitance generally limits the copper-oxide rectifier to power line and the lower audio frequencies. By reducing the size of the rectifier elements the capacity is reduced, so the frequency range can be extended somewhat.

Of course, the capacitance does not prevent the use of these rectifier-meter units as r.f. current indicators; it just changes the calibration. With increases in frequency, the capacity acts as a decreasing shunt reactance, so the current range changes rapidly. However, reasonably accurate readings on the original meter scales of some types of meters can be made up to 80 kc. without special calibration charts. Since you, as a serviceman, will be primarily interested in measuring operating volt-

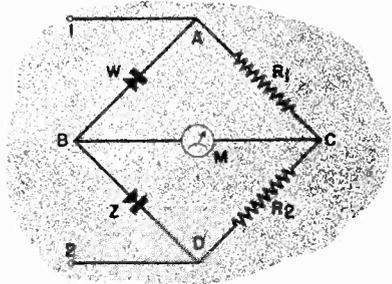


FIG. 15. The back-to-back bridge saves two rectifier elements.

ages and currents which are d.c., power line a.c. or low audio frequency a.c., you can usually ignore the frequency characteristics.

## CONVERSION DEVICES

**Thermocouple.** The need for a sensitive indicator for r.f. currents led to the development of the thermocouple, commonly found in transmitter installations and research laboratories.

This device works on an interesting principle—the fact that when a junc-

tion of two *dissimilar* metals is heated, a d.c. voltage is produced. Thus if a junction between a copper rod and an iron rod is heated in a flame, a d.c. voltage is produced; the two unjoined ends of the rods are now the d.c. terminals. This method of getting a voltage suggests a means of converting a.c. power through its heating effects.

We know that current flow through a resistance produces heat, so if we can bring a junction of dissimilar metals up to a resistor acting as the source of heat, the d.c. voltage so developed can be used to operate a d.c. meter. Figure 16 is a typical arrangement.

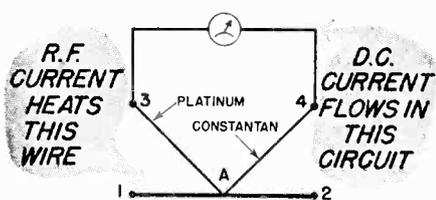


FIG. 16. The thermocouple is widely used for r.f. measurements.

The wire between terminals 1 and 2 is a resistance wire, called the heater. The wires between A and 3 and A and 4 are the thermocouple elements.

The heater wire is made short, and the ends are connected to large terminals so that heat developed near the ends will be rapidly conducted away. Hence, the heat is concentrated at point A, the center of the heater. The dissimilar metal junction, called the thermocouple, is placed at A. In some cases, the thermocouple is just near the heater, while in others it is welded to it. Since the voltage produced for a particular temperature depends on the metals used in the thermocouple, these metals are carefully chosen to give the greatest d.c. voltage. The wire between 3 and A may be platinum, while the wire between 4 and A may be constantan.

The current to be measured is per-

mitted to flow through the heater wire, between terminals 1 and 2. The resulting heat is proportional to the power dissipated in the resistance of the heater, which in turn makes the heat proportional to the square of the current. ( $P = I^2R$ ) The temperature rise heats the thermocouple junction at A, producing a d.c. voltage which will cause a d.c. current flow through the meter. Notice this is a *produced* d.c. and is not the current being measured, which is usually a.c.

The heating effect of a current flow through a resistance will occur regardless of frequency or wave form, so any a.f. or r.f. current can be measured by this device. In fact, the thermocouple is practically a standard r.f. current meter. To eliminate skin effect\*, the resistance wire is made in the form of a hollow tube in models intended for ultra-high frequencies.

These instruments are made in many ranges, from a few milliamperes up to practically any value wanted. An instrument sensitive to low currents is made by sealing the thermocouple and heater in a glass tube from which the air has been pumped. This is known as a vacuum thermocouple.

The use of a thermocouple has another advantage—the meter and thermocouple do not have to be mounted together. This is important, as currents frequently must be measured in hard-to-reach places, such as in a transmission line, transmitter tank circuit or antenna installation. It is impossible to run r.f. currents over long leads to a more convenient meter location without upsetting the circuit. Therefore, the thermocouple and heater unit is

\* Skin effect is the tendency of high-frequency currents to flow on the surface instead of through the center of the wire. This produces the effect of a change in resistance. By eliminating the center of the wire, all current must flow along the surface, so there is no change.

placed right in the circuit containing the current to be measured. Then the d.c. produced is fed over a feeder line to the meter, as shown in Fig. 17. Choke coils  $L_1$  and  $L_2$ , together with condensers  $C_1$  and  $C_2$ , filter r.f. currents out of the line to the meter. Should the thermocouple be connected in a circuit where one terminal of the heater is grounded, the filter probably would not be required. If the line to the

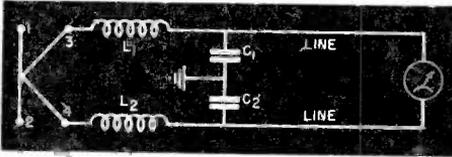


FIG. 17. The thermocouple permits remote meter readings.

meter passes through any r.f. fields, it may pick up some r.f. energy which could cause an undesired current through the meter. Another condenser (or a pair connected like  $C_1$  and  $C_2$ ) may be connected across the meter terminals to by-pass this r.f.

The meter is calibrated by passing known currents through the heater element. These currents are usually d.c. or low-frequency a.c., which can be more accurately determined than r.f.

As the deflection is proportional to the square of the a.c. being measured, the meter scale will be crowded at the lower end if a standard meter is used. That is, the deflection is smaller than normal for small currents, and is greater than normal for high currents. By using a meter with specially shaped pole pieces, it is possible to obtain a progressively decreasing d.c. sensitivity as the meter moves up-scale, so that the scale is more uniform.

Such a meter is shown in Fig. 18. The pole faces have a shape which creates a non-uniform field between them with the maximum lines between the closest points (1 and 2). When the

meter coil is in the position shown in Fig. 18, the twisting force caused by the reaction of its field with the meter field is a maximum, so the coil will give a large deflection as the current through it increases. The more the coil rotates to the right or clockwise, the smaller this twisting force becomes, so larger coil currents are needed to give the same amount of deflection. Thus we have the decreasing sensitivity at the upper end of the scale that we were looking for, and our scale becomes practically uniform over its whole length.

This special method of cutting the meter pole pieces increases the meter cost, but it is a means of getting an extended current range without sacrificing the ability to read small values accurately.

**Photoelectric Types.** This method is used only in experimental work, but you may meet it, so we'll go into it briefly.

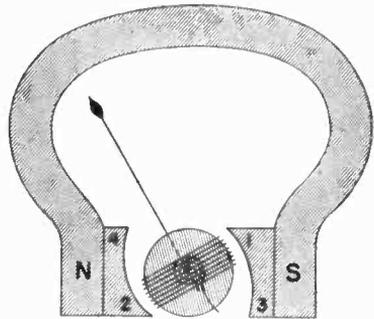


FIG. 18. Using pole pieces having a special shape produces a non-uniform magnetic field.

The amount of light obtained from an electric light bulb varies according to the amount of current flow through the bulb. By connecting a photovoltaic cell to a meter and exposing the cell to a light bulb, the light will produce currents through the cell and meter proportional to that through the bulb. Then we can calibrate the meter in terms of the current flow through the

bulb. The bulb can then be connected in the circuit carrying the current, and the photocell-meter used to indicate the current. This method is not very accurate—it's more useful to indicate when changes occur or when maximum current flows than it is to indicate the exact amount of current.

### TRUE A.C. METER MOVEMENTS

You recall the D'Arsonval meter cannot be used directly on a.c. because it cannot follow the rapid reversals of current. Even if it could, it would reverse its direction when the a.c. reverses. This is because the coil field

19A. When current flows through the coil, the vane is magnetized by the coil's magnetic field, with a magnetic polarity opposite to that of the coil. Since opposite poles attract, the vane will be attracted into the coil until the magnetic force is balanced by the spring. The vane is soft iron, so it does not become a permanent magnet. Hence, when the current flow through the coil reverses, the reversal of the coil magnetic field also reverses the polarity of the magnetic vane and the attraction is maintained.

Figure 19B shows another type of magnetic-vane meter. Two pieces of

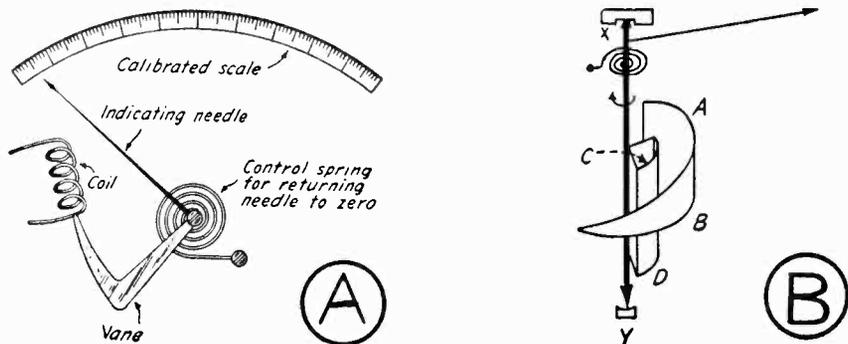


FIG. 19. Here are two common types of magnetic vane movements.

reverses with the a.c. flowing through it, but the fixed magnetic field continues to have the same polarity. If we can arrange the meter so both magnetic fields reverse at the same time the a.c. reverses, the magnetic turning force will always be in the same direction and the meter will indicate just as if d.c. were applied. There are two meters that do this, the magnetic-vane and the electro-dynamometer types.

**Magnetic-Vane Meters.** This meter works in a different manner from the D'Arsonval movement. Instead of using a fixed magnet and a moving coil, the coil is fixed and a soft iron vane is the moving element, as shown in Fig.

soft iron are used in this type, with the coil wound around both. The triangular piece *AB* is bent in a cylindrical shape and is mounted on the inside of the coil form. The rectangular piece *CD* is also bent in a cylindrical shape. This piece is connected to the spindle *XY*, to which is attached the meter pointer and restoring spring.

When current flows through the coil, both pieces of iron are magnetized. As the coil field is along the axis of *XY*, the top and bottom edges of vanes *AB* and *CD* will become the magnetic poles. The edges *C* and *D* will have the same polarity as *A* and *B* respectively. Thus, *C* and *A* may be north poles, while *D* and *B* may be south poles. This

means like poles are near each other and will repel. The soft iron pieces are so placed and shaped that *CD* rotates to the right (clockwise). Again, the movement will be to the point where the spring balances the magnetic force.

When the coil field reverses, both magnetic poles reverse, so the magnetic opposition remains the same as before.

Although several other types of magnetic-vane movements have been developed, they all have similar characteristics. To produce the strong magnetic field required, the coil must have a large number of turns. The coil thus has considerable inductance, which will upset the calibration if high-frequency currents are measured. Also, there is a limit to the frequency with which the vane magnetism can be changed, so these meters are used only for d.c. and low-frequency a.c. measurements.

The inductance of the meter coil makes it hard to use shunts for extending the range, because the coil reactance changes with frequency — so these meters usually have only one range. In spite of all its shortcomings, low cost and rugged construction have made the magnetic-vane meter very popular for use where great accuracy is not required.

**Electrodynamometer.** This meter is a moving-coil type, but is quite different from the D'Arsonval meter. No fixed magnet is used; instead, the field is produced by an electromagnet, and the moving coil has an air core. Figure 20 shows a typical electro-dynamometer. Coil *A* is the field coil, which may be wound on any suitable coil form. The inner coil *B* is the moving coil, made so it can rotate within coil *A*. The two coils are connected in series. When current flows through the coils, both develop magnetic fields proportional to the current strength and the number of turns on each coil. This

causes coil *B* to rotate until the magnetic force is balanced by the spring, as in other meters.

When alternating current is measured, the reversal of the direction of current reverses both magnetic fields, so the repulsion remains the same as before. Hence, the meter can be used either for a.c. or d.c.

The moving coil has a tendency to waver back and forth a while before

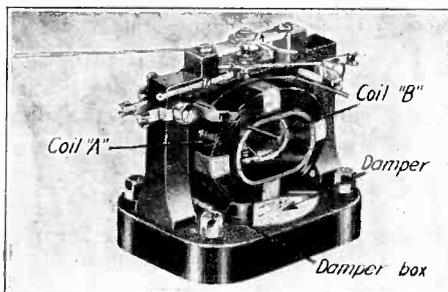


FIG 20. This electro-dynamometer does not use a permanent magnet.

settling down to the correct reading, just as with other meter movements. This wavering is damped by a vane attached to the moving coil and fixed to rotate through a closed chamber known as a damper box. When the coil moves, the damper vane also moves, compressing the air ahead of it in the damper box. The compressed air acts as a cushion, so that the pointer comes slowly up to the correct reading and then stops.

Because of the inductive effects of the coil windings, the frequency range of such meters is limited. The usual types hold their calibration for frequencies up to 130 cycles; special meters can have ranges up to 1000 cycles or more, but will be accurate only for frequencies very close to the frequency for which they were built.

The original electro-dynamometers were used as standard instruments, as they could be calibrated directly from their dimensions. Modern commercial

electrodynamometers are calibrated by comparison with a standard, however, just like other meters.

As the same current flows through both of the series-connected coils and as the deflection depends on the strength of both magnetic fields, the deflection is proportional to the square of the current for direct current. For an alternating current, the meter deflection is proportional to the square of the effective or r.m.s. value. Since an alternating current of any r.m.s. value will produce a magnetic field equivalent to that produced by a direct current of the same value, the meter scale is exactly the same both for d.c. and low-frequency a.c. This makes it possible to compare the effective values of currents regardless of wave form.

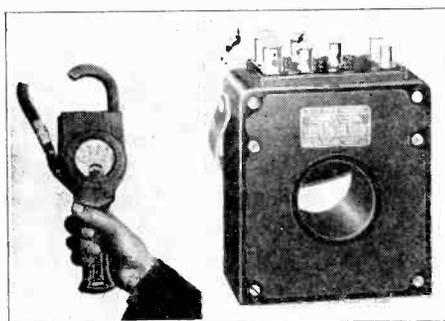
Large currents cannot be measured easily with an electro-dynamometer, because the moving coil cannot be made of exceedingly large wire or have heavy leads going to it. Although shunts can be used to a certain extent, a current transformer is the most common way of extending the range of this meter.

**Current Transformers.** As you will recall, a transformer will step up or down an a.c. voltage according to the ratio of turns on the primary and secondary windings. Also, neglecting losses in the windings and core, the primary power will equal the secondary power. This means that if the secondary voltage is stepped up, the secondary current will be reduced in the same ratio so that the power ( $I \times E$ ) will be the same. Hence, a transformer which steps up voltage by a ratio of 1 to 3 also steps down current by a ratio of 3 to 1. For example, if the primary voltage and current are 110 volts and 4 amperes, the primary wattage is 440 watts. If the secondary voltage is stepped up three times, to 330 volts, the secondary current must be one-third the primary current

(1-1/3 amps.), so that the secondary power will also equal 440 watts.

Therefore, by designing a step-up voltage transformer with low losses and a small primary, we have a step-down current transformer in which a large primary current flow will result in a small secondary current which can be indicated on a meter. This secondary current is proportional to the primary current, so the meter can be calibrated in terms of primary current.

Current transformers are usually found where very large currents are



*Courtesy Weston*

FIG. 21. Current transformers. The core of the one at the left is snapped around the current-carrying wire. The one at the right is installed permanently, passing the power line through the center. These transformers are usually used on high-current power lines, where currents of 25 amperes to thousands of amperes are to be measured.

measured. Where the current is several thousand amperes, the current transformer may be just clamped around the conductor carrying the current, allowing this conductor to act as a single-turn primary winding. Typical current transformers are shown in Fig. 21.

Although commonly used with electro-dynamometers, current transformers can be used with any other a.c. meter having the required characteristics, such as the proper range and proper impedance. However, since the type of meter used may have an effect on the calibration, the current transformer and meter combination have to be calibrated as a unit.

**Hot-Wire Meter.** Another method which has been used to measure current depends on the fact that metals expand when heated. Any kind of current (a.c. r.f. a.f. or d.c.) with any wave form will heat and expand a resistance wire through which it passes; the amount of expansion will be a measure of the heat produced, and therefore of the current flowing. We can use this principle in a meter by making the expansion cause movement of a needle.

Figure 22 shows a method of doing this. The wire which is heated is shown between *A* and *B*. This wire is held under tension by an adjustable spring (not shown). Attached to this wire at *C* is a metallic cord *D* which passes around the pulley *P*, continuing to the spring *S* which is fastened at *E*. The pulley and support *E* are made of insulating materials to prevent grounding the supply circuit through the meter frame or case. When a current flows through the wire *A-B*, it heats and stretches, sagging downward. This

allows the spring *S* to pull the cord *D* around the pulley *P*, thus rotating the pulley to the right. The meter pointer *X* is attached to the pulley and is thus moved over a calibrated scale.

This meter measures only high cur-

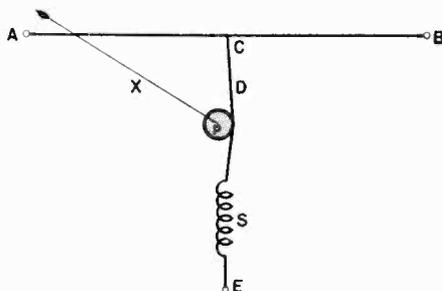


FIG. 22. The mechanism of a hot-wire ammeter.

rents and is easily burned out. Elaborate means of compensating for surrounding temperature changes must be used. Also the amount the wire stretches when a given current passes through it changes with repeated use, so such meters are no longer very popular.

## Voltmeters

Voltages, like currents, are measured by meters. In fact, a current meter can be used for voltage measurements if properly calibrated. You will recall that a certain voltage is necessary to force the required current through the resistance of the current meter. We'd be justified in saying that meter deflection is proportional to the applied voltage, and in calibrating our meter accordingly.

As an example, a 1-milliamperemeter may have a resistance of 200 ohms. The voltage required for full-scale deflection is  $E = I \times R$ , or  $.001 \times 200 = .2$  volt. Hence, we can use this meter

as a 0-.2 voltmeter. This range is too low for ordinary service purposes, but we can increase it by increasing the meter resistance.

Since increasing the resistance of the meter movement itself is not practical, resistance is added in *series* with the movement, so that a higher applied voltage is needed to drive full-scale current through the meter. If 800 ohms external resistance were added to the 1-ma. meter in the example above, 1 volt would be needed to give a full-scale reading.

By adding sufficient resistance, any desired voltage range can be obtained.

As with current meters, a single meter having several ranges is desirable, so a low basic range is usually chosen and then additional resistors are connected in series to extend the range further. A typical switching arrangement is shown in Fig. 23. Resistor  $R_1$ , together with the meter resistance, sets the basic voltage range. Then other resistors are added by the switch to obtain the desired extensions.

Notice that a current flow through the meter is being interpreted as voltage because of the known resistance of the meter. This current must come from the source being measured, in addition to that taken by the load, because the voltmeter is connected across

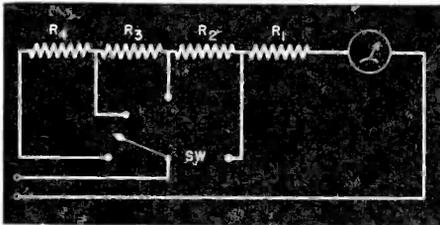


FIG. 23. Voltmeter ranges are extended by using multiplying resistors in series.

the terminals where the electrical pressure (voltage) is to be measured.

When the source has a low internal resistance (as is the case with a good battery) and the source is not supplying appreciable current to a load, the current taken by the voltmeter scarcely matters. However, *if there is appreciable resistance in the source or between the point of voltmeter connection and the source, the readings will vary with the sensitivity of the voltmeter.* Let's learn more about this now.

### VOLTMETER SENSITIVITY

As a practical example of a typical radio circuit, consider Fig. 24. Suppose that resistor  $R_1$  has a value of 50,000 ohms and the tube plate-cathode resistance is 25,000 ohms. The

total resistance is then 75,000 ohms, so 4 ma. flows from the 300-volt source ( $300 \div 75,000 = .004$  ampere). The tube plate-to-cathode voltage will be 100 volts, and the drop in resistor  $R$  will be 200 volts.

Now suppose we connect a voltmeter as shown. Will it indicate 100 volts? This depends on the current taken by the meter, which must flow through resistor  $R_1$ , in addition to the plate current of the tube. In other words, the meter is in parallel with the tube resistance, so our circuit is really that shown in Fig. 25. A few practical cases will help you see what happens.

*Case I.* If the voltmeter in Fig. 24 has a 100-volt range which requires 10 ma. for full-scale deflection, its resistance  $R_M = E \div I$ , or 10,000 ohms. This resistance is in parallel with the tube resistance  $R_P$  of 25,000 ohms, as shown in Fig. 25, so the resulting parallel combination of meter and tube is

$$R = \frac{R_P \times R_M}{R_P + R_M} = \frac{25,000 \times 10,000}{25,000 + 10,000}$$

7100 ohms, approximately. This value is in series with the 50,000-ohm resistor  $R_1$ , so the total circuit resistance has changed from 75,000 ohms to about 57,100 ohms. The new total current from the 300-volt source is about 5.25 ma. ( $300 \div 57,100 = .00525$  ampere). This current through  $R_1$  gives a voltage drop of 262.5 volts, leaving only 37.5 volts for the tube and meter parallel combination. In other words, the tube plate voltage is now only about 7/57 of the total, instead of 1/3. (The ratio is 7100 to 57,100 instead of 25,000 to 75,000.) This means that the voltage measured by the meter is about 38 volts, instead of 100 volts. The connection of the voltmeter has upset the voltages so that more drop exists across  $R_1$  (now about 262 volts) and less across the tube, giving a false indication of the conditions existing without the meter.

*Case 2.* Suppose we now use a voltmeter having a 100-volt range which only requires 1 ma. for full-scale deflection. The meter resistance is now 100,000 ohms. Figuring current and voltages as we did in *Case 1*, we find that the plate voltage now measures 85 volts, much closer to the original 100 volts.

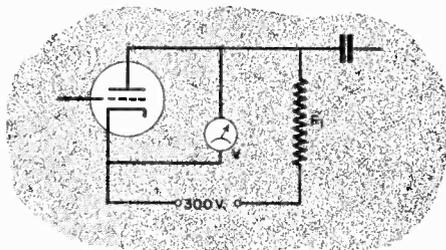


FIG. 24. A typical voltage-measuring problem. The accuracy depends on the ohms-per-volt sensitivity of the meter.

*Case 3.* A 100-volt meter requiring only 0.1 ma. has a resistance of 1,000,000 ohms. Such a meter will indicate about 98 volts if placed in the circuit of Fig. 24. The error now is only about 2%, well within the normal tolerances allowed for this kind of circuit.

Obviously, in any circuit having appreciable series resistance, the lower the current needed to give deflection, the more nearly correct the results.

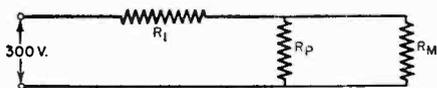


FIG. 25. Notice how the voltmeter resistance is in parallel with the tube plate-cathode resistance.

For this reason, voltmeters used in radio service work require 1 ma. or less for full-scale deflection.

**Ohms-Per-Volt.** The term "ohms-per-volt" is used to express voltmeter sensitivity. The smaller the current needed for full-scale deflection, the greater the sensitivity of the meter. Also, the smaller the current for a given

voltage, the greater the meter resistance must be. ( $R = E \div I$ ) By dividing the total meter resistance by the basic full-scale voltage range, the ohms-per-volt rating will be obtained. Thus a 2-volt meter having 2000 ohms resistance is a 1000 ohms-per-volt meter. Ohm's Law will let you figure ohms-per-volt ratings easily. Notice that this rating increases as meter sensitivity increases.

Now here's something you should remember. Service manuals usually give you radio circuit voltages (plate-to-cathode voltages, etc.) which are taken with 1000 ohms-per-volt voltmeters. But if you are using a more sensitive meter — say 20,000 ohms-per-volt — the readings you get when you measure these circuit voltages *will be higher than the manual says for any high-resistance circuits, but will be more nearly correct for the actual circuit operating conditions.* (You just learned why this happens.) Just remember this — set voltages will be *higher* than their service manual ratings in resistance coupled circuits, if you measure them with a voltmeter having a sensitivity greater than 1000 ohms-per-volt. How much higher, of course, depends upon the sensitivity of the meter, and the resistance in the circuit where it is connected.

Knowing the ohms-per-volt rating of the meter makes it easy to extend the range. Just multiply the ohms-per-volt rating by the number of volts the range is to be extended. For example, if a 1000 ohms-per-volt meter is to be extended from 100 volts to 500 volts, the extension is 400 volts. Multiplying 400 by 1000 gives 400,000 ohms as the needed extra series resistance.

## A.C. VOLTMETERS

Any of the standard alternating current meters can be used as an a.c. voltmeter in exactly the same manner

just described. For service work, the copper-oxide rectifier and D'Arsonval meter combination is generally used. This type has better sensitivity characteristics than any except the vacuum tube voltmeter, which will be taken up in another lesson.

The copper-oxide rectifier type a.c. voltmeter usually has a sensitivity of 1000 ohms-per-volt, requiring 1 ma. for full-scale deflection. Even lower sensitivities are permissible in service work, where the usual a.c. meter readings are filament voltage, power line voltage, rectifier plate voltage and output voltages during alignment. The sources for these first three measurements have low series resistance, so the current drawn by the meter does not greatly

matter. For output readings, only an indication of the maximum voltage, not the exact amount, is usually desired.

Of course, the copper-oxide unit has the same faults described before; it is easily overloaded and has shunting capacity. When overloaded, one element usually burns out so readings fall to half-normal. The effect of the capacity of the copper-oxide rectifier is more pronounced with a voltmeter than with a milliammeter, because of the series resistances. However, the copper-oxide types are reliable up to 1000 cycles or so, which is sufficient for output purposes, as most signal generators are modulated at about 400 cycles.

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## Ohmmeters

Since we can now measure voltages and currents, we can use Ohm's Law and calculate the resistance in a circuit or device. However, this is a cumbersome method; a more speedy way of finding the resistance is desirable.

Suppose we make up a circuit consisting of a known fixed voltage source in series with a current meter. When we connect a resistor into this circuit, the amount of current flow will be inversely proportional to the value of the resistor. Then we can calibrate the meter in terms of resistance, and we will have a resistance indicator or ohmmeter.

Figure 26 shows how such a meter is designed. The resistance  $R_2$  is adjusted so that its resistance plus that of  $R_1$  and the meter is just sufficient to permit a full-scale deflection on the current meter (using a known battery voltage) when the test prods are touched together. Then when the cir-

cuit is opened and the unknown resistance  $R_x$  is inserted, the meter will indicate some lower value, depending on the added resistance.

As the battery voltage will decrease with age and use, and also since the meter and resistance values may be off somewhat, resistor  $R_2$  is adjustable. Actually,  $R_1$  and  $R_2$  could be a single resistor, but if adjustments are not correctly made, the meter could be damaged. Hence,  $R_1$  is fixed and has a value sufficient to prevent meter damage.

This adjustable resistor is set for a full-scale meter deflection, which adjusts for zero ohms on the scale.

As the battery ages, its voltage drops, so  $R_2$  is decreased to restore zero. This change in voltage and resistance upsets the calibration, however; the ohmmeter accuracy will vary directly with the percentage change of battery voltage. Thus, a 10% drop in

battery voltage will cause a 10% error in the ohmmeter reading. Fortunately, battery voltages are relatively stable over long periods of time, so replacements are not too frequently required.

By designing the ohmmeter for the rated battery voltage, advantage can be taken of the fact that battery voltages are somewhat higher when new. Then the ohmmeter will read a few per cent high with fresh batteries, and will then drop to exact readings and gradually read lower as battery voltage decreases. This has the effect of halving the error, as a 10% battery voltage change now produces a 5% variation each way from the correct reading. The sizes of  $R_1$  and  $R_2$  are usually chosen so that when  $R_2$  can no longer be adjusted for zero ohms, the percentage of error has exceeded that set by the manufacturer, so a new battery must be obtained.

The ohmmeter is definitely a service instrument—it does not have sufficient accuracy for laboratory work. However, in service work, parts may normally vary as much as 20% anyway,

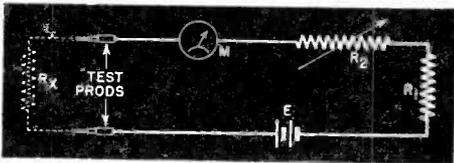


FIG. 26. This series ohmmeter is best for medium and high-resistance values.

so an error of a few per cent in measurement is not of great importance.

The fact that voltage variations will upset the ohmmeter is one of the reasons that power packs are not more widely used with ohmmeters. Only a well-regulated power pack will have a sufficiently stable output voltage.

The type of ohmmeter we have been considering is known as the series type, as the battery, meter and resistances

are in series. This type can easily be recognized, as the “zero-ohms” position is at the right of the meter scale, with increasing resistance values to the left. This is just the reverse of the ordinary meter scale.

As with other instruments, the range is of importance. In service work, the wide range of resistances met (from .05 ohm for r.f. transformers to sev-



FIG. 27. A typical series ohmmeter scale having a 3000-ohm center-scale value.

eral thousand megohms for leakage measurements) presents quite a problem. Let's investigate this further.

**Useful Range.** Having made an ohmmeter using some particular meter and total resistance, just what range of values will it indicate? Let us suppose we are using a 1-ma. meter and a 3-volt battery. This means the meter +  $R_1$  +  $R_2$  resistance is adjusted to 3000 ohms so 1 ma. can flow. You will see that the meter will read mid-scale (half the current) when a resistance of 3000 ohms is being measured, as we then have doubled the circuit resistance and still have the same voltage. Now, the most useful portion of the meter scale will be between values about 100 times the mid-scale value and 1/100 the mid-scale value. Thus, if the mid-scale value is 3000 ohms, the ohmmeter will be best calibrated and easiest read between 300,000 ohms and about 50 ohms. Higher and lower values can be read on this scale, but the readings become hard to make. Actually, this range is probably readable up to about 500,000 ohms, and values of 10 ohms or so may be estimated.

Figure 27 shows a typical ohmmeter scale based on these values. The

crowding at the low end is caused by the fact that the meter resistance is becoming much larger than the resistance being measured, so that changes in the latter have less and less effect on the current flow.

At the other end of the scale, the current is primarily determined by the external resistance. When the external resistance gets large, adding more resistance doesn't change the current much. Thus, at the low-resistance end, a scale division may represent only 50 ohms, while at the other end, several thousand ohms may be indicated by the same space on the scale.

The series ohmmeter is excellent for medium and high ranges, but is not the best type for low ohmic values. For measuring low resistances, we can lower the series resistance, but the lowest normal battery voltage is 1.5 volts, so there must be enough resistance to limit the current at the full-scale value. By reducing the meter resistance through the use of high current meters or shunts, or by using voltage dividing circuits to get lower voltages, somewhat better results can be obtained. However, this requires complex circuits and means a high current flow, which in turn means short battery life and possible damage to the device being tested.

**Low-Range Ohmmeter.** A shunt-type ohmmeter circuit, shown in Fig. 28, is used to measure low values of resistance. The resistor  $R$  is adjusted until the meter reads full-scale with the test leads separated. When the test leads are connected to the resistance being measured, a lower meter reading results, as the unknown resistance  $R_x$  is connected in parallel with the meter and acts as a shunt across the meter. Notice these two important facts: (1) zero ohms on the shunt meter is at the *left*, exactly opposite to a series ohmmeter; (2) if the test

probes are held together, the meter is short-circuited and cannot read. You will see that the "zero adjustment" on a shunt meter is actually the full-scale adjustment.

This circuit is excellent for low resistances. By using a low-resistance meter, the battery current flow will be primarily determined by resistance  $R$ , which can be fairly high. For example, if a 27-ohm, 1-ma. meter were used with a 1.5-volt battery, resistor  $R$  would have to be adjusted to about 1473 ohms to limit the current to 1 ma.

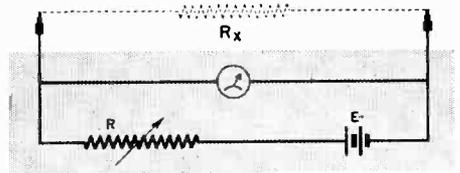


FIG. 28. This shunt ohmmeter is excellent for low-resistance values.

Since the resistance of  $R$  is so high compared to that of the meter, shunting the latter with even a very low-resistance device will have very little effect on the current drawn from the battery. We can assume the current flow is a constant amount and just divides between the meter and the unknown  $R_x$ .

A resistance equal to the meter resistance gives mid-scale deflection (since the current then divides equally between the meter and the unknown resistor). The best scale range is again from about 100 times to 1/100 the center scale value.

**High-Range Ohmmeters.** As mentioned before, the series ohmmeter is a medium and high-resistance indicator. However, there is a practical limit to the range, which is determined by the current required for the meter. Until recent years, service instruments were usually 1-ma. meters—and such meters require very large power supplies to measure very high resistances. For

example, using a 1-ma. meter to obtain a useful scale above 50 megohms requires 500 volts or so.

Servicemen were thus limited to ranges readable up to 1 or 2 megohms. However, when more sensitive meters became generally available, one of the first results was an extension of the upper ohmmeter range. Meters with 50 to 200-microampere ranges (.05 to .2 ma.) permit the construction of ohmmeters of 20 to 50-megohm ranges, using no more than 45 volts.

Even these meters aren't sensitive enough to measure small amounts of leakage in condensers or in shielded cables. Laboratories formerly used "meggers" for this purpose. These megohmmeters were high-voltage devices with a hand-operated 500-volt generator and a special meter movement.

In recent years, vacuum tube circuits have been developed which measure values up to 1000 megohms and higher. These have replaced "meggers" generally; they will be taken up elsewhere in your Course.

**Using Ohmmeters.** Since the ohmmeter depends on its own battery for operating power, it is not used like a

voltmeter or current meter. The latter two measure operating conditions, so the radio circuit being checked must be connected to its normal source of power and must be turned on. On the other hand, the circuit *must be turned off* when an ohmmeter is being used. This is an advantage at times, as the circuit need not be in operating condition when an ohmmeter is being used.

When making tests for circuit continuity, the ohmmeter is connected between selected points, as you will learn in other lessons. Then, when the defective circuit has been located, the various parts are individually checked. When a certain part is to be checked, remember that any parallel path in the set can give false readings. Hence, the part being checked should be disconnected from other parts if the results appear unusual.

The shunt-type meter draws current continuously from its battery as long as it is connected. Therefore, it always has a switch or other means for disconnecting the battery when readings are not being made. Remember to turn off this meter when not in use; otherwise, the battery will run down quickly.

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## Multimeters

As you have probably noticed, the same kind of meter can be used for voltage, current and resistance measurements. The only differences are the ways in which resistances are arranged as shunts and multipliers, the use of rectifiers for a.c., and the variable resistances and battery used on ohmmeters. Naturally, it would be very handy to have all these functions combined in a circuit using a single meter. That's exactly what the modern service

multimeter is.

There are many types of multimeters, having the same basic functions but differing in appearance, method of switching ranges, meter sensitivity and sometimes in the number and extent of the ranges available. These can be separated into two types, those using rotating switches for selecting ranges and functions, and those using plug-in or pin jacks for this purpose. Let's consider the latter type first.

## PIN-JACK TYPE

Figure 29 shows a typical circuit. There are five controls on this instrument—two switches and three ohmmeter adjustments. Switch  $SW_1$  is used to set the meter for a.c. or d.c. measurements.

There are eighteen jacks on the tester into which the test leads can be plugged. By choosing the proper pair of jacks, the proper range is obtained automatically. When a different range is needed, another pair of jacks is used.

Resistors  $R_1$ ,  $R_2$  and  $R_3$  serve as a series shunt across the meter for d.c. current measurements. The jack

tifier in the circuit. The jack marked "AC±" is the common a.c. voltage jack so one test lead is plugged in here for a.c. readings. However, at times you have to make a measurement in a circuit containing both a.c. and d.c. As the d.c. would cause false a.c. readings, it is blocked out by a condenser  $C$ , connected to a jack marked "output." This jack is used as the common jack whenever the a.c. only is to be read. The other test lead is plugged into the "A.C. VOLTS" jack corresponding to the proper range. Resistors  $R_8$  to  $R_{11}$  are the necessary voltage multipliers. Resistor  $R_{12}$  is

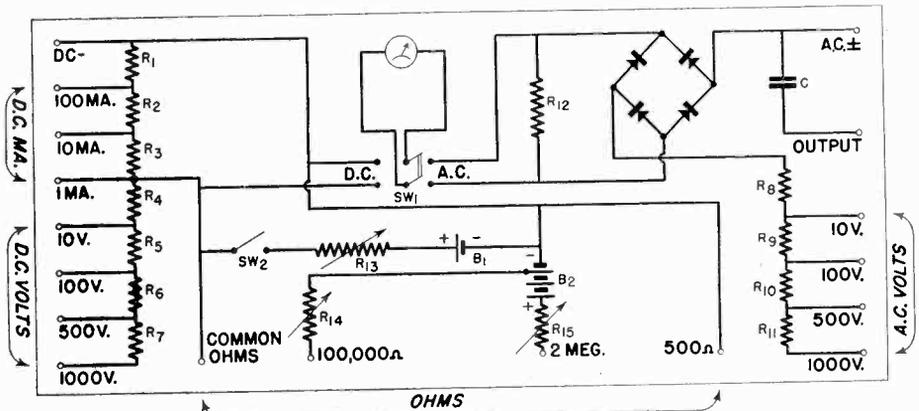


FIG. 29. A multimeter of the pin-jack type.

marked "DC —" is used for all d.c. ranges so is called the "common terminal"; one test lead is plugged here while the other is placed in the current jack marked for the desired range.

Resistors  $R_4$  to  $R_7$  are voltage multipliers. The current shunt resistors  $R_1$ ,  $R_2$ ,  $R_3$  remain in the circuit to give a basic 1-ma. meter movement (sensitivity is 1000 ohms-per-volt). Again the "DC —" jack is the common terminal; one test lead goes here while the other is plugged into the desired d.c. voltage jack.

For a.c. voltage measurements, switch  $SW_1$  is thrown to the AC position, thus inserting a copper-oxide rec-

a correction resistor, used to adjust the a.c. range according to the meter and copper-oxide rectifier characteristics.

Although the ranges of the d.c. and a.c. voltages are the same, they do not usually use the same meter scale, because the rectifier characteristics change somewhat with the amount of current flowing through it. This usually means that the a.c. scale will be crowded near zero. Separate scales are therefore on the meter face for each purpose, direct current and voltage, a.c. voltage, and resistance. A typical scale is shown in Fig. 30.

Both series and shunt ohmmeter ranges are provided. The "Common

Ohms" jack is one terminal for all ohmmeter ranges. Switch  $SW_2$  is closed for shunt ohmmeter measurements. (This switch must be open for all other measurements.) This places  $R_{13}$  and battery  $B_1$  in the circuit. The "500 $\Omega$ " jack is used for this shunt ohmmeter.

There are two series ohmmeter ranges, using a tapped battery  $B_2$  and separate zero adjustments. The a.c.-d.c. switch must be in the d.c. position for all ohmmeter measurements.

Practical multimeters of this type may have fewer or different ranges, but the circuits will be essentially like the one just described.

### SELECTOR SWITCH TYPE

This multimeter eliminates most of the jacks by using a multiple selector switch to select the desired range and function. A typical one is shown in Fig. 31. Switches  $SW_1$  and  $SW_2$  are ganged together. In position 1, the highest voltage range is obtained, through multipliers  $R_1$ ,  $R_2$ ,  $R_3$  and  $R_4$ . Notice that only switch  $SW_2$  is in the circuit at this setting, while at other settings  $SW_1$  is in alone or in combination with  $SW_2$ .

In positions 2, 3 and 4 of the selector switches we have the remaining voltage ranges; positions 5, 6 and 7 are the current ranges; position 8 is a shunt ohmmeter and position 9 is a series ohmmeter. Other ranges can be provided by adding other switch contacts.

Note that the same multiplying resistors are used both for a.c. and d.c. voltages. This is possible where similar ranges and sensitivities are provided. Of course, the a.c. ranges will be marked on different scales.

Where the d.c. voltage ranges are at sensitivities greater than 1000 ohms-per-volt, it is standard practice to shunt the a.c. ranges so they remain 1000 ohms-per-volt. This requires separate multipliers for d.c. and a.c. and

usually requires more switch sections.

The test leads are plugged into the jacks marked "DC—" and "+" for d.c. voltage, d.c. current and ohmmeter measurements. The jacks "+" and "AC  $\pm$ " are used for a.c. voltages, while the jacks "+" and "output" are used for a.c. readings when d.c. must be blocked out. This jack is marked "output" because it is usually used for making output voltage measurements in the plate circuit of the power tube.

**Tolerance.** How accurate do our measuring devices have to be? The answer depends on the type of work being done.

In a research or design laboratory, highly accurate measurements may be desired. Here, expense usually does not matter; it is worth the price of hand-calibration and extreme care to make sure of success in the particular work being done. Further, precision parts

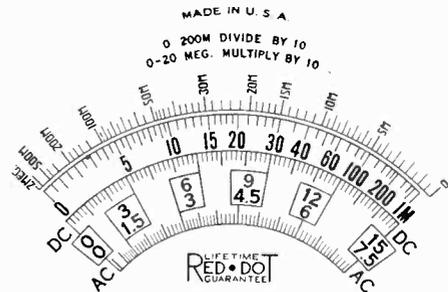


FIG. 30. Typical multimeter scales. Don't worry about reading this one—you will get instructions with any meter you buy. From the top down, the scales are: series ohmmeter; shunt ohmmeter; d.c. voltage and current; and last, a.c. voltage.

are usually used in research, so the observed results can be checked closely by calculations.

If this research is being carried on for a radio set manufacturer, some particular circuit design may be developed. Then, parts will be varied over wide limits to see just what variations can be permitted and still obtain rea-

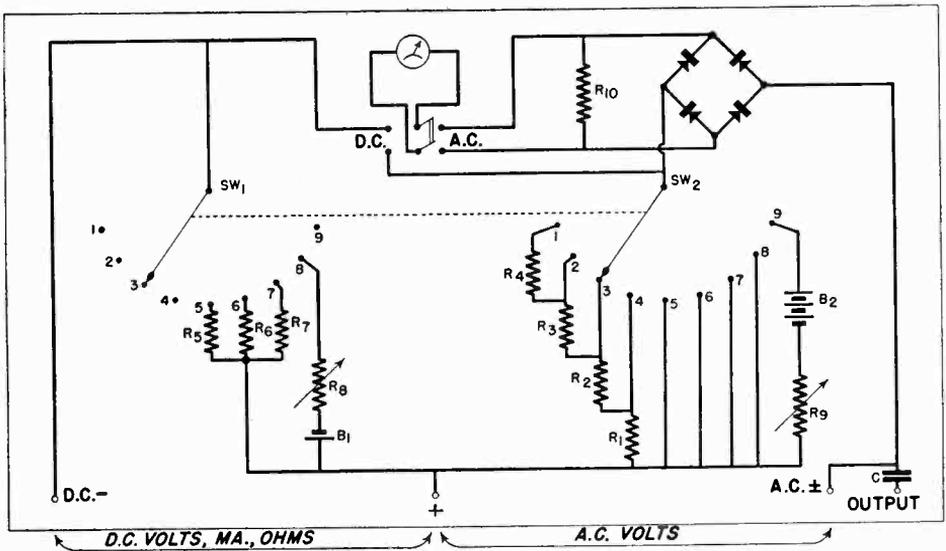


FIG. 31. A multimeter of the selector switch type.

sonably good results. There is a very important reason for this—the question of cost. Precision parts are practically hand-made, and therefore quite expensive. But if wider tolerances are permissible, it will be possible to use cheaper mass-production parts, thus lowering the cost of the set.

When a radio set is made from parts having tolerances of 10% to 20%, the manufacturer is interested in over-all results, not the exact happenings in each stage. Thus, it is possible for readings to be off 30% to 40% from the expected value in many instances, yet the receiver may give average results for its type. Therefore, there is little reason for the serviceman to use

meters having an accuracy better than 2%. Far wider variations will arise in the radio circuit itself.

One of the things you must learn in service work is to interpret the readings obtained, considering the circuits where measurements are being made. In one circuit, variations of 50 volts or so may be expected, while in another a few volts variation may indicate trouble. After gaining a little experience, your growing knowledge of circuits and their operation will soon help you determine just what to expect. Just remember that a meter cannot point out the trouble; it can only tell you what circuit conditions are, from which you in turn figure out the cause.

THE N. R. I. COURSE PREPARES YOU TO BECOME A  
**RADIOTRICIAN & TELETRICIAN**  
(REGISTERED U.S. PATENT OFFICE) (REGISTERED U.S. PATENT OFFICE)

# Lesson Questions

Be sure to number your Answer Sheet 28FR-3.

Place your Student Number on every Answer Sheet.

Send in your set of answers for this lesson immediately after you finish them, as instructed in the Study Schedule. This will give you the greatest possible benefit from our speedy personal grading service.

1. Is the resistance of a shunt used to extend a 1-ma. meter to 100 ma. more, less, or the same as the shunt used to extend the range to 10 ma.?  
*Less*
2. Will the current indicated by a milliammeter in the cathode circuit of a screen grid tube be an accurate measurement of the plate current?  
*No*
3. Give the five rules for connecting a current meter.  
*In series, highest range, meter resist. less than circ. resist proper position proper polarity.*
4. What will happen if you attempt to measure 60-cycle a.c. with a D'Arsonval meter alone?  
*gives an average a.c. current which is zero.*
5. Which types of meters can be used to measure high audio frequency currents accurately, regardless of wave form?
6. Suppose a copper-oxide rectifier voltmeter reads about 55 volts when you are checking a 110-volt power line. What is the probable trouble?  
*burnt or shorted*
7. In measuring voltage in high-resistance circuits, will a 20,000 ohms-per-volt meter or a 1000 ohms-per-volt meter give more accurate circuit voltage measurements?  
*20,000*
8. Which type of ohmmeter, when turned on, draws current from its battery even though the test leads are not touching each other?  
*Shunt type*
9. Is the "zero" adjustment for a shunt type ohmmeter made when the test leads are held together or when they are separated?  
*separated*
10. Suppose you are measuring in a circuit where both a.c. and d.c. exist together and you wish to measure only the a.c. Would you use: 1, the "AC±" multimeter jack; or 2, the "OUTPUT" jack, to exclude the d.c.?

## DIPLOMACY

*Diplomacy* in its true sense is simply courtesy. A true diplomat is one who shows a deep appreciation of the other fellow's feelings.

A diplomat knows when to agree with you—and how to disagree. He will agree on trivial matters regardless of his own opinions, thereby winning your good will. When it comes to something important, however, he will bring you to his way of thinking—*painlessly*. You will enjoy doing what he requests because he makes you feel that you are an important person.

Now turn matters around—and you be the diplomat. In conversation, be considerate of the other fellow's feelings and pet beliefs. Don't contradict people flatly—nobody appointed you to go around correcting the mistakes of others.

People like the man who thinks before he talks—who doesn't make rash or crude statements. And if you can get people to like you, you can count on them to help you—to give the sort of cooperation that will mean success for you.

So learn diplomacy—and practice it in all your contacts with people.

J. E. SMITH

**VACUUM TUBE VOLTMETERS**

**CATHODE RAY OSCILLOSCOPES**

29 FR-2



**NATIONAL RADIO INSTITUTE**

**WASHINGTON, D. C.**

**ESTABLISHED 1914**

# STUDY SCHEDULE NO. 29

For each study step, read the assigned pages first at your usual speed. Reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind, then answer the Lesson Questions for that step. Study each other step in this same way.

1. A Basic D.C. Vacuum Tube Voltmeter . . . . . Pages 1-5

The vacuum tube voltmeter is a radio stage with a means of indicating the voltage fed to it. Considerable ingenuity is used in getting the indications accurately and to eliminate effects of the v.t.v.m. on the circuit where voltages are to be checked and vice versa. You study a simple d.c. type to learn some of the problems and how they can be overcome. Answer Lesson Questions 1, 2 and 3.

2. Improved D.C. Vacuum Tube Voltmeters . . . . . Pages 5-9

Improvements in sensitivity, so small voltages could be measured, led to the development of several more elaborate types for d.c. measurements. The bridge type you will study is one of the leading circuits in v.t.v.m. service instruments today, so study this section carefully. Answer Lesson Question 4.

3. A.C. Vacuum Tube Voltmeters . . . . . Pages 9-13

We go back to our basic d.c. circuit and learn how it can be adapted to measure a.c. Many new problems now come up, particularly restrictions on the frequency range caused by test leads. Even the calibration requires care, as the wave shape will upset results. Answer Lesson Questions 5 and 6.

4. Peak Vacuum Tube Voltmeters . . . . . Pages 13-17

Again, sensitivity needs lead to improved circuits—where not only more sensitivity, but freedom from test lead effects can be obtained. The diode rectifier type and the slide-back circuits in particular are important. Answer Lesson Question 7.

5. Special Vacuum Tube Voltmeters . . . . . Pages 17-20

Here we take up vacuum tube indicators which use several stages of amplification. The tuned r.f. types are the "signal tracers" used by many service shops because of their ability to trace the signal through a radio, thus leading right to the trouble. There is also an important section on calibration of the v.t.v.m.—a procedure necessary at frequent intervals to maintain accuracy. Answer Lesson Question 8.

6. The Cathode Ray Oscilloscope . . . . . Pages 20-28

This is a truly remarkable instrument—one which lets you "see" the signal. Here you learn how we get a spot of light, how to move the spot and how to make an exact tracing of the signal. Answer Questions 9 and 10.

7. Mail Your Answers for this Lesson to N.R.I. for Grading.

8. Start Studying the Next Lesson.

# VACUUM TUBE VOLTMETERS

## CATHODE RAY OSCILLOSCOPES

### A Basic D. C. Vacuum Tube Voltmeter

WHILE nearly all kinds of service work can be done with a standard service multimeter, some of the more difficult jobs can be done more quickly by using special instruments, such as the vacuum tube voltmeter and the cathode ray oscilloscope. For years, these instruments have been standard laboratory and factory equipment. As they are particularly suited for many special jobs, more and more of them are finding their way into the better equipped service shops. Especially as television and high-fidelity receivers become more widespread, there will be an increasing need for knowledge of the uses and limitations of these instruments. Both will be covered in this lesson.

► We will start with the vacuum tube voltmeter—originally developed as an r.f. voltmeter but which was soon used for d.c. measurements as well.

You recall that ordinary voltmeters affect the circuit whose voltage they measure—because they draw current and because they add the meter resistance to the circuit. Fortunately, in ordinary service work these effects are not usually serious, or can be allowed for. In other fields, however, they often make involved calculations necessary.

Even in a simple circuit like Fig. 1, the voltmeter will not indicate the true voltage across  $R$  unless its resistance is many times higher than that of  $R$ , as the voltmeter current through  $R_1$  causes an additional voltage drop. As

you have learned, the voltmeter ohms-per-volt sensitivity determines the extent of the difference between the voltmeter reading and the actual voltage when the meter is removed.

Nowadays, standard meters have sensitivities of 1000 ohms-per-volt, which is acceptable for ordinary service work. For measurements in circuits containing reasonably high resistances, higher sensitivities up to 25,000 ohms-per-volt are available. But where very high resistances are used, as they are in so many radio applications, increased voltmeter sensitivity is needed. That's why d.c. vacuum tube voltmeters are used.

#### BASIC TRIODE V.T.V.M.

Fundamentally, a vacuum tube voltmeter is just an ordinary radio tube stage having an indicator instead of the usual plate load. Figure 2A shows a typical triode circuit. An  $E_g-I_p$  characteristic curve for this type tube is shown in Fig. 2B. By using the proper grid bias, we can operate on the straight part of the characteristic

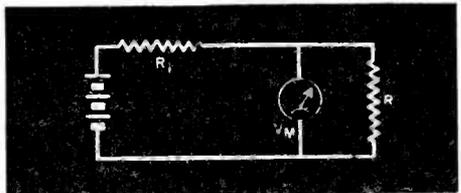


FIG. 1. The measurement depends on the voltmeter current, compared to that drawn by  $R$ .

(point 1), over a curved portion (point 2), at cut-off (3), or even beyond plate current cut-off (4), depending upon just what circuit action we want.

Suppose we operate on the straight part of the characteristic, the tube then acting as a class A amplifier. There will be a no-signal plate current, indicated by the meter in the plate circuit. We can make the meter read mid-scale for this current, either by using a shunt resistor across the meter or by adjusting the plate and grid voltages.

Now if we connect the test probes to a d.c. voltage source, current will flow through  $R_1$ , causing a voltage drop across it. Since this voltage drop is between the grid and cathode, it will either increase or decrease the plate current, depending on the polarity of the drop. By calibrating the plate current meter properly, we can make it indicate the voltage of the source that  $R_1$  is connected across, so we can make a simple d.c. vacuum tube voltmeter out of a standard radio tube amplifying circuit.

### CIRCUIT LOADING

Why is our vacuum tube voltmeter (or "v.t.v.m.," as radio men call it) any better than an ordinary meter? We put the resistor  $R_1$  across the source of voltage being measured—just as we'd have put the meter resistance across the source if we used an ordinary voltmeter. The difference is the fact that  $R_1$  can be made very large. The only real limit on the size of  $R_1$  is the grid current which may result from contact potential and gas within the tube. If resistor  $R_1$  is too large, these tiny currents would cause a positive voltage drop across it which could nullify the C bias and cause erratic changes in plate current. However, resistor  $R_1$  can be as high as 10 or 15 megohms. Just compare this with the

few thousand ohms introduced by the ordinary meter, and you can see that our v.t.v.m. will have far less effect on the circuit whose voltage we're measuring!

For example, if resistor  $R_1$  is 15 megohms and 3 volts across it will give a full-scale meter deflection, the sensitivity of our v.t.v.m. is 5 megohms-per-volt. Commonly, sensitivities of from 1 to 5 megohms-per-volt are obtained on the basic range in commercial types. This is obviously many times higher than even the best 25,000 ohms-per-volt D'Arsonval meter.

► Of course, the simple circuit shown in Fig. 2 needs improvement. The calibration depends upon the characteristics of the particular tube used; no provisions are shown for extending the range; the device will be affected by any stray a.c. voltages; and the circuit does not make full use of the meter sensitivity. Let's see how these limitations can be cured.

### CALIBRATION LINEARITY

No two radio tubes (even of the same type) have  $E_g-I_p$  characteristics which are exactly the same. Since the amount of plate current change for a particular applied grid voltage depends upon the shape of this curve, our v.t.v.m. would probably have to be recalibrated each time the tube burned out and a new one was put in. Furthermore, the shape of the characteristic is greatly affected by the supply voltages. As the batteries run down, the calibration will be thrown off.

One way to solve this difficulty is to use a relatively high plate load resistance in series with the meter. This makes the tube characteristic more nearly straight and less dependent on small supply voltage variations.

An even better scheme is to put a self-bias resistor (like  $R_2$  in Fig 3) in the cathode lead. Now, with this ar-

agement, if the plate current drops because of tube aging or battery supply reduction, the bias voltage will also drop. This tends to keep the plate current more nearly constant. As this bias changes with plate current variations caused by the applied voltage, this produces a bias which tends to oppose the applied voltage. The tube is forced to act as if it had a lower amplification factor, but the stage characteristics are made relatively independent of the tube and supply voltages. This method (called degenera-

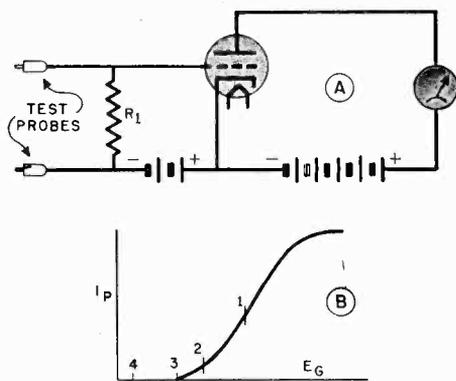


FIG. 2. An ordinary amplifier stage makes an excellent d.c. vacuum tube voltmeter, as the plate current is proportional to grid voltage.

tion) sacrifices some sensitivity, but the benefits outweigh the loss.

### KEEPING OUT A.C.

Since the circuit in Fig. 2 is an amplifier, any a.c. voltage applied across resistor  $R_1$  will cause an a.c. variation of the plate current. This will not affect the D'Arsonval meter, which indicates only the average d.c. plate current. However, if this a.c. exceeds the bias, the grid will swing positive and grid current will flow, which will cause false readings.

Figure 3 shows how a.c. troubles can be avoided. An extra resistor  $R_3$  and a condenser  $C$  have been added in the grid circuit. These parts act as an

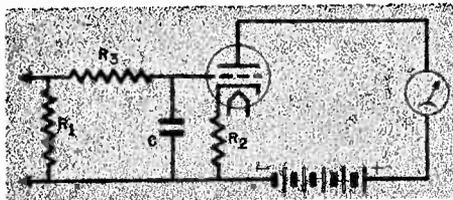


FIG. 3. The filter  $C-R_3$  prevents a.c. ripples from affecting the d.c. voltage readings.

a.c. filter. The applied a.c. is dropped across  $R_3$ , as its resistance is large compared to the low reactance of  $C$ . This reduces the level of any a.c. input voltages at the grid of the tube. Should these voltages still be high enough to swing the grid positive,  $R_1-R_3-C$  act as a grid leak and condenser, automatically biasing the tube. This scheme lets us measure the d.c. voltage in a circuit containing both a.c. and d.c., without the a.c. affecting the reading.

### EXTENDING RANGES

Since we might want our v.t.v.m. to measure d.c. voltages ranging from just a fraction of a volt to 1000 volts or more, we need some means of extending the meter range.

Figure 4 shows the same method we used with an ordinary voltmeter. Resistor  $R_1$  is the grid resistor of our vacuum tube voltmeter circuit and sets the basic range when the selector switch is in position 1. Then when the selector

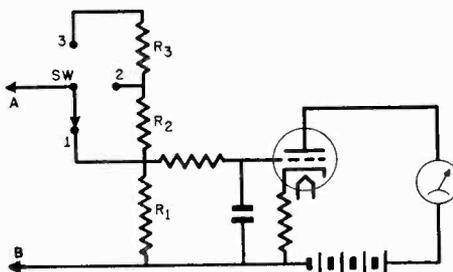


FIG. 4. The addition of series resistances extend the range, like the multipliers used with ordinary voltmeters.

switch is moved to position 2 or 3, the multiplying resistors  $R_2$  or  $R_2$  plus  $R_3$  are added in series.

Since  $R_1$  will probably be 1 or more megohms,  $R_2$  and  $R_3$  must have very high resistance values in order to extend the range. Very large resistances are not always available in compact, stable and exact sizes. Also, when the multiplying resistors are extremely high, any small amount of leakage across the selector switch or in the circuit will upset the voltage division.

► A better circuit is shown in Fig. 5. When the selector switch is in position 1, resistors  $R_1$ ,  $R_2$  and  $R_3$  are all across the grid-cathode of the tube, and all the input voltage is fed into our v.t.v.m. Switch position 1, therefore, sets the meter to its basic range.

When the selector switch is moved to position 2, only part of the input voltage is applied to the input of the v.t.v.m., as the circuit acts as a voltage divider. The exact amount of division depends upon the ratio of resistors  $R_1$  plus  $R_2$  to the total of  $R_1$  plus  $R_2$  plus  $R_3$ . Similarly, when the selector switch is moved to position 3, the voltage across  $R_1$  will be all that causes the meter deflection. Proper choice of resistance permits any desired range extension.

This circuit has the advantage of using easily obtainable resistor sizes. Also, it presents a constant d.c. impedance to the supply source regardless of the range chosen (provided there is no excessive leakage in the grid circuit of the vacuum tube). The ohms-per-volt sensitivity of this v.t.v.m. goes down as the range is increased, but this does not greatly matter—it is still far higher than the average meter.

### MEASURING CURRENT

Can a v.t.v.m. be used to measure current? Yes—very easily. Just insert

a known resistor in series with the circuit carrying the current and measure the voltage across the resistor with the v.t.v.m. Ohm's Law ( $I = E \div R$ ) does the rest. Of course, the known resistor should be made as small as possible so it won't have much effect on the circuit in which you insert it.

The range can be varied by using different resistors. For example, if 5 volts will give a full-scale reading, we can get a 50 ma. range by using a 100-ohm resistor ( $.05 \times 100 = 5$  volts) or a 5-ma. range by using 1000 ohms.

### MEASURING RESISTANCE

You can also measure resistance with a vacuum tube voltmeter. All you need is a known voltage source, as shown in Fig. 6. The resistor  $R_1$  is a part of the v.t.v.m. and is, of course, a known resistance. The resistor  $R_x$  is the unknown one we want to measure. Here's how we do it.

First, the battery is connected between terminals A and B. This puts the battery right across resistor  $R_1$ , and we get a deflection on our v.t.v.m. corresponding to the battery voltage. Usually the battery voltage is chosen or varied to give a full-scale meter deflection.

When the unknown resistance  $R_x$  is inserted in the circuit, the battery current flows through both  $R_x$  and resistor  $R_1$ . This divides the voltage so

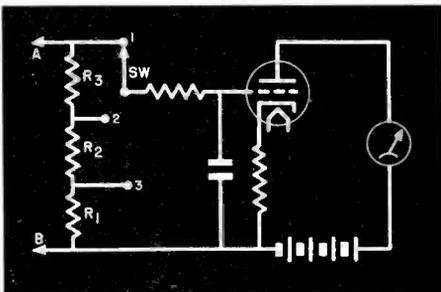


FIG. 5. A voltage divider method of extending the range.

the voltage drop across resistor  $R_1$  is less than the total battery voltage. If resistor  $R_x$  exactly equals resistor  $R_1$ , the voltage divides in half and we get a half-scale meter deflection. You remember that this is just how a regular ohmmeter works. Therefore, we can calibrate the v.t.v.m. scale in ohms, exactly as we do with any other ohmmeter.

The arrangement in Fig. 6 is much better than an ordinary ohmmeter for measuring high resistances. Resistor  $R_1$  may be 10 megohms, which would give us a 10-megohm center-scale reading on our meter. Since we can read 100 times the center-scale value on an ohmmeter, this means we can readily measure unknown resistors as high as 1000 megohms, using only 3 volts or so as  $E$ . This means we can extend the range to far higher values and still use reasonable values of voltage. The method is the same we used with an ordinary ohmmeter—we add additional voltage and an additional resistance in series with it at point X. Thus, if the original circuit measures 1000 megohms with a 3-volt battery, a 30-volt battery will give a 10,000-megohm range, while 300 volts will increase the range to 100,000 megohms. Compare this sensitivity with that of an ordinary ohmmeter, where 300 volts may give a range of only 10 or 15 megohms!

The limit to the amount we can extend our range is the leakage resistance across insulation between jacks,

terminals and mountings for the various parts. To measure very high resistances, you must use extremely good insulating material and just as little of it as possible.

► Low resistances can also be measured with a v.t.v.m. by shunting  $R_1$

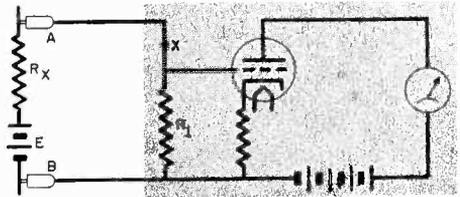


FIG. 6. Using the v.t.v.m. as an ohmmeter.

with a small resistor or replacing it with a low ohmic value to lower the center-scale value. By setting the  $R_1$  value at, say, 100 ohms, the mid-scale ohmmeter reading will be 100 ohms and we can calibrate our meter accordingly.

**V.T.V.M. as a Multimeter.** Since we can use a v.t.v.m. to measure voltage, current and resistance, it is possible to make a multimeter using this basic circuit. By using switches, the proper ranges and functions can be obtained.

This arrangement would give us an extremely sensitive multimeter—which, however, would need frequent recalibrations and adjustments. Whenever the tube is changed, a complete recalibration may be necessary.

## Improved D. C. Vacuum Tube Voltmeters

The circuits so far covered have been simple. They will work satisfactorily in measuring fairly high voltages. Now let's see how better d.c. vacuum tube voltmeters can be made.

**Meter Sensitivity.** So far, we've

been letting the normal plate current of the tube give our meter a mid-scale deflection. This has allowed us to neglect polarity in making measurements. However, it has also meant that we've had to use a fairly insensitive meter.

For example, if the normal plate current is 10 ma., we must use a 20-ma. meter to have the normal plate current cause a mid-scale deflection. Then, the applied voltage must cause a 10-ma. plate current change to give us a full-scale deflection. Even if the tube has a high mutual conductance, a 10-ma. plate current change requires a fairly high grid voltage, so the basic range of our v.t.v.m. is high. This means low voltages produce only small changes in plate current and are not easily read.

But if we can have our meter read zero when no voltage is being measured, we can use a much more sensitive meter—1 ma., for example. To do so, we must find some means of having zero plate current when there is zero grid input. How can this be done?

One way is to apply a higher bias and to work the tube at or near the plate current cut-off point. This is frequently done, but has the disadvantage of making the tube operate over a curved part of its characteristic. This means that plate current—and therefore meter readings—will not be linear for different applied voltages. Even worse, our meter will deflect least for small voltages, where we want the greatest sensitivity. A better plan is to let the tube operate on the straight part of its characteristic and use a bucking circuit to get zero plate current.

## BUCKING CIRCUITS

Figure 7 shows one common bucking circuit. The normal plate electron flow of the tube (solid arrows) and the electron flow caused by the bucking battery  $E$  (dotted arrows) flow through the meter in opposite directions. Resistor  $R$  is carefully adjusted so that the current from  $E$  exactly equals the tube current. The meter then has equal and opposite currents

flowing through it, so it reads zero. This is called a bucking circuit, as the extra current “bucks” or cancels the plate current through the meter.

Now when a voltage is applied to the grid of the tube, the plate current of the tube changes, while the bucking current remains the same. Then the meter indicates the difference between the two currents—in other words, the actual change in the tube plate current.

You can readily see that in using such a circuit we must be sure that the voltage applied to the grid makes it more *positive*. A more positive grid causes an increase in plate current, which will make the meter deflect up-scale, as we want it to.

As you have learned, the important advantage of such a circuit is that we can use a more sensitive meter in the plate circuit, instead of the 10 or 20-ma. meter needed in simpler vacuum tube voltmeters. If the tube has such a characteristic that a 1-volt change

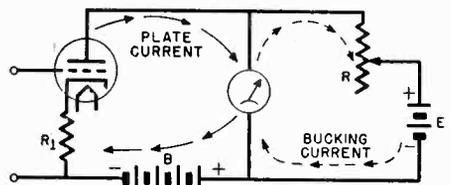


FIG. 7. The bucking current cancels the plate current flow through the meter.

on the grid causes a 1-ma. change in plate current, and a 1-ma. meter is used, the basic range of our improved v.t.v.m. is 1 volt. (By using a more sensitive meter, we can get an even lower basic range.) Furthermore, this 1-volt scale is now spread over the entire meter scale instead of half of it, which alone doubles the scale sensitivity.

There are many variations of this bucking circuit, but all have disadvantages. If  $R$  is not adjusted almost per-

fectly, the meter may be overloaded and burned out. This means the meter must be shunted until  $R$  is adjusted. Further, the separate battery is a nuisance—among other things, its voltage decreases as it grows older. Several better circuits, using the bridge principle, have been developed. Let's see how they work.

### D.C. BRIDGE V.T.V.M.

You will study the bridge circuit in more detail later, so here we'll just go into the basic principles briefly. A typical bridge is shown in Fig. 8.

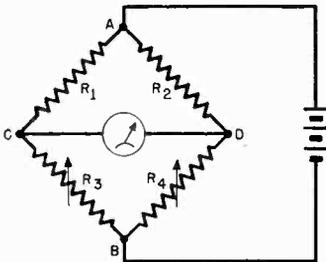


FIG. 8. A resistance bridge.

When a voltage is applied across terminals  $A$  and  $B$ , current flows through  $R_1$  and  $R_3$ , as well as through  $R_2$  and  $R_4$ . If we make  $R_1$  equal  $R_2$  and  $R_3$  equal  $R_4$ , the resistance of  $R_1$  plus  $R_3$  will equal  $R_2$  plus  $R_4$ , so the current through path  $A-C-B$  is equal to that through path  $A-D-B$ . This makes the voltage drop across  $R_1$  equal to that across  $R_2$ , so the potential of point  $C$  with respect to  $A$  is the same as that of  $D$  with respect to  $A$ . Hence, points  $C$  and  $D$  are at the *same potential* (no voltage difference between them)—and *no current* flows through the meter. The bridge is said to be “balanced” when the meter reads zero.

When either pair of resistors is made unequal ( $R_1$  not equal to  $R_2$ , or  $R_3$  not equal to  $R_4$ ), there will be a voltage difference between  $C$  and  $D$ , so current will flow through the meter.

The bridge is now unbalanced.

To use this bridge principle in our v.t.v.m., we set up the circuit shown in Fig. 9. This is much the same as the circuit in Fig. 8, except that we've replaced variable resistor  $R_3$  with a tube and bias resistor  $R_6$ . Resistors  $R_1$  and  $R_2$  are equal in value.

With no input on the grid, we balance this circuit by adjusting  $R_4$  until no current flows in the meter. As you just learned, this means that  $R_4$  equals the plate-cathode resistance of the tube plus the resistance of  $R_6$ .

Now if we apply an input to the grid through terminals  $X$  and  $Y$ , the resulting grid voltage changes the tube plate current. This current change means the plate-cathode tube resistance has changed—so the bridge is no longer balanced and current flows through the meter. Calibrating the meter in terms of voltage input to the grid gives us our v.t.v.m.

**Regulating the Power Supply.** It is usually preferable to use a power pack for our bridge type v.t.v.m., instead of a battery. But such a power supply must be very well regulated. Changes in the supply voltage will shift operation of the tube to a different part of its characteristic and, as you know, this destroys the calibration of the v.t.v.m.

One common method of regulation,

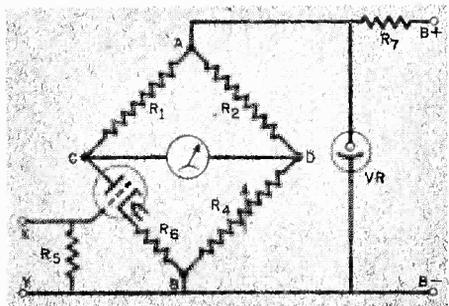


FIG. 9. Using a tube in a bridge in place of a resistance leg.

using a voltage regulator tube  $VR$  and a series resistor  $R_7$ , is shown in Fig. 9. This tube has the property of passing a much higher current when the voltage across it increases slightly. The voltage drop this current causes in resistor  $R_7$  then drops the voltage across  $VR$ . The net effect is to keep the voltage across  $VR$ , which is the voltage supplied to the v.t.v.m., very nearly constant. Under normal conditions, a modern regulator tube can maintain the voltage across its terminals within 2 or 3 volts of its rated value.

**An Improved Circuit.** An even better bridge type v.t.v.m. circuit is shown in Fig. 10. Here another tube, exactly like the first one, is used in place of resistor  $R_4$  of our original bridge circuit.

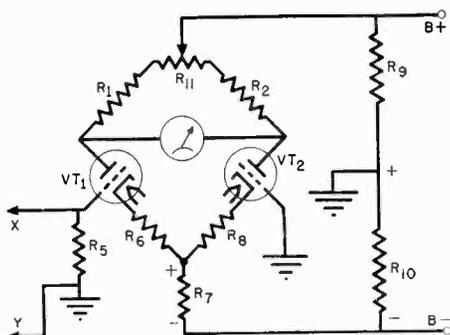


FIG. 10. A balanced tube bridge.

The use of two tubes tends to cancel the effects of any change in supply voltage. In other words, if the supply voltage increases, the plate current to both tubes increases correspondingly, and the bridge remains in balance, thus eliminating the need for a regulator tube.

► Resistors  $R_6$  and  $R_8$  are used to make the tubes bias themselves. This also helps keep the circuit in balance.

► An unusual feature of the circuit in Fig. 10 is the use of  $R_7$  to get greater stability and higher sensitivity. Both plate currents flow through this re-

sistor, producing a voltage having the polarity indicated, so more self-biasing is produced. However, this bias voltage is high enough to cut the tube currents off except for the fact that it is bucked out by a positive voltage from  $R_{10}$ . (Trace the grid circuits to ground, to the junction of  $R_9$  and  $R_{10}$ , then through  $R_7$  to each cathode to see this.) Now the resulting net bias is about that of each cathode resistor.

When a voltage is applied to the input of  $VT_1$ , so that terminal  $X$  is made positive with respect to terminal  $Y$ , the plate current of  $VT_1$  increases. This causes an increased flow of current through resistor  $R_7$ , thus biasing the grid of  $VT_2$  more negatively. (The positive  $R_{10}$  drop remains fixed, but the negative  $R_7$  drop increases, so the net effect is more negative bias on  $VT_2$ .) Therefore, the plate current of tube  $VT_2$  goes down.

This means that the plate resistance of tube  $VT_1$  goes down at the same time the plate resistance of  $VT_2$  goes



Courtesy R. C. A.

FIG. 11. The RCA-Rider Volt-Ohmyst Jr. is a typical d.c. type v.t.v.m. using the balanced bridge circuit.

up. Therefore, this bridge becomes much more unbalanced for a given input than does the circuit in Fig. 9, and so is more sensitive. Also, the v.t.v.m. in Fig. 10 can be made more linear-reading than the Fig. 9 v.t.v.m. by choosing tubes whose characteristics balance each other.

An exact balance of the bridge (zero meter adjustment) is obtained by adjusting  $R_{11}$ . Varying this resistor varies  $R_1$  and  $R_2$ , adding to one and subtracting from the other. This varies the tube plate voltages, so the tube resistances change also. If the ratio of  $R_1$  to  $R_2$  is made the same as the ratio of  $R_3$  to  $R_4$ , the currents in the two paths will adjust themselves so that the same voltage drops occur, even though the currents are unequal. This ratio is usually given as  $\frac{R_1}{R_2} = \frac{R_3}{R_4}$ .

Notice that it is not necessary that  $R_1 = R_2$  or  $R_3 = R_4$ , just so the ratio of values is properly chosen.

### A COMMERCIAL BRIDGE V.T.V.M.

A typical commercial bridge instrument is the RCA Junior Volt-Ohmyst shown in Fig. 11. Its circuit is basically the same as that of Fig. 10, with provisions to extend ranges and measure resistance.

The Junior Volt-Ohmyst may be used as a multi-range d.c. type v.t.v.m., as a low- and high-range ohmmeter, or as an a.c. voltmeter. Selector switches permit changeover to the desired range and function. The a.c. voltmeter incorporated in the instrument is a standard copper-oxide rectifier type—not an a.c. vacuum tube voltmeter, so this is a d.c. type v.t.v.m.

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## A. C. Vacuum Tube Voltmeters

You recall that most standard a.c. voltmeters have distributed inductance and capacity which ruin calibration when you try to use them on high-frequency a.c. Highly sensitive a.c. vacuum tube voltmeters have been developed which are relatively free from frequency errors and limitations, even at radio frequencies. Let's see how they work.

### A TRIODE A.C. TYPE V.T.V.M.

You learned that the basic d.c. v.t.v.m. shown in Fig. 2A won't indicate on a.c. because the tube is worked on a straight part of its characteristic. Thus, an a.c. grid input produces an a.c. plate current varying around the normal plate current—and the *average* of this a.c. plate current (which is all the meter can read) is just the same as the normal plate current. (See Fig.

12.) Therefore, the meter reads only the normal plate current as long as we operate the tube on the straight part of its characteristic.

But by changing the bias so as to move the operation to the curved part of the characteristic, or to plate current cut-off, we get the detector action shown in Fig. 13. Now there is little or no plate current when no signal is applied to the grid. When an a.c. voltage is applied, rectified pulses are produced which have an average different from the no-signal plate current. This difference in average plate current can be read on a meter.

**Sensitivity.** With this change in operating point, the circuit in Fig. 2A will read both d.c. and a.c. voltages, provided the d.c. voltage has a polarity which makes the grid positive. How-

ever, the meter scale is not at all linear, because of the curve of the tube characteristic near plate current cut-off. This means very little meter deflection is obtained for small voltages, so the scale is crowded and difficult to read at the low end.

As the applied voltage is increased, the meter deflection becomes more linear, so this circuit is satisfactory for fairly large voltages. Farther on

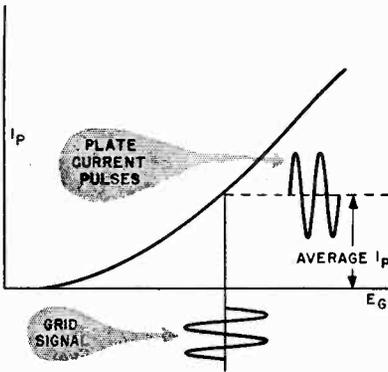


FIG. 12. The d.c. average of the plate current does not change when an a.c. voltage is applied to a class A amplifier.

in this lesson we'll take up circuits for measuring low voltages.

**Frequency Range.** With the proper tube and circuit arrangement, we can measure very high radio frequencies on our a.c. vacuum tube voltmeter. The limit to which we can go before the scale calibration becomes inaccurate is determined primarily by the input circuit of the v.t.v.m. and the circuit where measurements are being made. The input capacity of the tube is in parallel with the grid resistance. As frequency goes up, the reactance of this input capacity goes down. This reduces the impedance between the input terminals, so our v.t.v.m. will eventually start loading the circuit across which it is connected.

This loading is not so serious when the v.t.v.m. is connected across a tuned

circuit. Since the v.t.v.m. is primarily a capacitive load, it will detune the resonant circuit. If we retune the circuit with the v.t.v.m. in place, we can automatically cancel this effect and make our measurements. Of course, the circuit must be retuned to the original frequency when the v.t.v.m. is removed.

**Test Lead Effects.** In measuring r.f. voltages, it is desirable to use a shielded test lead to connect our v.t.-v.m. to the circuit being measured. Shielding prevents the leads from picking up hum or other stray voltages which would produce false readings. However, the capacity between the shielded lead and the shield is connected in parallel with the tube input capacity. This increases the load introduced into the measured circuit, so

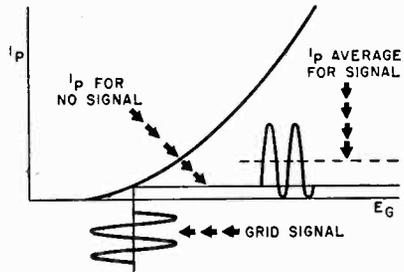


FIG. 13. By moving to the curved portion of the characteristic, there will be a d.c. average change from the no-signal value, when an a.c. voltage is applied.

a shielded lead further limits the frequency range.

**Test Lead Resonance.** The test leads become resonant whenever their physical length equals a quarter wavelength at the frequency where measurements are being made. At a frequency of 75 megacycles (4 meters), leads about 3 feet long would be a  $\frac{1}{4}$  wavelength line.

As you'll learn in another lesson, a quarter-wave line has an unusual voltage distribution. If we apply a high

voltage to one end, we get very little or no voltage from the other end. Obviously, this would make our v.t.v.m. reading highly inaccurate!

While this quarter-wave effect occurs only at resonance, the test leads begin to affect the voltage at frequencies considerably lower. In general, a v.t.v.m. using test leads cannot be used

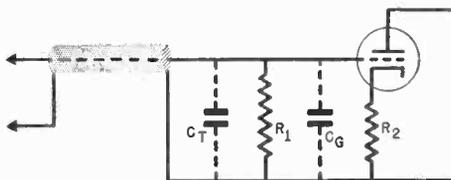


FIG. 14. The input circuit of an a.c. type v.t.v.m.

for voltages with frequencies greater than one or two megacycles, unless special calibration charts are provided. This is still quite an improvement over the standard a.c. voltmeter. Later on, we will see how voltages can be measured at higher frequencies.

**Extending A.C. Ranges.** The input circuit for the basic range of our v.t.v.m. is similar to the circuit shown in Fig. 14 where we have input resistance  $R_1$  shunted by  $C_T$  and  $C_G$ , the test lead capacity and the tube input capacity respectively.

If we try to use the voltage-dividing circuit shown in Fig. 15 which worked so well for d.c., we find that  $C_G$  causes trouble. Moving the selector switch to

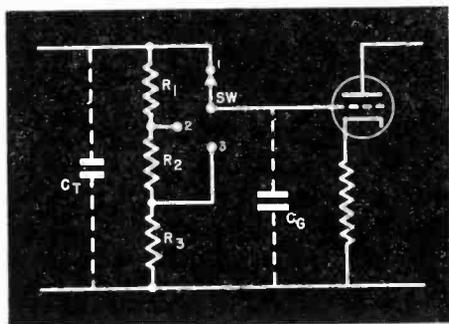


FIG. 15. The tube capacity upsets the voltage division as the switch is rotated.

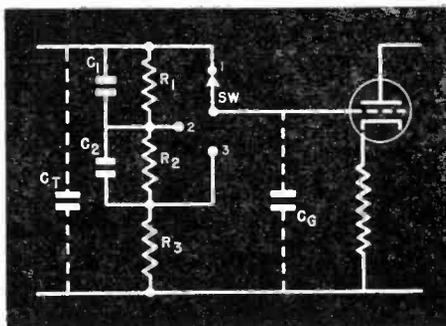


FIG. 16. This capacity voltage divider gives uniform a.c. voltage division.

position 2 shunts  $C_G$  across resistors  $R_2$  and  $R_3$ . Since the reactance of  $C_G$  varies with frequency, the impedance of  $C_G$  in parallel with  $R_2$  and  $R_3$  will vary with frequency, while the resistance of  $R_1$  remains constant. This means that the voltage division across the divider also varies with frequency—so the range we get at position 2 (or at position 3) depends on the frequency of the input voltage.

We can clear up this difficulty by using a capacitive voltage divider like that shown in Fig. 16. When used to measure a.c. voltages, the division of voltages across this divider is determined by the reactances of the condensers. In position 2, the a.c. voltage division is determined by the reactances of  $C_1$  and  $C_G$ . In position 3,  $C_2$  is added to  $C_1$ , so the combined reactance to that of  $C_G$  is the divider. Since these reactances all vary in the same manner when the input frequency varies, the division of voltages remains the same, and the various ranges of our v.t.v.m. are therefore very nearly independent of frequency.

► This v.t.v.m. circuit may also be used to measure d.c. voltages. In this use, the condensers have no effect and the resistors determine the voltage division for the instrument's ranges, just as they did in the circuit shown in Fig. 5.

## A.C. WAVE SHAPES

You recall that an a.c. cycle has a peak value, an average value and an effective value. Because we are usually most interested in the r.m.s. value of a sine wave, we calibrate an a.c. type of v.t.v.m. to indicate r.m.s. values.

But the meter of an a.c. type v.t.v.m. actually operates from the *average* of the plate current. Now the average plate current is not the same as the effective or r.m.s. value—so we have a meter which really operates on one value of an a.c. wave but is calibrated to indicate another.

This is perfectly all right as long as the ratio of the r.m.s. value to the average value is constant—as it is for any regular wave shape. If you want to determine the average value of the a.c., all you have to do is multiply the meter scale reading by the proper multiplying factor. The peak value can also be determined in the same way (using a different factor, of course).

But you must remember that the exact ratios between the r.m.s., average and peak values of a wave depend on the wave shape. Figure 17 shows the relationships existing in various kinds of waves. Notice that all these waves have a *different* peak-to-average-to-r.m.s. relationship. (The special wave of Fig. 17E is similar to television control pulses and has widely different r.m.s. and average values, depending on the height and width of the pulses as well as their spacing.)

**Distorted Waves.** The meter of a v.t.v.m. intended for service work is usually calibrated to read r.m.s. values when the instrument is measuring a pure sine wave. The readings may be in error if such a v.t.v.m. is used to measure a distorted wave, because the distorted wave will probably not have the same average-to-r.m.s. relationship as a sine wave.

Suppose we start with a sine wave, but one receiver stage distorts the wave as shown in Fig. 18. One-half the wave is practically a square wave, while the other half remains a sine wave. This is commonly the result of an amplifying tube operating too near the current cut-off point.

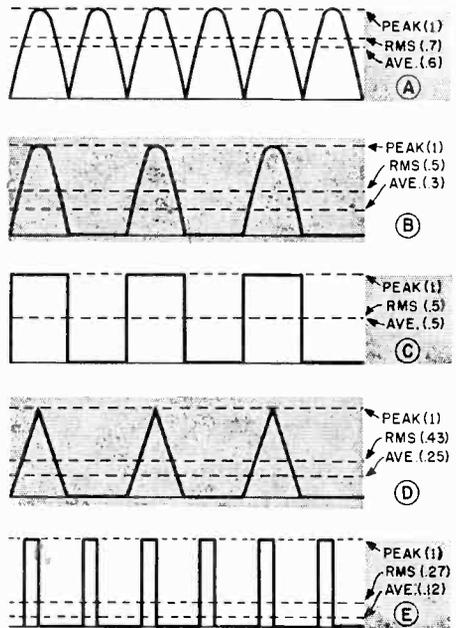


FIG. 17. The shape and size of the wave pulse determines the relationship between the peak, average and r.m.s. values.

If we apply our test leads in such a manner that the sine portion of this wave makes the grid of the v.t.v.m. more positive, we'll get an indication which corresponds to the effective value of the sine wave.

On the other hand, if we reverse the test leads, the rectifying action of the v.t.v.m. will cut off the sine wave portion and we will get an indication corresponding to the distorted half of the cycle. This will give us an entirely different reading from the first one, be-

cause the relationship between the peak, average and effective values will be different. Distorted waves of this kind cause more error with a *peak* v.t.v.m. (to be studied later) than with *average* types (such as we have already discussed), because the peak is always very different from the r.m.s. value while the average and r.m.s. values are usually fairly close to each other.

► Reversing the v.t.v.m. leads may produce different readings even on an undistorted wave, because of the difference in capacity to ground between leads or parts in the v.t.v.m. The different readings caused by reversing leads is an effect known as "turnover."

Usually a shielded cable is provided to connect the v.t.v.m. to the radio being tested, with the shield used as one of the probes. Since the shield is always connected to an r.f. grounded

point in the radio, the hot probe will always be the probe for measuring voltage, and it will not be possible to reverse the probes. Remember, however, that each amplifying stage reverses the phase of a signal 180°—

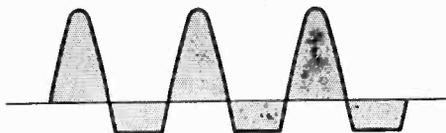


FIG. 18. A distorted wave, such as might be produced by operating a tube at the wrong point on the characteristic.

which produces exactly the same effect as reversing the leads—so it is still possible for distortion to produce an error in the reading as we go from stage to stage. Fortunately, in servicing, we are generally interested more in the *presence* of voltage than in the exact amount, so this problem is mostly limited to laboratory work.

## Peak Vacuum Tube Voltmeters

A diode rectifier can be used like a copper-oxide rectifier, with a d.c. meter to make an a.c. voltmeter. The diode will give the voltmeter a wider frequency range, but otherwise has no advantages. Yet, a simple change in this diode circuit will make it one of the most valuable v.t.v.m. types.

### PEAK-RESPONDING DIODE V.T.V.M.

The diode circuit just mentioned would have a bypass condenser across the meter, to prevent a.c. flow through the movement. By moving this condenser to the position shown in Fig. 19 and using a resistor  $R_1$ , to set up an R-C time-constant circuit, we will have a peak-responding v.t.v.m.

When the plate of diode VT is made positive by the voltage source, con-

denser  $C$  is charged.

When the cycle reverses, the tube stops conducting and  $C$  starts to discharge through  $R_1$ . However,  $R_1$  is so large that condenser  $C$  does not have time to discharge very much before the cycle reverses and charges it up again. When fully charged,  $C$  has a voltage equal to the source voltage peak. After some of this charge has leaked off during the half cycle the tube does not conduct, the voltage drops somewhat. When the cycle reverses again, the tube can't conduct until the source voltage becomes higher than the reduced condenser voltage. When the tube does conduct, the condenser is charged to full source voltage almost at once.

The net result of this circuit action

is that  $C$  always has a voltage very near the peak voltage of the source. Since  $C$  is in parallel with the series combination of  $R_1$  and the meter, this combination also has practically all the peak voltage across it all the time. This peak voltage causes a current flow through the meter—so we have a v.t.v.m. which indicates peak voltages.

► Unfortunately,  $R_1$  has to be so high in resistance that very little current flows through the meter. A very small current change occurs for even large voltage changes so the meter does not make a good indicator. However, we can easily solve this problem.

If we remove the meter altogether,  $R_1$  will have practically all the peak

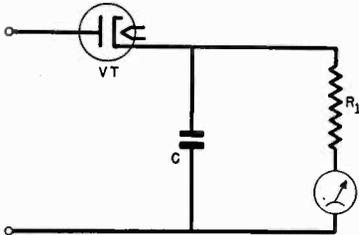


FIG. 19. A simple diode peak-indicating a.c. type v.t.v.m.

source voltage across it all the time. You'll notice that the action of the tube and condenser makes the voltage across  $R_1$  a d.c. voltage. Therefore, all we have to do is put a d.c. type v.t.v.m. across  $R_1$ , and we'll have a sensitive way of measuring the peak voltage of the a.c. source.

► Since this circuit draws current from the source only at the peaks of the applied voltage, this v.t.v.m. acts as if it had a very high resistance. When it is used to measure high frequency, the amount it loads the voltage source depends mostly on the capacity between the input terminals. The plate-cathode capacity of the tube is in series with condenser  $C$ , across the terminals, but condenser  $C$  offers low reactance to r.f., so the loading

depends primarily upon the internal tube capacity.

**An Improved Circuit.** An even better circuit is shown in Fig. 20. Here's how it works.

During the half of an input a.c. cycle that makes terminal  $A$  positive with respect to terminal  $B$ , the diode conducts. Electrons flowing through the tube charge  $C_1$  to the peak source voltage, with the polarity shown.

When the input cycle reverses and the tube ceases to conduct, the voltage on the condenser and the input voltage combine to cause an electron flow through  $R_1$ . However, the time constant of  $C_1$  and  $R_1$  in series is so high that  $C_1$  discharges only a little during this time.  $C_1$  is thus only a little below the source voltage when the cycle reverses again. As soon as the source voltage rises above the voltage on  $C_1$ , the tube conducts again, and  $C_1$  is rapidly charged up to full peak voltage.

Thus this circuit acts in much the

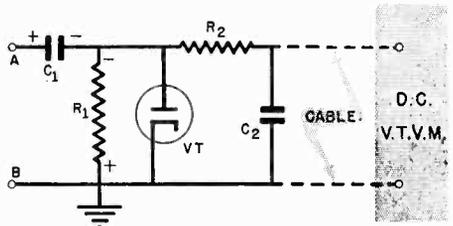
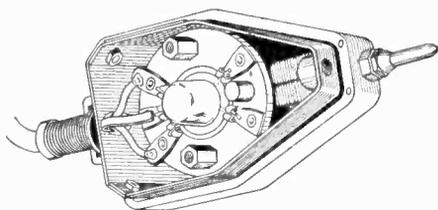


FIG. 20. An improved peak v.t.v.m.

same way as the previous one—almost the full peak voltage remains on  $C_1$  all the time, and the circuit draws current from the source only when the peak value exceeds the  $C_1$  voltage. Thus, the circuit draws but little current from the source.

► The voltage indications are obtained from the d.c. type vacuum tube voltmeter which is connected through filter  $R_2$ - $C_2$  so that it measures the voltage across  $R_1$ . Any r.f. across  $R_1$  is prevented from getting to the v.t.v.m. by the a.c. filter  $R_2$ - $C_2$ .

On the half cycle when terminal  $B$  is positive, the voltage across  $C_1$  and the source voltage combine to force a small current through the high resistance,  $R_1$ . The resulting voltage drop across  $R_1$  has a d.c. average which operates the v.t.v.m. On the other half cycle, the source voltage and that across  $C_1$  buck or oppose each other. Hence, there is practically no current through  $R_1$  on this half cycle.



Courtesy P. R. Mallory Co.

FIG. 21. An acorn type tube, mounted right in the probe, provides a minimum of input capacity, allowing measurements up to several hundred mc.

► At low frequencies, the time between reversals of the input voltage may become so great that nearly all the charge leaks off  $C_1$ . The circuit then ceases to be a peak-indicating device. However, it is easy to give  $C_1$  and  $R_1$  such a high time constant that the frequency at which this effect occurs is quite low.

► The amount this circuit loads the measured circuit depends on the plate-cathode capacity of the tube and on stray capacity between the terminals and between  $C_1$  and ground. By using a tube with extremely low capacity, such as one of the modern acorn type tubes, we can reduce this capacity so that very high frequencies can be measured. In fact, we can mount the tube right in the test probe, and so eliminate most of the difficulties with test leads that you learned about earlier in this lesson. Figure 21 shows a tube mounted in this fashion. Using such a probe and reducing stray ca-

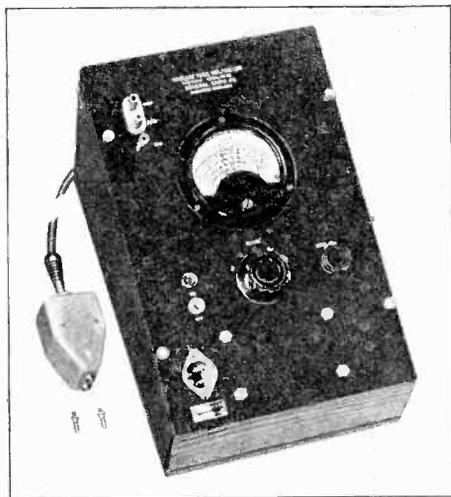
pacities in the circuit to a minimum, we can measure voltages at several hundred megacycles with this instrument.

There are three important things to remember about this type of v.t.v.m.:

1. It can be made fairly sensitive.
2. It can be used at extremely high frequencies.
3. It will not load the measured circuit much.

► It is possible to change the range of a v.t.v.m. of this type by using a multi-range d.c. vacuum tube voltmeter.

► The peak-responding type of v.t.v.m. is more sensitive than average-



Courtesy General Radio Co.

A vacuum tube voltmeter capable of measuring r.f. voltages up to several hundred megacycles. An acorn type tube is mounted right in the probe unit to reduce input capacity and make such high-frequency measurements possible.

indicating types, because the peak value of a wave is almost always reasonably high and so provides enough voltage to operate the meter—while the average voltage may be quite low. However, since a peak v.t.v.m. is usually equipped with a meter calibrated to indicate r.m.s. or effective values, it may give incorrect readings on distorted waves.

## SLIDE-BACK V.T.V.M. CIRCUITS

There is another kind of v.t.v.m. used to measure peak values. It is known as the "slide-back" type.

A typical circuit is shown in Fig. 22. In using it, the slider of potentiometer  $P$  is first moved to point 1. The voltmeter  $V_M$  then reads zero. Next, with no signal applied (terminals  $A$  and  $B$  shorted together), the small tube current is read on meter  $I_M$ . This is not quite zero, because contact potential within the tube causes a small amount of current flow. The exact amount of current does not matter, just so we remember it.

Then the voltage to be measured is applied between terminals  $A$  and  $B$ . This causes a current flow through the meter  $I_M$  and through resistor  $R_1$ . Resistor  $R_1$  and condenser  $C_1$  can be so chosen that this current will cause a meter deflection proportional to the peak value of the applied voltage.

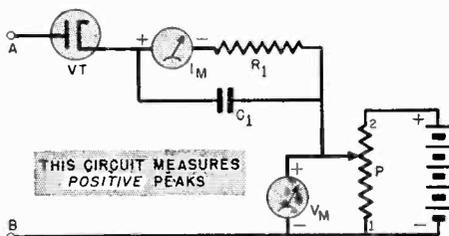


FIG. 22. A positive-peak slide-back v.t.v.m.

Now, with the signal still applied, potentiometer  $P$  is adjusted to bring the pointer of current meter  $I_M$  back to its initial position. In other words, the potentiometer  $P$  is used to introduce a bucking voltage from the battery into the circuit. When this bucking voltage just brings  $I_M$  back to its initial reading, it is equal to the peak signal voltage. Then all we have to do is read the bucking voltage on meter  $V_M$ , and we know what the peak voltage is.

► This circuit has several advantages. We don't have to calibrate meter  $I_M$ ,

because we use only its initial pointer position. Also, the characteristics of the tube and other parts aren't very important. In fact, the accuracy of the measurement in a properly adjusted circuit depends only on the accuracy of the d.c. meter  $V_M$ .

Some precautions must be taken in using this slide-back meter, however. Principally, you must be sure to adjust the potentiometer just enough to bring  $I_M$  back to its initial reading and

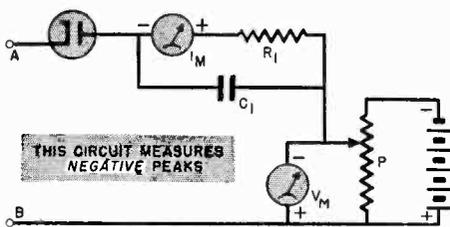


FIG. 23. A negative-peak slide-back v.t.v.m.

no farther, to prevent errors in the reading.

**Negative Peak Indicator.** The circuit in Fig. 22 works whenever terminal  $A$  is made positive with respect to terminal  $B$ . Therefore, this device indicates on d.c. as well as a.c.

Should the polarity be reversed or should we want to measure the other half of the a.c. cycle, we can reverse the terminals  $A$  and  $B$ . Sometimes it is not a good plan to do this, as there may be capacity effects that would affect the readings. Also, it may be necessary or desirable to ground terminal  $B$ .

These possible difficulties make it better to leave terminals  $A$  and  $B$  connected as before, and change the circuit as shown in Fig. 23. Here the diode, the meter  $I_M$  and the slide-back battery have been reversed. Now, when terminal  $A$  is negative ( $B$  is positive), the diode passes current and the instrument works as before—except that it measures the negative peaks.

**Using Slide-Back Voltmeters.** As

there must be a small tube current so that the meter  $I_M$  will indicate, there is some rectification of the extreme tips of the a.c. cycle. This produces an error of about .5 volt, which is appreciable on low-voltage measurements but is negligible on high voltages. This error must be allowed for when measuring voltages below about 25 volts.

► To protect the instrument, always operate it in the following manner:

Short the input terminals and make a note of the initial  $I_M$  reading. (All practical types have an extra biasing battery and potentiometer with which you can adjust the initial current to a small value.) Then apply the maximum slide-back battery voltage *before* connecting to the unknown voltage source. After the unknown voltage is applied, adjust the potentiometer until meter  $I_M$  rises from zero to its initial reading. This procedure protects the current meter.

**Triode Slide-Back V.T.V.M.** You can use a triode tube as a slide-back

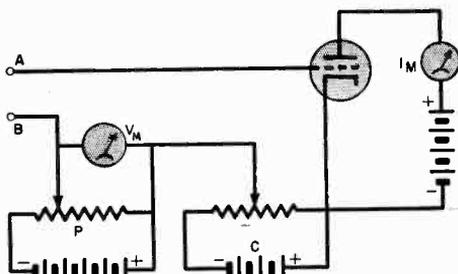


FIG. 24. A triode slide-back v.t.v.m.

v.t.v.m., as shown in Fig. 24. With terminals A and B shorted, the C bias is adjusted to give some particular plate current  $I_M$ . Then the maximum slide-back voltage is introduced by potentiometer P. With the unknown voltage now applied to terminals A and B, the potentiometer setting is reduced until the meter  $I_M$  returns to its original reading. The  $V_M$  reading is then equal to the peak voltage applied.

This circuit can only be used to measure positive peaks. However, it causes less loading of the source of voltage, as the grid circuit draws no current from the source.

## Special Vacuum Tube Voltmeters

### TUNED RADIO FREQUENCY V.T.V.M.

Some of the peak vacuum tube voltmeters we've studied can be made to have a basic range of about 1 volt, which is sufficient for many r.f. measurements. However, to measure the small voltages in the r.f. stages of a receiver, we need a much more sensitive instrument.

Furthermore, all the a.c. types studied so far have one fault. They measure *any* input signal, whether it's the one we want or not. Noise pulses, hum, or additional r.f. voltages that happen to be in the circuit will be measured right along with the signal whose voltage we're trying to measure.

Now, we can overcome this difficulty by using a tuned circuit. As you know, a resonant circuit has the ability to separate frequencies. By tuning a resonant circuit to the desired frequency and using it in our v.t.v.m. ahead of the d.c. indicator, we can make measurements at just the frequency we want.

A resonant circuit also makes it possible to use several amplifying stages ahead of the rectifier in our peak v.t.v.m., coupling from stage to stage just as in a radio receiver. This, of course, gives us the extra sensitivity we need for measuring small voltages.

► What kind of amplifier shall we use? A superheterodyne? No—because

the superheterodyne will itself produce interfering signals unless extreme care is exercised. Instead, we use tuned radio frequency stages in our amplifier. Although the stage gain with a t.r.f. circuit is not as great as with a superhet, the former gives us a more reliable instrument.

► We don't use a tuned circuit at the input of the vacuum tube amplifier. If we did, as soon as we connected the input across the radio part whose voltage we wanted to measure, the tuned circuit would either be detuned by the reactive component of that part, or be so loaded by the resistive component that the  $Q$  of the circuit would drop. Either condition would cause a loss of amplification, and so make the calibration of our v.t.v.m. inaccurate.

Instead, we use a special untuned input circuit which feeds into an amplifying tube. The tube then feeds into a tuned coupler which connects to the next tube in line. The special input circuit, by sacrificing some amplification, eliminates the effects of the shielded cable and tube input capacities. This means the input loading is reduced and the input will not detune the signal source.

Figure 25 shows the input circuit. Capacity  $C_T$  represents the input tube capacity combined with the shielded cable capacity. We'll first analyze its operation with the switch  $SW$  in position 1.

The circuit is connected to the point of measurement by two leads. One is a grounded lead (usually clipped to the chassis of the radio under test). The other is a shielded cable about 3 feet long with a test probe at the end. The test probe of the shielded cable is the secret of this device. Inside the insulating sleeve of the probe is a small mica condenser  $C_1$ , usually between 1 and 4 mmfd., which connects the probe point  $P$  and the "hot" inner cable lead.

This probe capacity is in series with the signal source and capacity  $C_T$ . Since  $C_1$  is much smaller than  $C_T$ , and capacities in series give a value less than the smallest, the maximum capacity across the circuit where measurements are made will be approximately the probe capacity. This value is so small that practically no detuning of the signal source will occur at low or medium radio frequencies.

When probe point  $P$  is connected to an r.f. or i.f. terminal, the source voltage divides between the probe capacity  $C_1$  and capacity  $C_T$ . If the input capacity  $C_T$  is 100 mmfd. and the probe-to-cable capacity  $C_1$  is 2 mmfd., 1/50 of the source voltage is applied to the tube input. If the amplifier has a gain of 50,000, the net over-all gain of the v.t.v.m. is  $50,000 \div 50$ , or 1000 times. Thus, we sacrifice some of the gain to use the probe capacity and avoid detuning and loading effects—but we still have considerable amplification left.

The voltage division will remain the same for all frequencies which are low enough to allow the cable to act essentially as a capacity (no resonant effects). The average limit is about 5 mc. Since most of the service measurements normally made with this instrument will be below 2 mc., this is not much of a limitation. In a factory-built unit, the tuning range is restricted to those frequencies where reasonably flat response can be obtained.

The probe capacity allows us to use a capacity voltage divider to obtain extra ranges. Turning switch  $SW$  to position 2 puts  $C_2$  in parallel with  $C_T$ . If  $C_2$  is 900 mmfd. and  $C_T$  is 100 mmfd., the total shunting capacity is now 1000 mmfd. Assuming the probe-to-cable capacity  $C_1$  is 2 mmfd., the voltage division is now in the ratio of 2 to 1000, so only 1/500 of the actual voltage is applied to the input

tube. This means we've increased the voltage range to 10 times what it was in position 1.

The output of this first stage feeds into a resonant circuit, which in turn feeds the second tube. It is possible to cover a wide frequency range by using plug-in coils or a wave band switch in the resonant circuits.

► This circuit still offers some problems. First, the amplification obtained over a wide frequency band is not constant, but varies according to circuit conditions. Further, the amplification obtained from a tuned circuit varies with the age of the parts, because moisture absorption and physical changes upset the inductance, capacity

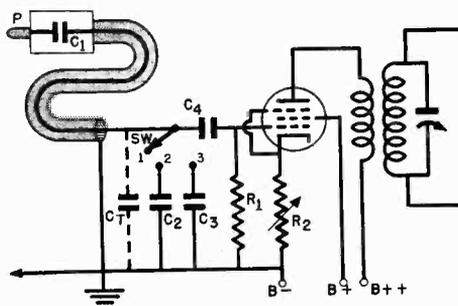


FIG. 25. The input circuit of a tuned circuit v.t.v.m. This is the basic circuit used in signal tracers.

and Q factor values. Finally, the amplification of a tube itself varies with age and with supply voltage changes.

You may wonder just what amplification has to do with the v.t.v.m. reading. The answer is—our v.t.v.m. is calibrated with the amplifier in place, and a change in amplification changes the calibration. For example, suppose our amplifier gives us an amplification of 1000. We apply a certain signal voltage and get, say, a full-scale deflection. Then the amplification changes to 800 for some reason. Now if we apply the same signal voltage, we get—not a full-scale deflection

—but only .8 of full scale. If our scale is calibrated on the basis of a 1000 amplification, obviously our reading is going to be wrong when the amplification changes to 800.

Since extreme accuracy is not needed in service work, however, many instruments of this type, known as signal tracers, have been developed and manufactured. They indicate voltage with reasonable accuracy and are very good for comparative purposes.

For example, if the voltage at the input of a radio stage is measured and found to be 2 volts and then the voltage at the output is found to be 10 volts, the gain of this particular stage would be  $10 \div 2$  or 5. If there is any error in the reading of the voltage by the signal tracer, the same error will be in both these readings, and the errors will cancel. Thus, the actual gain of a radio stage can be determined with a tuned v.t.v.m., even if neither of its voltage measurements is accurate. Incidentally, for greatest accuracy in making comparative measurements of this sort, always make them with the switch SW set to the same range for both measurements if possible.

## CALIBRATING VACUUM TUBE VOLTMETERS

A vacuum tube voltmeter must be recalibrated at frequent intervals, because of tube and voltage changes. Commercial instruments are provided with adjustments for this purpose.

In general, you calibrate a v.t.v.m. by applying a known voltage input, then adjust the instrument until it reads correctly. For d.c. vacuum tube voltmeters, ordinary dry cells are usually used as the known voltage source. The battery voltage should be measured with an accurate d.c. voltmeter while calibrating the v.t.v.m., however, as brand-new batteries may

be as much as 10% higher than their rated voltage. Naturally, any error in the calibrating voltage will cause a similar error in the v.t.v.m.

► When an a.c. vacuum tube volt-meter can be calibrated on 60 cycles, the circuits shown in Fig. 26 can be used. The accuracy depends on the standard meter—if  $V_M$  is a copper-oxide type the accuracy may only be about 5% unless the meter has been hand-calibrated or checked against a standard.

Figure 26A shows a simple circuit which can be used if the test voltage is near the full-scale reading of  $V_M$ . (As you will recall, meters are more accurate at full scale.) If the test voltage applied to the v.t.v.m. must be considerably lower than full scale of the meter, you should use the circuit of Fig. 26B. Here, rheostat  $R$  is adjusted to give  $V_M$  a nearly full-scale deflection. Then the desired test voltage is taken from voltage divider  $R_1$ - $R_2$ .

The accuracy with which the test voltage is known in Fig. 26B depends on the tolerances of  $R_1$  and  $R_2$ , as well as on the meter  $V_M$ . Precision resistors, within ½% to 1% of rated values should be used.

If the meter cannot be calibrated on 60 cycles, a frequency within the range of the device must be used. A signal generator, with an amplified output, might be used as the source. The meter will have to be a thermocouple type, since that is the only

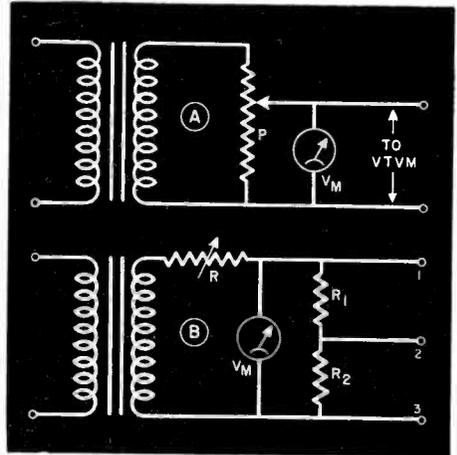


FIG. 26. Two methods of getting an a.c. voltage for calibration purposes.

standard meter having any accuracy for r.f. measurements. A circuit like Fig. 26B would be used, as the thermocouple cannot be used on very small voltage values. The resistors will have to be non-inductive (have no frequency errors) and the loading of the v.t.v.m. must be small.

Where a commercial instrument is being calibrated, the manufacturer's instructions should be carefully followed. They will usually tell you just how calibrations can be made with simple apparatus.

► Commercial instruments usually have an accuracy of from 1% to 5%. Should it be impossible to reach calibration, try a new tube in the v.t.v.m. You may have to try several tubes before finding one which will permit calibration.

## The Cathode Ray Oscilloscope

Very often in radio work it would be extremely helpful if you could actually see the form of the a.c. waves you're working with. You seldom have a pure sine wave of current or voltage, you

know—the radio circuits always distort them somewhat (and sometimes a great deal). Figures 17 and 18 show some of the wave forms you might meet in a radio or television set.

Now, a.c. meters are almost always calibrated on the assumption that they are going to measure pure sine waves. If they're used to measure distorted waves, their readings will usually be incorrect. Often this doesn't make much difference—but sometimes it's very important to know just what kind of a wave form you're measuring.

If you go into radio design, you'll find it necessary to draw graphs of resonance curves, frequency response, phase shift, and many other circuit characteristics. Such curves can be drawn by making many measurements and doing a lot of laborious work—but it would certainly be a lot easier if there were an instrument that drew them for you automatically!

There is such an instrument. It will let you see the exact shape of the wave form in radio circuits. It will draw the curves mentioned above for you—and even show you changes in them as they occur.

This remarkable instrument is called the "cathode ray oscilloscope." Every radio manufacturer—every radio laboratory—considers it indispensable equipment. More and more servicemen are finding it a valuable addition to their service bench.

In the rest of this lesson, you'll learn how the cathode ray oscilloscope works.

## FLUORESCENT SCREEN

Certain chemicals, when bombarded by electrons, will give off a glow of visible light. This phenomenon is known as "fluorescence." It is the principle on which fluorescent lamps work.

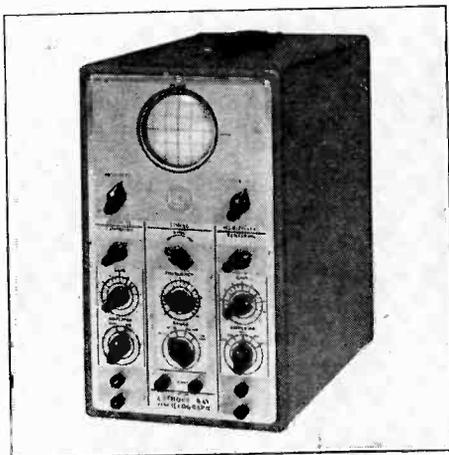
Now if we coat a transparent screen with these chemicals, and allow a thin beam of electrons to hit the screen, we will produce a dot of light on the screen. Further, if we can make the electron beam move over the screen in response to applied voltages, we can

make this dot of light trace out the wave form of the voltages. That's what a cathode ray oscilloscope does.

## THE ELECTRON GUN

As you know, we can easily get a good supply of electrons by heating a cathode. If we put a positive plate with a hole in it near the cathode, the electrons will move toward the plate. Some will hit the plate and others will go through the hole.

We can use this fact to produce the thin beam of electrons—or "electronic



Courtesy R. C. A.

A typical c.r.o. designed for radio servicing, communications and design laboratory work.

pencil"—that we want to move over our fluorescent screen. Figure 27 shows how.

In Fig. 27A, a filament  $F$  with an oxide-coated tip emits electrons from a concentrated spot. (A special heated cathode is sometimes used in place of this filament.) These electrons are attracted toward a highly positive cylindrical plate  $P_1$ . The filament is surrounded by another cylindrical electrode  $G$ . This electrode  $G$  is negative, and serves to concentrate the electrons and force them towards  $P_1$ . It acts like the grid in ordinary tubes.

Since  $P_1$  is highly positive, the electrons travel toward it at great speed and tend to go right down the length of the cylinder. Inside  $P_1$  is a disc with a hole in it. Some electrons strike the disc and are collected by the plate, but most of them are concentrated in the middle of the cylinder and pass through the hole in the disc.

We now have a thin beam of electrons coming through the hole in the disc—the electronic pencil we've been looking for. But as this beam goes away from the disc, it tends to spread out. We need something to keep it together and focus it on the fluorescent screen.

► The optical equivalent of our electronic pencil is shown in Fig. 27*B*. The lamp  $L$  acts as a source of light. This light is collected by reflector  $R$  and is concentrated in a single direction. The light rays would spread out as shown by dotted lines  $X$  and  $Y$  if it were not for the lens placed in their path. The lens serves to focus the rays to a point on the screen  $S$ .

► Returning to Fig. 27*A*, we find that cylindrical plates  $P_1$  and  $P_2$  act as an "electronic lens" to focus our electron beam. Plate  $P_2$  is even more positive than  $P_1$ , and is carefully placed with respect to  $P_1$ . The electrons coming through the disc hole are speeded up by the highly positive potential of  $P_2$ . Furthermore, an electric field exists between  $P_2$  and  $P_1$ .<sup>\*</sup> This electric field collects and focuses the electron beam on the screen.

By choosing the proper  $P_1$  and  $P_2$  voltages, the electron beam is focused to a spot on the screen  $S$ . By varying the voltage on  $G$ , the number of electrons in the beam can be varied. In

<sup>\*</sup>An electric field exists between any two objects having a voltage difference. This field is made up of electrostatic lines, which may be thought of as being similar to magnetic lines of force. Actually, magnetic fields are frequently used in cathode ray tubes for focusing.

this way we can adjust the intensity of the light spot. This system is often called an "electron gun," because it "shoots" electrons at the screen.

The color of the light spot depends on the chemicals used in the screen. White, yellow, green or blue light spots may be produced by using different chemicals. At present, green is the most popular color for direct observation, and blue for photographing the image.

## OBTAINING A DEFLECTION

Now that we have a spot of light produced by a pencil of electrons, we want to move the spot about on the screen and thus trace out wave forms.

This pencil of electrons consists of moving negative charges. If we place other electrical charges near this stream of electrons, we can divert its

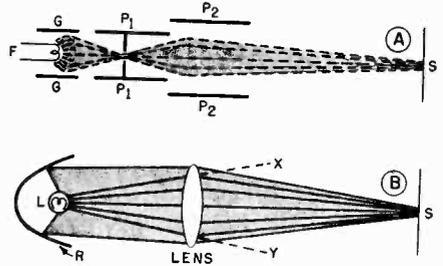


FIG. 27. Comparing an electronic focusing method with a light system.

direction of flow, so we place a flat plate on each side of the electron beam between plate  $P_2$  and the screen, as shown in Fig. 28. We don't want a difference in potential between these plates and  $P_2$ , as this would make the electron beam spread out, so we connect plate  $B$  to  $P_2$ .

As long as the flat plates are at the same potential with respect to each other, the electrons will travel onward in their normal direction, toward point  $I$  on screen  $S$ .

However, suppose we make plate  $A$  positive with respect to plate  $B$ . The

electrons in the beam will then be attracted toward plate *A*, and the beam will be deflected. The electrons do not go directly to plate *A*—they are traveling too fast. Instead, their direction is changed so that they strike the screen at point *2* rather than at point *1*.

The amount the electron beam is bent from its normal direction depends upon the voltage applied between plates *A* and *B*. Of course, if the polar-

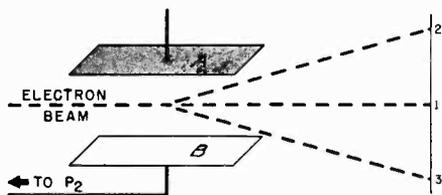


FIG. 28. How the electron beam is deflected by charged plates.

ity is reversed so that plate *A* is made negative with respect to *B*, the electrons will move in the other direction, toward point *3* on the screen.

Thus, placing these flat plates on each side of the electron beam gives us a means of making the electron stream move from point to point on the screen. The movement occurs as soon as the voltage is applied, so it will follow exactly any changes in voltage which may be impressed on the deflecting plates.

► Two sets of these plates are used—one set to move the beam up and down and the other set to move it from left to right, as shown in Fig. 29.

Imagine that the electrons are coming out of the paper toward you. If there is no voltage on the deflecting plates, the electrons will make a spot in the center of the screen, as shown in Fig. 29*A*.

A voltage difference between plates *c* and *d* will cause a vertical (up and down) deflection, so these plates are called the *vertical deflecting plates*. If the voltage changes rapidly enough,

a vertical line will be formed, as shown in Fig. 29*B*.

Plates *a* and *b* cause a horizontal deflection, so are called the *horizontal deflecting plates*. An a.c. voltage between these plates will form a horizontal line, as in Fig. 29*C*.

Any a.c. above 20 cycles per second will produce these lines of light, because of the persistence of human vision. The eye continues to see light for a fraction of a second after it has been cut off, so the rapidly moving spot “blends” into a line.

Now that we can move the beam about on the screen, how do we get a complete picture of the wave form?

### SWEEP VOLTAGE

Suppose we apply a sine wave voltage to plates *c-d*, with no voltage on plates *a-b*. The spot of light starts at

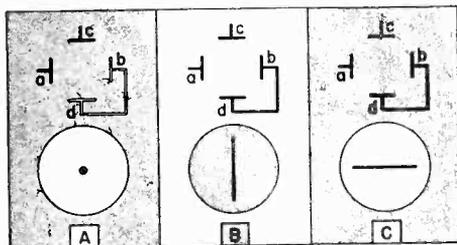


FIG. 29. By applying an a.c. voltage to one set of plates at a time, we change from a spot of light to either a vertical or horizontal line of light.

the no-voltage position. (Fig. 29*A*), goes up until the voltage reaches a peak, goes down until the voltage reaches a negative peak, and repeats this motion as long as the voltage is applied. This gives us the vertical line shown in Fig. 29*B*, but it doesn't show us what the wave form looks like.

What we want is some means of making the spot move at a regular speed sideways at the same time it is moving up and down. Then the beam can trace out the wave form. Now we

know how to make the spot move sideways—we apply a voltage to plates *a-b*.

You might think of applying the sine voltage to both sets of plates at the same time. However, this would give us a tilted line or a series of circular patterns instead of a picture of the wave we are interested in.

We want a voltage which will move the spot sideways until one cycle has been traced out, then sweep the spot back to the starting point to begin the process all over again. In other words, we want a voltage that increases regularly, then drops to zero when the correct point is reached.

Figure 30 shows just such a voltage. From *a* to *b* it increases regularly, then snaps down to *c* at once, and repeats the cycle from *c* to *d* to *e*. Because of its shape, this voltage is called a “saw-tooth” wave. We call such a voltage a “sweep voltage” when we apply it to a

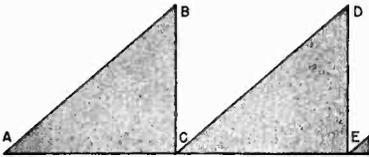


FIG. 30. The sweep voltage must increase steadily and linearly to a maximum, then drop back quickly to the starting level.

cathode ray oscilloscope, because it sweeps the spot back and forth.

**Sweep Generator.** We can get this special wave from a circuit like that in Fig. 31. You recall that a condenser-resistor combination has a time constant—it takes time for the condenser to reach a full charge when a voltage is applied through a resistor. The condenser voltage builds up gradually from zero, somewhat like the curve *a-b-c* of Fig. 31*B*.

Battery  $B_1$  charges condenser  $C$  through the resistance  $R$  in Fig. 31*A*. If it were not for tube  $VT$ , the con-

denser would charge up to the battery voltage and nothing else would happen. However, tube  $VT$  is a special gas tube (like a thyratron) in which no plate current will flow until a certain critical plate voltage is applied. Then the tube suddenly begins conducting.

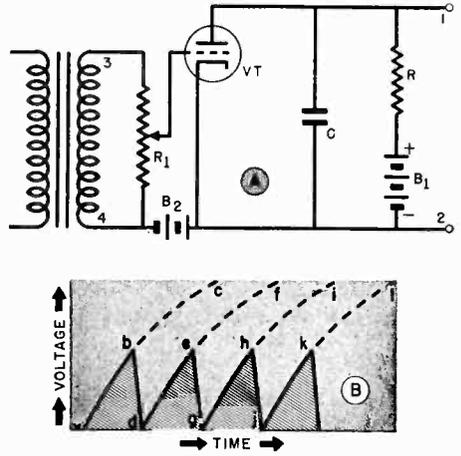


FIG. 31. How a sweep generator produces the desired sweep voltage.

Because of the gas, it becomes practically a short circuit.

As the condenser starts charging through  $R$ , its voltage follows the path *a-b-c*. When *b* is reached, the condenser voltage is sufficient to make  $VT$  conduct. Condenser  $C$  then discharges immediately through the tube, so the condenser voltage drops to zero, or from *b* to *d*. This means there is no longer any plate voltage on  $VT$ , so the tube stops conducting.

Battery  $B_1$  again tries to charge the condenser through  $R$ , along curve *d-e-f*. At *e*,  $VT$  again breaks down and the cycle is repeated. Thus the actual voltage across the condenser follows the *a-b-d-e-g-h-j-k* curve and has approximately the desired shape.

We want the *a-b*, *d-e*, *g-h* and *j-k* portions of the condenser voltage to be as straight as possible. This means the voltage source must be high enough

to make the tube *VT* act quickly, which will keep these points on the more nearly straight part of the condenser charging curve.

The frequency of this wave can be varied by changing the resistance of *R* or the capacity of *C*. This changes the time constant, and hence the frequency with which *VT* breaks down.

We can take this saw-tooth voltage off the condenser from terminals 1 and 2. A blocking condenser is usually used, feeding into a resistor, so the following circuits will not affect the shape of the sweep voltage. As you've learned, feeding a pulsating d.c. (which our sawtooth wave is) through a blocking condenser gives an a.c. input to the next circuit.

**Using the Sweep Voltage.** When we apply only the sweep voltage to the horizontal deflecting plates, the line in Fig. 29C is produced. The spot moves at a steady rate from left to right, then snaps back to the left to repeat the trace.

Suppose we now apply a sine wave voltage to the vertical plates, starting this wave and the sweep at the same time. Figure 32 shows what happens. The voltage wave shown at *A* is the sweep voltage, while *B* represents the voltage applied to the vertical deflecting plates. The combination of these two voltages produces the wave *C* on the screen.

The spot will normally be in the center of the screen, about at point *e*. As the sweep voltage now has an a.c. form because of the blocking condenser through which it was fed, points 1 to 9 of *A* represent negative potentials on the right-hand horizontal deflecting plate which corresponds to plate *b* of Fig. 29. Points 9 to 17 represent positive voltages on this plate (with respect to plate *a*).

Therefore, when the sweep starts at point 1, plate *b* is negative, so the spot

moves toward plate *a* or to the extreme left. This is point *a* in Fig. 32C. The effect of the changing sweep voltage is to make the right-hand plate less negative, then positive, so the spot is moved to the right.

When the sweep voltage is at position 1, the vertical voltage is at position 2 (no voltage). The spot is therefore far over to the left at position *a*. When the sweep voltage increases to 3, the vertical voltage increases to 4. Now the sweep voltage moves the spot horizontally, and the vertical voltage moves it vertically at the same time.

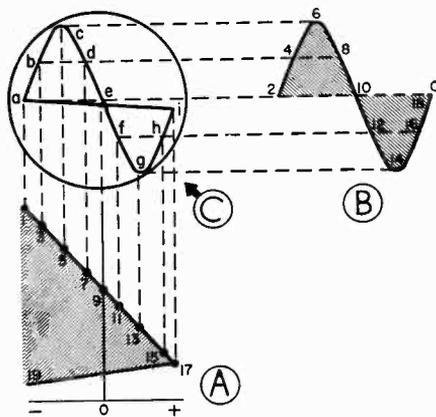


FIG. 32. How applying the sweep and a sine wave causes the sine wave to be reproduced.

The combined effect of the two voltages moves the spot to position *b*, (where lines drawn from the two voltages intersect). Similarly, voltages at 5 and 6 move the spot to *c*, and so on. When the sweep voltage reaches 17, the condenser discharges, and the sweep voltage drops swiftly to point 19, then starts to build up again and repeats the whole process.

These voltages move the spot so quickly that we see a line, instead of moving spots, on the screen. The line from *i* to *a* on the screen is produced by the swift drop in sweep voltage from 17 to 19.

Thus, the sweep voltage has changed the screen trace of the voltage on the vertical plates from a vertical line to a spread-out "picture" which is exactly like the applied wave. Regardless of the shape or distortion of the wave applied to the vertical plates, it will be reproduced exactly.

## SYNCHRONIZATION

We started the sweep at the same time as the sine wave and had it finish at the same time. Hence, both are of the same frequency.

These frequencies must be "locked" together to prevent the cathode ray pattern from drifting about. We do this by feeding some of the input signal to the thyratron tube grid at points 3 and 4 of Fig. 31A. The sweep generator is first adjusted to a frequency just

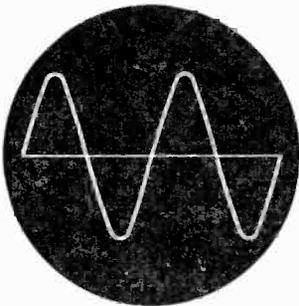


FIG. 33. When the sweep frequency is half the vertical deflecting frequency, two cycles will be produced.

slightly *below* the right frequency. Then, the signals applied to the thyratron swing the grid positive, making the tube conductive and discharging the condenser at just the right moment to lock the sweep in step with the incoming wave. In other words, this forces the sweep to "jump" to the right frequency. This is called synchronization. This control will not work if the sweep frequency is too far from the right value. It is necessary to

adjust the sweep frequency *below* the right value so the sweep won't ever discharge by itself and thus get out of step with the synchronizing grid voltage.

It is also possible to make the sweep lock at a frequency which is an exact fraction of the applied voltage frequency. This gives two or more cycles on the screen, as shown in Fig. 33. You will learn more about this in another lesson.

## FREQUENCY RANGE

Very high frequencies can be applied directly to the deflection plates of a properly designed c.r.o. However, about 30 volts are needed to give a deflection of about 1 inch on the screen. This means that voltages of 75 to 100 volts or so may be required for a reasonably large figure on the screen—far higher than are usually available. For this reason, an amplifier for the input to the vertical plates is built into the c.r.o.

This is usually a resistance-coupled amplifier of from 1 to 3 stages, designed to have a wide frequency response. Even so, the amplifier limits the frequency range. Oscilloscopes made for radio service work are mostly used at audio frequencies, and so have a range up to about 100 kc. Special amplifiers are used in laboratories and in television work to gain a wide frequency range.

A volume control is used on the amplifier so that the deflection can be adjusted within the limits of the screen.

## A TYPICAL C.R.O.

Figure 34 shows a block diagram of a typical oscilloscope designed for service work. Let's go through the sections starting with the c.r.o. tube.

**The C.R.O. Tube.** The tube is

shown here as you would find it on a schematic diagram. Starting from the filament, we have a cathode, the element  $G$ , the plate  $P_1$  (shown here with a grid symbol), plate  $P_2$ , the vertical deflecting plates and the horizontal deflecting plates.

The cathode connects to the junction of  $R_3$  and  $R_4$ . The grid element  $G$  connects to the slider of  $R_3$ , which provides a variable negative bias used to control the intensity of the spot on the screen.

Plate  $P_1$  connects to the slider of

of the earth.

**Vertical Amplifier.** The voltage being checked is fed in at the terminals  $V$  at the left. Switch  $SW_1$  can be set to feed this signal directly to the c.r.o. through  $C_1$ , or into the vertical amplifier. This amplifier is designed for frequencies up to 100 kc. in most service instruments. It has a volume or gain control.

**Horizontal Amplifier.** The signal fed to the horizontal deflecting plates may be an external signal or may come from the sweep generator. The switches

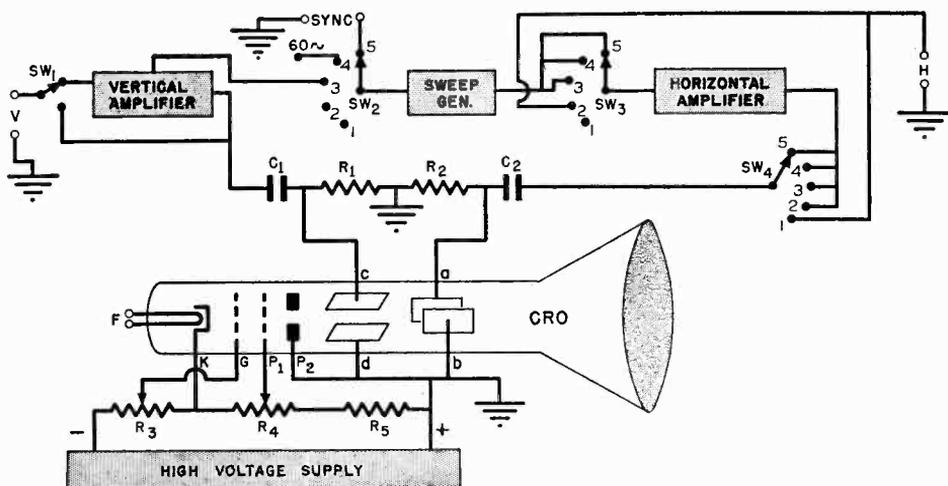


FIG. 34. A block diagram of the "innards" of a typical c.r.o.

$R_4$ , which provides a variable positive voltage, used to focus the spot accurately on the screen. Plate  $P_2$  connects to the positive supply terminal, so that the maximum voltage difference exists between  $P_2$  and the cathode. The power supply furnishes 1000 volts or more for the tube.

One plate of each pair of deflecting plates is connected to  $P_2$ . The application of a signal requires a ground here, which means the positive power supply terminal is grounded instead of the negative as in radio sets. The position of the ground does not affect the power supply in any way, as it is independent

$SW_2$ ,  $SW_3$  and  $SW_4$  are ganged together, on the same shaft, and control the signal paths to the horizontal plates.

► In position 1, an external signal is fed in at the  $H$  terminals at the right, and passes through position 1 of switch  $SW_4$ , then through  $C_2$  to the c.r.o. tube.

► In position 2, the external signal from terminals  $H$  comes through  $SW_3$  into the horizontal amplifier, and from there through  $SW_4$  to the c.r.o. tube. This provides amplification, controllable by the gain control on this amplifier.

► Positions 3, 4 and 5 feed the sweep

generator signal through  $SW_3$  to the horizontal amplifier and from there through  $SW_4$  to the c.r.o. tube. These three positions provide varying sources of synchronizing voltage for the sweep generator.

► In position 3, part of the signal from the vertical amplifier is fed through  $SW_2$  to the sweep input. This is the most used position of these switches, as this provides a "lock" with the incoming signal for the vertical plates.

► Position 4 feeds 60 cycle a.c. into the sweep generator, permitting a lock with the power line frequency.

► Position 5 connects the sweep to external terminals  $SYNC$ , so that an external frequency source can be used to control the sweep.

**Centering Controls.** It is impossible to locate the tube elements in every tube so as to get the spot in the exact center of the screen when no deflecting voltages are applied. However, this can be corrected electrically as shown in Fig. 35.

Instead of connecting  $P_2$  and deflecting plates  $b$  and  $d$  to the maximum positive point as in Fig. 34, this connection is now made to the junction of added resistors  $R_6$  and  $R_7$ .

Resistors  $R_1$  and  $R_2$  are now grounded for a.c. through condensers  $C_3$  and  $C_4$  and their d.c. paths return to potentiometers  $R_8$  and  $R_9$ .

Resistors  $R_6$  and  $R_7$  are now part of the voltage divider, and the same voltage exists across  $R_8$  and  $R_9$ , which are in parallel. When the sliders on  $R_8$  and  $R_9$  are centered, the arms are at the same potential as the  $R_6$ - $R_7$  junction. There is now no difference in po-

tential between the pairs of deflecting plates.

Adjusting  $R_8$  will make plate  $c$  either positive or negative with respect to  $d$ , depending on the direction of movement from the center of the control.

Similarly  $R_9$  will bias plate  $a$  with

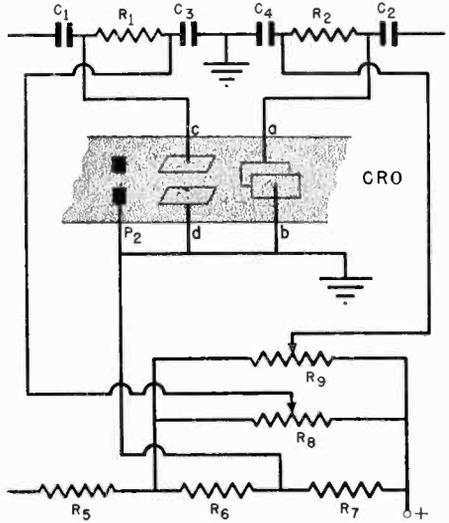


FIG. 35. Centering controls correct for an off-center spot of light.

respect to  $b$ . Thus it is possible to move the spot up or down or to the right or left, until it is exactly centered. These controls are found on commercial instruments, and are called the centering controls.

After the beam is centered and focused, and the intensity is adjusted to reasonable brilliancy, we can apply the external signals and the sweep voltage (if used) through  $C_1$  and  $C_2$  to the c.r.o. tube.

# Lesson Questions

Be sure to number your Answer Sheet 29FR-2.

Place your Student Number on every Answer Sheet.

Send in your set of answers for this lesson immediately after you finish them, as instructed in the Study Schedule. This will give you the greatest possible benefit from our speedy personal grading service.

1. What ohms-per-volt sensitivity is commonly obtained on the basic range of d.c. type vacuum tube voltmeters?

25,000

2. What is the purpose of  $R_3$  and  $C$  in Fig. 3?

acts as an a.c. filter and improves accuracy

3. What is the purpose of  $R_2$  in Fig. 3?

del. of bias

4. What is the purpose of  $R_7$  in Fig. 10?

del. stability

5. Suppose an a.c. type v.t.v.m. is connected across a tuned circuit. How can you compensate for the capacity loading which detunes the circuit?

Remove the tuned circuit from the circuit

6. Why do we get an error when we try to measure a distorted wave with an a.c. type v.t.v.m.?

change to rms, average d.c. level

7. What determines the accuracy of a slide-back v.t.v.m. when properly adjusted?

current source IM

8. What are the advantages of using tuned circuits in signal tracers?

eliminates unwanted signals, greater sensitivity

9. What is the purpose of the sawtooth voltage in a c.r.o.?

10. When synchronizing the sweep of a c.r.o. so as to make the pattern "lock in," is the sweep frequency adjusted above, below, or the same as the synchronizing signal frequency?



## “WISHERS” AND “DOERS”

How often have you said, “I wish I had more money?” Thousands of times, possibly. But do you realize that if you are living in a town of, let us say, 5000 inhabitants, there are exactly 4999 others in your town who are saying exactly the same thing?

And yet, of these 5000 “wishers,” only about 100 are going to do something about it. The others are going to continue being “wishers.”

Now, any man who shows enough “get-up-and-go” spirit to undertake this Course proves that he is not a mere “wisher.” When you enrolled, you showed that you wanted to be a “go-getter.” Your job now is to keep going forward on the road you have mapped out for yourself.

Every lesson in this Course, every radio job you work hard to get, is a step along this road. So don't let yourself wish that the lessons were easier, or that you could become successful without studying, or that radio jobs would come looking for you. Stay out of the class of the “wisher,” and stay in the class of the “doer.”

*J. E. Smith*

# MEASUREMENTS AT AUDIO AND RADIO FREQUENCIES

30FR-1



**NATIONAL RADIO INSTITUTE**

**WASHINGTON, D. C.**

**ESTABLISHED 1914**

# STUDY SCHEDULE NO. 30

For each study step, read the assigned pages first at your usual speed. Reread slowly one or more times, finish with one quick reading to fix the important facts firmly in your mind, then answer the Lesson Questions for that step. Study each other step in this same way.

1. Introduction . . . . . Pages 1-2  
Read this once carefully to get an idea of the measurements covered in this lesson, as well as the required accuracies in various radio fields.
2. Inductance of Air-Core Coils . . . . . Pages 2-7  
The simple methods discussed here can be carried out with a minimum of equipment. These methods require more work and are less accurate than some of the other methods but are useful where extreme accuracy is not required. Answer Lesson Question 1.
3. Inductance of Iron-Core Coils . . . . . Pages 7-8  
The method is similar to that used for air-core coils, with the addition of a direct current flow through the coil to provide the proper core flux density. This permits the measurement to be made under conditions simulating actual operation. Answer Lesson Questions 2 and 3.
4. Condenser Capacity . . . . . Pages 8-11  
The basic methods developed for coils are used to determine the capacity of solid-dielectric and electrolytic condensers, as explained here. Answer Lesson Question 4.
5. Resonance Methods of Measuring L and C . . . . . Pages 11-15  
By setting up a resonant circuit, the inductance and capacity values can be accurately found. Also, the coil r. f. resistance and distributed capacity can be measured and the coil Q calculated. This method is very valuable where r. f. circuit components are being checked. Answer Lesson Questions 5, 6 and 7.
6. Bridge Measurements . . . . . Pages 16-23  
The most commonly used method of measuring inductance and capacity makes use of bridge circuits. These circuits have the advantage of covering wide ranges, using only a few standard parts, and have a high degree of accuracy. Answer Lesson Questions 8 and 9.
7. Frequency Measurements . . . . . Pages 24-36  
There are many ways of checking frequencies, such as by comparison with a frequency standard, using a wave meter, tuning fork or reed, etc. You will find this section very interesting—giving an insight into the way service equipment is calibrated. Answer Lesson Question 10.
8. Mail Your Answers for this Lesson to N. R. I. for Grading.
9. Start Studying the Next Lesson.

# MEASUREMENTS AT AUDIO AND RADIO FREQUENCIES

**I**N past lessons you have studied inductance, capacity, impedance, reactance, power factor and the Q factor possessed by coils and condensers at audio and radio frequencies. Now we are going to learn practical ways of actually measuring these properties.

Such measurements are made by using a circuit which allows the part in question to be compared directly or indirectly with a standard part. The National Bureau of Standards, in Washington, D. C., has a set of resistors, a set of coils and a set of condensers whose resistance, inductance and capacity values are accurately known. This government agency and other large laboratories calibrate radio parts for manufacturers and others engaged in research work. These calibrated parts, in turn, are used as standards in rating ordinary radio parts.

Usually some ratio circuit is used in making comparisons, so that fewer standard parts—which are very expensive—are needed. If the ratio test circuit is put together on the workbench, some simple calculation is required, using either Ohm's Law or simple multiplication or division. However, factory-made, precalibrated capacity bridges, Q factor meters, inductance bridges and other measuring apparatus are available with direct-reading dials which can be read as easily as ordinary meter scales.

► The method used to make the measurements depends on the accuracy desired and on the apparatus available. If coils had only induc-

tance, condensers only capacity and resistors only resistance, measurement would be vastly simplified. Instead, coils have distributed capacities and both a.c. and d.c. resistances; condensers have resistances and inductances; while resistors may have inductances which, though small, are quite effective at higher frequencies. Measurements must either take all these factors into account or be somewhat in error.

► The accuracy with which you will wish to make measurements depends on the radio field you enter. If you become a design or research engineer, you must take into account all the properties of each radio part in your circuits. You will need to calculate carefully the values of parts to be used, then insert parts (whose values have been accurately determined) in a trial circuit to prove the design works. You will then vary parts values to find the tolerances permissible. The laboratory measuring procedures you will use require careful control and very expensive equipment.

If you enter production work in a radio factory as a laboratory technician, factory serviceman or inspector, you will need to make only semi-laboratory measurements. These do not require such complex procedures nor such expensive equipment, for you will usually make comparison measurements to see if parts are alike or within tolerance limits, rather than find absolute values directly. In other words, you will compare one coil with another similar one to find their inductance difference instead of to find

the actual inductance value accurately.

If you make servicing or radio communications your career, you will rarely make even these simplified measurements. Usually the radio trouble will be some simple breakdown which you can find from voltage, current or resistance measurements, and you can determine replacement parts values from the wiring diagram or previous experience with similar models. Even so, knowledge of ways of measuring coils and condensers, using ordinary shop equipment, will help you in special cases of unusual trouble and also in understanding more clearly the relations between these units.

► In this lesson we will discuss both simple methods and basic laboratory techniques. To do so, we here present a number of methods. Some use formulas, and in some cases we show how

the formulas are obtained. We do not intend that you try to memorize the methods, the formulas, or the process of getting the formulas—they are presented here for your future reference and you can come back to this lesson should you need this information. We cover the field here, so you will have some idea of the measurements required in all branches of radio and will not have to start in absolutely green should you go into any field of radio requiring this information. You will probably never use all the methods described, no matter where you go, so just learn the basic principles of the methods and answer the lesson questions.

► Let us now see how measurements are made. We'll start with air-core inductances, then take up iron-core coils, condensers, resonance methods and bridge methods.

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## Inductance of Air-Core Coils

Inductance values of air-core coils are rarely given on service diagrams. However, just any coil cannot be used as a replacement, particularly for an r.f. transformer, where the inductance value may determine the tuning range, and the Q factor may affect both the sensitivity and selectivity. The practical serviceman would therefore replace a defective air-core coil with an exact duplicate ordered from the receiver manufacturer or a radio supply house. If a duplicate were unobtainable, he would send the original coil to a coil manufacturer specializing in winding coils for service purposes. The serviceman is thus likely to need to find the inductance of an air-core coil only when building something for experimental purposes (a wave trap, for instance).

To the factory man or engineer, on the other hand, inductance and Q factor are very important and must be found time and time again. Let's see how they can be measured. We'll use basic methods which are suitable for other r.f. measurements as well.

► The inductance of an air-core coil can be found from its physical dimensions—such as length and diameter of the coil, the number of turns and size of wire. You can then use these values with an inductance design chart or formula to determine the inductance within a few percent. These charts or formulas are found in engineering handbooks and radio publications.

However, at the time you need them, charts are not always available; they cannot be used when tolerance limits are being determined, nor do they give

the Q factor. So now let us study actual inductance measurements.

► We can find the inductance of an air-core coil by determining the coil reactance and calculating the inductance, by direct comparison methods, or by using bridge or resonant circuit methods. Since the last two can also be used to measure capacity and resistance, they will be covered in another section of this lesson. Let us now take up the reactance and direct comparison methods, which use equipment found in any laboratory and in many service shops.

### REACTANCE METHODS

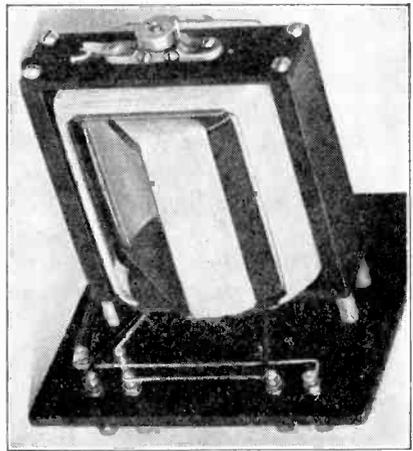
A very simple method which can be used with reasonable accuracy on air-core coils is to apply a known a.c. voltage to the coil and determine the current flow, as in Fig. 1. From Ohm's Law for a.c. circuits, you know that the impedance of the circuit equals the voltage divided by the current flow ( $Z = E \div I$ ). Also, you know that the impedance  $Z$  is a combination of the a.c. resistance and the reactance of the coil. ( $Z = \sqrt{R^2 + X_L^2}$ ). If you neglect the a.c. resistance, the impedance is the same as the reactance of the coil. Once you find the inductive reactance, you can find the inductance from the formula  $X_L = 2\pi fL$ , which is easier to use if arranged in the form  $L = X_L \div 2\pi f^*$ .

To find the inductance of the coil by this method, you must know the frequency and neglect the a.c. resistance. The frequency will be that of the voltage source, which will probably be a signal generator when an air-core coil is being checked. For accuracy, the frequency used to determine the reactance by this method must be

\*  $L$ =inductance in henrys  
 $X_L$ =inductive reactance in ohms  
 $f$ =frequency in cycles  
 $2\pi=6.28$ , a constant

somewhere near the normal working frequency of the coil. Since air-core coils are used in r.f. circuits, radio frequencies must be used. Usually the frequency band in which the coil is designed to work is known, so you can choose some frequency in or near this band. (Remember that if the frequency is too low, the coil will have very little reactance and an extremely high current will flow.)

► This method is quite simple, since you just measure the voltage and current and perform a calculation. How-



*Courtesy General Radio*  
Interior view of a standard variable inductance.

ever, it is only accurate enough for a very rough approximation. Remember, you neglect the a.c. resistance of the coil, which may be appreciable. Also, since you use the frequency in your calculation, any error in the calibration of the signal generator will cause a similar error in your result. Further, you must use accurate a.c. meters, because any inaccuracies in current and voltage measurements will likewise affect the final result. Only a thermocouple type current meter and an r.f. vacuum tube voltmeter are accurate enough for r.f. measurements.

► You must also be careful how you use your meters. In the circuit shown

in Fig. 1A, the current meter  $I_M$  actually measures the current through the coil, but the voltmeter  $V_M$  measures both the voltage drop across the coil and the drop across the current meter. If the current meter has an appreciably high resistance compared to the coil reactance, the measured voltage may be considerably different from the actual voltage across the coil, as it will be the sum of the coil voltage and the voltage drop across the current meter.

You can make the voltage measurement more accurate by placing the voltmeter directly across the coil, as shown in Fig. 1B. Now, however, the current meter measures both the current through the coil and the current through the voltmeter, so the current reading will be accurate only if the voltmeter has a high ohms-per-volt sensitivity.

Which connection is better depends on the meters used. At radio frequen-

tance, while Fig. 1B can be used if the voltmeter has a high ohms-per-volt sensitivity. When a v.t.v.m. is used, Fig. 1B is the better circuit.

► Another practical difficulty with this method is that the signal generator may not have sufficient output. If so, use a single stage amplifier to increase the output sufficiently.

Obviously, with so many chances of error, this method is not extremely accurate. However, it may be good

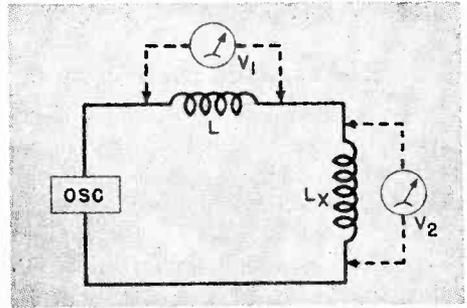


FIG. 2. The comparison method used for an air-core coil.

enough if accuracy within 10% to 20% is permissible.

### COMPARISON METHODS

If a known inductance is available, you can place the unknown inductance in series with the calibrated one, apply an r.f. voltage to the combination (as shown in Fig. 2), and get the voltage readings. The voltage drops are proportional to the impedances. If we ignore the coil a.c. resistance, then the voltages are proportional to the reactances, which in turn are proportional to the inductances, so we can say that

$$\frac{V_1}{V_2} = \frac{L}{L_X} \quad \text{or} \quad L_X = \frac{V_2 L}{V_1}$$

Thus, to find the inductance of  $L_X$ , multiply the inductance of  $L$  by the voltage  $V_2$  and divide by the voltage  $V_1$ . For best re-

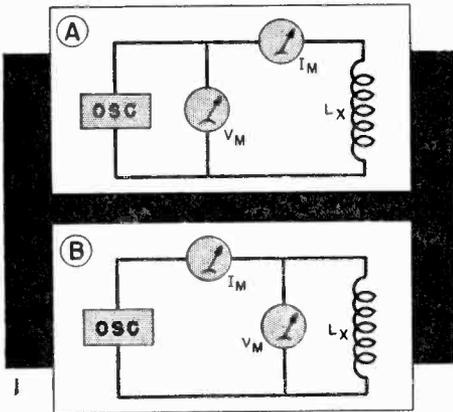


FIG. 1. Two methods of connecting meters to measure voltage and current.

cies, the current meter must be a thermocouple or hot-wire type, as other a.c. meters are usable only at power and audio frequencies. Fig. 1A can be used if the current meter has low re-

sults, the standard should be near the value of the unknown, so the voltmeter readings will be similar.

► If a variable standard inductance is available, use it as  $L$ , then adjust the variable standard inductance until the voltage drops across the two coils are equal. When they are equal, the impedances of the two coils are equal and, neglecting resistance again, you can read the unknown inductance directly from the calibrated scale on the variable standard inductance. This method has many advantages over the reactance method. For example, you do not have to know the frequency of the source—and, with the variable standard, if identical voltmeters are used across each coil or the same one is moved from coil to coil, even the accuracy of the voltmeter does not matter. However, some error may be caused by neglecting the resistances of the two coils, which may be quite different.

**Resistor Comparison.** If a standard inductance is not available and great accuracy is not required, the method shown in Fig. 3 can be used. Here you use a variable resistor as the variable standard and adjust it until the two voltmeters read the same value.

To make the meters read alike, you may have to adjust the frequency of the a.c. source until the coil reactance falls within the resistance range. If the voltmeter reading across the coil is higher than the reading across the resistance, even with maximum resistance, you must either lower the frequency or use a higher resistance.

Again, equal voltages mean the impedances of the two devices are equal. If you neglect inductive and capacitive effects in the resistance and also neglect resistance within the coil, you can say that the resistance of  $R$  is

equal to the reactance of  $L_X$ . Then you can use the reactance formula to determine the inductance. Since the reactance of  $L_X$  is equal to  $R$ , the formula becomes  $L_X = R \div 6.28f$ , where  $L_X$  is in henrys,  $R$  is in ohms and  $f$  is in cycles per second.

► This method has the advantage of not requiring a standard inductance. You do not even need a calibrated resistor, because you can measure the resistance with an ohmmeter after equal voltages are obtained. However, it does require the use of either two identical voltmeters or one meter with a very high ohms-per-volt rating such as a v.t.v.m.

The accuracy is affected by neglecting both the resistance of the coil and the inductance and capacitive effects in the resistance unit, and also depends

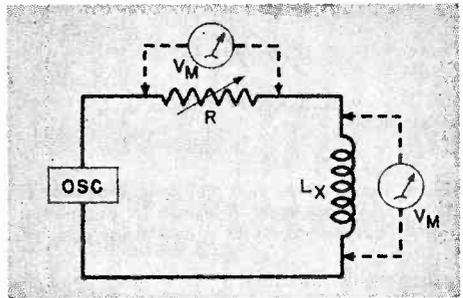


FIG. 3. How to use a variable resistance as the standard in the comparison method.

upon the accuracy with which the resistance and frequency are determined. However, this method eliminates current measurements and does not require accurate meters, so it is rather commonly used when errors of 10% to 20% can be permitted.

**Three-Voltage Method.** By making three voltage measurements and using a graph, you can improve con-

siderably the accuracy of the method of Fig. 3. The measuring circuit is shown in Fig. 4A. Here,  $R$  need not be variable, but its resistance must be known (or measured) accurately. Preferably,  $R$  should have a resistance such that  $E_R$  is somewhere near the value of the voltage  $E_Z$ , but this is not necessary.

After making the three voltage measurements shown in Fig. 4A, you must draw a graph. To do this, first mark a horizontal line on a sheet of

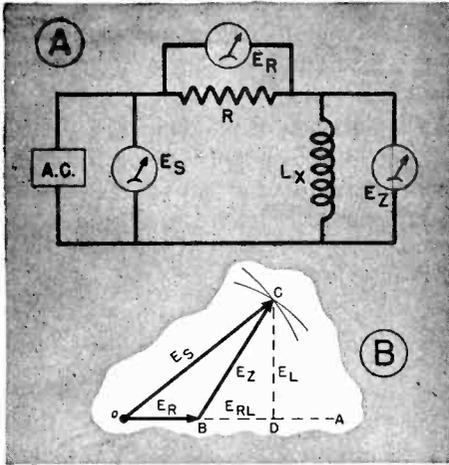


FIG. 4. The three-voltage method. Not only is this method more accurate, it can be used when only a voltmeter is available.

paper, such as the line  $O-A$  in Fig. 4B. Along this line, measure off a distance proportional to the voltage found across resistor  $R$ . Any satisfactory scale can be used, such as 10 volts to the inch. After you have measured the proper length along this line from point  $O$  with a ruler, mark point  $B$ . The distance  $O-B$  is then proportional to the voltage  $E_R$ .

Now, take a compass (a drawing instrument similar to a pair of dividers, with a pencil attachment) and adjust the compass so that the distance between its point and its pencil is proportional to the voltage  $E_S$ , using

the same scale of volts-per-inch used for laying out the line  $O-B$ . With the compass point set at point  $O$ , draw an arc or semicircle with the compass.

Next, adjust the compass so that the distance between its point and pencil is proportional to voltage  $E_Z$ , using the same volts-per-inch scale as before.

With the compass point now set at point  $B$ , draw an arc with the compass that crosses the arc you made from point  $O$ . The point where they cross is point  $C$  in Fig. 4B. Now, draw lines from point  $C$  to point  $B$  and from point  $C$  to point  $O$ . The line  $O-C$  is then proportional to the voltage  $E_S$ , while the line  $B-C$  is proportional to the voltage  $E_Z$ .

Next, drop a vertical line (line  $C-D$ ) from point  $C$  at right angles to line  $O-A$ . You can use the right ( $90^\circ$ ) angle of a triangle to do this.

You have now formed a triangle having as its sides the lines  $B-C$ ,  $B-D$  and  $C-D$ . As line  $B-C$  represents the voltage across the coil impedance, the line  $C-D$  is proportional to the voltage drop across the inductive reactance alone, while the distance  $B-D$  is proportional to the voltage drop across the coil a.c. resistance. The combination of these two voltages, of course, makes up the total voltage drop across the coil, which you have found to be  $E_Z$ .

Now since the voltage  $E_L$  is caused by the reactance alone, dividing this voltage by the current flow will give you the true coil reactance. The voltage  $E_L$  can be determined by measuring the line  $C-D$  with a ruler, then converting the number of inches into volts, using the same volts-per-inch scale you used in drawing the other lines.

The current is, of course, equal to the resistor voltage  $E_R$  divided by  $R$ . Compute this current, divide  $E_L$  by it,

and you have the coil reactance. You can now find the true inductance from the formula  $L = X_L \div 6.28f$ .

► This method eliminates one error by taking the coil a.c. resistance into account. Using identical voltmeters (or the same one), your accuracy depends only on the accuracy with which the resistance is found, the accuracy with which the frequency is known, and the care used in drawing the graph. Make the graph as large as possible (fewer volts-per-inch scale) for greatest accuracy.

► The three-voltage method also lets you determine the Q factor of the coil at the frequency of measurement. You

know the Q factor is equal to the reactance of a coil divided by its a.c. resistance ( $Q = X_L \div R$ ). As the coil resistance and coil reactance are effectively in series, with the same current through each, the reactance and resistance are proportional to their voltage drops. Therefore, you can find the Q factor directly by dividing voltage  $E_L$  by voltage  $E_{RL}$ .

**Other Methods.** You can also measure inductance by using various bridge circuits and resonant circuits. Since these methods can also be used to measure capacity, they will be discussed by themselves later in this lesson.

---

## Inductance of Iron-Core Coils

You cannot determine the inductance of an iron-core coil except by electrical measurements, as its physical size does not indicate its inductance. A serviceman would probably replace a defective coil with a duplicate, but it might be necessary for him to measure the inductance to be sure the coil replacement is proper.

The methods of measuring the inductance of an iron-core device are somewhat similar to those already discussed for air-core coils. However, you must be careful to measure the inductance under the operating conditions which will exist in the radio receiver, transmitter or other apparatus, because the actual amount of inductance depends on the core flux density. If the coil is used in a circuit where d.c. flows through it, you must approximate the same d.c. flow during tests. You must also have about the same a.c. flow as in normal operation, and the frequency used for measurement must be within the normal range of the device.

► If the iron-core coil is used in a circuit where no d.c. normally flows, its inductance can be measured by the methods used for an air-core coil as long as a normal a.c. voltage and frequency are employed. With a transformer, you must remember that you are determining the inductance of only the one coil which is connected in the measuring circuit. The inductance of the other windings must be determined separately. Also, to find the actual operating inductance of any winding, you must connect the normal loads to all the windings of the transformer—even to those not being measured.

### POLARIZING CURRENT

In most radio uses, an iron-core coil will have both d.c. and a.c. flowing through it. As you know, you must send normal amounts of both a.c. and d.c. through such a coil to measure its inductance under working conditions. A circuit for doing so is shown in Fig. 5.

In this circuit, coil  $L_x$  is the induct-

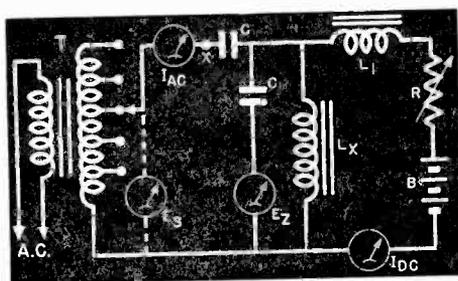


FIG. 5. Iron-core coils require a polarizing current, so this circuit is used.

ance under test. The battery  $B$  causes d.c. to flow through the coil; this current is read on  $I_{DC}$  and adjusted to the required value by resistor  $R$ .

A.C. is applied through transformer  $T$  and condenser  $C$ . Condenser  $C$  prevents d.c. from flowing through the secondary of transformer  $T$ , while choke coil  $L_1$  prevents a.c. flow through the battery and  $R$ . Thus, d.c. flows only from battery  $B$  through  $R$ ,  $L_1$  and  $L_X$ , while a.c. flows only from the transformer through condenser  $C$  and coil  $L_X$ .

Condenser  $C$  should have a large capacity, so its reactance will be negligible. Also,  $L_1$  must be a high-inductance coil so the a.c. flow through  $L_1$ - $R$ - $B$  (in parallel with  $L_X$ ) will be negligible.

After the d.c. has been adjusted to

approximate the amount which normally flows through the coil when in use, a tap on the secondary of transformer  $T$  is chosen which produces about the normal a.c. voltage drop across coil  $L_X$ , as indicated by  $E_Z$ . (This meter must measure a.c. only, so blocking condenser  $C_1$  is placed in series with it.) Then the current meter  $I_{\Delta O}$  is read. Dividing the voltage  $E_Z$  by the alternating current gives the impedance of the coil  $L_X$ . If the a.c. resistance of the coil is assumed negligible, the impedance will be approximately the same as the reactance of the coil, and the inductance can be found from the formula  $L = X_L \div 6.28f$ .

► However, most iron-core coils are low-Q devices, having large windings with appreciable a.c. resistance which cannot be ignored. To eliminate the large error introduced when assuming the impedance equals the resistance, the best way to measure the inductance of such coils is the three-voltage method already described. To use it, first insert a known resistance at point  $X$ , then adjust the d.c. to the proper value and measure: 1, the voltage  $E_S$ ; 2, the voltage across the resistor at  $X$ ; and 3, the voltage  $E_Z$ . You can then use exactly the same procedure followed for Fig. 4.

## Condenser Capacity

We normally deal with three kinds of radio condensers; air-dielectric condensers (usually variable condensers), solid-dielectric condensers, and electrolytics.

The serviceman is interested primarily in the last two types. (Measurement of an air-dielectric condenser is almost never required in servicing, so we will cover this type later.) Next

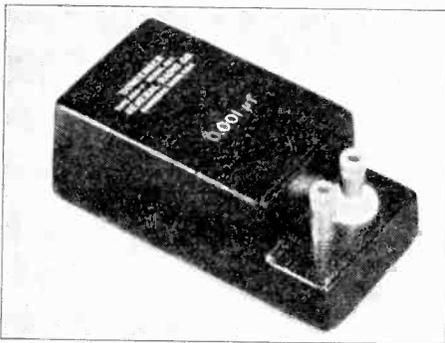
to tubes, solid-dielectric and electrolytic condensers cause more troubles than any other parts. While ohmmeter measurements or part substitution will show up most of these defects, condenser measurements are required frequently enough so that test equipment manufacturers have brought out capacity testers for servicemen. These

will be described shortly, when we take up bridge circuits.

### SOLID-DIELECTRIC CONDENSERS

Let us first see how to measure capacities of ordinary solid-dielectric bypass condensers—paper, mica, ceramic, and similar types. The methods we will now discuss are satisfactory for capacities above .01 mfd.

**Reactance Methods.** Condenser capacity can be measured in almost



*Courtesy General Radio*

A standard fixed condenser, which has been accurately determined and sealed within a protective container.

exactly the same manner as coil inductance, by measuring the applied a.c. voltage and the current flow. It's a wise precaution to determine first whether the condenser is short-circuited, since the alternating current meter can be damaged by excess amounts of current. Also, be sure the applied a.c. voltage peak is well below the peak voltage rating of the condenser.

After measuring the a.c. voltage and current, you can find the impedance by dividing the voltage by the current. Since the a.c. resistance of the normal solid-dielectric condenser is negligible at low frequencies, you can say that the reactance equals the impedance and then use the formula for capaci-

tive reactance,  $X_c = \frac{159,000}{fC}$  (where

$f$  is in cycles per second and  $C$  is in microfarads), to find the capacity. This is usually converted to the han-

dier form  $C = \frac{159,000}{fX_c}$ , where  $X_c$  is

the capacitive reactance in ohms.

As you know, the larger the condenser, the lower the reactance, hence the greater the current flow for a fixed voltage. This means it is possible to make a simple capacity meter with a basic circuit like that in Fig. 6. Using a fixed frequency and a fixed applied voltage, the current meter scale can be calibrated directly in terms of capacity. Shunts can be used to give different current ranges and extend the capacity range. However, the larger the condenser the lower the reactance, and hence, the higher the current. This means the condenser size range is limited by the current range and too large a condenser can cause the meter to be ruined by excess current. A shorted condenser used in this circuit will also ruin the meter. All condensers must first be checked with an ohmmeter to prevent such damage.

► You can also measure condenser capacity with the comparison circuit

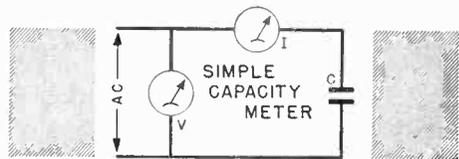


FIG. 6. Circuit for solid-dielectric condensers. They must be in good condition or the current meter can be ruined.

shown in Fig. 7, which requires a vacuum tube voltmeter and a condenser  $C_s$  of known capacity. You compare the voltages across the two condensers. As you know, the capacitive reactance increases as the capacity is decreased.

Since the voltage drop depends on the reactance, this means the larger voltage drop will appear across the smaller condenser. In other words, the voltage across each condenser is inversely proportional to its capacity.

Therefore, you can say that the ratio of the voltage across the known condenser to that across the unknown condenser equals the ratio of the capacity of the unknown to that of the

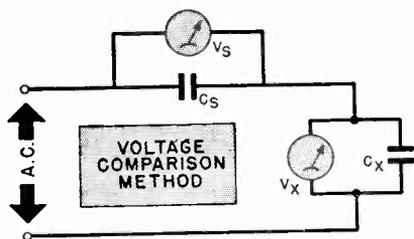


FIG. 7. The comparison method for condensers.

known condenser. Putting this in a

$$\text{formula, } \frac{V_s}{V_x} = \frac{C_x}{C_s} \text{ or } C_x = \frac{C_s V_s}{V_x}.$$

Thus, to find the capacity of the unknown condenser, multiply the capacity of the known condenser by the voltage across it, then divide this product by the voltage across the unknown condenser. The answer is the capacity of the unknown condenser in the same units as the known condenser (usually in microfarads).

If  $C_s$  is adjusted so the two voltages are equal, then the capacities are equal. Thus, if you use a calibrated variable condenser at  $C_s$ , you can make the voltages equal (so  $C_s$  equals  $C_x$ ) and read the value of  $C_x$  directly from the calibration of  $C_s$ .

A vacuum tube voltmeter should be used for making measurements—unless the capacities are quite large and the test frequency is lower than 10 kc., when an ordinary copper-oxide rectifier type meter can be used. If the resistance of the meter is many times

higher than the reactance of the condensers, one meter will suffice. But if the meter has such low resistance that it may affect the results, two identical meters should be used.

## ELECTROLYTIC CONDENSERS

The capacity of an electrolytic condenser should be measured with a d.c. voltage applied, because practically all electrolytics need a d.c. voltage to keep the film formed on the anode of the condenser. If an a.c. voltage of more than 3 or 4 volts is applied by itself, the condenser may be ruined. Also, the condenser power factor must usually be taken into consideration, since in electrolytics the series resistance is relatively high.

Fig. 8 shows a circuit for checking an electrolytic. The three-voltage method is used. As the voltmeters must read a.c. only, they are isolated by blocking condensers.

► First, adjust the d.c. voltage to equal some value higher than the peak of the a.c. to be applied. This value plus the applied a.c. peak value must be below the working voltage rating of the condenser. After waiting a

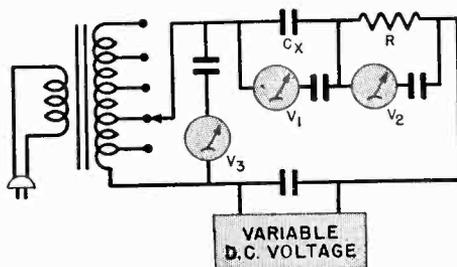


FIG. 8. The polarizing and testing circuit for electrolytics.

reasonable length of time for the d.c. to form the condenser, apply an a.c. voltage from the transformer. (This a.c. voltage must not exceed the d.c. voltage.) Now, measure voltages  $V_1$ ,  $V_2$  and  $V_3$ . Next, construct the triangle shown in Fig. 9 by the same

method used in Fig. 4, using  $V_2$  as the horizontal reference vector. Finally, drop a vertical line from the point  $P$  to form the line  $V_x$ , and extend the line  $V_2$  to form  $V_R$ .

The line  $V_x$  represents the true voltage across the condenser reactance. Measure the length of this line, convert it into volts, and divide the voltage thus found by the actual current

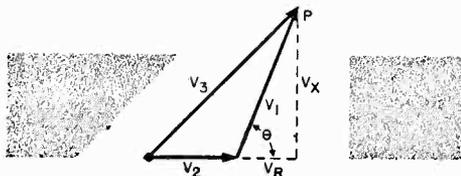


FIG. 9. The three-voltage triangle for electrolytics.

flow in the circuit (which is equal to voltage  $V_2$  divided by  $R$  in ohms). The result is the condenser reactance, which you can readily convert to capacity by using the reactance formula.

**Power Factor.** The power factor is found by dividing the resistance of a part by its impedance. This result is multiplied by 100 to express the power factor as a percentage.

A resistor has a power factor of 1, or 100%, because the resistance is equal to the impedance. On the other hand, a perfect coil or a perfect condenser should have zero resistance and

thus zero power factor. Therefore, a coil or condenser with high power factor has considerable series resistance, and so will not act as much like a perfect inductance or perfect capacitance as will a similar device with a lower power factor.

After having made the triangles shown in Fig. 9, you can determine the power factor of the condenser by measuring the line  $V_R$  (which represents the voltage drop across the condenser resistance), then dividing this amount by the voltage  $V_1$ . Multiply by 100 to give the power factor as a percentage. The smaller the power factor, of course, the better the condenser. Measured on 60-cycle a.c., a good electrolytic should have a power factor of about 6%. If the percentage is above 25% or 30%, you should replace the condenser.

► Another point of interest is the  $Q$  of the condenser. The larger the  $Q$  factor, the better the condenser. This factor may be obtained by dividing the value of  $V_x$  by the value of  $V_R$ . If angle  $\theta$  is close to  $90^\circ$ , the  $Q$  factor is approximately equal to one, divided by the power factor.

► The circuit in Fig. 8 can be used also to measure the capacity of solid-dielectric condensers. To do so, short-circuit the d.c. power pack terminals and make measurements with a.c. only.

## Resonance Methods of Measuring L and C

While most of the circuits discussed so far are satisfactory for measuring power supply and audio components, they are not always usable or reliable at radio frequencies. Resonant circuits are usually better for r.f. measurements, both because they will measure small inductances and capacities accurately, and because these measure-

ments can be made regardless of the a.c. resistance or distributed capacity of the device.

In general, to make a resonant circuit measurement, you insert the inductance or capacity to be measured in a circuit which you tune to resonance. Then, since you know the values of the other circuit components,

the value of the inductance or capacity under test may be easily determined. Either a series or a parallel resonant test circuit may be used. In a series resonant circuit, a maximum current flow through the circuit or a maximum voltage across one of the parts will show when you reach resonance. In a parallel resonant circuit, a maximum voltage across the circuit, maximum circulating current, or minimum line current will all indicate resonance.

► To set up one of these measuring circuits, you need: a signal source capable of delivering about 5 watts of power, with a calibrated frequency output; some means of making measurements at radio frequencies (such as a thermocouple ammeter or an r.f. type vacuum tube voltmeter); and either a calibrated precision condenser or a known inductance. For resistance measurements, a variable resistor calibrated for r.f. will be necessary.

Fig. 10 shows two typical circuits. In Fig. 10A, you know you have reached resonance when the circulating current (measured by the thermo-

couple type meter) is at a maximum. The indicator in Fig. 10B is a v.t.v.m. which will show a maximum voltage across the condenser at resonance.

In both circuits, the coupling to the oscillator is adjusted to the minimum value which will give a reasonable meter reading. Very frequently an electrostatic shield *S* (shown by dotted lines) is used between the link coil and the coil in the resonant circuit to eliminate capacity coupling effects.

## CAPACITY MEASUREMENTS

Suppose coil *L* is a known inductance. You can then measure capacity by inserting the unknown condenser in place of condenser *C<sub>s</sub>* and adjusting the frequency of the oscillator until resonance is indicated by a maximum meter reading. The resistor *R* should be adjusted to zero resistance if the circuit in Fig. 10A is used.

Knowing the frequency and the inductance, you can determine the condenser capacity from the fact that, at resonance, the capacitive reactance equals the inductive reactance. This

gives you a formula 
$$C = \frac{25,330}{f^2 L}$$

(where *C* is the unknown capacity in microfarads, *f* is the frequency in kilocycles, and *L* is the inductance in microhenries) from which you can calculate the unknown capacity.

In this method the a.c. resistance has no effect (because you use a reactance balance, not an impedance balance), and the accuracy depends entirely upon the accuracy with which the inductance and frequency are known. Even the meter accuracy does not matter, since you use it only to indicate a maximum. However, in Fig. 10B, the input capacity of the v.t.v.m. is included in the calculated value. This capacity is small in a well-de-

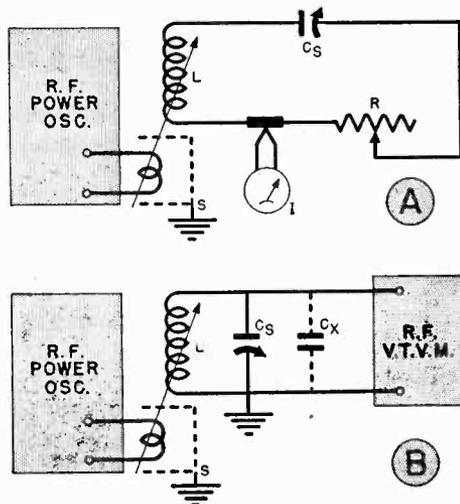


FIG. 10. Resonance methods for measuring inductance and capacity.

signed unit, but the measurement is in error by this amount. When doubtful about this capacity, Fig. 10A should be used if possible.

► If you have a calibrated condenser instead of a known inductance, you can use any coil which will give resonance and find the unknown capacity by a substitution method, provided the unknown capacity is within the range of the known capacity. First, put the unknown condenser in place of  $C_s$  and



*Courtesy General Radio*

The front panel of a standard variable condenser.

adjust the frequency of the oscillator until resonance is indicated. Then replace the unknown condenser with the standard calibrated condenser and adjust the latter until resonance is again indicated, using the same coil and the same frequency. The capacity of the unknown condenser can now be read directly from the scale of the calibrated condenser. We do not need to know the frequency nor the inductance value and make no calculation. Even the v.t.v.m. input capacity will not matter, as it will be present in both cases so the standard condenser has exactly the capacity of the unknown. Only the accuracy of the standard condenser determines the accuracy of the result, so this method can be very precise.

► Another method of using a calibrated condenser is particularly valuable where the unknown condenser is very small. Leave the calibrated condenser in the circuit and adjust the frequency of the oscillator so that resonance will be obtained with nearly maximum capacity of  $C_s$ . Call this  $C_s$  value  $C_1$ . Next, shunt condenser  $C_s$  with the unknown capacity  $C_x$ , as in Fig. 10B. This increases the capacity of the circuit, so reduce the value of  $C_s$  until resonance is again obtained with the same frequency. Call this value  $C_2$ , and find the value of the unknown condenser  $C_x$  by subtracting  $C_2$  from  $C_1$ . In other words, the capacity of the unknown condenser is equal to the difference in the two values of the standard condenser. If a vernier (a geared tuning arrangement or a calibrated trimmer) is used on the standard condenser  $C_s$ , you can measure values as low as 2 micro-microfarads accurately by this method.

## INDUCTANCE MEASUREMENTS

Either of the circuits in Fig. 10 can be used to measure inductance if you use a standard condenser and set resistor  $R$  at zero. The unknown coil is used as the inductance  $L$ . Bring the circuit to resonance, either by using a fixed capacity  $C_s$  and varying the oscillator frequency, or by leaving the oscillator frequency fixed and varying  $C_s$ , then calculate the inductance from

$$25,330$$

the formula:  $L = \frac{25,330}{f^2 C}$  (where  $L$  is

in microhenries, frequency is in kilocycles and  $C$  is in microfarads).

Since the oscillator link coupling may affect the inductance somewhat, weak coupling should be used. You can easily tell when the coupling is too tight, because you will get a double resonant hump (two maximum peaks) rather than a single sharp peak indica-

tion on your meter. Again, the a.c. resistance has no effect; the accuracy of the measurement depends only on the accuracy with which the capacity and the frequency are known.

### COIL A.C. RESISTANCE

Almost always, the a.c. resistance in a tuned circuit will be in the coil. You can neglect the resistance in the average tuning condenser.

You can use the circuit shown in Fig. 10A to measure the coil resistance. First, set resistor  $R$  to zero, bring the circuit to resonance and note the current reading on the meter. Then adjust resistor  $R$  to a value which reduces the current meter reading to one-half its first reading. This value of  $R$  is equal to the a.c. resistance of the coil.

You can easily see why. At resonance, the coil inductive reactance and the capacitive reactance cancel each other, and the circuit current is determined only by the source voltage and the a.c. resistance. When resistor  $R$  is increased to a value which cuts the current in half, the resistance in the circuit must have doubled, because you have not changed the source voltage. Therefore, this value of resistor  $R$  is equal to the a.c. resistance of the coil. Of course, this method requires that resistor  $R$  be variable and be calibrated at the frequency used in making the measurement.

If a variable resistor is not available, then a slight change in the method will allow you to use a fixed resistor to measure the coil resistance. To do so, first measure the current flow with the resistance out of the circuit. Call this  $I_1$ . Then, insert a fixed resistor  $R$  in the circuit, re-measure the current and call the new reading  $I_2$ . The a.c. resistance of the

$$\text{coil} = \frac{I_2 R}{I_1 - I_2}$$

The fixed resistor should have a value reasonably close to the expected coil resistance, which may range from a fraction of an ohm to 10 or 15 ohms. If it is too large,  $I_2$  will become a very small reading, difficult to accurately determine on the meter scale.

### COIL Q FACTOR

As you know, the Q factor of the coil is equal to its reactance divided by its a.c. resistance. If you know the inductance, you can compute the reactance ( $X_L = 6.28fL$ ), find the resistance by one of the methods just described and so find the Q of the coil at that particular frequency.

You can also find the Q of the coil with the circuit shown in Fig. 11. You

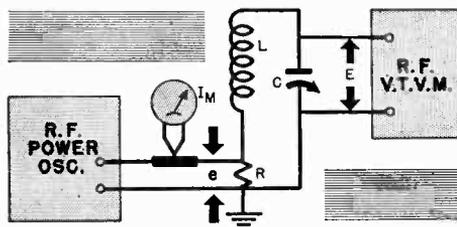


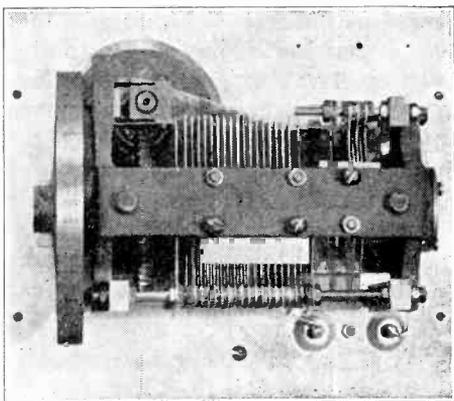
FIG. 11. A circuit for determining Q.

do not need to know the inductance or resistance of the coil, nor do you need a calibrated condenser. This circuit works on the basic principle of resonant voltage step-up, making use of the fact that, at resonance, the voltage across the condenser or the coil equals Q times the source voltage. Thus, you can divide the voltage across the condenser by the source voltage and obtain the Q of the circuit—which is the coil Q when the resistance is all in the coil.

You can easily measure the voltage across the condenser with an r.f. vacuum tube voltmeter, but measuring the source voltage in the resonant circuit is more difficult. The best way is to insert a known small resistance in the resonant circuit and use the voltage

drop across it as the source voltage (Fig. 11). Resistor  $R$  has very low resistance (a fraction of an ohm) which is only a fraction of the expected coil resistance value. By measuring the current flow from the oscillator through  $R$  with the meter  $I_M$ , you can find  $e$ , which equals  $I_M \times R$ .

Tune the circuit to a maximum reading on the v.t.v.m., divide this voltage by  $e$ , and you have the  $Q$  of the circuit. While this does not give the a.c. resistance or coil reactance, it does give you a way to compare coils for merit. If



*Courtesy General Radio*

Interior construction of a standard variable condenser.

you know either the reactance or the r.f. resistance, you can find the other

from  $Q = \frac{X_L}{R}$ . Of course, the circuit

resistance found from this formula includes the resistance of  $R$ , but this is usually small enough to neglect. If you wish to be very accurate, you can subtract it from the total resistance found.

► Of course, the reactance and a.c. resistance both vary with frequency, so the  $Q$  found holds good only near the frequency used. If the  $Q$  at some other frequency (or over a band of frequencies) is required, then you will have to tune the oscillator to these other frequencies, retune for resonance, and divide the new value of  $E$  by the new  $e$  to get the  $Q$  at these points.

### DISTRIBUTED CAPACITY

Occasionally you may want to know the distributed capacity between the turns of a coil. You can measure this with either of the circuits in Fig. 10. First, tune the circuit to resonance at some frequency which uses near maximum capacity of condenser  $C_s$ . Call this capacity  $C_1$ . Now, without changing the r.f. oscillator frequency setting, adjust  $C_s$  to tune the circuit to the second harmonic of the oscillator. Record this value of  $C_s$  (which will be about one-quarter the first value) as  $C_2$ . The distributed ca-

capacity of the coil,  $C_o$ , equals  $\frac{C_1 - 4C_2}{3}$ .

# Bridge Measurements

The bridge circuit is the most widely used means of measuring inductance and capacity values. It has the advantage of being a ratio device, so that the values of parts far different from the standard can be checked. Just a few standard parts permit a very wide range of measurement. Let's see how these circuits work.

## THE WHEATSTONE BRIDGE

You have met the basic bridge circuit in other lessons. Essentially, it is a connection of four resistors, with a voltage supplied to two terminals and a meter between the other two terminals, as in Fig. 12. To see how the circuit works, let us presume that  $R_X$  is an unknown resistor, and that  $R_s$  is adjusted until a value is found which makes the meter read zero.

The fact that the meter reads zero means that points  $B$  and  $C$  are at the same potential. Tracing the circuit, you will see that this means the voltage drop across  $R_1$  equals that across  $R_s$ , and the drop across  $R_X$  equals that across  $R_2$ . In other words, assuming that a current  $I_1$  flows through resistors  $R_1$  and  $R_2$ , and that another current  $I_2$  flows through  $R_s$  and  $R_X$ , the circuit is so adjusted that:

$$R_X \times I_2 = R_2 \times I_1$$

and

$$R_s \times I_2 = R_1 \times I_1$$

If you divide the upper equation by the lower equation, the currents will cancel and you will have:

$$\frac{R_X}{R_s} = \frac{R_2}{R_1}$$

This says that the ratio of  $R_X$  to  $R_s$  is the same as the ratio of  $R_2$  to  $R_1$ . Carrying this a step further, you have:

$$R_X = \frac{R_2}{R_1} \times R_s$$

This is a very important equation. It shows that if you make  $R_1$  equal to  $R_2$ , then  $R_X$  will be equal to the value of  $R_s$  which will bring the bridge into balance (give zero meter reading). But if  $R_1$  and  $R_2$  are not equal, this equation further shows that  $R_X$  will be equal to the ratio of  $R_2$  to  $R_1$ , multiplied by the value of  $R_s$  which balances the bridge. This means that even if resistor  $R_s$  has only a limited range of variation, you can still measure an unknown resistor  $R_X$  of almost any value merely by choosing the proper ratio between resistors  $R_1$  and  $R_2$ . This, of course, gives you a highly flexible measuring circuit.

In practical bridge circuits, resistors  $R_1$  and  $R_2$  are made so that they can

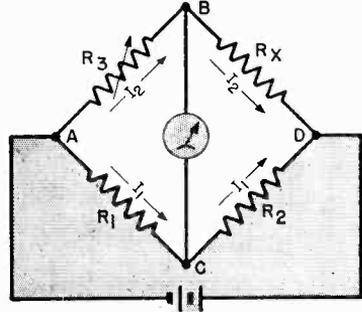


FIG. 12. The basic Wheatstone resistance bridge.

be varied in decimal ratios to each other—such as 1 to 1, 1 to 10, 1 to 100, 1 to 1000, or 1000 to 1, 100 to 1, or 10 to 1. Usually,  $R_1$  is an accurate fixed resistor, and  $R_2$  a series of fixed resistors arranged with a switch so the ratio can be varied in steps. Then the only continuously variable resistor will be  $R_s$ , and even for this, fixed resistors in a decade box arrangement may be used.

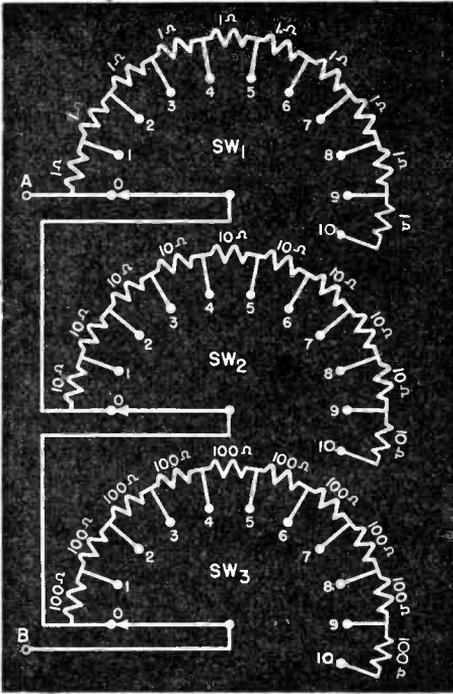


FIG. 13. A decade box.

► As shown schematically in Fig. 13, a decade resistance box is a series of accurate resistances connected to a series of contacts. Various resistances can be obtained by switching to various contacts. The name comes from the fact that each switch section has ten contacts, so there are ten steps (decade means a group of ten) in each section.

With the particular arrangement shown, any resistance between 0 ohms and 1110 ohms can be obtained in 1-ohm steps. (The switches are shown set in the 0 ohm position.) If, for example, switch  $SW_1$  is moved over to contact 4, there will be 4 ohms resistance between terminal A and terminal B. If  $SW_2$  is then set to contact 6, 60 ohms will be added to the 4 ohms and you will have 64 ohms. If  $SW_3$  is moved to contact 8, 800 ohms will be added in the circuit; you would then have a total of 864 ohms, etc.

The values of the resistances need not be 1, 10 and 100 ohms each; they can be small fractions of an ohm or much larger values, depending on the design of the decade box.

► In using a bridge, the ratio of  $R_2 \div R_1$  is adjusted to a likely value ( $R_X$  may or may not be approximately known) and  $R_3$  is adjusted to make the meter read zero. If the bridge cannot be made to balance, other ratios are tried until the zero reading is found. Then the value of  $R_X$  is the  $R_2 \div R_1$  ratio multiplied by  $R_3$ .

► Of course, the meter will read until the bridge is balanced. Since the amount of meter current depends on the amount of unbalance, it may be large if the bridge is far from balance. Further, the direction of current flow through the meter depends on whether the unknown is larger or smaller than the ratio times  $R_3$ .

In order to let the meter pointer swing in either direction, the meter springs are adjusted so zero is in the center of the scale. By noting which way the meter deflects, you can tell what adjustment is needed to balance the bridge. Thus, a deflection to the left may mean an increase in  $R_3$  is needed for a balance, while a deflection to the right may mean  $R_3$  should be decreased. It may be the other

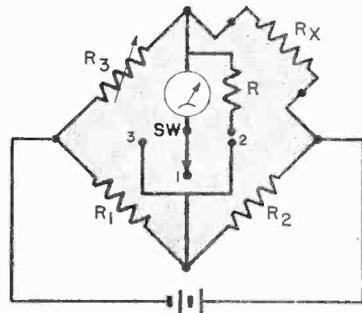


FIG. 14. For meter protection, switch position 2 is used until the bridge is nearly balanced.

way around, depending on how the battery is connected to the bridge terminals.

To prevent ruining the meter with excess currents, a switch like *SW* in Fig. 14 is usually used. This is a three-position switch that can be thrown to the right or the left from its normal resting position. Position 1 is the normal position and opens the meter circuit. (A spring returns the switch to this position when it is not held in one

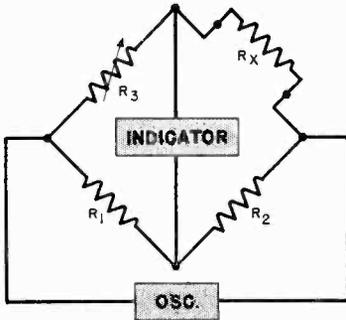


FIG. 15. This basic a.c. bridge is just like the d.c. bridge except for a change in the voltage source and the indicator.

of the other positions by the operator.) In making readings, the switch is first thrown to position 2, closing the meter circuit and connecting a shunt across the meter at the same time. This makes the meter insensitive and protects it from excess current. The bridge is adjusted until the meter shows little or no deflection, then the switch is moved to position 3, which places the sensitive meter directly in the circuit, and a final balance is found.

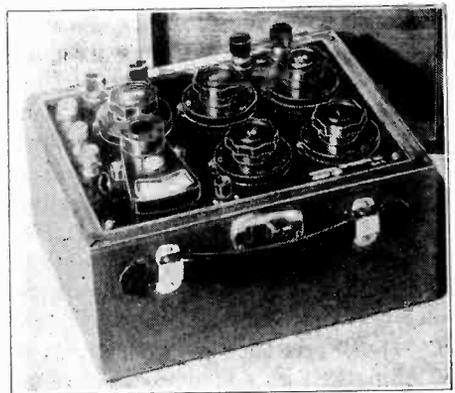
Until the bridge is almost balanced, the switch is closed only long enough to see whether the meter pointer moves rapidly away from zero. Then, as the balance point is neared, the current drops so that a reading can be taken safely. Even so, the switch is allowed to return to position 1 while any adjustment is made.

## A. C. BRIDGES

It is not necessary to use a d.c. power supply with a bridge. You can use an a.c., a.f. or r.f. voltage if you wish—but you must use an indicating device suitable for the kind of voltage used. For an audio voltage, an a.c. meter or a pair of headphones may be used, connected in the position shown in Fig. 15. Balance will be shown by a zero meter reading or minimum sound in the phones. R.F. voltages require a v.t.v.m. as the indicator.

Generally speaking, an a.c. bridge is more useful than a d.c. bridge, for we can measure coils and condensers as well as resistors with it. The a.c. bridge comes to balance when the impedances of the arms are in the proper ratios.

You might think that you have to use coils in all four arms of the bridge to obtain an inductance balance, and capacities in all arms for a capacity balance. The fact is, the ratio arms  $R_1$  and  $R_2$  can be resistances as long as they do not have appreciable inductance or capacity. Then a standard inductance can be used in place of  $R_3$  for inductance measurements, or a



Courtesy Industrial Instruments Co.

A typical bridge, having a built-in decade box. The decade can be used separately, if desired, as it is connected to a pair of terminals on the panel.

standard capacity in place of  $R_3$  for capacity measurements.

**Capacity Bridge.** The bridge for capacity measurements shown in Fig. 16A looks just like the resistance bridge of Fig. 15 except for the substitution of the two condensers. This bridge will balance when the impedances of the condensers have the rela-

$$Z_{CX} = - \frac{R_2}{R_1} \times Z_{CS}.$$

We can find the capacity of  $C_X$  if we assume that the impedance of a condenser is approximately the same as its reactance (for air- or solid-dielectric types the resistance is negligible) and remember that the reactance is inversely proportional to the capacity,

$$(X_C = \frac{1}{6.28fC}).$$

Substituting this relationship in the above formula and simplifying, you

get  $C_X = \frac{R_1}{R_2} \times C_S$ . Notice that this

capacity formula has *just the opposite ratio of resistances* from the formula used with the resistance bridge.

**Inductance Bridge.** On the other hand, you should use the *same* resistance ratio for the inductance bridge (Fig. 16B) as for a resistance bridge, because inductive reactance is proportional to inductance. The formula for

this bridge is:  $L_X = \frac{R_2}{R_1} \times L_S$ .

**Fixed Standards.** In the circuits shown in Fig. 16, the standard parts are variable. Each circuit is used just like a resistance bridge, with ratio arm adjustments made if the range of the variable is not great enough to balance the bridge. Often, however, variable standards are not available. If they are not, fixed coils or condensers can

be used as  $L_S$  or  $C_S$  and the ratio arms can be adjusted to balance the bridge. When this method is used,  $R_2$  should be a continuously variable resistor rather than a step or decade box type.

While theoretically you need only one fixed standard part for this bridge, in practice you will find a fine balance cannot be easily obtained, and the bridge becomes inaccurate if the ratio between  $R_1$  and  $R_2$  becomes very large. Therefore, it is usually best to have several fixed standard parts available and use the one which gives a balance most nearly in the ratio of 1 to 1. If the value of the unknown part cannot be guessed, use a fixed part of any

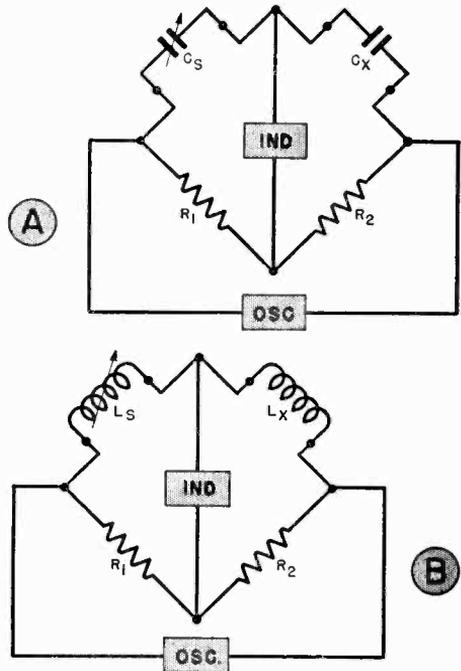


FIG. 16. The bridge at A is a capacity bridge, while B shows an inductance bridge.

value and determine the approximate ratio. Then use a smaller or larger part to bring the ratio down to a point where a fine balance may be easily obtained.

The fact that a fixed inductance or

capacity can be used in the bridge makes the instrument very practical. Fixed standards are comparatively cheap and readily available. For ordinary service work, readily obtainable parts of 2% to 5% tolerance may be entirely adequate for checking the radio parts—which often have tolerances as high as 20%.

**Wagner Ground.** When you measure high-impedance parts (small-

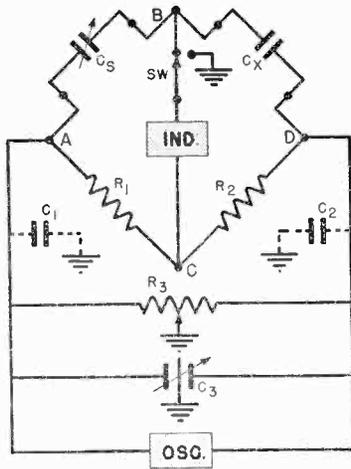


FIG. 17. Balancing to ground to eliminate body capacity effects.

capacity condensers or high-inductance coils), particularly at high frequencies, you may find that body capacity unbalances the bridge. For example, putting on the phones used as an indicator—or even bringing your hand close to the bridge—may change your readings. Grounding the bridge directly would solve this difficulty, but would also present us with another one just as bad, for the stray capacities  $C_1$  and  $C_2$  shown in Fig. 17 would then shunt the ratio arms and would themselves determine the ratio setting.

The best solution is to balance the entire bridge with respect to ground. One way to do so is to use a system known as a Wagner ground, shown in

Fig. 17. A low-resistance potentiometer  $R_3$  is connected across the bridge and the variable arm is grounded. As you see, this puts a low-resistance shunt across  $C_1$  and  $C_2$ . To operate a bridge using this ground, first balance the bridge in the usual manner as well as possible. Then, throw switch  $SW$  to the ground position and adjust  $R_3$  to give a minimum indication. (Notice that grounding one terminal of the indicator through switch  $SW$  makes another bridge of  $R_1$ ,  $R_2$  and the two sections of  $R_3$ , as this connects the indicator between  $C$  and ground, and through the ground to the arm of  $R_3$ .)

The bridge is balanced with respect to ground when a minimum indication is obtained. You should then throw the switch back to its normal position and readjust the bridge to give a minimum indication. If necessary, repeat this process until further adjustments cause no upset of the bridge.

► Since the trouble is caused by capacity to ground, the resistance of  $R_3$ ,

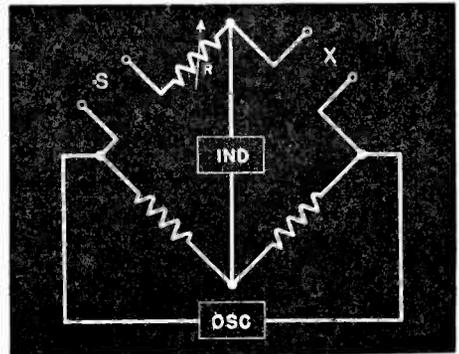


FIG. 18. Using  $R$  permits the bridge to be balanced for the resistance of the unknown part while its reactance is balanced by that of the standard part. This gives a direct measure of the a.c. resistance of the unknown part and permits the calculation of the  $Q$ .

may be too high to permit a fine balance at the higher radio frequencies. For high-frequency work, some bridges use the split stator condenser  $C_3$  shown



Courtesy General Radio

Using a capacity bridge of the type having a tuning eye indicator. This bridge will measure the power factor as well as the capacity.

in Fig. 17. This is a condenser with two stator sections and a single common rotor section; as the condenser is rotated, the capacity of one section increases while that of the other decreases. Its capacity can thus be varied to make the capacitive reactance between  $A$  and ground and that between  $D$  and ground have the same ratio as the  $R_1 \div R_2$  ratio, thus balancing the bridge with respect to ground. The balancing procedure is exactly like that used with the Wagner ground—except, of course, you vary  $C_s$  instead of  $R_s$ .

### IMPROVED BRIDGE CIRCUITS

The basic bridge we have discussed so far is excellent if you can neglect resistance in the coil or in the condenser being checked and the resistance in your standard. But a zero current indication can be obtained only if the impedances of the arms are exactly

equal; if the unknown has appreciable resistance as well as reactance, you can get a *minimum indication* but not a true *zero indication*. Also, this causes some error in adjusting the bridge. However, a rather simple means can be used to get a true null (zero) balance and, at the same time, determine the  $Q$  factor of the coil or condenser.

Usually the standard coil or condenser is chosen to have very low resistance, so if a resistance unbalance exists, it will usually be the fault of the unknown coil or condenser. As shown in Fig. 18, you need only add resistance in series with the standard coil or condenser to bring the bridge back into balance. To use this circuit, insert the coil or condenser standard and the unknown as indicated, and adjust the standard or the ratio arms to give a minimum indication. Then, adjust  $R$  for a further reduction. Repeat the

standard and  $R$  adjustments until the indication cannot be reduced further. When the *least possible* signal is heard or indicated, the reactive components of the standard and unknown are balanced, and so are the resistive components of the unknown and resistor  $R$ . Multiplying the value of resistor  $R$  by

$$X_C = \frac{1}{6.28fC}, \text{ so } Q_c = \frac{X_C}{R} = \frac{1}{6.28fCR}$$

As the power factor is approximately 1 — for low power factor condensers,  $Q$  the power factor is  $6.28 fCR$ .

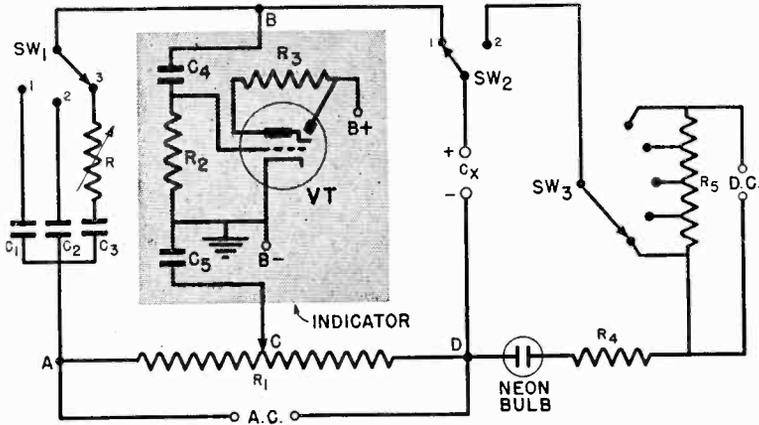


FIG. 19. A simplified diagram of a typical capacity bridge of the type made for radio servicemen. The bridge "corners" are A, B, C and D. Condensers  $C_1$ ,  $C_2$  and  $C_3$  are the standard parts, selected by  $SW_1$ . Resistor  $R_1$  makes up the A-C and C-D legs of the bridge. When it is varied, it adds resistance in one leg and takes it out of the other at the same time. Condensers  $C_4$  and  $C_5$  are just blocking condensers to isolate the indicator, which is a magic eye tube VT operating from the voltage across  $R_3$ . An adjustable d.c. supply is at the right, for leakage tests. To use, the unknown condenser is connected to  $C_x$  and  $SW_2$  is thrown to the right to apply the rated d.c. from  $SW_3$ . The neon bulb will flash or glow, the rate of flash depending on the leakage. Assuming a satisfactory leakage value (and no short-circuit causing a continuous glow),  $SW_2$  is thrown to the left,  $SW_1$  is set to use the proper standard, and  $R_1$  is adjusted for a maximum closure of the eye indicator, indicating a minimum voltage across  $R_3$ . Should the adjustment of  $R_1$  approach but not pass through a balance when moved all the way to the A end, the wrong standard is selected by  $SW_1$  or the  $C_x$  condenser is open. The same indication at the D end shows the wrong standard or a shorted  $C_x$ . When checking large condensers, the  $SW_1$  setting including  $R$  in the circuit is used, and  $R$  is adjusted for further closure of the eye. This resistor is calibrated to show the power factor directly.

the bridge ratio will give the amount of resistance in the unknown part.

Thus, you can determine the capacity or inductance and the a.c. resistance of the unknown part at the same time. If you want to find the Q factor of an inductance, find the inductive reactance from the formula  $X_L = 6.28fL$ , then divide this reactance by the resistance. Similarly, you can find the Q of a condenser by dividing the reactance by the a.c. resistance. However, the reactance of the condenser is

► The circuit shown in Fig. 18 is the basis for several capacity bridges designed for radio servicemen. A typical instrument is shown in Fig. 19. By calibrating  $R$  in terms of power factor and inserting a d.c. power supply to polarize electrolytic condensers, a bridge of this kind can be used to measure capacities, leakage and power factor for condensers ranging from the smallest air and mica types up through large electrolytic condensers. The capacity and power factor are in-

licated directly on a calibrated scale.

**Comparing L with C.** In the basic bridges covered so far, coils have been compared with coils and condensers with condensers. There are many variations of these bridges, found mostly in laboratories. Some have resonant circuits in their arms while others are designed for certain special purposes.

Another type, which balances a condenser against a coil, is very useful. A basic form of this bridge, using a standard condenser (which is more likely to be available than a standard coil) is shown in Fig. 20. Since the

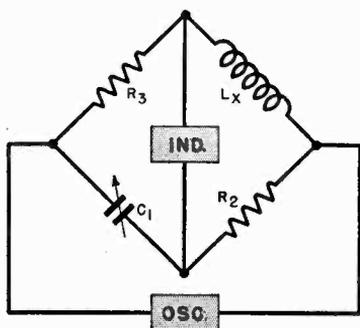


FIG. 20. This bridge balances a capacity against an inductance.

capacitive reactance is opposite in phase to the coil reactance, the balancing condenser must be in the position shown. The value of

$$L_x = R_2 R_3 C_1.$$

Fig. 21 shows the Owen bridge, a more common form of this circuit, which also takes the a.c. resistance of the coil into account. The capacity  $C_2$  is used to balance the coil resistance

$R_x$ . The a.c. resistance of the coil is found from the formula

$$R_x = \frac{C_1}{C_2} \times R_3,$$

while the inductance is found by the formula in the preceding paragraph.

This bridge is easily converted into a standard capacity bridge, so it can be used to check both inductance and capacity, using only a standard condenser.

**Bridge Limitations.** Most a.c. bridges work over a limited audio to low r.f. range. If the frequency is

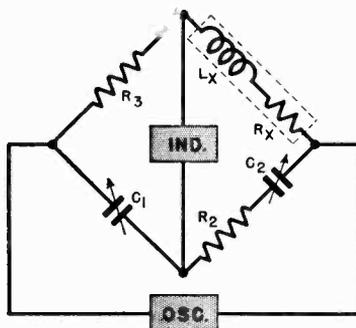


FIG. 21. The Owen bridge.

increased, complete shielding of each element becomes necessary, and many precautions must be observed, to prevent stray fields and coupling between parts from upsetting results. Although high-frequency r.f. bridges are available, they are costly and are found only in large laboratories. For ordinary purposes, the resonance method is usually used when very high frequencies are involved.

# Frequency Measurements

The number of times an event repeats itself within a specified length of time is its frequency—usually measured in cycles per second. Therefore, the first requirement for accurate measurement of frequency is a highly accurate way to tell time. The fundamental unit of time is determined by the rotation of the earth. By observing the positions of certain stars, astronomers determine this period very exactly. From these observations extremely accurate standard clocks are calibrated. Now if you were to operate an oscillator in such a way that an electric clock is driven from its output, you could compare the time kept by the electric clock with that kept by a standard clock and have a very precise way to measure both the frequency of the oscillator and its ability to maintain that frequency. You could then use the oscillator output as a standard of frequency.

Primary standards of frequency of this kind are found in the National Bureau of Standards and other large laboratories. They are crystal oscillators, because this type has the great-

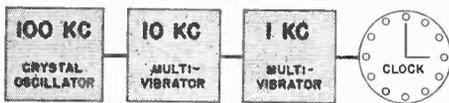


FIG. 22. How an electric clock is run from a crystal oscillator.

est frequency stability, and are kept at constant temperatures to make them as stable as possible.

Usually these oscillators produce frequencies of 50 or 100 kilocycles. Either frequency is too high to drive an ordinary clock, so it is stepped down by a chain of multivibrators. (The multivibrator is a circuit which

can produce an accurate fraction of a frequency fed into it.) Fig. 23 shows how the crystal oscillator is used to drive a chain of multivibrators which finally produces a signal capable of driving a clock. The clock used follows frequency exactly, so comparing its time with that of a standard clock shows us the frequency of the crystal oscillator and also indicates any variations in this frequency. By making adjustments to keep the clocks together, the output of the oscillator can be used as a frequency standard.

**The Multivibrator.** This interesting circuit does not “vibrate”—it is

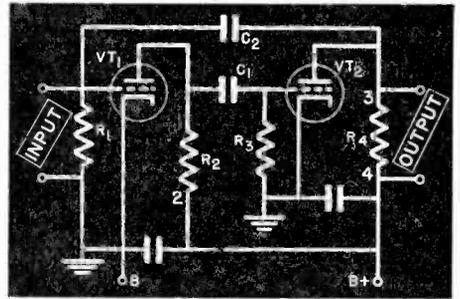


FIG. 23. A multivibrator is a resistance-coupled amplifier which can oscillate.

an oscillator consisting of a two-stage resistance-coupled amplifier like that shown in Fig. 23. Notice that condenser  $C_2$  provides a feedback path from the plate circuit of the tube  $VT_2$  to the grid circuit of tube  $VT_1$ . Each tube reverses phase by  $180^\circ$  so two tubes produce a  $360^\circ$  reversal. This means the voltage across  $R_4$  is back in phase with the input of  $VT_1$ , so this feedback path allows the circuit to oscillate.

► By inserting a bias, we can make one tube draw slightly more plate current than the other when the circuit is first turned on. For this explanation,

we will assume that  $VT_1$  draws more current than  $VT_2$ . The initial rush of current through  $R_2$  produces a voltage pulse across it, with a polarity such that the end of  $R_2$  marked 1 is negative. This pulse passes through  $C_1$  and is developed across  $R_3$ , driving the grid of  $VT_2$  negative and so cutting off its plate current. This causes a rapid decrease in the voltage drop across  $R_4$ ; you can consider this decrease to be a voltage pulse with a polarity such that the end of  $R_4$  marked 3 is less negative (more positive than normal). This pulse, in turn, passes through the coupling condenser  $C_2$  and is developed across  $R_1$ , making the grid of  $VT_1$  positive and so increasing its plate current.

The plate current of  $VT_1$  increases until the tube becomes saturated. When this occurs, the current becomes constant, and so does the voltage drop across  $R_2$ . In other words, there is no longer a voltage pulse across  $R_2$  and, naturally, no pulse voltage developed across  $R_3$ . With the disappearance of the pulse voltage across  $R_3$ , the grid of  $VT_2$  is able to go back to its initial bias. (This does not happen instantaneously, however, because  $C_1$  has been charged and must discharge through  $R_3$  before the grid of  $VT_2$  can reach its original bias; this discharge, of course, takes a little time.)

As the grid of  $VT_2$  approaches its initial bias, plate current again starts to flow through the tube. This increasing current creates another voltage pulse across  $R_4$ . Because the current is increasing, this time the pulse has a polarity such that the end of the resistor marked 3 is negative. This pulse, passing through  $C_2$ , drives the grid of  $VT_1$  negative, and stops the flow of plate current through the tube. This causes a voltage pulse across  $R_2$ , with a polarity such that

point 1 is more positive; passing through  $C_1$ , this pulse makes the grid of  $VT_2$  positive, and the current through  $VT_2$  increases.

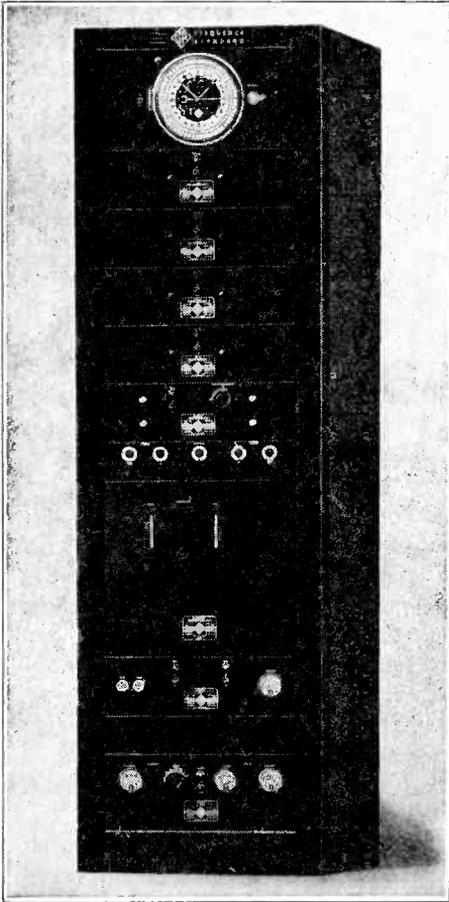
Now it is the turn of the current through  $VT_2$  to increase until saturation is reached. When the tube does saturate,  $VT_1$  is able to pick up again, while  $VT_2$  shuts off. This process is repeated over and over, with current flowing through first one tube, then the other. The net effect of this action, as far as output is concerned, is that voltage pulses appear across  $R_4$ .

► The rate at which these pulses occur—that is, the frequency of oscillation of the circuit—depends on the time constants of the  $C_1$ - $R_3$  and  $C_2$ - $R_1$  combinations. As you just learned,  $C_1$  must discharge through  $R_3$  before  $VT_2$  can pick up; similarly,  $C_2$  must discharge through  $R_1$  before  $VT_1$  can pick up. By adjusting the capacities and resistances of these combinations, you can obtain almost any fundamental frequency you want—from a very low one to a frequency well up in the r.f. range.

The output of this oscillator is rich in harmonics—in fact, the output wave may be triangular or even square in shape because there are so many.

► Although the multivibrator is fairly stable, its frequency can shift somewhat with changes in tube characteristics, operating voltages, etc. However, it will “lock in” with control pulses fed in the input, and will then produce its fundamental just as accurately as the control pulses are maintained. Equally as important—it will lock with a control pulse which is some multiple of its fundamental. For example, a 1-kc. multivibrator can be controlled by a 10-kc. control unit, as it will ignore all pulses in the input except the one it is locked with.

You see, at some time during the cycle of operation the voltage on the



*Courtesy General Radio*

This rack contains a frequency standard, employing a 50-kc. crystal and two multivibrators. By checking the clock against standard time signals, the accuracy of the frequency standard can be determined. By using the multivibrator outputs, frequencies from one cycle per second up to several megacycles can be obtained.

grid of  $VT_1$  has such a value that the tube can almost conduct current. Now suppose you feed in a 10-kc. signal across  $R_1$ . This signal adds to the voltage already present across  $R_1$ . If you adjust the amplitude of this signal carefully, you can make it increase the voltage on the grid of  $VT_1$  just enough to make the tube start conducting a tiny fraction of a second ahead of the time it would have started

if the signal had not been fed in. The circuit then proceeds through its normal operating cycle, unaffected by the input voltage, until it reaches the precise point where the input voltage is again able to make  $VT_1$  conduct. Since the fundamental frequency of the circuit is 1 kc., this point will not arrive until 9 cycles of the input signal have passed by, as these intermediate pulses, when added to the voltage across  $R_1$ , are not of sufficient amplitude to start tube conduction. However, the feedback voltage is continuously building up, and the tenth cycle of the input signal will make  $VT_1$  conduct again. Thus, the multivibrator locks in step with every tenth cycle of the input frequency, but ignores the cycles in between.

► In other words, the oscillation of the vibrator is precisely controlled by the input voltage, which overcomes the tendency of the circuit to wander and shift frequency. Thus, the fundamental and all its harmonic frequencies will be just as accurately controlled as is the input voltage. Further, the circuit can be controlled by frequencies as high as the tenth harmonic of the fundamental. Therefore, you can lock the circuit at the input side with one of its harmonics, but use the fundamental as the output frequency, and so get an output which is any fraction (up to, say, a tenth) of the input frequency.

It is this characteristic that makes the multivibrator useful in the circuit in Fig. 22. The crystal produces a 100-kc. voltage which controls the 10-kc. (fundamental frequency) multivibrator. The 10-kc. multivibrator in turn controls a 1-kc. multivibrator. By bringing out taps from the multivibrators in Fig. 22, you can obtain voltages with frequencies maintained just as accurately as the crystal oscillator frequency itself.

► Because it has so many harmonics, you can obtain from the 10-kc. multivibrator frequencies of 10 kc., 20 kc., 30 kc., etc., all the way up to 1000 kilocycles. Similarly, the 1-kilocycle multivibrator produces 1-kilocycle steps

burn brightest when resonance is reached).

When the wave meter has no indicator, you must depend on the meters associated with the source as indicators. Practically all r.f. power oscillators and power amplifiers have plate current meters. When a wave meter is brought near a resonant circuit and is itself tuned to resonance, it will absorb energy from the source. The plate current rises when this power is being furnished, so the point of maximum rise in plate current is the point of resonance; provided you are not too closely coupled, so as to cause a broad response or stop the oscillator.

► To use a wave meter, bring its coil close to an inductance in the circuit producing the frequency to be checked. If it is physically impossible to do this, feed the output of the circuit into a coil and bring the wave meter near this coil. Then, vary the tuning condenser dial until resonance is indicated and read the frequency off



*Courtesy General Radio*

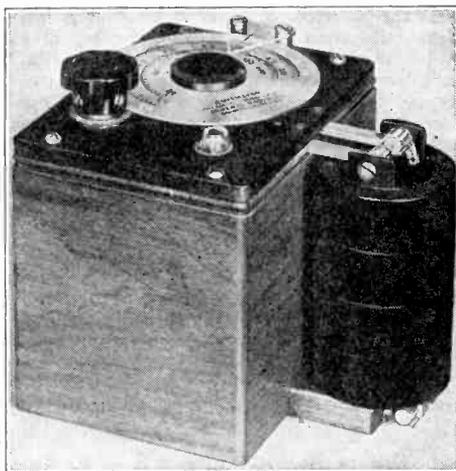
An engineer reading the clock on a primary standard of frequency.

all the way up to 100 kilocycles. If you wish, you can feed these outputs into other multivibrators and so get a large number of extremely accurate frequencies.

Then, the output from such a primary frequency standard can be used to calibrate secondary standards—such practical radio devices as signal generators, wave meters and other frequency determining devices. Let us now see just how such secondary standards are used in practical radio work.

## THE WAVE METER

A very common frequency-indicating device is a wave meter. This is nothing more than a coil-condenser resonant circuit. Many have some means of indicating when they are tuned to resonance—such as a vacuum tube voltmeter, a thermocouple current meter in series with the resonant circuit, or a pilot lamp (which will

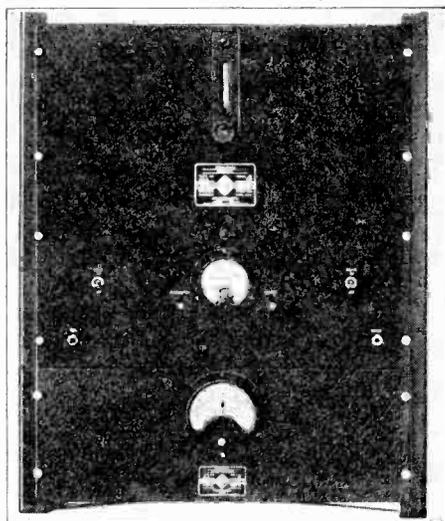


*Courtesy General Radio*

The condenser for this wave-meter is inside the box, while the round black object at the side is one of the plug-in coils. A pilot lamp on the panel is used as the resonance indicator. The condenser dial is directly calibrated in frequency, different scales applying to the different coils.

the dial. You may have to use a calibration chart if the condenser dial is not calibrated directly in frequencies.

Wave meters will maintain their accuracy of calibration for quite a long time if made from precision parts.



*Courtesy General Radio*

A frequency monitor used to check the frequency deviation (or frequency change) of a transmitter. An extremely accurate crystal oscillator produces a practically constant frequency. The crystal output is mixed with the transmitter carrier and the resulting beat frequency operates the deviation meter. A transmitter shift of one-half cycle can be observed on the meter, allowing the operator to keep the transmitter accurately adjusted to its assigned frequency. The use of this kind of equipment makes the carrier frequency of broadcast transmitters useful sources of known frequency for calibrating radios and service type signal generators.

However, a wave meter must be carefully made to be at all accurate—and even then will not show small variations in frequency less than 1% of the fundamental. For this reason, they are now used primarily for experimental purposes where an approximate frequency measurement is all that is required.

## FREQUENCY COMPARISON

The radio serviceman often makes frequency comparisons. The signal generator normally available for service work is accurate enough for all practical alignment purposes, but not for precision measurements if they should be necessary. Furthermore, the service instrument may drift, so its frequency scales are usually not accurately calibrated after a period of time.

However, a signal generator may be easily recalibrated by comparing its output with the frequencies of broadcast stations. These stations make handy frequency standards, since they are required by law to maintain their frequencies within 20 cycles of the assigned frequency.

All you need to recalibrate a signal generator (or any other oscillator with a frequency in the broadcast range) is a receiver and a knowledge of the exact frequencies of several stations, which you can find in any log book. Tune the set to a particular broadcast station as accurately as possible, using an output indicator. Then, tune the signal generator near the same frequency and feed its output into the receiver at the same time as the incoming program. You will hear a high-pitched squeal which will change to a lower and lower frequency as you tune closer to the broadcast frequency. If the signal generator is tuned past the broadcast frequency, the pitch will rise from a low-frequency to a high-frequency squeal.

At the point where the squeal is of the lowest frequency or disappears entirely, the frequency of the signal generator is identical with that of the broadcast station. This is known as the zero beat method of determining frequency.

After finding the zero beat point, all you need do is mark the **signal**



*Courtesy General Radio*

Using a heterodyne frequency meter to check a transmitter frequency. This device uses a stable variable oscillator and a crystal oscillator. The crystal and its harmonics are beat against the variable oscillator. The zero beat points, as indicated in the headphones, are the points where the frequency of the variable oscillator is exactly the same as the crystal or a crystal harmonic. Thus, the crystal oscillator serves to check the calibration of the variable oscillator and allow corrections to be made. Then, the output of the variable oscillator can be beat against the output of any other oscillator or transmitter within its range and will indicate the frequency of this transmitter or oscillator.

generator dial or adjust its trimmers so the frequency marking corresponds with the frequency of the broadcast station to which you are tuned. Then repeat the process with other broadcast stations, and you will be able to calibrate the signal generator (or oscillator) dial very accurately for broadcast frequencies.

**Using Harmonics.** By using harmonics of the signal generator output, you can calibrate it over the frequency bands which are lower in frequency than the broadcast band, and thus have available an accurately calibrated signal generator for making intermediate frequency alignments.

You recall that if an oscillator produces a fundamental frequency of 300 kilocycles, its harmonics will be at 600 kilocycles, 900 kilocycles, 1200 kilocycles and 1500 kilocycles. The 300-kc. frequency is below the broadcast band, but all of the above-men-

tioned harmonics are within the broadcast band. Therefore, if you tune the receiver to a broadcast station which has one of these particular frequencies, then tune the generator to exactly 300 kilocycles, you will get a zero beat from the harmonic of the oscillator beating with the incoming signal. (If the receiver dial is calibrated accurately, we need only tune in the signal generator harmonics at the proper points on the dial. However, the zero beat method is more accurate.)

In a similar manner, other points in the frequency range from 100 to 500 kilocycles can be checked by beating harmonics against known broadcast station frequencies.

► You should tune in harmonics at more than one place on the dial, unless you are certain of the fundamental frequency within a few kilocycles. For example, if you set the receiver

dial at 900, harmonics of 90, 100, 112.5, 128.5, 150, 180, 225, 300 and 450 kc. would all cause zero beats. You must tune away from 900 and see where the next signal comes in, to identify the fundamental frequency accurately. Thus, if you get signals at 900 and 990 kc., the fundamental must be 90, because this is the only one of this group which would have harmonics coming in at these two points. Similarly, harmonics of 100 would come in at 900, 1000, 1100 kc., etc.

Remember, you can find the fundamental frequency by locating any two consecutive points on the dial where harmonics can be heard, and then finding the frequency difference between these dial settings. This difference is equal to the fundamental frequency of the signal generator.

You will have to be careful to identify each and every harmonic to be sure that you have the right fundamental. Thus, if you use 900 kc. as one setting and find another signal at 1350, you'll have to be very certain there is no signal in between. Harmonics of 112.5, 150, 225 and 450 kc. will all come in at both 900 and 1350 kc.

If you hear no signal between 900 and 1350, the fundamental must be 450 kc., because 1350 minus 900 is 450. In other words, you have picked up the second and third harmonics of this frequency. On the other hand, if there are intermediate signals between 900 and 1350-kc. settings, you must determine where they are and must find the frequency difference between any two consecutive ones to find the fundamental frequency.

► This method lets you correct or calibrate a signal generator with sufficient accuracy for most measurement purposes. If the procedure is followed carefully, the accuracy will be that of the broadcast station fre-

quencies, which are usually accurate within one part in 100,000, or better, so the inaccuracy will be within a small part of 1%.

Broadcast stations themselves use accurate means of frequency determination to maintain their frequencies within the above-mentioned limits. Naturally, this maintenance of frequency is very important—for, if they drifted about, they might interfere with each other.

### ✓ DETERMINING FREQUENCY OF AN UNCALIBRATED OSCILLATOR

The method just discussed can be used for other purposes than correcting an approximately calibrated oscillator. You can also use it, for ex-

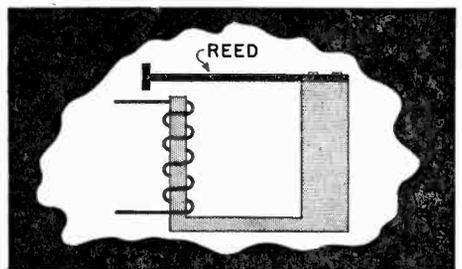


FIG. 24. A vibrating reed is shown here. A somewhat similar arrangement can be used to operate a tuning fork, which is a two-pronged "fork" made so its mass and springiness cause it to be mechanically resonant to some desired frequency.

ample, to calibrate an oscillator which is not calibrated at all.

To do so, just feed the output of the oscillator into a receiver and set the oscillator dial to a zero beat setting with a broadcast station, or depend on the receiver dial indications if it is accurately calibrated. Then tune the receiver slowly to higher frequencies and find the next zero beat setting. The difference between the two receiver frequencies which give successive zero beat indications is, of course, the fundamental frequency of the par-

ticular oscillator setting you chose.

By marking this oscillator dial setting with the frequency found, then repeating the process for other oscillator settings, you will be able to calibrate your oscillator accurately. Of course, as you learned in the previous section, you must be careful not to miss any harmonics as you tune the receiver.

### AUDIO FREQUENCY MEASUREMENTS

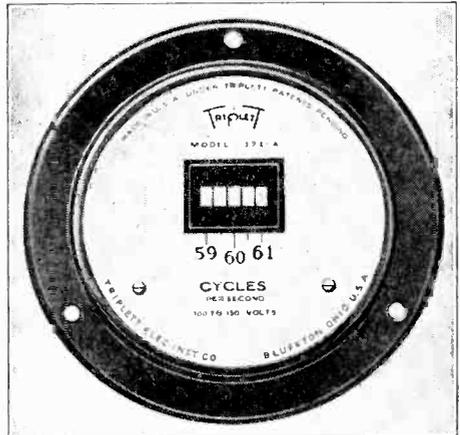
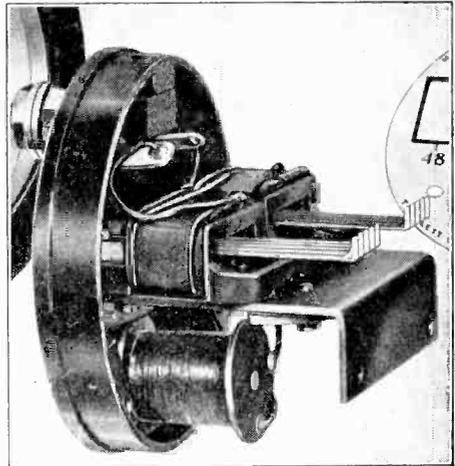
The standard frequency source shown in Fig. 22 can be used to check audio frequencies by adding another multivibrator or two to the chain. This will give frequencies of 100 cycles or 1 cycle per second, depending upon whether one multivibrator or two are added, which can be used to calibrate an audio signal generator or an audio wave meter. Either of these devices can then be used to determine the frequency of an audio voltage.

► There is another source of standard low frequencies—a vibrating reed or tuning fork. These devices have a natural frequency of vibration which depends on mass and springiness. When forced into vibration by an electromagnetic device like that shown in Fig. 24 the fork or reed will vibrate with the greatest amplitude when the frequency of the voltage applied to the coil is the same as the natural frequency of the device.

This fact has been used as the basic principle of a simple frequency indicator, using the less-expensive reeds. Several reeds, each with a different natural frequency, are mounted so that they are driven by an electromagnet. When an a.c. voltage is fed into the electromagnet, the reed with the natural frequency closest to the frequency of the a.c. vibrates with the greatest amplitude.

Of course, such a device would re-

quire a great many reeds to cover the entire audio range, because a different reed would have to be used for every few cycles. It is therefore usually used only to measure frequencies within a small range, particularly those near power line values. As you know, the standard a.c. power line in the United States is a 60-cycle line. Power companies use vibrating-reed meters a great deal in checking the line frequency so as to keep it accurately controlled. (Incidentally, the accuracy of electric clocks used in

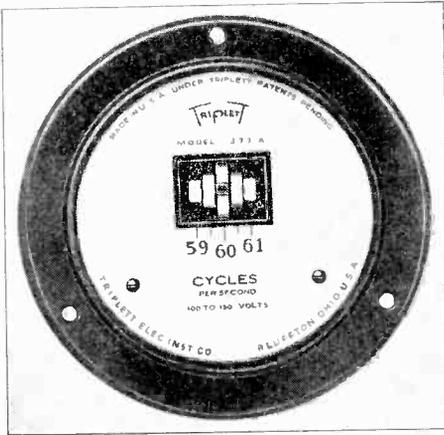


*Courtesy Triplett Elec. Inst. Co.*

**FIG. 25.** An interior view and a face view of the vibrating reed type frequency indicator.

homes depends entirely on the accuracy of the power line frequency. The fact that they keep such good time shows how well this frequency is controlled.)

A typical vibrating-reed meter is shown in Fig. 25. The ends of the reeds are painted white, so you can see when they vibrate. Fig. 26 shows



Courtesy Triplett Elec. Inst. Co.

FIG. 26. How the reeds indicate the frequency. As the 60-cycle reed appears largest, the frequency is 60 cycles.

its appearance when a frequency of 60 cycles is being measured. The 60-cycle reed appears the largest because it is vibrating at its natural frequency and so has a large amplitude of vibration. The reeds on each side are also vibrating, but not as much.

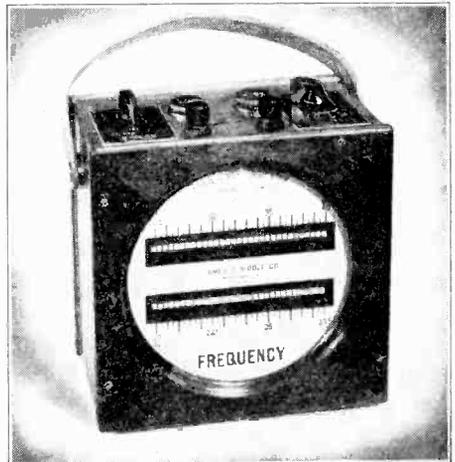
► While we are on the subject, another type of power line frequency meter is shown in Fig. 27. This meter has two coils ( $L_1$  and  $L_2$ ), with a magnetic vane mounted so it is free to turn in their fields. When an a.c. voltage is applied to the terminals, current will flow through both coils. The external circuit parts  $R_1$ ,  $L_3$ ,  $R_2$  and  $L_4$  are so proportioned that equal currents flow in  $L_1$  and  $L_2$  when the applied a.c. has a 60-cycle frequency. The normal tendency of the magnetic

vane in the meter is to line up along the axis of one or the other of the coils, but since both have equal fields, it stands still, halfway between the two axes.

Changes in applied voltage will not move the vane, because the voltage drops will merely increase or decrease in proportion in both halves of the circuit. However, as you know, coil reactance goes up with frequency. So, if the frequency increases, the drop across  $L_3$  will increase, producing a greater current through coil  $L_1$ ; at the same time, the reactance of  $L_4$  will increase, allowing less current to flow through  $L_2$ . Since  $L_1$  now has the stronger field, the magnetic vane will swing to the right to line up more nearly with the axis of coil  $L_1$ . On the other hand, if the frequency decreases, opposite effects occur and the vane will move to the left.

Properly designed, this instrument will measure very small frequency changes. A picture of a typical instrument is shown in Fig. 28.

► While vibrating-reed and magnetic



Courtesy James G. Biddle Co.

A portable laboratory type vibrating reed frequency indicator. Note the large number of reeds, providing a wide frequency range in one-quarter and one-half cycle steps.

vane meters are found mostly in power stations, many laboratories also use them to check audio frequencies within their ranges.

### THE WIEN BRIDGE AS A FREQUENCY INDICATOR

Any resonant bridge circuit can, of course, be used like a wave meter to determine frequencies. For audio frequencies, a bridge known as the

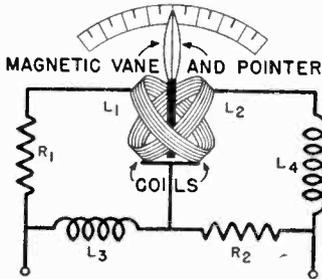


FIG. 27. The magnetic vane frequency meter.

Wien bridge (which is also very popular for measuring capacitivities) is particularly useful. Its circuit is shown in Fig. 29.

This bridge can measure frequencies because of the actions of the arms containing the condensers. Notice  $C_1$  is in shunt with  $R_3$ , while  $C_2$  is in series

with  $R_4$ . As you can see, the impedance between terminals  $A$  and  $C$  can drop to a very low value at high frequencies because of the drop in reactance of  $C_1$ . However, no matter how high the frequency, the impedance between  $C$  and  $D$  can never drop below the value of  $R_4$ . On the other hand, if the frequency is decreased, the impedance between  $A$  and  $C$  will never go higher than the value of  $R_3$ , while that between  $C$  and  $D$  can go on up to practically an open circuit.

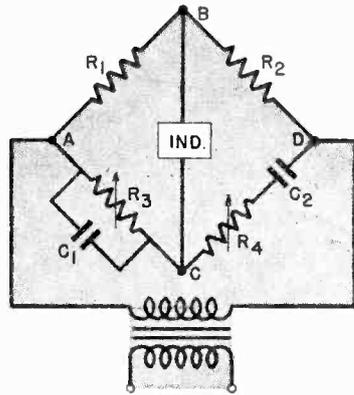
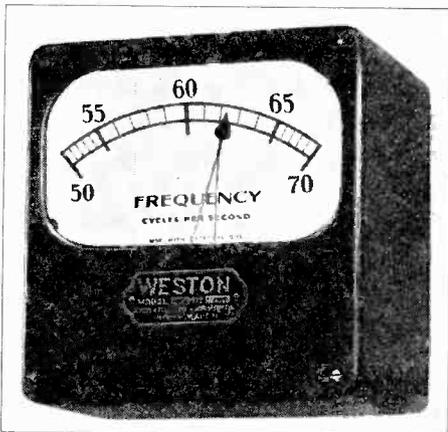


FIG. 29. The Wien bridge used as a frequency indicator.

Therefore, as you move from some one particular frequency in either direction, the reactances of the two arms will behave in different manners. In fact, they act much as a coil and condenser would—therefore, in effect, this is a resonant circuit. We won't go into the engineering behind this bridge, but if resistor  $R_1$  is twice  $R_2$ ,  $C_1$  equals  $C_2$  and  $R_3$  equals  $R_4$ , the frequency of balance of the bridge will be

$$f = \frac{159,000}{R C}$$

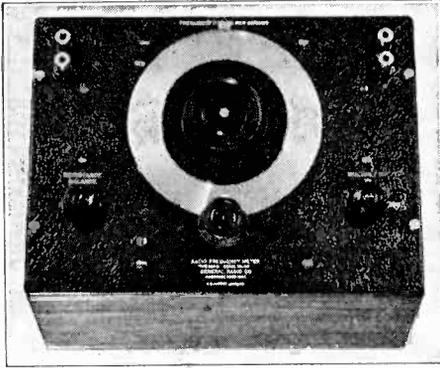
where  $f$  is in cycles,  $C$  is in microfarads and  $R$  is in ohms. In this case,  $R$  is the value of either  $R_3$  or  $R_4$ , and  $C$  is the value of either of the condensers.



Courtesy Weston

FIG. 28. A face view of a typical magnetic vane frequency meter.

By varying the resistances of  $R_3$  and  $R_4$  (but keeping them equal), or by using different sizes of condensers (but keeping them equal also), we can change the frequency to which the bridge will be balanced. Thus, we can use the bridge to determine the fre-



*Courtesy General Radio*

**FIG. 30. A typical laboratory type frequency meter using the Wien bridge.**

quency of the source connected to it, provided we know the capacity of the condensers and the value of the resistors. A typical audio frequency bridge is shown in Fig. 30. The main dial varies the resistors, which are ganged together.

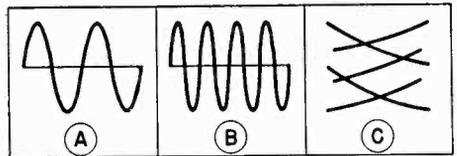
### USING THE C. R. O. TO DETERMINE FREQUENCIES

From your previous study of the cathode ray oscilloscope, you know that you can get an image on the face of the tube which will be an exact reproduction of the applied voltage. Furthermore, when the sweep frequency is locked in with the incoming signal frequency, you'll get a single cycle of the incoming wave. Now, let's see how you can use these characteristics of the c.r.o. to measure frequencies.

As you know, a stationary pattern

can be obtained if the sweep frequency is a sub-multiple of the incoming frequency. The number of complete cycles in the pattern gives the relationship between the sweep frequency and the incoming frequency. For example, if the incoming wave is 1000 cycles and the sweep is set to 500 cycles, you will get a 2-cycle pattern as shown in Fig. 31A. Similarly, a wave four times the sweep frequency makes four complete cycles, as shown in Fig. 31B, and so on up to 15 or 20, depending on the size of the c.r.o. screen.

When the sweep frequency is adjusted the other way so it is above the incoming signal frequency, you no longer get an image of the incoming signal. Instead, there will be a series of more or less horizontal lines on the c.r.o. screen. When there are two stationary lines, the sweep frequency is twice the incoming signal; when there are three lines, the sweep is three times the frequency, etc. Fig. 31C shows five lines, so the sweep fre-



**FIG. 31. The number of cycles or lines in the c.r.o. pattern can be used to indicate frequency.**

quency is five times the incoming frequency.

Thus, if the sweep generator in the c.r.o. is calibrated, you can easily determine the approximate frequency of the incoming signal from the setting of the sweep controls and the pattern formed. You can calibrate the sweep by feeding known frequencies into the c.r.o. and noting the setting of the sweep frequency control necessary to produce stationary patterns.

## LISSAJOUS FIGURES

Since the sweep of the c.r.o. may drift in frequency and thus be of varying accuracy, the calibrated sweep method is used only for rough measurements. When you want greater accuracy you should use Lissajous figures to measure frequencies with the c.r.o.

To use Lissajous figures, you must have an accurate source of known frequency. The unknown frequency is fed to the vertical plates of the c.r.o. tube, while the known standard is fed to the horizontal plates. An excellent source for the standard frequency is a.c. from a 60-cycle power line, which is accurately controlled by the power company.

If the frequency of the unknown and the known frequencies are exactly equal, you will get a single slanting line, an ellipse, or a single circle on the c.r.o. screen. The possible pattern

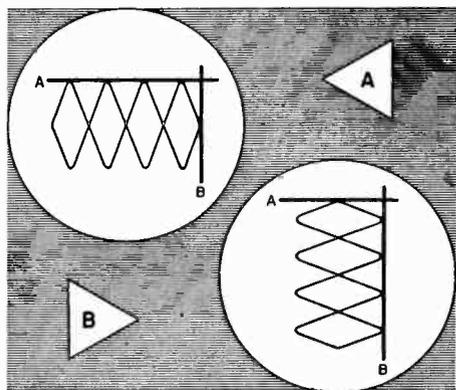


FIG. 32. Using imaginary lines A and B to determine the frequency of a Lissajous figure.

differences are due to phase shifts. However, any pattern which is a single line, oval or circle shows that the two a.c. voltages have the same frequencies.

On the other hand, if you get a

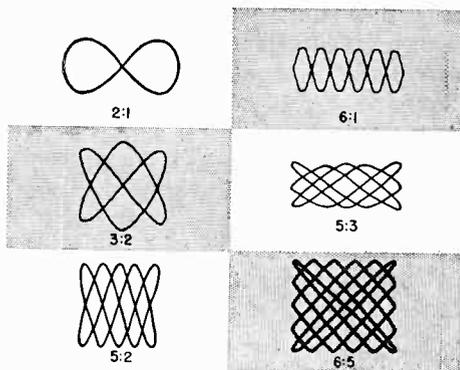


FIG. 33. Several typical Lissajous figures showing frequency ratios.

series of closed loops, the voltages have different frequencies. The position and number of the loops give the ratio of the applied frequencies; even fractional ratios can be determined in this way.

For example, suppose you get a pattern like that shown in Fig. 32A. To determine the frequency ratio, draw or imagine the lines A and B. Notice the number of times each line is touched by the pattern. Line A is touched four times, while line B is touched once. This means the frequency ratio is 4 to 1—that is, the signal applied to the vertical plates is four times that on the horizontal plates. Thus, if a 60-cycle voltage is applied to the horizontal plates, the vertical plate voltage has a frequency of 240 cycles.

If the image is turned around, as shown in Fig. 32B, so the figure touches the line A once and the line B four times, the ratio will be 1 to 4. This means the horizontal signal frequency will be four times the vertical signal frequency; if the horizontal voltage is still a 60-cycle signal, the vertical voltage must be a 15-cycle signal.

A number of typical patterns are shown in Fig. 33. In all cases, count-

ing the number of times the loops touch the imaginary lines *A* and *B* will give the frequency ratios. These ratios, multiplied by the frequency of the horizontal voltage, equal the frequency of the vertical voltage. Thus,

if the horizontal voltage is 60 cycles a pattern with a ratio of 5 to 3 shows that the frequency of the vertical voltage is  $5/3 \times 60$ , or 100 cycles. This method is quite accurate if the pattern is studied carefully.

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THE N. R. I. COURSE PREPARES YOU TO BECOME A  
**RADIOTRICIAN & TELETRICIAN**  
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# Lesson Questions

Be sure to number your Answer Sheet 30FR-1.

Place your Student Number on every Answer Sheet.

Send in your set of answers for this lesson immediately after you finish them, as instructed in the Study Schedule. This will give you the greatest possible benefit from our speedy personal grading service.

1. What factor is found when you divide the coil reactance by its resistance ( $X_L \div R$ )? *Impedance*
2. When measuring the inductance of an iron-core coil, what value of polarizing (d.c.) current should flow through it? *Normal operating d.c. current*
3. When measuring the inductance of an iron-core coil which has a high resistance, would you use: 1, The reactance method; or 2, the three-voltage method, if the true inductance is to be found? *2*
4. Which of the following condensers requires a polarizing voltage when checking the capacity: 1, Air dielectric; 2, paper; 3, ceramic; 4, electrolytic; 5, mica? *4*
5. When using the resonance method of measuring inductance, what condition is indicated by a double-hump (two peaks) reading on the meter? *Too large a capacitance*
6. Using a resonant circuit like the one in Fig. 10, what four factors of a coil can be measured or figured? *Q, a.c. reactance, a.c. resistance, distributed capacity, inductance*
7. When finding the Q using the circuit of Fig. 11, what basic principle is being used in the measurement? *resonance step up*
8. Using the decade box of Fig. 13, to which contacts would you set switches  $SW_1$ ,  $SW_2$  and  $SW_3$  so that you would have 471 ohms between terminals A and B?  *$SW_1 - 1$   $SW_2 - 7$   $SW_3 - 4$*
9. When a bridge circuit is balanced, does the indicator show: 1, Maximum current; or 2, minimum current? *2)*
10. Suppose you are using a wavemeter, consisting of a coil and a condenser, which has no built-in indicator. If this wavemeter is brought near an oscillator tank circuit, what indication will be obtained on the oscillator plate current meter when the wavemeter is tuned to resonance?



## VARIED INTERESTS

Don't go stale! Keep a fresh, active interest in everything you do. When you work or when you play—work hard, play hard. When you study, concentrate on study. When you stop working, or playing, or studying—let go. Forget about it. The best vacation is merely a complete change from what you have been doing. Loafing is not a vacation—it is merely boredom.

There is nothing better for an office worker after hours than a brisk walk, a swim or a round of tennis. There is nothing better for an outdoor worker than a quiet hour with a book or a good newspaper, or listening to radio programs. But don't do the same things all the time. Vary your life as much as you can. Cultivate a keen interest in the world of which you are a part.

Keeping alert keeps you young. Keeping interested keeps up your energy. Together these things will make you able to learn more in a given period of time. Finally, the more you learn, the more easily you learn.

*J. E. Smith*

**TUNING INDICATORS AND  
AUTOMATIC FREQUENCY CONTROLS  
FOR RADIO RECEIVERS**

31FR-1



**NATIONAL RADIO INSTITUTE**

**WASHINGTON, D. C.**

**ESTABLISHED 1914**

# Study Schedule No. 31

For each study step, read the assigned pages first at your usual speed. Reread slowly one or more times. Finish with one quick reading to fix important facts firmly in mind, then answer the Lesson Questions for that step. Study each other step in this same way.

**1. Making Receivers Easy to Tune; Current-Controlled Visual Tuning Indicators; Voltage-Controlled Visual Tuning Indicators - Pages 1-8**

Here is a wealth of practical information which you'll appreciate even more once you begin working on receivers which have tuning indicators. Give particular attention to the section on cathode ray tuning indicators, because you will encounter these more than any other types. Answer Lesson Questions 1, 2 and 3.

**2. Unique Tuning Aids; Essential Sections of an A.F.C. System; The Discriminator Circuit - - - - - Pages 8-13**

Here we have brief but interesting descriptions of one-of-a-kind tuning indicators which are to be found on only a few models of receivers. One reading now will be enough, because you can always refer to this section later. Study thoroughly every bit of the information on a.f.c. systems, however, because this basic information applies also to f.m. receivers. Answer Lesson Questions 4, 5 and 6.

**3. Review of Phase; Discriminator Circuit Action; The Oscillator Control Circuit - - - - - Pages 13-19**

Study this material slowly because every word counts. It isn't exactly essential to understand all this theory in order to fix a receiver having an a.f.c. system or repair an f.m. receiver, but this knowledge will definitely help a lot in most jobs of this type. Furthermore, any man who understands how an a.f.c. system works is really deserving of promotion from the ranks of a beginning student to an advanced group. There are thousands of actual servicemen who do not understand the material you will study here.

**4. Typical A.F.C. Circuits; Adjusting A.F.C. Systems - - - Pages 19-28**

Here are practical examples from actual receivers, to illustrate the theory you learned in the first part of this lesson. Take it slow and easy, because fast reading will get you nowhere in material of this nature. Answer Lesson Questions 7, 8, 9 and 10.

**5. Mail your Answers for this Lesson to N.R.I. for Grading.**

**6. Start Studying the Next Lesson.**

# Tuning Indicators and Automatic Frequency Controls for Radio Receivers

## Making Receivers Easy To Tune

THE tuning dial of a selective, modern receiver may require adjustments which are accurate to within one-hundredth of an inch movement of the dial pointer on its scale, if the desired signal is to be tuned in properly, free from distortion. The effects of improper tuning upon the wave form of a signal are clearly shown by the diagrams in Fig. 1.

Receiver designers recognize the difficulties involved in tuning a receiver with this high degree of accuracy, and for this reason have provided a number of tuning aids on the receiver. Large dial scales are provided, with correspondingly longer pointers, to make small movements of the pointer more readily distinguishable. Two-speed adjusting knobs, one turning the tuning condenser assembly directly and the other through a mechanical speed-reducing or vernier mechanism, further simplify accurate tuning.

A dial designed for accurate tuning is of little value if the listener has to rely upon his ears as a guide for proper tuning. The reasons for this are quite simple; loudness is not a dependable guide because about equal loudness is secured for several kilocycles on either side of resonance, due to A.V.C. action. Clarity of reception is likewise a poor guide, for the average individual is not capable of telling immediately whether or not distortion is present. (In a receiver having A.V.C. but no tuning indicator we could, of course, tune for maximum volume with minimum background noise, but few people realize that this can serve as a guide for correct tuning.)

*Tuning Indicators.* The introduction of automatic volume control practically forced radio designers to develop another improvement, the visual tuning indicator. Fortunately, there is in A.V.C. receivers a voltage or current which either reaches a maximum or minimum value when the receiver is correctly tuned. This voltage or current can be made to operate a simple meter-type indicator having either a needle pointer or a shadow indicator; other types of indicators used for this purpose include a cathode ray indicator tube (often



FIG. 1. These diagrams, copied from curves produced on the screen of a cathode ray oscilloscope in the N. R. I. laboratory, show what happens when a selective radio receiver is carelessly tuned. At A is the audio output curve secured when the receiver is correctly tuned to a carrier modulated with a single pure sine wave; there is no distortion. The slightly-distorted audio output wave form produced when the receiver is tuned about 2 kc. above the input R.F. carrier frequency appears at B, while the severely distorted wave form at 10 kc. off-tune appears at C.

called a magic eye), a miniature electric lamp providing flash tuning, or an audio beat oscillator which provides an audible indication of incorrect tuning.

Visual tuning indicators are by no means a perfect solution to this problem of tuning, for even with the best types of indicators the average broadcast listener finds it difficult to tune exactly at all times. Correct tuning requires patience and careful watching of the tuning indicator; a person who is relaxing to enjoy a radio program does not care to bother with the details of proper tuning.

Radio engineers recognize that slight mistuning is relatively unimportant during the first few moments of listening. After the radio receiver has been in operation for an hour or so, however, even small amounts of distortion become objectionable although they cannot consciously be recognized by the listener. Distortion appears to affect the nervous system, making a person tired of listening even to a good program.

Since the success of the entire Radio industry, including broadcasters and receiver manufacturers alike, depends upon having a large listening public, it is only natural that radio receiver engineers continued their search for devices and methods which would simplify the tuning of receivers. Their goal was a system which would permit even a small child to tune in the receiver just as well as could a radio expert.

*Automatic Frequency Controls.* As you know, in a superheterodyne receiver it is the local oscillator frequency which, more than anything else, determines the value of the I.F. signal. When this I.F. signal frequency is different from the I.F. value of the receiver, due to improper tuning, distortion will be present and sensitivity will be poor.

The superheterodyne receiver circuit is well suited for automatic tuning; even though it may be as much as 5 kc. off resonance, the tuning can be corrected merely by adjusting the oscillator trimmer condenser or by changing the inductance of the oscillator coil.

With this basic superheterodyne principle in mind, radio engineers produced a circuit for developing automatically a voltage which is proportional to the amount by which the oscillator is off resonance. This voltage is fed into a special vacuum tube circuit which is connected to the oscil-

lator and has the peculiar ability of being able to change the inductance or capacity of the oscillator circuit just enough to produce the correct I.F. signal value. This system of automatic frequency control is commonly called A.F.C.

When a receiver is equipped with A.F.C., the listener merely tunes in the desired signal as best he can without any particular attention to accuracy of tuning; the A.F.C. system then automatically completes the tuning procedure.

Automatic frequency control sys-

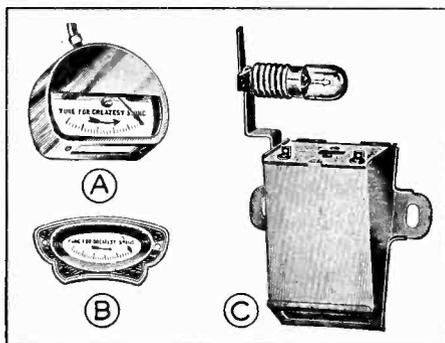


Fig. 2. Examples of current-controlled tuning indicators using an inexpensive milliammeter movement which consists essentially of a solenoid (coil) of about 1000 ohms resistance, inside which moves a long, pointed steel vane to which is attached the indicating pointer. The greater the D.C. current through the solenoid, the greater is the force which pulls the vane into the solenoid and moves the pointer. At C is a shadowgraph tuning indicator using a milliammeter movement; its operating principle is illustrated in Fig. 3.

tems are used chiefly in those larger receivers which have some type of automatic push button tuning system; in fact, A.F.C. is essential in a receiver having the electro-mechanical (motor-driven) type of automatic tuning system in order to correct for slight inaccuracies in the mechanical action of the tuning mechanism and compensate automatically for oscillator frequency drift.

Tuning aids and automatic frequency control systems provide additional problems for the Radiotrician.

The operating principles of the devices in general use for tuning aids must be thoroughly understood before service can be rendered, so these principles will now be taken up.

### Current-Controlled Visual Tuning Indicators

*Meter-Type Indicator.* Meter-type tuning indicators are ordinarily connected into the plate supply lead going to one or more A.V.C.-controlled stages. The average D.C. plate current drawn by a screen grid or pentode tube with normal C bias is about 5 milliamperes; this plate current drops practically to zero when the A.V.C. system increases this negative C bias to 40 or 50 volts. A 0-5 ma. milliammeter movement can therefore be used as a meter-type tuning indicator.

In Fig. 2A is shown one form of a meter-type tuning indicator. This unit is attached to the receiver chassis in such a way that the face of the meter can be viewed through a hole in the receiver panel. An escutcheon plate like that shown in Fig. 2B is often used with this meter to improve its appearance, and a pilot lamp is sometimes mounted below the meter to illuminate the scale through the window which can be seen in Fig. 2A. The no-current position of the pointer is at the extreme right, as shown in these views. With the meter connected to indicate D.C. plate currents of an A.V.C.-controlled stage, the current through the meter is a maximum when no signal is tuned in, and the pointer will be at the extreme left. Tuning in a carrier signal causes A.V.C. action to decrease the D.C. plate current for the A.V.C.-controlled stages, and the tuning meter pointer therefore moves toward its zero position at the extreme right. The listener tunes the receiver for a maximum deflection to the right, or in other words,

tunes for greatest swing to the right, for this corresponds to at-resonance conditions in the receiver circuit.

There are a number of variations of the simple meter movement just described, but all depend upon the same operating principles. In one case a small, round black disc is mounted on the end of the meter pointer; this disc moves over a series of concentric circles, alternately black and white, like those on rifle targets. The no-current

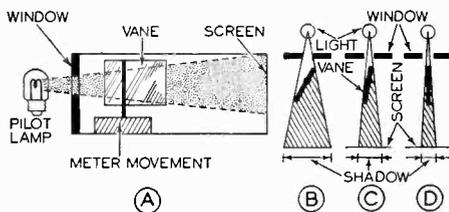


FIG. 3. In this shadowgraph tuning indicator, a large current causes the meter movement to place the vane in position B, giving a maximum-width shadow. Tuning in a station reduces the current, making the vane take various positions, such as at C and D; the listener tunes for minimum shadow, since this corresponds to exact resonance for any station.

position of the disc is at the center of the target, and the listener therefore tunes to get this pointer as close as possible to the center; manufacturers' instructions say: "Tune for a bull's eye."

*Shadowgraph Tuning Indicator.* Perhaps the most widely used variation of the tuning meter movement is the shadowgraph tuning indicator shown in Fig. 2C. This contains a frosted glass or celluloid screen which is mounted so as to be visible from the front of the receiver panel. Behind this screen is a pilot lamp, the light from which passes through a small window and past a small rectangular vane attached to the meter movement before reaching the screen. When meter current is a maximum, this vane blocks most of the light, casting a shadow on the screen. As resonance is approached when tuning in a station, the meter movement ro-

tates the vane towards its zero position, giving a minimum-width shadow. Details of this arrangement are shown in Fig. 3A, and the effects of vane position upon the width of the shadow are shown in Figs. 3B, 3C and 3D.

**Circuit Connections.** When a tuning meter is to be actuated by the D.C. plate current of only one A.V.C.-controlled stage, connections into the plate supply lead of this A.V.C.-controlled tube may be made as shown

circuit must be greater than for the tuning indicator in Fig. 4A. The meter movement should be selected to give nearly full-scale deflection to the left for a no-carrier-signal condition, in order to give maximum meter movement when a station is tuned in. Notice that by-pass condensers  $C_{P1}$  and  $C_{P2}$  are used in this arrangement to keep R.F. and I.F. currents out of the tuning indicator.

### Voltage-Controlled Visual Tuning Indicators

**Cathode Ray Tuning Indicator.** A special cathode ray tube which acts essentially as a high-resistance voltmeter is used in many receivers as a voltage-controlled visual tuning indicator. The D.C. voltage developed across the load resistor of a diode second detector-A.V.C. tube in a receiver is used to control the cathode ray tuning indicator, since this voltage varies with carrier level.

The operating principle of a cathode ray tuning indicator tube is essentially the same as that of the cathode ray oscilloscope tube. Electrons striking a surface coated with a fluorescent material (such as willemite) cause the surface to glow with a greenish light. The diode detector load voltage determines the amount of surface which glows at any time, and thus we have a simple and highly attractive tuning indicator.

In a cathode ray tuning indicator an oxide-coated cathode, heated by a filament, is mounted in the center of a cone-shaped anode having a coating of fluorescent material on its inner surface, as indicated in Fig. 5A. When a D.C. voltage is applied between the cathode and the anode, the electrons emitted by the heated cathode will strike the inner surface of the anode, called the *target*, and the entire inside of the circular cone or tar-

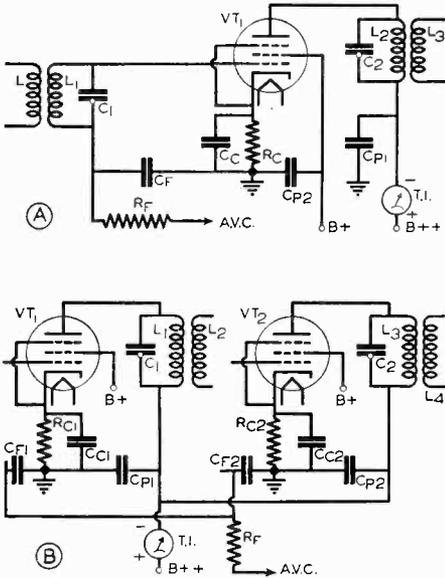


FIG. 4. Connections for a meter-type tuning indicator for a single A.V.C.-controlled stage (A) and for the common plate supply lead to two A.V.C.-controlled stages (B).

in Fig. 4A. Notice that by-pass condenser  $C_{P1}$  prevents R.F. and I.F. currents from passing through the tuning indicator. The resistance of the tuning indicator drops the D.C. plate voltage about 5 to 10 volts.

When the D.C. plate current to more than one A.V.C.-controlled amplifier stage is passed through the tuning indicator, connections into the plate supply lead going to these A.V.C.-controlled tubes may be as shown in Fig. 4B. Naturally the current range for the indicator in this

get will glow with a greenish light. In addition, a thin, vertical dart or vane is located between the target and the cathode, as shown in Fig. 5B. This dart is known as the *control*

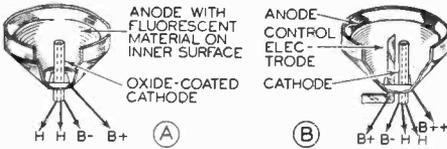


FIG. 5. Essential elements of a cathode ray tuning indicator tube.

*electrode*, for it determines how much of the target will glow.

When the control electrode is connected to the cathode, it has no electrical influence upon the electrons emitted by the cathode, but does have a physical blocking effect which prevents electrons from reaching that portion of the target directly behind it. We thus have a shadow in back of the control electrode, as indicated in Fig. 6A. Making the control electrode negative with respect to the cathode, as in Fig. 6B, widens this shadow, for the control electrode now has a repelling effect on the emitted electrons. When the control electrode is made positive with respect to the cathode, however, it serves to speed up the emitted electrons and to make them bend towards the shadow region, narrowing the shadow as indicated in

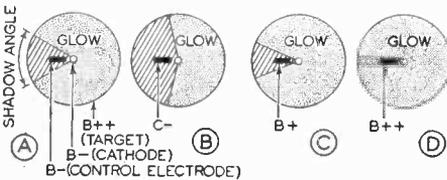
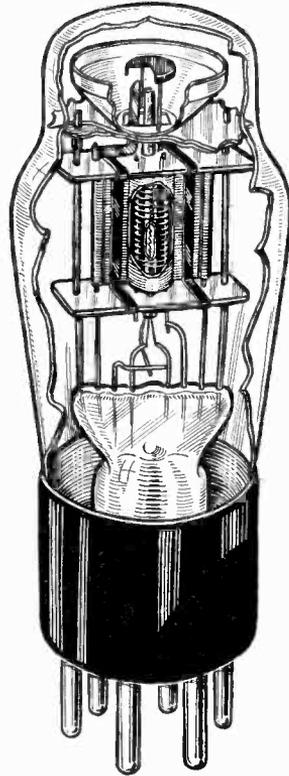


FIG. 6. Variation of shadow angle with control electrode potential in a typical cathode ray tuning indicator tube.

Fig. 6C. When the control electrode is highly positive, as in Fig. 6D, the shadow becomes very narrow or disappears entirely.

When a receiver having a diode de-

tector is tuned to resonance, the diode load voltage increases in a negative direction with respect to the chassis or ground. If we want the shadow angle to close up as a station is tuned in, we must convert this increasing negative voltage into an increasing positive voltage for the control electrode of the cathode ray indicator



Courtesy National Union Radio Corp.  
FIG. 7. Cutaway view showing construction of 6G5 cathode ray tuning indicator tube.

tube. This can quite easily be done by introducing a triode amplifier between the detector load and the control electrode of the indicator tube; in an actual cathode ray indicator tube like that shown in Fig. 7, this triode amplifier is built into the same envelope as the target system.

The schematic circuit diagram in Fig. 8 gives in simplified form exactly the same information as the pictorial sketch in Fig. 7. In addition, this

diagram shows that the target connects directly to the B++ D.C. supply source, while the plate of the triode tube connects to this source through resistor  $R$ , which is usually 1 megohm. The common cathode for the two tubes is grounded to the receiver chassis, and the A.V.C. voltage is applied between the control grid of the triode section and the cathode.

When a receiver which has a cathode ray tuning indicator is tuned be-

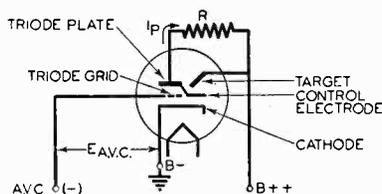


FIG. 8. Schematic symbol for a cathode ray tuning indicator tube. External resistor  $R$  is always used between the triode plate and the target. Circuit variations have little effect upon target current, since the target receives a constant D.C. supply voltage.

tween stations, there is no carrier signal and consequently there is no A.V.C. voltage developed across the detector load resistor. Under this condition the C bias for the triode section of the indicator tube is zero ( $E_{AVC}$  is zero) and D.C. plate current  $I_P$  for the triode section depends upon the B++ supply voltage, the value of resistor  $R$  and upon the plate-to-cathode resistance of the triode section. Since this plate-to-cathode resistance is quite low for zero grid bias, current  $I_P$  is high, and creates a high voltage drop across  $R$ . The triode plate voltage is consequently considerably less than the target voltage (because of this drop across  $R$ ). The control electrode (connected directly to the triode plate) likewise has considerably less voltage than the target when no station is tuned in, and the shadow angle is therefore large.

When the receiver is tuned to a desired station carrier signal (is tuned to resonance), the A.V.C. voltage de-

veloped across the detector load increases, driving the triode grid of the indicator tube negative and increasing the D.C. plate resistance of the triode section. The result is reduced D.C. plate current and a lowered voltage drop across resistor  $R$ . The triode plate voltage and the control electrode voltage become more nearly equal to the target voltage, and the increasing positive voltage on the control electrode decreases the shadow angle on the target. This shadow angle will be a minimum when the receiver is tuned exactly to resonance for a particular station. For distant stations the minimum shadow angle will be considerably greater than for local stations.

It is possible to adjust a cathode ray indicator tube circuit to make the shadow vanish for either small or large A.V.C. voltages. With circuits designed so the shadow closes on low A.V.C. voltages, however, the tuning in of strong carrier signals will completely close the shadow before resonance occurs and an exact indication of resonance will not be possible. On the other hand, if the circuit is designed so the shadow will completely close only for strong local carrier signals, then the change in shadow angle for weak signals may be inadequate for accurate tuning. (This is not a serious drawback, however, for it is ordinarily easy to tune in a *weak* station accurately by ear.)

The A.V.C. voltage is, of course, the primary control upon the shadow angle. The angle for zero A.V.C. voltage is governed essentially by the design of the tube, but the amount which the shadow closes for a given increase in A.V.C. voltage is determined by the target voltage and by the value of series resistor  $R$ .

Curve 1 in Fig. 9 tells how the shadow angle varies with A.V.C. voltage when the supply voltage is low

(100 volts) and the ohmic value of  $R$  is low (.5 megohm). Observe that the shadow angle is 90 degrees for zero A.V.C. voltage, and the shadow is completely closed ( $0^\circ$  shadow an-

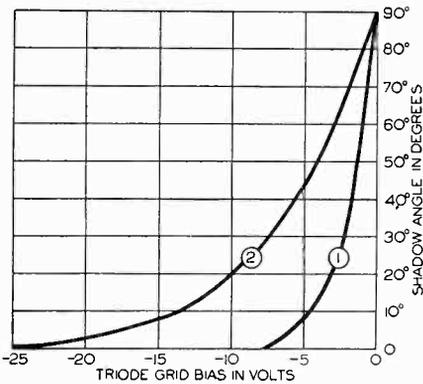


FIG. 9. This graph shows how shadow angle varies with grid voltage for a 6G5 cathode ray tuning indicator tube. Curve 1:  $R = .5$  megohm and  $B++$  voltage = 100 volts; curve 2:  $R = 1$  megohm and  $B++$  voltage = 250 volts.

gle) at an A.V.C. voltage of about  $-8$  volts (the curves in Fig. 9 apply to a 6G5 tube). Curve 2 in Fig. 9 is for a high plate supply voltage and a high value of  $R$ ; in this case the shadow does not close until the A.V.C. voltage has driven the triode tube grid to  $-23$  volts.

We can draw the following conclusions from the curves in Fig. 9: 1, to make the shadow close at a more negative grid bias value, increase the supply voltage to the tube; 2, to make the shadow close at a less negative value (so it will close for weak carrier signals), reduce the supply voltage to the tube. To secure the desired open shadow angle, usually about 90 degrees, adjust resistor  $R$  when no carrier signal is present. With these facts in mind, you will be able to adjust a cathode ray indicator tube for satisfactory operation on either local or distant stations. Most receivers are adjusted for a characteristic curve approximating curve 2 in Fig. 9, for this gives a reasonable change in

shadow angle at all reasonable carrier levels.

*Adding a Cathode Ray Indicator Tube to a Receiver.* A suitable circuit is shown in Fig. 10A; all parts which are added to the original receiver circuit are shown in heavy lines. Note that the grid of the 6G5 cathode ray tuning indicator tube is connected through a 1-megohm resistor  $R_1$  to the A.V.C. voltage source in the receiver. This connection could be to the minus terminal of the diode detector load, but since this terminal may furnish as much as  $-40$  volts when a strong local station is tuned in, it is better to place voltage divider  $R_2$ - $R_3$  across the diode load and make a connection to this.

The larger the value of  $R_2$  with respect to that of  $R_3$ , the lower will be the maximum A.V.C. voltage applied to the indicator tube.  $R_1$  prevents the 6G5 tube from shorting the diode load

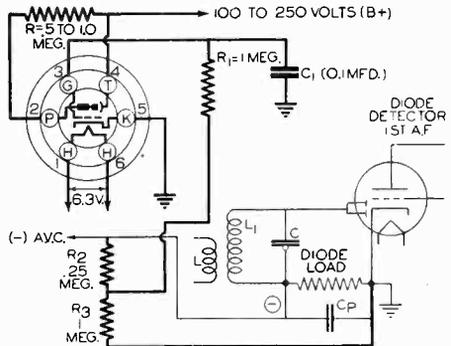


FIG. 10A. This diagram shows you how a cathode ray indicator tube would be connected between the A.V.C. supply terminal and ground in a receiver. The only extra parts required, in addition to the type 6G5 indicator tube and its standard 6-prong socket, are resistors  $R$ ,  $R_1$ ,  $R_2$  and  $R_3$ , and condenser  $C_1$ . You will sometimes find the control electrode and target represented as shown here rather than as in Fig. 8, so learn to recognize both types of diagrams.

or the voltage divider; in addition,  $R_1$  serves with  $C_1$  as an A.F. filter which prevents the shadow from flickering at an A.F. value.

*Wide-Angle Tuning.* The shadow angle for a no-signal condition in the

average cathode ray tuning indicator tube is  $90^\circ$ , but this can be increased approximately to  $180^\circ$  by inserting an ordinary triode amplifier tube between the A.V.C. supply terminal and the indicator tube in the manner shown in Fig. 10B.

When there is no carrier signal in the diode second detector circuit of a receiver, there is no A.V.C. voltage developed across the diode load, and consequently the grid of  $VT_1$  in Fig. 10B will be at zero bias. The plate

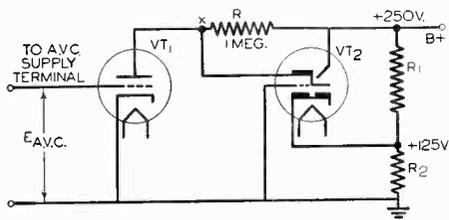


FIG. 10B. A shadow angle of approximately  $180^\circ$  can be secured with this cathode ray tuning indicator circuit.

current of this tube is then high, and the plate-cathode resistance of  $VT_1$  drops so low that point  $x$  is nearly at ground potential. Since the cathode of  $VT_2$  is  $+125$  volts with respect to ground, the triode plate and the control electrode of  $VT_2$  are  $-125$  volts with respect to the cathode of this tube; under this condition the shadow angle is approximately  $180^\circ$ .

When a highly negative A.V.C. voltage is applied to the grid of  $VT_1$  (tuning the receiver to a powerful local station would do this), the plate current of  $VT_1$  drops nearly to zero; under this condition the D.C. resistance of  $VT_1$  is so much higher than the 1-megohm resistance of  $R$  that point  $x$  practically assumes B+ potential of  $+250$  volts. This places the triode plate and the control electrode at about  $+125$  volts with respect to the cathode of  $VT_2$ , giving zero shadow angle. Weaker signals give shadow angles between  $0$  and  $180^\circ$ ,

with the change in shadow angle while tuning in a station being about twice as great as for an ordinary indicator circuit; this, of course, makes more accurate tuning possible.

### Unique Tuning Aids

Although the milliammeter and the cathode ray tuning indicator tube are the most widely used tuning aids, a number of other unique methods have been developed. Three of these will be considered at this time: 1, the G.E. Colorama indicator, using red and green lamps; 2, the neon lamp flasher; 3, the zero beat tuning indicator.

*G. E. Colorama Tuning Indicator.* The circuit diagram of the Colorama tuning indicator used in a number of General Electric receivers is shown in Fig. 11. The A.V.C. voltage developed by the diode detector in the receiver is applied to the control grid of an ordinary type 6C5 triode tube (designated as the Colorama tuning indicator tube). In the plate circuit of this tube is the primary of the Colorama tuning reactor (an iron-core transformer) while across the secondary of this transformer is an arrangement of three green lamps in series and four red lamps in series-parallel, connected as indicated in Fig. 11. Condenser  $C52$ , across the primary of the transformer, serves to filter out any A.F. components in the A.V.C. voltage;  $R28$  and  $C53$  are power pack supply filters.

When the A.V.C. voltage is low or zero, as it is when the receiver is tuned off resonance or between stations, the C bias on the 6C5 tube is zero and D.C. plate current for this tube is therefore quite high. This high plate current flowing through the primary of the Colorama tuning reactor saturates its iron core. *This core saturation lowers the reactance of the secondary winding, so that it is*

practically a short-circuit path for 60-cycle alternating current. Under this condition the green lamps are in effect shunted by the 15-ohm resistor  $R_{30}$ , and the current which is sent through the bank of red lamps and  $R_{30}$  by secondary winding  $S$  on the receiver power transformer is high enough to light all of the red lamps. We thus have the red lamps glowing whenever the A.V.C. voltage is reduced by tuning between stations.

Now let us see what happens when a station is tuned in. Tuning the receiver to resonance increases the negative A.V.C. voltage applied to the grid of the 6C5 tube, and the plate current of this tube drops. The core of the Colorama tuning reactor becomes less saturated as primary current drops, and consequently the inductance of secondary winding  $L_{28}$  increases. The reactance of this secondary winding at 60 cycles is correspondingly increased, reducing the shunting effect of  $R_{30}$  and thus increasing the resistance of the entire circuit across transformer winding  $S$ . Less current now flows through the red lamps, and they begin to grow dim.

At a critical negative A.V.C. voltage, full current (.150 ampere in this case) flows through the green lamps, lighting them brightly, while the current through a red lamp is only half of the circuit current, because of the series-parallel connection. Resistor  $R_{30}$  controls the maximum current which can flow through the red lamps when the receiver is off-tune, while resistor  $R_{29}$  controls the current which the green lamps can draw when the receiver is tuned to a station, and also controls to a certain extent the current for the red lamps. These resistors must be properly adjusted if the red lights are to illuminate the station selector dial when the receiver is off resonance, and the green lights

are to illuminate the dial when the receiver is at resonance.

*Neon Lamp Flasher Indicator.* A single neon lamp connected into a circuit which causes it to glow only when a station is tuned in, serves as the tuning indicator for a number of Silvertone receivers. The circuit diagram is shown in Fig. 12; an ordinary I.F. transformer with weak coupling

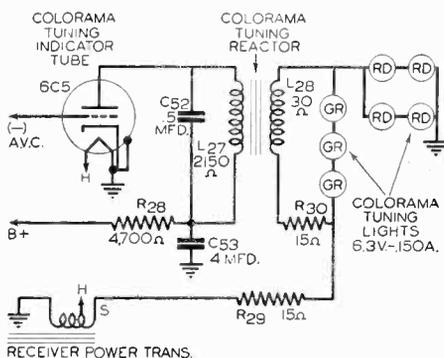


FIG. 11. Circuit diagram of the Colorama tuning indicator used in General Electric Models E-101, E-105 and E-106 receivers. The green lamps light up when the receiver is properly tuned; red lamps illuminate tuning dial whenever receiver is off tune. The reactance values given here for the primary and secondary ( $L_{27}$  and  $L_{28}$ ) of the Colorama tuning reactor are for 60 cycles A.C. Ohmmeter measurements will therefore give considerably lower values.

between the tuned primary and tuned secondary is connected between the primary of the last I.F. transformer and the 6B7 duo-diode-pentode which is designated as the flasher tube. Primary circuit  $L_1-C_1$  of this extra I.F. transformer forms a series resonant circuit which is connected across condenser  $C$  in the last I.F. transformer circuit, with the result that the small I.F. voltage drop across condenser  $C$  is applied to  $L_1-C_1$ . At resonance this series resonant circuit acts as a low resistance, and hence does not affect the plate load of the last I.F. stage. Loose coupling between  $L_1$  and  $L_2$  insures higher selectivity for resonant circuits  $L_1-C_1$  and  $L_2-C_2$  than could be secured in the I.F. amplifier, for

we are not concerned with fidelity in the tuning indicator circuit.

The I.F. voltage developed across resonant circuit  $L_2-C_2$  is rectified by the diode section of the 6B7 flasher tube, and a rectified voltage is developed across the 1-megohm diode load resistor. The control grid of the pentode section is connected to the negative terminal of this load resistor.

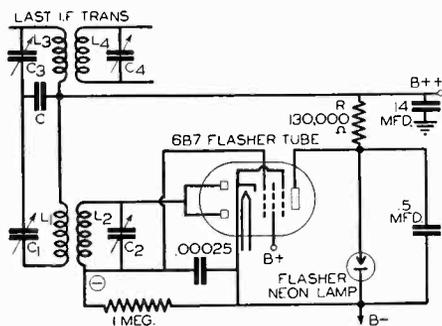


FIG. 12. Neon lamp flasher indicator circuit as used in Silvertone Models 1722 and 1732 receivers. The neon flasher lamp glows only when the receiver is tuned correctly.

The plate of the pentode section receives its D.C. voltage through 130,000-ohm resistor  $R$ ; between the plate and the cathode is the flasher neon lamp which glows only when its terminal voltage exceeds a definite value. Furthermore, the terminal voltage of this neon lamp is equal to the supply voltage minus the voltage drop in resistor  $R$ .

When the receiver is tuned off resonance, little or no I.F. voltage is applied to the diode section of the 6B7 flasher tube. As a result, no rectified voltage exists across the 1-megohm resistor, the C bias on the pentode section is practically zero, and plate current of the pentode section is high. This plate current produces a large voltage drop across resistor  $R$ , making the neon lamp voltage too low for it to glow. Since resonant circuits  $L_1-C_1$  and  $L_2-C_2$  are highly selective, the receiver must be tuned exactly to

resonance before enough I.F. signal can get through these circuits to excite the diode section and produce sufficient negative voltage across the 1-megohm resistor to reduce the pentode plate current to a low value. Reducing the plate current reduces the voltage drop across  $R$  and raises the neon lamp voltage. The lamp therefore glows when the receiver is properly tuned to a station. When the receiver is tuned rapidly from station to station, the lamp flickers each time a station is passed; this is why this particular arrangement is known as a flasher indicator.

*Zero Beat Tuning Indicator.* If the I.F. carrier signal produced by the mixer-first detector of a receiver is mixed with a locally-produced signal having exactly the correct I.F. value, and this mixing occurs just ahead of the second detector, an audio beat (low-frequency audio note) will be heard in the loudspeaker whenever the receiver is improperly tuned to a station.

When the receiver is properly tuned, the difference between the two I.F. signals will be less than 30 cycles, and the beat note will therefore be inaudible. When the receiver is tuned off a station, there will be no incoming I.F. signal to beat with the locally produced signal, and consequently no audio beat will be heard. A zero beat indicator of this type is entirely satisfactory provided the listener does not object to the squeal which occurs while tuning in a station. Furthermore, this indicator is effective regardless of whether the I.F. amplifier is peaked or band-passed. A zero beat tuning indicator is also of great value when tuning in continuous wave code signals, for it provides an audible tone for these signals.

The special local oscillator circuit which is used as a zero beat tuning indicator in one receiver is shown in

Fig. 13. The oscillator tank circuit is made up of  $L_1$  and  $C_1$ , with  $C_1$  being adjusted to make this circuit produce a frequency exactly equal to the I.F. value of the receiver. A.C. plate

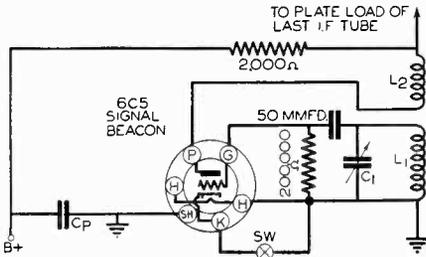


FIG. 13. Circuit of the R.F. oscillator which serves as a zero beat tuning indicator in the Grunow Model 12A-1241 receiver.

current flowing through coil  $L_2$  induces in  $L_1$  the feed-back voltage required for oscillation. The A.C. plate current produced by this oscillator flows through a 2,000-ohm resistor which is also in the plate supply lead of the last I.F. amplifier stage. The result is that an A.C. voltage is superimposed on the D.C. plate voltage applied to this last I.F. stage. Two signals, the exact I.F. signal produced by the oscillator and the I.F. signal produced by tuning the receiver to a station, reach the second detector. After detection, we have left the desired audio signal and a beat note equal to the difference between the two I.F. signals; when the receiver is accurately tuned to a station, this difference will be zero, the beat note will disappear and we have what is known as zero beat.

### Essential Sections of an A.F.C. System

The two essential sections of an A.F.C. system are: 1, the discriminator, which usually produces a positive D.C. control voltage for below-normal I.F. signal frequencies, and a negative D.C. control voltage for above-normal I.F. signal frequencies with

the value of this voltage being proportional to the amount of error in tuning; 2, the oscillator control circuit, which converts the D.C. control voltages (produced by the discriminator) into changes in the effective inductance shunting the oscillator coil, thereby compensating for errors in tuning. The positions of these sections with respect to the other stages of a conventional superheterodyne receiver having A.F.C. are clearly shown by the box diagram in Fig. 14. Note that the I.F. amplifier feeds both the discriminator and the second detector.

### The Discriminator Circuit

*Typical Discriminator Circuit.* The easiest way to see how the discriminator circuit in an A.F.C. system can furnish a D.C. control voltage of suitable polarity is to consider an actual discriminator circuit, as given in Fig. 15. We will first "get our bearings" by analyzing this circuit in a general manner, after which we will be ready to consider in detail the manner in which the D.C. control voltage is produced.

In the input circuit for the discriminator tube in Fig. 15 we find parallel resonant circuit  $L_P-C_P$  serving as the plate load for the last I.F. stage, and inductively coupled to series resonant

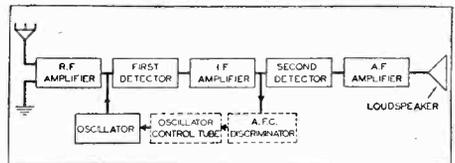


FIG. 14. The two sections shown in dotted lines must be added to a superheterodyne receiver in order to secure automatic frequency control.

circuit  $L_1-L_2-C$ . Both of these resonant circuits are tuned to the exact I.F. value of the receiver. The coil in the series resonant circuit is split into two sections of equal size  $L_1$  and  $L_2$ , and

the plate (high R.F.) terminal of the parallel resonant circuit is connected to the common terminal of these coils through D.C. blocking condenser  $C_B$ .

The discriminator tube is a double diode rectifier tube (a 6H6 tube is most generally used), with the plate of each diode section connected to one terminal of the series resonant circuit and the cathodes connected together through equal-value resistors  $R_1$  and  $R_2$ . Each resistor is shunted by a by-pass condenser ( $C_1$  and  $C_2$ ) and one cathode is grounded directly, hence

age which we will designate as  $e_B$ . This voltage acts in series with  $L_1$ ,  $L_2$  and  $C$  in our series resonant circuit, causing a current to flow through the two coil sections. This current develops I.F. voltage  $e_1$  across coil section  $L_1$  and develops I.F. voltage  $e_2$  across coil section  $L_2$ . For any condition of tuning, voltage  $e_1$  will always be equal to  $e_2$  in magnitude.

We can now see that I.F. voltage  $e_1$  acts in series with I.F. voltage  $e_P$  on diode section  $D_1$ , and the resulting rectified electron current  $i_1$  flows

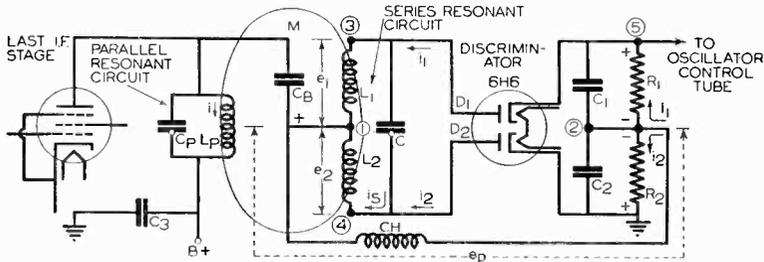


FIG. 15. Conventional discriminator circuit.

both cathodes of the discriminator tube are at ground I.F. potential.

Notice now that the I.F. voltage developed across the parallel resonant circuit in the last I.F. stage is applied across choke coil  $CH$ , this I.F. voltage being designated as  $e_P$  (condenser  $C_B$  provides a direct path for I.F. current from the plate end of the parallel resonant circuit to point 1 and one terminal of the choke coil, and the path from the other end of the parallel resonant circuit is through by-pass condenser  $C_3$  to the receiver chassis, through the chassis to the grounded end of  $R_2$  and then through by-pass condenser  $C_2$  to point 2 and the other terminal of the choke coil). This I.F. voltage  $e_P$  developed across choke coil  $CH$  is in turn applied to points 1 and 2 in the discriminator circuit.

Current  $i$ , flowing through the coil of the parallel resonant circuit, induces into secondary coil  $L_1$ - $L_2$  a volt-

age through  $R_1$ , developing across this resistor a D.C. voltage having the polarity indicated in Fig. 15. Likewise,  $e_2$  and  $e_P$  act upon diode section  $D_2$ , and rectified electron current  $i_2$  produces a D.C. voltage across  $R_2$  with the polarity shown.

Since we are dealing with A.C. voltages in this discriminator circuit, phase must of course be taken into account when we consider their combined effects.

*Correct Tuning.* When the phase difference between  $e_1$  and  $e_P$  is the same as that between  $e_2$  and  $e_P$ , the net voltages acting upon diode sections  $D_1$  and  $D_2$  will be equal in magnitude; equal values of rectified current will then flow through the two resistors, making the D.C. voltage drop across  $R_1$  equal to that across  $R_2$ . The net D.C. control voltage produced across these two resistors (between point 5 and ground) will be

zero since these voltage drops are of opposite polarity, and the A.F.C. system will have no effect upon the oscillator. This is, of course, the condition for correct tuning of the receiver to a station.

**Incorrect Tuning.** When the frequency of the received I.F. signal does not correspond to the resonant frequency of the tuned circuits in the

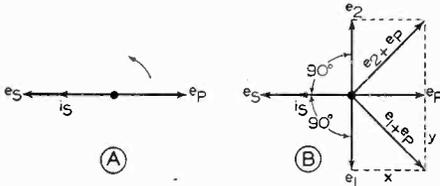


FIG. 16. Vector diagrams showing conditions in the discriminator circuit when the receiver is correctly tuned.

discriminator (because of incorrect tuning),  $e_1$  and  $e_2$  no longer have the same phase relationship with  $e_p$ , and consequently the A.C. voltages acting upon the diode sections are different in magnitude. Unequal rectified currents through  $R_1$  and  $R_2$  produce unequal voltage drops which, when combined, leave the desired D.C. control voltage to act upon the oscillator control circuit and correct for the error in tuning.

Since any detailed analysis of the action of an A.F.C. system must necessarily involve phase, it will be well at this time to review the fundamental facts about phase which you have already studied.

### Review of Phase

All the information you need know about the phase relationships of voltages and currents for the parts in an A.F.C. circuit is given in Chart 1, page 27. The essential facts presented by this chart, particularly by the vector diagrams in the fourth column, are:

**Resistors:** The voltage across a resistor is in phase with the current through it.

**Coils:** The voltage across a perfect coil leads the current through it by  $90^\circ$  (the current therefore lags the voltage by  $90^\circ$ ).

**Condensers:** The voltage across a perfect condenser lags the current through it by  $90^\circ$  (the current therefore leads the voltage by  $90^\circ$ ).

**Transformers:** The voltage induced in the secondary winding of a transformer (the open-circuit secondary voltage) is  $180^\circ$  out of phase with the primary voltage.

### Discriminator Circuit Action

When the I.F. Signal Frequency is Correct. We will first analyze the discriminator circuit in Fig. 15 for the condition where the input I.F. signal frequency exactly corresponds to the I.F. value of the receiver. In this case parallel resonant circuit  $C_p-L_p$  and series resonant circuit  $L_1-L_2-C$  will be resonant to the incoming signal. To determine the net A.C. voltages acting on diode section  $D_1$ , we must add together A.C. voltages  $e_p$  and  $e_1$  vectorially, so as to take into account the phase relationship of these two voltages. Likewise we must add together  $e_2$  and  $e_p$  vectorially to find the net A.C. voltage acting upon diode section  $D_2$ . First of all, we must choose some voltage or current for reference purposes; since  $e_p$  is common to all circuits under study, we shall use it as our reference voltage. To fix this fact in mind, we draw our vector  $e_p$  along the reference line in our vector diagram, as in Fig. 16A, using any convenient scale to determine its length.

The voltage  $e_s$  which is induced in the secondary of the discriminator transformer is  $180^\circ$  out of phase with the voltage across the primary (see diagram  $P$  in Chart 1), and consequently we can say that  $e_s$  is  $180^\circ$  out of phase with  $e_p$ . We therefore place vector  $e_s$  on our vector diagram

in the opposite direction from  $e_P$ , as indicated in Fig. 16A.

Since the frequency of induced secondary voltage  $e_s$  is exactly the same as the resonant frequency of series resonant circuit  $L_1-L_2-C$ , this circuit acts like a resistance at resonance and  $e_s$  will send through the circuit a current  $i_s$  which is *in phase* with  $e_s$ , as shown in Fig. 16A. Current  $i_s$  flows through  $L_1$  and  $L_2$ , developing across each of these coil sections an A.C. voltage which leads the current by  $90^\circ$  (see diagram *H* in Chart 1).

Rather than confuse the diagram in Fig. 16A by adding more vectors to it, let us redraw it in Fig. 16B and place vectors  $e_1$  and  $e_2$  on this new diagram. Before these vectors can be drawn, however, one other factor must be taken into account. For any direction of current flow  $i_s$  through  $L_1$  and  $L_2$ , one of the coil voltage drops will be opposite in polarity ( $180^\circ$  out of phase) with the other coil voltage drop in so far as  $e_P$  is concerned. This is because  $e_P$  acts in opposite directions through the two coils (it acts in the direction from point 1 to point 3 through  $L_1$  and from point 1 to point 4 through  $L_2$ ). For this reason, if we show voltage drop  $e_1$  as leading  $i_s$  by  $90^\circ$ , we must show  $e_2$  as lagging  $i_s$  by  $90^\circ$ , making  $e_1$   $180^\circ$  out of phase with  $e_2$ . (We could just as well make  $e_2$  lead  $i_s$  by  $90^\circ$  and make  $e_1$  lag  $i_s$  by  $90^\circ$ , for the same results would be secured.) Now we can draw in vector  $e_1$  and  $e_2$ , as shown in Fig. 16B.

The next step is to find the net A.C. voltage acting upon diode section  $D_1$  through resistor  $R_1$ . As was said before, this voltage will be equal to  $e_P + e_1$  with phase taken into account. We can add these two voltages quite easily on the vector diagram in Fig. 16B; we simply complete the parallelogram (rectangle) of which  $e_1$  and  $e_P$  form two sides, as indicated by the dotted lines  $x$  and  $y$  in Fig.

16B, and then draw a line from the center of our vector diagram to the intersection of these dotted lines. This line now represents the vectorial sum of voltages  $e_P$  and  $e_1$ , so we label this vector in this way. The length of vector  $e_1 + e_P$  corresponds to the magnitude of the A.C. voltage acting upon diode section  $D_1$ . In a similar manner we determine that vector  $e_2 + e_P$  is the net A.C. voltage acting upon diode section  $D_2$ .

These net A.C. voltages acting upon the diode sections will always be exactly equal when the I.F. signal frequency corresponds exactly to the I.F. value of the receiver. The diode sections  $D_1$  and  $D_2$  will consequently pass currents of equal value, and the D.C. voltage developed across  $R_1$  by diode current  $i_1$  will exactly equal the D.C. voltage developed across  $R_2$  by diode current  $i_2$ . Note that these currents flow in opposite directions through  $R_1$  and  $R_2$ , making the voltage drops across these resistors have the polarities indicated. The net D.C. voltage developed across the two resistors (between point 5 and ground) for the oscillator control circuit is therefore zero whenever the I.F. signal is of the exact value.

*When the I.F. Signal Frequency is High.* When the I.F. signal frequency entering the discriminator circuit is higher than the I.F. value of the receiver (higher than the resonant frequencies of  $L_P-C_P$  and  $L_1-L_2-C$ ), we will naturally expect the discriminator circuit to produce a D.C. control voltage for the oscillator control circuit. Let us see how this is done.

Primary voltage  $e_P$  will again be used as our reference voltage for the vector diagram. The induced voltage  $e_s$  will again be  $180^\circ$  out of phase with the primary voltage  $e_P$ , so we draw the vectors for these two values as in Fig. 17A.

Since the frequency of  $e_s$  is higher

than the resonant frequency of series resonant circuit  $L_1-L_2-C$ , the reactance of coils  $L_1-L_2$  will be greater than the reactance of condenser  $C$ , and the entire series resonant circuit will act as an inductance whose reactance is equal to the difference between the above-mentioned reactances. We thus have voltage  $e_s$  acting upon a coil; if there were no resistance in this circuit, we could say that circuit current  $i_s$  lags the applied voltage by  $90^\circ$  (Chart 1). Since this series resonant circuit has a certain amount of A.C. resistance, however, and since the net inductance is quite low near resonance, the current  $i_s$  will lag the voltage  $e_s$  by less than  $90^\circ$ . For explanation purposes, let us assume that series resonant circuit current  $i_s$  lags  $e_s$  by  $45^\circ$ . We therefore draw vector  $i_s$  lagging behind  $e_s$  by  $45^\circ$ , as indicated in Fig. 17A.

Regardless of the phase relation between  $i_s$  and  $e_s$ , the voltage drops across coil sections  $L_1$  and  $L_2$  will lead current  $i_s$  by  $90^\circ$ , and since one voltage drop acts in the opposite direction from the other in so far as  $e_p$  is concerned, we can say that A.C. voltage  $e_1$  leads circuit current  $i_s$  by  $90^\circ$ , and A.C. voltage  $e_2$  lags  $i_s$   $90^\circ$ . We will leave Fig. 17A as it is, and redraw the vectors in Fig. 17B for the complete vector diagram. Vectors  $e_1$  and  $e_2$  are now placed on this diagram.

Again we add  $e_1$  and  $e_p$ , taking phase into account, to find the net A.C. voltage acting upon diode section  $D_1$ . We do this by drawing lines  $x$  and  $y$  in Fig. 17B to complete the parallelogram having  $e_1$  and  $e_p$  for two of its sides. The diagonal line drawn from the center of the vector diagram to the intersection of these dotted lines now represents the net A.C. voltage,  $e_1 + e_p$ , acting upon diode section  $D_1$ .

In the same manner we add A.C.

voltages  $e_2$  and  $e_p$  in Fig. 17B, getting the somewhat longer vector  $e_2 + e_p$  as the net A.C. voltage acting upon diode section  $D_2$ . We can see immediately that vector  $e_2 + e_p$  is longer than vector  $e_1 + e_p$ ; this means that a higher voltage acts upon diode section  $D_2$  than upon  $D_1$  in Fig. 15, and a higher rectified current therefore flows through  $R_2$  than through  $R_1$ . The D.C. voltage drop across  $R_2$  will therefore be greater

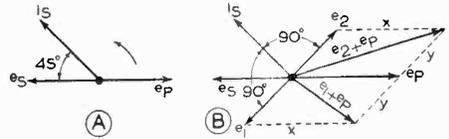


FIG. 17. Vector diagrams showing conditions in the discriminator circuit when the I.F. signal frequency is higher than the I.F. value of the receiver.

than the D.C. voltage drop across  $R_1$ , and only a part of the drop across  $R_2$  will be canceled out by the drop across  $R_1$ . This leaves point 5 negative with respect to ground and this negative D.C. voltage constitutes the D.C. control voltage applied to the oscillator control circuit. The higher the I.F. signal is above the I.F. value of the receiver, up to a certain limit, the higher will be this negative D.C. control voltage.

*When the I.F. Signal Frequency is Low.* If we repeat our analysis of the discriminator circuit in Fig. 15 and draw a vector diagram for the condition where the I.F. signal frequency is lower than the I.F. value of the receiver, we would find that net A.C. voltage  $e_1 + e_p$  would be larger than  $e_2 + e_p$ , with the result that diode section  $D_1$  produces a higher rectified voltage across  $R_1$  than diode  $D_2$  does across  $R_2$ . The drop across  $R_2$  would cancel out only a part of the drop across  $R_1$ , with the result that the D.C. control voltage between point 5 and ground would have the same polarity as the drop across  $R_1$ . The

net D.C. control voltage is therefore *positive* with respect to the ground or chassis when the I.F. signal frequency is *lower* than the I.F. value of the receiver.

*S Curve for Discriminator Circuit.* The manner in which the D.C. control voltage produced by the discrimi-

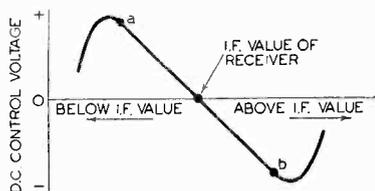


FIG. 18. S-curve showing output characteristics of a discriminator circuit.

nator varies with the I.F. signal frequency is usually expressed by radio engineers in the form of a graph like that in Fig. 18. The curve on this graph is called an *S curve*. Note that a positive D.C. control voltage is produced for I.F. signal frequencies below the I.F. value of the receiver, and a negative D.C. control voltage is produced for I.F. signal frequencies above the I.F. value; this verifies the results of our analysis of the circuit in Fig. 15. Observe also that the D.C. control voltage increases quite uniformly as we go above or below resonance, up to points *a* and *b*. The part of the curve between these two points is the operating range for the discriminator; slightly beyond *a* and *b*, further deviations from the resonant frequency do not give additional increases in D.C. control voltage. As a result, there is insufficient control beyond these points and A.F.C. action is not complete. From a practical standpoint, this means that I.F. signal frequencies between points *a* and *b* will be properly corrected by the A.F.C. system, while frequencies outside this range will not be "pulled in" satisfactorily.

## The Oscillator Control Circuit

The D.C. control voltage developed by the discriminator (sometimes called the *discriminator voltage*) must be converted by the oscillator control circuit into an action which will increase or decrease the oscillator frequency the correct amount to make the I.F. signal frequency exactly equal to the I.F. value of the receiver. Since the frequency developed by the oscillator circuit is essentially controlled by the capacity of the oscillator tank circuit condenser and the inductance of the tank coil, this D.C. control voltage must be converted into a capacity or inductance which, when applied to the oscillator resonant circuit, will give the necessary change in frequency. In a practical oscillator control circuit, the vacuum tube is usually made to act as an inductance, for this produces a more uniform A.F.C. action over the entire band.

Before taking up an actual oscillator control circuit, let us see how we can tell when a vacuum tube is acting as a coil. We know that when a coil is connected to an A.C. voltage source, the current drawn by the coil will lag the voltage by  $90^\circ$ . Suppose we had a box with two terminals, and some unknown electrical device inside. If we connected this box to an A.C. voltage source and found that the current through the box lagged the voltage by  $90^\circ$ , we would immediately say that the box acted like a coil. As far as the A.C. voltage source is concerned, this box is behaving as an inductance.

When we increase the inductance of an ordinary coil, the reactance of the coil increases and as a result, the current through the coil is reduced. In other words, if the coil draws a low current, it has a high inductance; if the coil draws a high current, it has a low inductance. If, now, our imaginary box draws a high current which

lags the voltage by  $90^\circ$ , we would say that this box has a low inductance; if the box draws a low current, we would say that it has a high inductance. You will soon see that we can consider the oscillator control circuit as an imaginary box having two terminals which are connected across one of the coils in the oscillator resonant circuit.

Now let us review a few fundamental facts about coils. When one coil is connected in parallel with the

Note that the D.C. control voltage produced by the discriminator is applied to the control grid of the type 6J7 oscillator control tube through resistor  $R_2$ . This D.C. control voltage, acting in series with the automatic C bias voltage produced by  $C_O$  and  $R_O$ , determines the average D.C. plate current for the oscillator control tube. Turning now to the oscillator circuit, coil  $L_O$  and tuning condenser section  $C_O$  form the oscillator tuned circuit, with trimmer condenser  $C_{PD}$  serving

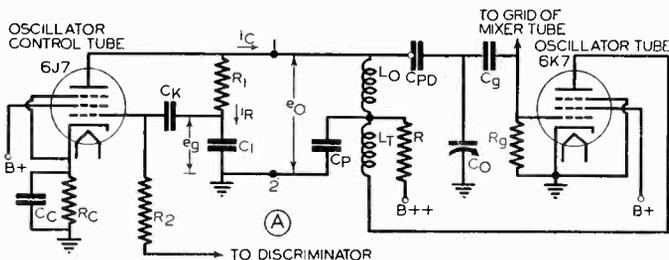


FIG. 19A. Conventional oscillator control circuit (to the left of points 1 and 2) and receiver oscillator circuit which it controls (to the right of points 1 and 2).

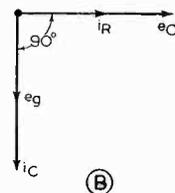


FIG. 19B. Vector diagram for oscillator control circuit.

other, the inductance of the combination is less than that of either coil. Increasing the inductance of one of these coils increases the inductance of the combination, and likewise, decreasing the inductance of one coil decreases the inductance of the combination. An increase in inductance lowers the resonant frequency of a resonant circuit; a decrease in inductance increases the resonant frequency.

*General Analysis of Oscillator Control Circuits.* Now we are ready to investigate the behavior of the oscillator control circuit, to see how the D.C. control voltage produced by the discriminator can make this circuit act like an inductance and can change its effective inductance to correct for errors in tuning. A typical oscillator control circuit, along with the special superheterodyne oscillator circuit upon which it acts, is shown in Fig. 19A.

as the oscillator padder. Condenser  $C_g$  is the R.F. grid coupling condenser, and serves together with  $R_g$  to produce the rectified D.C. grid voltage for the 6K7 oscillator tube. Coil  $L_T$  is connected between the plate and grid circuits of the oscillator, and therefore provides feed-back; the upper end of this coil is grounded for R.F. current through by-pass condenser  $C_p$ . The oscillator tube receives its D.C. plate voltage through feed-back coil  $L_T$ , while the oscillator control tube receives its D.C. plate voltage through oscillator tank coil  $L_O$ .

Resistor  $R_1$  and condenser  $C_1$  in the oscillator control circuit are connected directly across the oscillator coil  $L_O$  in so far as R.F. currents are concerned, and form what is known as the *phase-shifting network*. This network receives the full R.F. voltage developed by the oscillator circuit, for  $C_p$  is an R.F. by-pass condenser. The resist-

ance of  $R_1$  is so much greater than the reactance of condenser  $C_1$  that we can consider this phase-shifting network as essentially a resistance. This means that the oscillator coil voltage  $e_o$  will send through  $R_1$  and  $C_1$  an R.F. current  $i_R$  which is essentially in phase with  $e_o$ .

Let us again resort to vector diagrams, to avoid the need for keeping phase relationships in mind while studying the circuit. We will use  $e_o$  as our reference vector, and it is therefore drawn as shown in Fig. 19B. Vector  $i_R$  can now be placed, also along the reference line since it is in phase with  $e_o$ .

The flow of R.F. current  $i_R$  through  $C_1$  develops across this condenser an R.F. voltage  $e_g$  which lags  $i_R$  by  $90^\circ$  (see diagram  $L$  in Chart 1). We therefore draw vector  $e_g$  lagging behind  $i_R$  by  $90^\circ$ . This R.F. voltage  $e_g$  is applied to the control grid of the oscillator control tube through D.C. blocking condenser  $C_g$ , causing A.C. plate current  $i_o$  to flow through the oscillator control tube. This plate current will be in phase with the A.C. grid voltage, for this is a conventional amplifier circuit action in which a positive increase in the grid input voltage results in an increase in the plate current. We therefore draw vector  $i_o$  in phase with vector  $e_g$ , as in Fig. 19B. Our vector diagram now shows clearly that R.F. current  $i_o$  lags oscillator coil voltage  $e_o$  by  $90^\circ$ . If we consider that part of the circuit to the left of points 1 and 2 in Fig. 19A as a device having two terminals, we can readily see that the current  $i_o$  which is drawn by this device will lag by  $90^\circ$  the voltage  $e_o$  which is applied to the device. We have thus shown that the oscillator control circuit acts as an inductance shunting the oscillator coil  $L_o$ .

Now let us see how the D.C. control voltage produced by the discrimina-

tor will affect the value of this inductance shunting  $L_o$ . If the D.C. control voltage which is applied to the grid of the oscillator control tube through  $R_2$  is positive with respect to ground, the net bias on this grid will become less negative, increasing the D.C. plate current of the oscillator control tube. Since this 6J7 tube is operated on the curved portion of its  $E_g-I_p$  characteristic curve, the increase in D.C. plate current produced by a positive D.C. control voltage moves the operating point for the tube to a steeper portion of the  $E_g-I_p$  characteristic. This means that a given A.C. voltage  $e_o$  on the grid of the tube will produce a larger A.C. plate current  $i_o$ , making this stage act as a smaller inductance. If the D.C. control voltage is negative, the net C bias voltage on the 6J7 tube becomes more negative. This shifts the operating point nearer to plate current cutoff, and  $e_o$  produces a low A.C. plate current, making this stage act as a larger inductance. Clearly, *the D.C. control voltage produced by the discriminator serves to change the effective inductance of the oscillator control circuit.*

Now, for the first time, we can consider the action of the entire A.F.C. system. When the receiver is properly tuned, so that the I.F. signal frequency corresponds to the resonant frequency of the discriminator resonant circuit, no D.C. control voltage is produced by the discriminator and the only C bias acting upon the oscillator control tube is that produced by  $C_g$  and  $R_c$ . This bias sets the operating point for the tube at a point which allows a medium value of A.C. plate current  $i_o$  to flow. The inductance corresponding to this A.C. plate current, acting in shunt with the oscillator inductance  $L_o$ , has been allowed for in the design of the oscillator, and consequently the A.F.C. system can

be considered inactive when the receiver is properly tuned.

When tuning is such that the I.F. signal frequency is below the I.F. value of the receiver, the discriminator produces a positive D.C. control voltage which increases the A.C. plate current and decreases the inductance effect of the oscillator control tube, thereby decreasing the shunt inductance across the oscillator tank coil and increasing the oscillator frequency.

When the I.F. signal frequency is higher than the correct value for the I.F. amplifier, the discriminator will produce a negative D.C. control voltage which makes the net negative C bias on the oscillator control tube more negative. The result is a decrease in A.C. plate current  $i_0$ ; this corresponds to an increase in the effective inductance shunting oscillator inductance  $L_0$ , and this action of course lowers the oscillator frequency just enough to compensate for incorrect tuning.

The A.F.C. system cannot correct completely for errors in tuning, for then there would be no D.C. control voltage for correction purposes. The response of the I.F. amplifier is always broad enough to allow for small errors in tuning, however, and consequently the A.F.C. action is satisfactory for all practical purposes.

### Typical A.F.C. Circuits

Although the circuits in Figs. 15 and 19 have given the basic principles of A.F.C. systems, you will find a great many variations of these circuits. Before analyzing a number of typical circuits to familiarize you with these variations, let us consider briefly those components of each section which are to be found in all A.F.C. systems.

The discriminator section can be identified on the circuit diagram of a

receiver having A.F.C. by the fact that it will have a *double-diode vacuum tube, with the two diode plates connected into a tuned circuit having a split or center-tapped coil which is fed with the I.F. amplifier output voltage both inductively and by a direct connection to the center tap.* Each diode section thus gets two distinct I.F. voltages: 1, half of the I.F. output voltage of this resonant circuit; 2, the I.F. amplifier output voltage. The phase relationship between the voltages determines what the resultant I.F. voltage acting on each diode section will be.

In general, the oscillator control tube will have C bias voltages produced by two different sources: 1, by a cathode circuit resistor; 2, by the discriminator, feeding through a resistance-capacitance filter which keeps out R.F. components. Furthermore, the oscillator control circuit will have a phase-shifting network, usually made up of a resistor in series with a condenser, the combination being in shunt with the coil of the oscillator resonant circuit. The condenser in this phase-shifting network will have a very low capacity, ordinarily from about 2 mmfd. to 20 mmfd. Occasionally the grid-to-cathode inter-electrode capacity of the oscillator control tube will be used in place of a separate phase-shifting condenser.

With these general ideas in mind, we are ready to consider a few of the unique variations of the basic A.F.C. circuit just studied.

*General Electric A.F.C. System.* In Fig. 20 is shown the circuit diagram of the A.F.C. system and associated circuits used in General Electric Models E-101, E-105 and E-106 all-wave superheterodyne receivers. The A.F.C. system follows very closely the fundamental circuit presented in Figs. 15 and 19, except that the discriminator in this General Electric

circuit also serves as second detector and A.V.C. tube.

The output of the I.F. amplifier feeds into the discriminator circuit, through a conventional split-secondary I.F. transformer with which you are already familiar (Fig. 15). Notice, however, that the resistor between the cathodes of the double-diode tube is divided into three sections rather than two. Point 2 is the

of low voltage rating, which provides an entirely satisfactory ground for I.F. and A.F. signals. The use of this condenser permits connecting point 1 through a 100,000-ohm resistor to a -3 volt terminal in the power pack voltage divider. In this way all A.V.C.-controlled tubes in the receiver get an initial C bias of -3 volts, eliminating the need for automatic C bias resistors in some of the

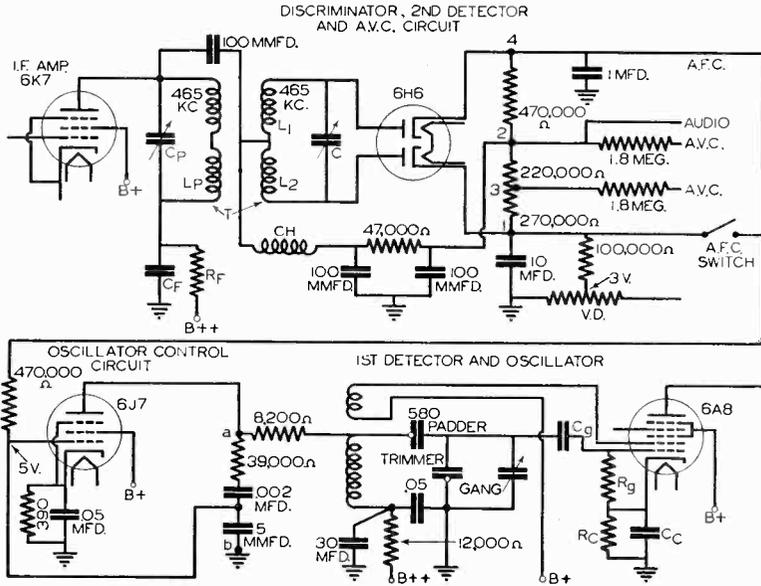


FIG. 20. Circuit diagram of the A.F.C. system used in General Electric Models E101, E105 and E106 receivers.

midpoint of this resistor network. Between this point and the upper cathode (point 4) is a 470,000-ohm load resistor, while between point 2 and point 1, the lower cathode, is a 220,000-ohm resistor in series with a 270,000-ohm unit, these together being approximately equal to the 470,000-ohm resistor. The tap at point 3 serves to provide a lower A.V.C. voltage for one or more of the A.V.C.-controlled stages in the receiver.

Ordinarily we would expect the circuit to be grounded to the chassis at point 1; instead, however, it is grounded through a 10 mfd. condenser

cathode circuits. The 100,000-ohm resistor and 10 mfd. condenser also serve to keep power pack hum out of the discriminator circuit.

The cathode of the upper diode section is grounded through a 1 mfd. condenser, and the cathodes of both diode sections are therefore at ground R.F. potential. It is not essential that condensers of equal capacity be across each diode load resistor. The A.V.C. sources feed through 1.8-megohm A.V.C. filter resistors.

A resistance filter made up of two 100 mmfd. condensers and a 47,000-ohm resistor is placed between point



tance applied to the oscillator resonant circuit.

With these facts in mind, you should have no difficulty in completing an analysis of this General Electric A.F.C. system. It reacts in exactly the same way as the circuits in Figs. 15 and 19 to I.F. signals which are above or below the I.F. value of the receiver.

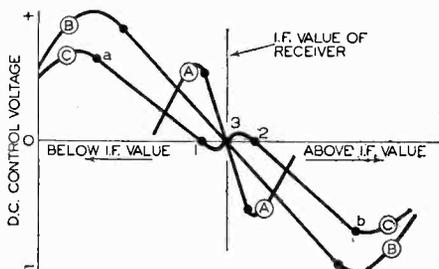


FIG. 22. Types of S-curves obtained with discriminator transformers having various degrees of coupling.

*Discriminator Transformer Design Problems.* The conventional discriminator transformer used in the circuits of Figs. 15 and 20 require careful design. Weak coupling between the primary winding and the split secondary winding makes these two tuned circuits highly selective, giving the S-curve marked A in Fig. 22. The steepness of the linear region of this curve indicates that stations will be pulled in very close to the I.F. value (to point 3); the small frequency difference between the ends of the linear region means that manual tuning must be quite accurate before A.F.C. will take hold, and the relatively low magnitudes of D.C. control voltages provided by curve A may not be sufficient for full correction.

Approximately critical coupling gives S-curve B, indicating greater D.C. control voltages and a greater frequency range over which A.F.C. will pull in a station; in fact, the linear range may be too wide, allowing an undesired station to be dragged

over an adjacent desired station. The linear section of the curve is less steep, indicating that A.F.C. will not correct as completely as with curve A, but a deviation of a few hundred cycles from resonance is ordinarily quite permissible.

When coupling is greater than the critical value, the two tuned circuits interact, giving a double-peak response curve which results in the undesirable S-curve at C in Fig. 22. This curve tells us that I.F. signals which are outside of the region between points 1 and 2 will be pulled in to a frequency corresponding to these points rather than to the correct I.F. value at point 3.

The curves in Fig. 22 show clearly that the design of a discriminator transformer must be a compromise between the various desired performance characteristics. In addition, the receiver must have a good A.V.C. system, in order that signals which are received at different signal levels will produce almost identical voltages in the discriminator circuit. If this were not true the amount of correction provided for mistuning would vary with the strength of the signal picked up.

*Silvertone (Sears Roebuck) A.F.C. System.* The manufacturers of Silvertone receivers, as well as a number of other manufacturers, have overcome these discriminator transformer design problems by using a triple-tuned circuit ahead of the discriminator, as shown in Fig. 23. The three resonant circuits,  $L_1-C_1$ ,  $L_2-L_M-C_2$ , and  $L_3-L_4-C_3$  are all peak-tuned to the I.F. value of the receiver. Circuit  $L_3-L_4-C_3$  is designed with the correct Q factor to give the desired S characteristic, and the coupling between  $L_M$  and  $L_3-L_4$  is made close enough to give the required voltages across  $L_3$  and  $L_4$  for the diode sections.

Another unique feature of this Silvertone A.F.C. system is the absence of the choke coil ordinarily found between points 1 and 2. In this particular circuit the choke coil is unnecessary because the required voltage is developed by the middle resonant circuit ( $L_2-C_2-L_M$ ) which is di-

lator control circuit. The connection is through a 1-megohm resistor and .1 mfd. condenser which serve as the A.F.C. filter. The time constant of this filter is greater than that of the A.V.C. filter, as is required.

The oscillator control circuit in Fig. 23 is of conventional design, with  $R_N$

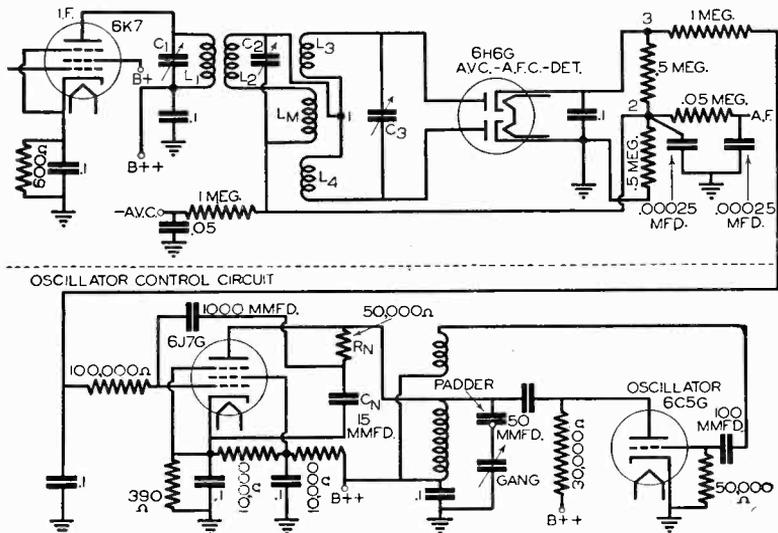


FIG. 23. Circuit diagram of the A.F.C. system used in the Silvertone model 4587 receiver.

rectly connected between points 1 and 2. Point 2, being negative with respect to ground and having a potential which is proportional to the rectified current flowing through the lower diode load resistor, serves as the A.V.C. terminal. The 1-megohm resistor and .05 mfd. condenser make up the A.V.C. filter which removes A.F. and I.F. components from the A.V.C. voltage and at the same time determines the time constant of the A.V.C. system. Point 2 also serves as the A.F. voltage supply terminal, with the .05 megohm resistor and the two .00025 mfd. condensers keeping I.F. signals out of the audio amplifier. Terminal 3 on the diode load resistor network is, of course, the point from which the D.C. control voltage for A.F.C. purposes is fed into the oscil-

lator control circuit. The connection is through a 1-megohm resistor and .1 mfd. condenser which serve as the A.F.C. filter. The time constant of this filter is greater than that of the A.V.C. filter, as is required.

The oscillator control circuit in Fig. 23 is of conventional design, with  $R_N$  and  $C_N$  serving as the phase-shifting network. The receiver oscillator is of the tuned plate type, with the phase-shifting network shunting the oscillator tuning coil. A somewhat similar discrimination circuit is also used in one RCA A.F.C. system; the essential difference between the RCA and Silvertone circuits is shown in Fig. 24. The first two tuned circuits in the discriminator transformer are the same in both cases, the difference being in the third tuned circuit. Note that in Fig. 24 coil  $L_M$  induces a voltage in the split-coil arrangement  $L_3-L_4$ , rather than in two separate coils as was the case in Fig. 23. Across  $L_3-L_4$  is fixed condenser  $C_3$  and variable inductance  $L$ , having an iron dust core whose position can be varied in order to change

the inductance of  $L$  and tune the discriminator circuit. Coil section  $L_3$  also has an adjustable iron dust core which can be adjusted during alignment of the receiver to make the inductance of  $L_3$  exactly equal to the inductance of  $L_4$  and thus make both coils develop the same I.F. voltage for the diode section.

*Effect of Preselector Upon A.F.C. Action.* Up to this point we have neglected the effects of the preselector upon the action of the A.F.C. system. Actually, however, the selectivity of the preselector is quite important with relation to that characteristic of an A.F.C. system which is known as "dragging." Consider, for example, a receiver having an S curve which indicates the ability to correct for signals as much as 5 kc. off tune. When a station is properly tuned in, then gradually tuned off resonance, the A.F.C. circuit will correct the oscillator frequency satisfactorily for about 5 kc. off resonance. Tuning farther off than this does not remove the discriminator control voltage entirely, as you can see by referring to the S curve in Fig. 18; actually the D.C. control voltage may be produced for a considerably greater deviation from resonance, but it will be insufficient to correct completely for the off-tune condition. This action of an A.F.C. system in making a partial correction outside of the efficient operating range of the system is referred to by engineers as "dragging."

When two stations are close together in frequency, and one of the stations is tuned in, it is perfectly possible that a receiver having A.F.C. and a broad preselector might not be able to pick up the other station. The station originally tuned in is held by the A.F.C. system because of excessive dragging action. With this broad preselector, the only way to hear the adjacent station would be to cut off

temporarily the A.F.C. action of the receiver in order to allow the adjacent station signal to take hold of the A.F.C. system; a highly selective preselector, however, *would not allow* an A.F.C. system to hang onto a powerful local station when the receiver is tuned to a weaker adjacent station.

Here is another interesting result of dragging action; when a receiver having A.F.C. is tuned carelessly to a desired weak station, so that the desired signal is being dragged in from a position considerably off tune, any fading of this desired signal may allow an adjacent station to take hold.

The remedy for excessive dragging involves improving the selectivity of the preselector enough so that when the tuning condensers are outside the desired range of A.F.C. action, the reduction in R.F. gain will reduce the I.F. signal enough to make the discriminator release its control over the oscillator control tube. Strictly speaking, the preselector should be band-passed so it will tune broadly in the frequency range over which A.F.C. is to act, but will have a sudden reduction in gain (high selectivity) immediately outside of this band-pass region.

Band-pass preselectors are rarely used in receivers, but sufficient selectivity should be incorporated in the preselector to prevent annoying dragging action. Because it is not easy to get high preselector selectivity in the high frequency bands of an all-wave receiver, A.F.C. action is often omitted in these bands. Any A.F.C. circuit is most effective for strong stations in the broadcast band.

*Philco Push-Pull A.F.C. System.* The automatic frequency control system used in some Philco receivers is unique in that when the receiver is properly tuned or the A.F.C. system is shorted out, the receiver oscillator circuit is entirely independent of the

A.F.C. system. It is therefore possible to align the oscillator and pre-selector independently of the A.F.C. system, whereas in the systems previously described the shunt inductance of the oscillator control tube always had to be considered when aligning the oscillator trimmer condensers.

A typical example of this unique Philco push-pull A.F.C. system, that used in the Philco model 37-9 all-wave superheterodyne receiver, is shown in Fig. 25; it is also known as a *magnetic tuning system*. As you can see, the type 6K7G tube in the final I.F. amplifier stage feeds into two entirely separate circuits, a 6Q7G second detector-A.V.C.-first A.F. tube circuit (not shown) and the conventional 6H6G double-diode discriminator circuit.

The two diode sections of the discriminator tube each receive two I.F. voltages in the usual manner; the I.F. voltage existing across the tuned plate circuit of the 6K7G tube is applied between the center tap of the split secondary winding and the cathodes of the 6H6G tube, and the I.F. currents flowing in the split secondary (due to the voltage induced by current in the tuned plate circuit) produce across the coil sections the other I.F. voltage acting on each diode section. The resulting rectified current flow develops the required D.C. discriminator control voltage across the two 2-megohm resistors connected between the cathodes. The polarity of this voltage depends, of course, upon whether the I.F. signal frequency is above or below the correct value, and the magnitude of this D.C. voltage depends upon the amount of error in tuning. This D.C. control voltage is applied across two 1-megohm resistors having their midpoint grounded, and hence points *x* and *y* will *always* be equal to each other in potential and of opposite polarity *with respect to ground*.

In this manner the discriminator output voltage is divided into two equal voltages of opposite polarity.

During off-tune conditions these equal and opposite voltages at *x* and *y* are fed to points *a* and *b*, at the input of the oscillator control tube. Note that there is a 490,000-ohm re-

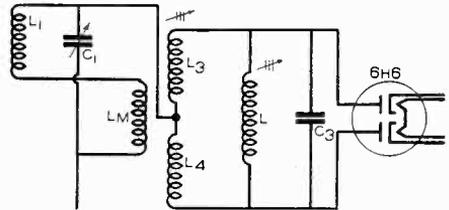


FIG. 24. Discriminator transformer connections used in RCA model 812K receiver.

sistor in each lead, with a .3 mfd. condenser shunted across the leads; these filters serve to remove all I.F. and A.F. signal components and at the same time delay the discriminator action enough to make it slower than A.V.C. action. The two .15 mfd. condensers with common terminals grounded serve to maintain a balance between the filtered A.C. currents. Switch  $SW_1$  is the A.F.C. ON-OFF switch; when in the OFF position it grounds points *a* and *b*. Switch  $SW_2$  closes only when the dial tuning mechanism is operated, and shorts points *a* and *b* to prevent A.F.C. from dragging one station beyond the dial setting for another station.

Observe now that a type 6A8G pentagrid converter tube serves as the first detector-mixer-oscillator tube for the receiver, with the first grid serving as the oscillator control grid and the second grid acting as the oscillator plate. The oscillator tuned circuit, made up of  $L_1$ ,  $C_1$ ,  $C_3$  and the 85-ohm resistor, is connected between the first grid and the chassis. The necessary feed-back is obtained by connecting the oscillator plate electrode through a 250-mmfd. condenser to the

lower end of coil  $L_1$ ; current flow through this section of  $L_1$  induces the feed-back voltage in the main section of  $L_1$ .  $C_3$  serves as the oscillator low frequency padder and  $C_2$  as the high frequency trimmer. Rectified grid current flowing through  $L_1$  and a 32,000-ohm resistor in the oscillator tuned circuit produces across this re-

into the discriminator circuit. This R.F. voltage  $e_R$  produces R.F. plate currents  $i_2$  and  $i_3$  which are *in phase* with  $e_R$ , are equal in value when both tube sections have the same C bias voltage, and flow in opposite directions through the two halves ( $L_2$  and  $L_3$ ) of the secondary winding of the oscillator control transformer.

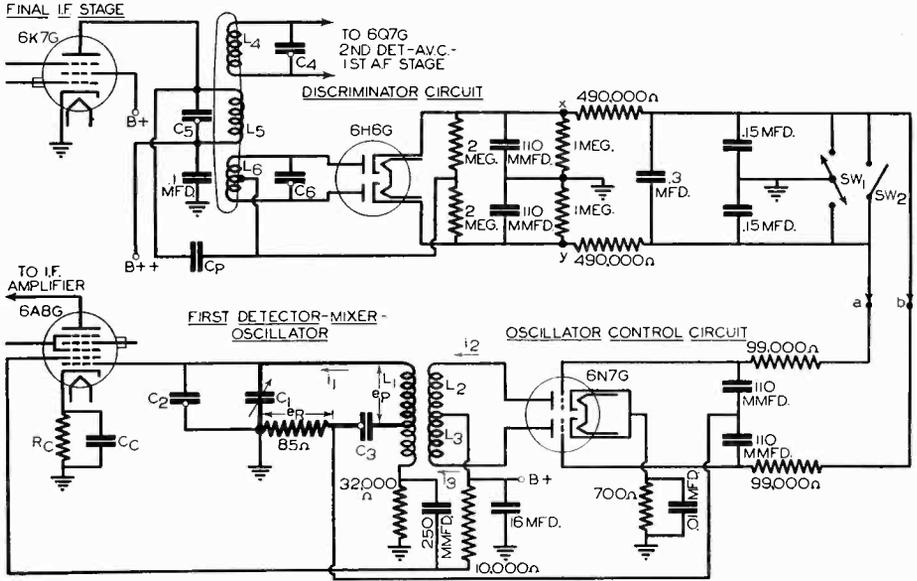


FIG. 25. Circuit diagram of the push-pull A.F.C. system (also known as magnetic tuning) used for the broadcast band on the Philco model 37-9 all-wave receiver. There is no A.F.C. action on short-wave bands in this set.

sistor a self-adjusting negative C bias which acts in series with the fixed C bias produced by the oscillator cathode resistor.

Locate the oscillator tuned circuit, shown in heavy lines in Fig. 25 and made up essentially of  $L_1$ ,  $C_1$ ,  $C_3$  and an 85-ohm resistor. The same R.F. current  $i_1$  flows through all of these parts, developing across the 85-ohm resistor an R.F. voltage  $e_R$  which is in phase with the current.

R.F. voltage  $e_R$  is applied to the two control grids of the 6N7G oscillator control tube through 110 mmfd. condensers. Note that 99,000-ohm resistors connected to these grid leads prevent the R.F. voltage from feeding

From your study of coils you will recall that increasing the amount of flux through a coil increases its inductance, and decreasing the flux lowers the inductance. If current  $i_2$  flowing through  $L_2$  produces a flux which *increases* (aids) the flux produced by  $i_1$  through  $L_1$ , then  $i_3$  flowing in the opposite direction through  $L_3$  will produce a flux which *decreases* (opposes) the flux produced by  $i_1$  through  $L_1$ . (This holds true only when all three flux-producing currents are *in phase*, as they are in this circuit.) Current  $i_2$  thus has the effect of increasing the inductance of  $L_1$  in the oscillator tuned circuit, while  $i_3$  decreases the inductance of  $L_1$ .

When  $i_2$  and  $i_3$  are equal, their effects upon the inductance of  $L_1$  will be equal and will cancel each other. This proves clearly that when the A.F.C. system is inoperative, or when the receiver is properly tuned, the oscillator control tube has no effect whatsoever upon the oscillator tuned circuit.

moving the operating point to a less steep region and reducing the A.C. plate current for that section. The D.C. control voltage thus makes  $i_2$  different from  $i_3$ . The effects of the larger current upon the inductance of  $L_1$  are not altogether cancelled out by the effects of the smaller current; the result is a net change in the induc-

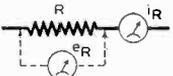
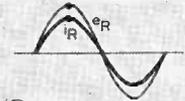
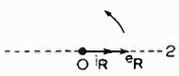
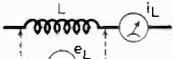
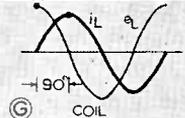
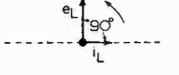
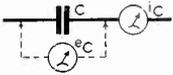
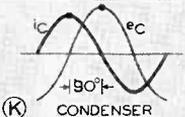
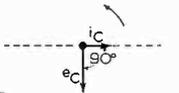
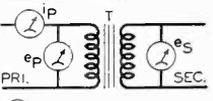
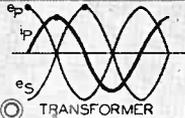
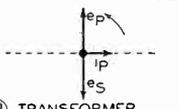
① SKETCHES	② SCHEMATIC SYMBOLS	③ CURRENT-VOLTAGE RELATIONS	④ VECTOR DIAGRAMS
 (A) RESISTOR	 (B) RESISTOR	 (C) RESISTOR	 (D) RESISTOR
 (E) COIL	 (F) COIL	 (G) COIL	 (H) COIL
 (I) CONDENSER	 (J) CONDENSER	 (K) CONDENSER	 (L) CONDENSER
 (M) TRANSFORMER	 (N) TRANSFORMER	 (O) TRANSFORMER	 (P) TRANSFORMER

CHART 1. This chart gives you important facts about the phase relationships of voltages and currents for resistors, coils, condensers, and transformers in A.C. circuits.

As you already know, improper tuning of a receiver causes D.C. voltages of equal value but opposite polarity with respect to ground to be developed by the discriminator; these are applied to the grids of the 6N7G oscillator control tube through 99,000-ohm resistors, making one grid less negative and the other more negative than the bias value determined by the 700-ohm cathode resistor. Making the net C bias of one triode section less negative moves the operating point on the  $E_g-I_p$  characteristic to a steeper region, increasing the A.C. plate current for that section; at the same time the net C bias on the other triode section becomes more negative,

tance of  $L_1$ , and this changes the frequency of the oscillator. In other words, one current *pushes* up and the other *pulls* down the inductance of  $L_1$ . The manner in which leads *a* and *b* are connected to the control grids of the 6N7G tube determines whether this control will be in the proper direction; if the control is such as to exaggerate errors in tuning, reversing the leads will correct the trouble.

### Adjusting A.F.C. Systems

Fortunately for Radiotricians, the adjustment of an A.F.C. system is quite simple. The only instruments needed are an ordinary signal generator capable of producing the I.F.

value of the receiver, and a sensitive low-range (0-10 volts) voltmeter. At least a 5,000 ohms-per-volt meter should be used; if this is not available, use an ordinary vacuum tube voltmeter or one which you assemble yourself from a type 31 tube and batteries, using a milliammeter as the indicator.

The signal generator is set at the I.F. value of the receiver and is connected to the input of the I.F. amplifier. The sensitive voltmeter is connected across the D.C. output terminals of the discriminator (points 1 and 4 in Fig. 20; between the chassis and point 3 in Fig. 23; between points *a* and *b* in Fig. 25). The I.F. amplifier stages are now aligned in the usual manner for a maximum reading of the voltmeter. (Do not touch the trimmer across the split secondary of the discriminator stage at this time unless the voltmeter deflection is too small for an accurate adjustment; in this case, adjust the trimmer in either direction just enough to get a suitable deflection.) When adjusting the tuned circuit ahead of this split sec-

ondary winding, it is a good plan to move the signal generator to the input of the previous tube; adjust this discriminator input circuit carefully for maximum output.

Now, without changing the voltmeter or signal generator connections, adjust the trimmer condenser in the split coil circuit for *zero meter reading*. This adjustment is quite critical, so be sure that the minimum reading is secured when the adjusting tool is removed. This is all there is to the adjustment of a conventional A.F.C. circuit.

You might think that with so simple an adjusting procedure, the information given in the first part of this lesson is unnecessary for the practical man. This is decidedly not true, however, for an understanding of how A.F.C. systems work is quite necessary when trouble develops due to failure of various parts in the system. If your work will involve the servicing of receivers having automatic frequency control, you will at some time or other find use for every bit of the information given in this lesson.

# Lesson Questions

Be sure to number your Answer Sheet 31FR-1.

Place your Student Number on every Answer Sheet.

Never hold up one set of lesson answers until you have another ready to send in. Send each lesson in by itself before you start on the next lesson. In this way we will be able to work together much more closely, you'll get more out of your Course, and you will receive the best possible lesson service.

1. Where are meter-type tuning indicators ordinarily connected in a receiver having A.V.C.? *Between the oscillator and the first IF amplifier.*
2. Why is an A.F.C. system essential in a receiver having an electro-mechanical (motor-driven) automatic tuning system? *to tune the receiver to the desired station.*
3. What voltage in a receiver is used to control a cathode ray tuning indicator? *D.C. from the A.V.C. circuit.*
4. How does core saturation affect the reactance of the secondary winding of the General Electric Colorama tuning reactor? *It decreases the reactance.*
5. Name the two essential sections of an A.F.C. system. *Motor and control circuit.*
6. In a receiver having A.F.C., what voltage serves to change the effective inductance of the oscillator control circuit? *A.C. from the oscillator.*
7. How can you identify the discriminator section on the circuit diagram of a receiver having A.F.C.? *It is the section containing the discriminator coil in tuned circuit.*
8. Will a highly selective preselector allow an A.F.C. system to hang on to a powerful local station when the receiver is tuned to a weaker adjacent station? *No.*
9. When the Philco push-pull A.F.C. system is inoperative or when the receiver is properly tuned, what effect does the oscillator control tube have upon the oscillator tuned circuit? *None.*
10. When adjusting an A.F.C. system a signal generator set at the I.F. value of the receiver is fed into the I.F. amplifier and a sensitive D.C. voltmeter is connected across the D.C. output terminal of the discriminator. How is the trimmer condenser in the split-coil discriminator circuit adjusted as regards the meter reading? *to meter reading.*



## COURTESY

Truly big men are always courteous. It is only "small" men, men with inferiority complexes, who are rude or thoughtless. And smaller than small are those who are over-courteous to their superiors and intentionally rude to those over whom they have some authority.

Practice courtesy in all your contacts, business as well as social. Establish courtesy as one of your life habits. Be polite and kind to every one you meet, rich and poor alike, and soon courtesy will become second nature for you.

The best place to test yourself is right at home. Are you always courteous to members of your immediate family, or do you shout at them on the slightest provocation? Are you considerate of their feelings, or do you delight in saying things and doing things which you know will hurt them?

If you develop the habit of being courteous to your own folks, your away-from-home courtesy will ring true. It won't appear "put-on," as is so often the case when a man reserves his courtesy for only special occasions.

*J.E. Smith*

**HOW TO USE CIRCUIT DIAGRAMS  
AND LOCATE CHASSIS PARTS**

32FR-2



**NATIONAL RADIO INSTITUTE**

**WASHINGTON, D. C.**

**ESTABLISHED 1914**

# Study Schedule No. 32

For each study step, read the assigned pages first at your usual speed, then reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind, then answer the Lesson Questions specified for that step. Study each other step in this same way.

**1. What a Circuit Diagram Tells; Principles of Radio Servicing** Pages 1-7

Circuits need not be completely analyzed; two groups of receiver defects; Surface Defects; Dead Receiver; Improperly Operating Receiver; trouble-locating questions; Distortion at High Volume Levels; Hum; Alignment Procedures; Subjects for Further Study; example for study is Philco Model 38-7 receiver. Answer Lesson Questions 1, 2 and 3.

**2. Identifying Stages on a Schematic Diagram; Analyzing Each Stage on a Schematic Diagram** - - - - - Pages 7-12

Read diagrams from left to right; identifying clues on diagrams; stage-identifying examples: Oscillator-Mixer-First Detector; I.F. Amplifier; Second Detector; A.V.C. Tube; First A.F. Amplifier Stage; Output Stage; Power Pack. Answer Lesson Questions 4 and 5.

**3. Identifying Stages on the Chassis** - - - - - Pages 12-20

Use of tube layout diagrams; identifying the circuit; locating the rectifier tube and a.f. output tube; identifying other tubes; General Suggestions. Answer Lesson Question 6.

**4. Locating Parts with a Pictorial Diagram; Locating Parts Without a Pictorial Layout Diagram; Tracing Tube Electrode Circuits** - - - - - Pages 20-24

Locating electrolytic filter condensers and identifying their leads; locating a resistor; use of circuit diagram alone for locating parts and checking continuity; Plate Circuits; Screen Grid Circuits; Control Grid Circuits; Suppressor Grid Circuits; Diode Detector Circuits. The important thing in this step is to refer to the diagrams as you study, so as to get practice in locating parts quickly with the aid of diagrams. Answer Lesson Question 7.

**5. General Rules for Checking Continuity of Electrode Circuits in A.C. Receivers; How to Locate Trimmer Condensers; Appraising Receiver Performance; Schematic and Chassis Wiring** - - - - - Pages 24-29

Rule for positive electrodes; rule for negative electrodes; rule for diodes; identification of trimmers; General Trimmer-Locating Suggestions; how performance is affected by the number of tuned stages, number of tubes, type of circuit, use of r.f. stage, type of loudspeaker and presence of band-pass circuits; practical chassis wiring. Answer Lesson Questions 8, 9 and 10.

**6. Mail your Answers for this Lesson to N.R.I. for Grading.**

**7. Start Studying the Next Lesson.**

# How To Use Circuit Diagrams and Locate Chassis Parts

## What a Circuit Diagram Tells

**A**S YOU know, a schematic circuit diagram is a symbolic means for showing the electrical connections in a radio set. A schematic diagram tells very little about the construction of each part, how the parts should be adjusted, where the parts are located on the actual chassis, or how the connections are actually made between various points; nevertheless, to a man possessing the fundamental training in radio theory and practice which you are now acquiring, this diagram can yield information of great value in connection with the servicing and maintenance of radio equipment.

There is no more need for a Radiotrician to analyze a complete receiver circuit diagram when repairing a defect, than there is need for a carpenter to study the complete building plans for a new home when repairing the front porch railing. Once you become familiar with the general principles of radio servicing, you will agree that a circuit diagram is a "reference text" rather than a "study text."

## Principles of Radio Servicing

When you tackle a defective receiver, you know that it must belong to one of the two general trouble groups in which all defective radio receivers can be placed: 1, *the receiver is dead*—it does not play at all; 2, *the receiver plays improperly*—it may howl, squeal, distort, lack selectivity, lack sensitivity, be noisy, have a faulty automatic tuning or A.F.C. system, or have any one of a host of other defects which are considered elsewhere in the Course.

*Surface Defects.* Occasionally the

trouble which results in a dead or improperly operating receiver is not in the chassis at all, but on the outside or on the *surface*, where it can be readily seen and corrected. Obviously you will want to check for surface defects before removing the chassis.

The practical radio serviceman generally starts in by checking the antenna system and the power line



Making a continuity test between the top caps of two tubes on the Philco model 38-7 receiver, using the multimeter section of a combination signal generator and multimeter.

cord, making sure that all tube caps are in position and not touching the shields, making sure that all extension wires and cable plugs are in place, and noting whether any tubes are obviously defective. He may even remove each tube and check it in a tube tester, to see if a simple replacement of a defective tube will clear up the trouble.

When a check for surface defects in this manner proves unsuccessful, one trouble-locating procedure is followed for the case of a dead receiver, and an entirely different pro-

cedure is chosen for an improperly operating receiver. In both of these procedures the schematic circuit diagram is brought into use.

*Dead Receiver.* When anything goes wrong, the natural first step is to ask yourself what the trouble could be. This applies to any machine, to any human being, and to any action, as well as to the radio receivers under discussion. We therefore ask: "Why does a receiver become dead?" There are a number of answers—one or more tubes may be defective, some part in the power pack or in the voltage supply system for the receiver may have opened up or become shorted, a signal circuit part may have opened or shorted, or a connection may have opened somewhere.

With a dead receiver, then, our first task is the locating of the defective tube, part or connection. Do we have to check all parts and connections until we find the one which is defective? Not if we reason out this problem a bit first. You know that a carrier signal modulated with an intelligence signal enters the antenna circuit of a receiver, with this intelligence signal working its way through a chain of stages to the loudspeaker. The receiver is dead because one or more links in this chain have failed. Radio men say that one or more stages have become defective; once they locate this defective stage, they have limited the task of locating the defective part to a smaller portion of the receiver.

As you will learn elsewhere in your Course, there are two ways of locating the defective stage in a dead receiver; we will present them here only briefly. In the *dynamic stage-by-stage elimination test*, a modulated signal generator is connected to the input of the second detector; if the modulation signal is heard in the loudspeaker, you know immediately that the detector and the audio am-

plifier are free of trouble. If no signal is heard, headphones are connected in turn to the output of the detector and to the output of each following audio stage. Failure to hear the output signal in the headphones at any stage identifies that stage as being defective, assuming that the power supply has been checked and found to be performing properly.

If the detector and the stages following it are found to be operating properly, the signal generator is advanced toward the antenna one stage at a time; the signal should become louder as this is done. Failure to hear the signal identifies the defective stage.

The simplest and speediest test for isolating the defective stage in a dead receiver is the *circuit disturbance test*. This depends upon the fact that when a tube is pulled out of its socket and quickly returned, or when the grid of a tube is touched or shorted to the chassis, or when the grid cap connection to a tube is removed and returned, a sharp change in the plate current for the tube takes place. This sudden change or shock is relayed through the receiver and is heard in the loudspeaker as a click, a thud or a squeal provided that all stages in the path of this disturbance are operating properly. Starting at the last audio stage and working toward the antenna stage, one of these methods for producing a disturbance is employed on each tube in turn. When you reach a tube which produces no indication in the loudspeaker while being shocked, you have isolated the trouble to that stage.

Once the defective stage is isolated, it is given a thorough check either with an ohmmeter or a voltmeter in a manner explained elsewhere in the Course, after making sure that the tube is good. Supply voltages for the tube are usually checked first.

Obviously you must know something about the circuit of a receiver in order to be able to isolate the defective stage and then isolate the defective part in that stage. You must be able to identify the second detector tube, which is the usual starting point for a stage isolation procedure. You must be able to locate the first, second and power audio tubes so you can follow the chain of stages in the proper order from the second detector to the loudspeaker; likewise you must be able to locate the I.F. and R.F. amplifier tubes in their proper order from the second detector to the antenna. You should know where to connect a voltmeter in order to check the main power supply. Once the defective stage is located, you must be able to locate and identify the various parts in the grid, plate, screen grid and suppressor grid circuits of the tube. You may be able to determine all these things by a study of the chassis itself, but this will take time; a schematic circuit diagram gives this information far more readily. If the service manual of the receiver also contains a tube layout diagram, showing the positions of the various tubes on the chassis and identifying their functions, your task is further simplified.

*Improperly Operating Receiver.* Expert Radiotricians will tell you that a dead receiver is generally easier to service than an improperly operating receiver. The true value of theoretical training, modern technique and practical experience becomes evident when it is necessary to service a receiver which squeals (oscillates), blocks, distorts, hums, is noisy, cuts off intermittently, lacks selectivity, lacks sensitivity, or lacks output volume. The value of circuit diagrams becomes increasingly more evident as servicing problems become more complicated.

A customer may bring in a receiver for repair, complaining that he can't understand the announcements or speeches as reproduced by the loudspeaker. After making sure that surface defects are not the cause of this distortion, a Radiotrician will automatically ask himself these questions:

1. *Under what conditions of receiver operation does distortion ordinarily occur? Does the receiver distort only when the volume level is high, only when the volume level is low, or at all volume levels and all settings of the receiver controls?*

2. *What are the general reasons for distortion which occurs under the conditions noted?*

3. *What are possible causes, in this particular receiver, for the type of distortion observed?* (Reference text "Radio Receiver Troubles—Their Cause and Remedy," which was sent you some time ago, covers many of the common causes for various types of distortion. This reference text will help you develop an ability to reason from observed effects like distortion to the logical causes.)

When a serviceman cannot answer these questions, he is compelled to test all stages and parts, using some systematic method of defect isolation. The checking of operating voltages for all tubes is an important part of this procedure; the correct voltages are usually given in the service manual.

*Distortion at High Volume Levels.* Let us assume that the receiver in question distorts only when the volume level is high. If it is an inexpensive midget table model receiver, this distortion may be natural and unavoidable, especially if it is most serious when powerful local stations are tuned in. Experience is a great help in determining whether or not an actual defect exists. Reference to the circuit diagram of the receiver will

often tell whether or not the receiver is capable of handling high volume levels. If you decide that the receiver is operating as well as can be expected for that particular type of circuit, you will have to explain the situation to the customer and point out that the volume level should be kept below the point at which distortion begins.

If the receiver is of normal design and you decide that it should handle the maximum volume level without distortion, the next question to answer is: What can cause distortion when the volume level is high? The Radiotrician knows that possible causes for this effect are overloading of the detector, overloading of the audio output tube, overloading of the loudspeaker, weak tubes (low emission), low supply voltages, or a defective loudspeaker. With these probable causes in mind, you should now proceed to eliminate them one by one until you reach the true cause of the trouble. Test the tubes first of all, then check the main supply voltage and the individual supply voltages for each suspected tube. You may have to refer to a circuit diagram in order to determine where the voltmeter connections should be made for each test.

Plate and screen grid supply voltages are usually measured between these electrodes and the cathode of the tube, while a C bias voltage is measured between the cathode and the chassis, as a rule. Quite often, manufacturers will specify, in service manuals, all electrode voltages with respect to the chassis. Reference to the schematic circuit diagram will tell you exactly where connections can best be made to measure a particular voltage.

Can you quickly locate the plate, screen grid and cathode terminals on an actual tube? Few people can, for

there are hundreds of different types of tubes in use today, each with different connections between the electrodes and the terminal prongs. Receiver manufacturers sometimes show tube sockets in pictorial form in the service manual, either on the schematic circuit diagram itself or on a separate chassis diagram. Tube manufacturers likewise supply socket connection diagrams for all tubes, so there should be no trouble in securing a tube diagram to serve as a guide.

When there are two or more stages in a receiver employing the same type of tube, how can you locate one of these tubes on the circuit diagram? The answer is quite simple; examine a few of the parts in the chassis which are connected to the tube in question, then determine which tube in the schematic diagram has these same parts connected to it.

Now you can check the values of each part in the defective stage against the values specified on the circuit diagram; check for changes in the resistance of a part and for shorted or leaky condensers, as any of these defects will alter the supply voltage enough to cause distortion when the volume level is high.

If you have followed the servicing procedure up to this point without locating the defect, refer to the schematic circuit diagram and determine whether the receiver has A. V.C. If the volume control is connected into the input of the audio system, you can be pretty sure that some of the preceding I.F. and R.F. amplifier stages are A.V.C controlled. Assuming this to be the case, you can be fairly certain that R.F., I.F., and second detector signal and supply circuits are all in order since there is no distortion at low volume levels on the particular receiver being analyzed.

You think a bit—could there be

overloading of an audio amplifier stage? As soon as you think of a reasonable cause of trouble, check up on it. An easy way to locate the overloaded stage is to turn up the volume control while tuned to a strong local station, so that distortion is clearly evident, and then connect headphones across the output of each audio stage in turn, starting with the second detector and ending at the loudspeaker terminals. Reference to the schematic circuit diagram and perhaps to the pictorial layout diagram will tell you where to make the headphone connections. The stage at which distortion is first noticed in the headphones will very likely be the overloaded stage. If no distortion is first noticed in the headphones up to the loudspeaker, then you can be pretty certain that there is a defect in the loudspeaker or its input circuit.

*Hum.* Now let us consider a receiver which has excessive hum. Naturally you turn on the set and make the usual inspection for surface defects. Let us assume that while doing this you note one tube with a blue glow between the cathode and plate. By noting the number of this tube, by referring to the parts layout diagram in the service manual, or by noting that there are no grid electrodes between the cathode and plate in the tube, you identify it as a rectifier. Obviously you have located a surface defect, for these tubes (with the exception of mercury vapor rectifier tubes) should not glow when operating properly; there is no sense in looking farther for other causes of hum until this trouble is cleared up.

A blue glow inside a vacuum type rectifier tube indicates the presence of gas; we say that the tube is *gassy*. The question is: Why did the tube become gassy? A logical answer is normal aging of the tube with constant use, for rectifier tubes are often

operated very nearly at their maximum capacity in receivers. On the other hand, there may be a short in the power pack which is making this rectifier tube deliver higher than normal current. It would not be safe to insert a new tube until the power pack has been examined for possible defects. Again you would refer to the circuit diagram, noting what parts are in the power pack circuit and noting what their values should be. From your study of condensers, you know that paper condensers should have very large resistances, above 20 megohms, and electrolytic condensers should have resistance above at least 250,000 ohms when properly tested. Since filter condenser failure is a very common cause of hum, and excess rectifier tube current, you check these first, then check the electrical values of other parts in the power pack or in the power supply system.

But suppose that the inspection for surface defects gave no sign of the trouble. In this case the Radiotrician would turn to the schematic circuit diagram with one thought in mind: What part of circuit in this particular receiver could become defective and cause hum? As he glances over the power section of the circuit diagram, he may notice a tuned filter; knowing that even small changes in the values of the condenser and choke coil can result in hum, he checks these parts carefully with his multimeter. Or he may notice a hum bucking coil indicated in the schematic symbol for the loudspeaker; this could well be the cause of the hum, so a careful check of this part is made. There are electrolytic condensers in the power pack, and they have an unfortunate tendency to become leaky with age, so he checks them next. There are resistance-capacitance filters in grid circuits for application of a C bias

voltage, so he checks these for leaky condensers and shorted resistors.

But this is not intended to be a lesson telling how to service a radio receiver; we have briefly considered these actual examples of servicing problems primarily to illustrate correct service technique and to show how the schematic circuit diagram and the other information contained in service manuals can be of help in servicing. You are becoming acquainted with the process of figuring or reasoning out probable causes of trouble when certain effects are observed—a process which we call “effect-to-cause reasoning.”

A knowledge of how radio circuits work is one essential requirement for successfully applying effect-to-cause reasoning; this knowledge you are now acquiring in your Fundamental Course. Experience with actual radio apparatus and continued reference to schematic circuit diagrams will eventually make effect-to-cause reasoning become second nature to you.

*Alignment Procedures.* We have by no means exhausted the Radiotrician's uses for circuit diagrams. One highly important use, in connection with the alignment of tuned circuits in a receiver, deserves particular attention, for it is often desirable to realign a repaired receiver to secure maximum selectivity and sensitivity.

In a conventional superheterodyne receiver having peak-tuned circuits, the I.F. stages are aligned first and then the oscillator is made to track the preselector. In an all-wave receiver the preselector-oscillator tracking adjustment must be made for each band. Before any of these adjustments can be made, however, you will first want to know how many trimmers there are to be adjusted, where each one is located, and what it does. A study of the schematic circuit diagram will give you this information. Furthermore, this analysis of the dia-

gram will reveal if there are any image rejection circuits, harmonic or code interference traps, A.F.C. circuit trimmers, or fidelity-equalizing trimmers. Once you spot these on the circuit diagram and note their functions, you can mentally omit them from your alignment procedure and concentrate upon the true circuit-aligning trimmers. You may note that certain oscillator coils have adjustable iron cores; this establishes the fact that they are the low-frequency oscillator adjustments.

The circuit diagram may reveal the presence of variable-selectivity I.F. transformers and band-pass circuits which can be adjusted either for peak response or for broad (band-pass) response. It would be difficult to determine the presence of a band-pass circuit from mere study of the receiver chassis, yet the schematic circuit diagram and your knowledge of circuit operation immediately suggests the correct aligning procedure. If you know radio theory and the usual design practices, there will be no need for you to follow detailed alignment instructions as given in service manuals.

Once you have determined what the correct alignment procedure should be, you must locate the trimmer condenser adjusting screws or nuts on the actual chassis; again the schematic circuit diagram can be of great help in identifying each trimmer. If a pictorial layout is available, do not overlook it during an alignment procedure; this diagram often reveals the exact locations of the various trimmers.

*Subjects for Further Study.* The preceding brief outline of the technique of radio servicing has shown you many uses for circuit diagrams and the associated information prepared by receiver manufacturers in the form of service manuals. We will now consider in some detail how these service manuals can be used to best advantage in radio servicing, and how the information needed for servicing

can be obtained directly from the receiver chassis when no service manual is available. The important subjects for further study are therefore:

1. How to identify the various stages on a schematic circuit diagram.
2. How to analyze each stage of a schematic circuit diagram.
3. How to identify the various stages on the actual receiver chassis.
4. How to locate various parts on the chassis with a pictorial layout diagram.
5. How to locate parts when no pictorial layout is available.
6. How to trace each tube electrode circuit with and without the help of a schematic circuit diagram.
7. How to locate the alignment trimmer condensers.
8. How to appraise the performance of a receiver.
9. Important differences between schematic diagrams and actual chassis wiring connections.

For the purpose of illustrating the various points covered in this lesson, we have selected a popular model of a well known receiver (Philco Model 38-7) and have reproduced, in the center pages of this lesson, the following material pertaining to this receiver: Fig. 1—the schematic circuit diagram as it appears in the service manual of the receiver; Fig. 2—the pictorial layout diagram for parts underneath the chassis, as given in the service manual; Fig. 3—pictorial layout of parts on the top of the chassis, as given in the service manual; Fig. 4—tube socket connections as seen when looking at the bottom of the chassis, and electrode operating voltages; Fig. 5—reproduction of an actual photograph of the top of the chassis, corresponding to the diagram in Fig 3; Fig. 6—reproduction of an actual photograph of the bottom of

the chassis, corresponding to the diagram in Fig. 2.

For convenience in referring to these diagrams while studying this lesson, it is suggested that you pry up the staples holding this lesson together, carefully remove the inside four pages (13, 14, 15 and 16), and bend back the staples again. When you have completed your study of this text-book, you can replace these diagrams in the same manner.

### Identifying Stages on a Schematic Diagram

When a Radiotrician first takes up a schematic circuit diagram, he makes a general survey before attempting to locate any individual circuits or parts, in order to determine the general lineup of stages in the receiver. When looking at Fig. 1, for example, he would immediately identify the circuit as that of a two-band superheterodyne receiver which is A.C. operated and has a simple tuned input circuit feeding directly into a pentagrid converter tube which acts as oscillator-mixer-first detector. He notices that this is followed by a single I.F. amplifier stage which feeds into a double-diode detector in which one section acts as the actual demodulator and the other serves as the A.V.C. tube. The diode demodulator feeds into a single audio amplifier stage which in turn feeds a pentode power output amplifier stage. He notes also that the power pack employs a full-wave rectifier tube. An experienced man could make this identification of stages almost at a glance, for familiarity with the tubes and circuits employed enables him to omit a great many of the steps in the reasoning process now to be described.

Circuit diagrams are invariably arranged so they can be read from left to right when tracing the path taken by intelligence signals from the antenna to the loudspeaker. We therefore start at the upper left of the dia-

gram in Fig. 1, where we find a terminal strip having provisions for two antenna connections and one ground connection. Observe that the signals pass through R.F. transformer 1, the secondary winding of which is tuned by one section of a ganged variable condenser. This tuned circuit feeds into a type 6A8G tube having five grids. Although on this particular diagram the designation *DET.-OSC.* identifies this stage as the detector-oscillator, you will find many circuit diagrams in which only the tube type numbers are given. We know, however, that the first stage must be either an R.F. amplifier stage or the first detector; we trace from the plate of this tube into I.F. transformer 13, and we know that tubes having five grids (pentagrid converters) are almost invariably used as combination oscillator-mixer-first detector tubes, so we feel safe in identifying this stage as the mixer-first detector of a superheterodyne circuit.

Another glance at input R.F. transformer 1 reveals that each of its windings is in series with the winding of another R.F. transformer marked 2, and that there are switches for shorting out one secondary winding. Clearly, then, we have a two-band receiver. Verification of this can be secured by tracing from the first grid of the 6A8G tube, which we know must be serving as the grid of the local oscillator; this goes to two tuning coils, 5 and 6, and one of these can be shorted out with a switch.

The second tube in our receiver lineup, a 6K7G, is fed by the secondary of the first I.F. transformer and feeds into the primary of the second I.F. transformer; it must therefore be an I.F. amplifier stage. We know that this second I.F. transformer (19) must feed either into a second detector or into another I.F. amplifier stage. The tube in question is a 6J5G triode, and triodes are seldom if ever used as I.F. amplifiers. We must examine this cir-

cuit more carefully to determine its true function (assuming that the *2ND DET.-A.V.C.* designation is not present). Note that the secondary voltage of second I.F. transformer 19 is applied between the grid of this triode and ground, with the cathode of the tube directly grounded. This looks much like an ordinary diode detector circuit, with the control grid of the triode serving as the diode plate. Resistors 20 and 21 are in the circuit, and now we note a volume control potentiometer connected across resistor 21, with condenser 24 in series to block direct currents. The plate and cathode of this triode tube must therefore be serving as the A.V.C. diode, since the plate is also fed by the second I.F. transformer through D.C. blocking condenser 23.

From the movable contact of the volume control we trace our signal path to the grid of a 6K5G triode, which can only be a first audio amplifier stage since it follows a diode second detector. This tube in turn feeds into a pentode type 6F6G tube having a loudspeaker as its plate load, and consequently this is the power output stage.

One glance at the *power transformer symbol* in the power pack circuit in Fig. 1 is sufficient to identify our circuit as that of an A.C. receiver, for power transformers are never used in universal or D.C. sets.

There are a great many other clues which will prove useful in identifying each stage when you first go over a circuit diagram to get the general lineup, and consequently you will seldom if ever have any difficulty in identifying a stage even though its function is not indicated on the diagram.

### **Analyzing Each Stage on a Schematic Diagram**

When the servicing procedure has advanced to the point where the defective stage is isolated, it is often

necessary to refer to the circuit diagram and analyze in detail that particular stage. Since all of the stages in a receiver are subject to failure, you should know how to analyze any one of the stages when necessary. To show how simple this can be when the analysis is made in the proper manner, we will go through the process for each stage in turn in the circuit of Fig. 1.

*Oscillator - Mixer - First Detector.* Below the sketch of band-changing switch 48 is a note indicating that this switch is shown in the broadcast band position. Comparison of this diagram with the individual switches in the R.F. input and oscillator coil circuits shows that these separate switch symbols are also in the broadcast band position. For the broadcast band setting, then, the short across the primary of antenna coil 1 makes this coil inactive. Signals picked up by the antenna (the RED terminal) thus pass through the primary of antenna coil 2 and through switch contacts A1-A2 to ground, inducing in the secondary winding an R.F. voltage which is applied through the inactive secondary of antenna coil 1 to the fourth grid of the 6A8G tube. The fact that the secondary of antenna coil 2 has a higher D.C. resistance than the secondary of antenna coil 1 verifies the fact that coil 2 serves for the broadcast band (a higher resistance usually indicates more turns of wire, giving the higher inductance required for the lower-frequency bands).

The only switch in the oscillator circuit, that for contacts A3-A4, is open during broadcast band operation, and consequently oscillator coil 6 acts in series with oscillator grid coil 5 in the oscillator tuned circuit. The feedback voltage required for oscillation enters this circuit through oscillator plate coil 5, which is in the second grid (oscillator plate) circuit of the 6A8G tube. The oscillator section of the gang tuning condenser connects

between the first grid and ground, and consequently is shunted across both coils in the tuned circuit.

For the short-wave setting of the band-change switch, contacts A1-A2 are open, making antenna coil 1 effective, and contacts A6-A7 are closed, shorting the secondary of antenna coil 2 and thereby making this transformer ineffective. Switch contacts A3-A4 in the oscillator circuit are connected together, shorting oscillator coil 6 and thus reducing the inductance in the oscillator tuned circuit to the value required for the higher-frequency band.

Condensers 7 and 8 acting with resistor 9 furnish the automatic C bias for the local oscillator. Condenser 7B is simply an oscillator coupling condenser, which serves to feed A.C. plate current into the tuned grid circuit during broadcast band operation. Condenser 7 also serves as the low-frequency padder in the oscillator circuit for broadcast band operation. Trimmer condenser 7A is the oscillator high-frequency trimmer for the broadcast band, and trimmer condenser 4B is the oscillator high-frequency trimmer for the short-wave band. There is no oscillator low-frequency padder on the short-wave range. High-frequency trimmer 4A is the only input circuit alignment adjustment; consequently we can assume that the input circuit or preselector is quite broad in frequency response on the short-wave band, and is adjusted only for the broadcast band.

The pictorial diagrams of the antenna and oscillator coils, at the lower left in Fig. 1, give coil connection information which is often helpful in tracing continuity through the coil circuits. The small numbers on these coil sketches correspond to the numbers on the schematic symbols for the corresponding coils.

*I.F. Amplifier.* We recognize the 6K7G I.F. amplifier tube as a pentode,

and learn from a tube chart that it is a super-control amplifier tube which can be A.V.C.-controlled. Tracing from the control grid or first grid of this tube through the secondary of the first I.F. transformer, we find the circuit to be through one-megohm resistors 15 and 27 to a tap on the voltage divider in the power pack, and then to ground. This means that the tube receives an initial negative C bias from the power pack. From the common connection of resistors 15 and 27 we note a lead which can be traced to ground through one diode section of the 6J5G tube (that formed by the plate and cathode), which means that the tube also receives an A.V.C. voltage.

There is a third winding on the first I.F. transformer, which places the suppressor grid at D.C. ground potential. The secondary of this I.F. transformer induces in this coil a signal voltage which can produce either regeneration or degeneration, depending upon the manner and upon the phase relationship of the secondary coil current with respect to the coil voltage. This third coil is connected in such a way that whenever the circuit tends to oscillate (regenerate), a degenerating voltage is applied the suppressor; likewise, when the circuit tends to degenerate during off-tune conditions, a regenerating voltage is applied to the suppressor. The third winding thus serves to stabilize the circuit during off-tune conditions.

Following the 6K7G tube is the second I.F. transformer, of conventional design. Trimmer condensers 13A, 13B, 19A and 19B permit peak adjustments of the two I.F. transformers.

*Second Detector.* The output voltage of the I.F. amplifier is applied to the grid and cathode of the 6J5G triode, with resistors 20 and 21 serving as the load for this diode detector circuit. Resistor 20 also acts in combination with condensers 19C and 19D

as an I.F. filter. Rectified diode current flowing through resistor 21 develops across it an A.F. voltage which is transferred through D.C. blocking condenser 24 to potentiometer 26. Resistor 32 and condenser 38, connected between a tap on this potentiometer and ground, provide automatic bass compensation. Switch 39 has one contact for shorting out condenser 38 when automatic bass compensation is not desired, and other contacts for tone control circuits. The movable contact on volume control potentiometer 26 feeds A.F. signals through condenser 28 to the grid of the 6K5G first audio amplifier tube.

*A.V.C. Tube.* The output voltage of the I.F. amplifier also acts upon the plate of the 6J5G triode tube through condenser 23. The grid and cathode of this tube thus serve as one diode, while the plate and cathode form another diode section which acts as the A.V.C. tube. Resistor 27 (trace down from the plate of the 6J5G tube) serves as the load for this second diode; the D.C. voltage required for A.V.C. purposes is developed across this resistor. The circuit from resistor 27 to ground is through the 8-ohm and 35-ohm sections of voltage divider 43, and consequently the D.C. voltage drop across these sections is applied to the A.V.C. controlled tubes in the form of a constant negative C bias, as well as a delay voltage for the diode A.V.C. tube. The A.V.C. voltage is applied to the two A.V.C.-controlled tubes (6A8G and 6K7G) through an A.V.C. filter made up of resistor 15 and condenser 3.

*First A.F. Amplifier Stage.* A tube chart tells us that the 6K5G tube is a high mu triode designed especially for use in resistance-capacitance coupled audio amplifiers. Resistor 36, coupling condenser 34 and grid resistor 35 make up the resistance-capacitance coupling network through which this tube feeds into the output

**stage.** A negative C bias for the first audio tube is obtained from tap 2 on voltage divider 43, with resistor 33 and condenser 30 serving as a filter.

**Output Stage.** According to a tube chart, the 6F6G tube in the output stage is a power amplifier pentode. The entire voltage drop across voltage divider 43 serves as the negative C bias for this tube, and likewise the entire output voltage of the power pack is applied to the plate of the tube through the primary winding of output transformer 41. Note that condensers 37 and 40 are connected in series between the plate of the output tube and ground, with the common connection of these condensers going to one contact on tone control switch 39. When the switch is set at this contact (at its extreme right-hand position), condenser 40 is shorted out and condenser 37 then by-passes the higher frequency components of the intelligence signal to ground, giving the effect of a boost in bass notes. When switch 39 is at some other setting, the extremely low capacity of condenser 40 (.008 mfd.) makes the reactance of this shunt path to ground so high that there is very little attenuation of any signal frequencies.

The voice coil of the dynamic loudspeaker, marked 42, is connected across the secondary of output transformer 41. The single-loop coil symbol drawn in series with the voice coil but in the opposite direction represents a hum bucking coil.

**Power Pack.** Obviously we have a full-wave rectifier circuit here, since the two plates of the 5Y4G rectifier tube are connected to the outer ends, 5 and 7, of the high voltage secondary winding on power transformer 46. Center tap 6 on this secondary winding is therefore the negative or B— terminal of the power pack; it traces through resistor 43 to the chassis and ground, making terminals 2, 3 and 4 on this resistor increasingly more

negative with respect to point 1, the chassis.

Point 3 on the rectifier tube filament winding is the high-voltage terminal of the rectifier system, so we will find the power pack filter network connected between this point and either the B— terminal or ground. The main filter employs condenser input (45) with loudspeaker field coil 44 serving as a choke and condenser 11A serving as the output filter condenser. The symbols indicate that condensers 45 and 11A are of the electrolytic type. The common terminal of field coil 44 and filter condenser 11A is the B+ or high-voltage output terminal of the power pack, and feeds directly to the plate of 6K7G I.F. tube, the 6K5G audio amplifier tube and the 6F6G output tube through the plate load of each tube.

A separate filter system is employed for the 6A8G pentagrid converter, however; this is made up of filter resistor 12 and electrolytic condenser 11. A voltage divider network (resistors 16 and 22) is connected between the output of this filter and ground, with the higher output voltage being fed through resistor 10 and the oscillator feed-back coil to the second grid (oscillator plate) of the 6A8G tube. The plate of the 6A8G tube receives this same high output voltage through the primary of the first I.F. transformer. A lower voltage is applied from the common junction of resistors 16 and 22 to the screen grids of the 6A8G and 6K7G tubes, with condenser 14 serving as the screen grid by-pass condenser.

Filament connections are indicated but not completed on this diagram. Observe that terminal 8 of the filament winding on power transformer 46 is grounded and that one filament terminal of each tube except the rectifier is likewise grounded. The other filament terminal for each tube is terminated in an arrow, as also is the

other filament winding terminal; this of course, indicates that all these terminals are connected together. All tube filaments except the rectifier are thus connected together in parallel across the filament winding having terminals 8 and 9.

Observe that two condensers, marked 47, are connected in series across the 110 volt A.C. power line, with their midpoints grounded. These condensers shunt to ground any noise interference signals which might otherwise enter the receiver.

The extreme left position of switch 39 is the off or open position for the receiver on-off switch which is in series with the primary of the power transformer. This power switch is closed for all other positions of switch 39. Moving this switch one contact to the right shorts out automatic bass compensation condenser 38, providing normal receiver operation. Moving the switch one more contact to the right places this condenser in the circuit, and the final position of the switch to the right shorts out condenser 40, giving a bass-boosting effect together with automatic bass compensation.

A note at the bottom of the schematic circuit diagram indicates that on some receiver models employing this circuit, a shadowgraph indicator is inserted in series with the plate supply lead to the I.F. amplifier tube, with a .05 mfd. condenser connected between the plate side of this meter and the chassis to keep R.F. current out of the meter. The small diagram at the lower right of the circuit diagram indicates that pilot lamp 49 is connected in parallel with the tube filaments across the filament winding on the power transformer. A note indicates that an extra pilot lamp (49X) is used on one model to operate a shadowgraph tuning indicator.

Remember that it will seldom if ever be necessary for you to analyze a complete receiver circuit as we have

just done. Modern servicing techniques isolate the trouble to a particular stage or section, making it necessary to analyze that one small part of the receiver circuit. Furthermore, do not expect to be able to analyze a diagram or even a part of a diagram completely right from the start. As you acquire additional knowledge and experience, you will find it easier and easier to secure the information which schematic diagrams can give you.

### Identifying Stages on the Chassis

By including top-of-the-chassis layout diagrams along with the schematic diagrams in some service manuals, receiver manufacturers have made it quite easy to identify each tube and its function. A typical diagram of this type, in which the position of each tube is clearly indicated with respect to easily recognizable parts such as the power transformer and the tuning condenser, is shown in Fig. 3. With this layout at hand, it is a simple matter to perform either a dynamic stage-by-stage elimination test or a circuit disturbance test when hunting for the defective stage.

A tube layout diagram has other uses in radio servicing. Since octal-base tubes are used in practically all modern receivers, any tube will fit into any socket. The sockets on the chassis are ordinarily not marked for the proper tubes, and consequently the tube layout diagram serves as a valuable guide for replacing tubes when all are removed for testing, or in order to clean the chassis. The wise Radio-trician usually does not depend upon a tube layout diagram, however; he removes only one tube at a time, returning it or replacing it with a new tube before testing the next tube.

But how do we go about identifying the stages on the chassis when no schematic diagram and no tube layout diagram are available? All we

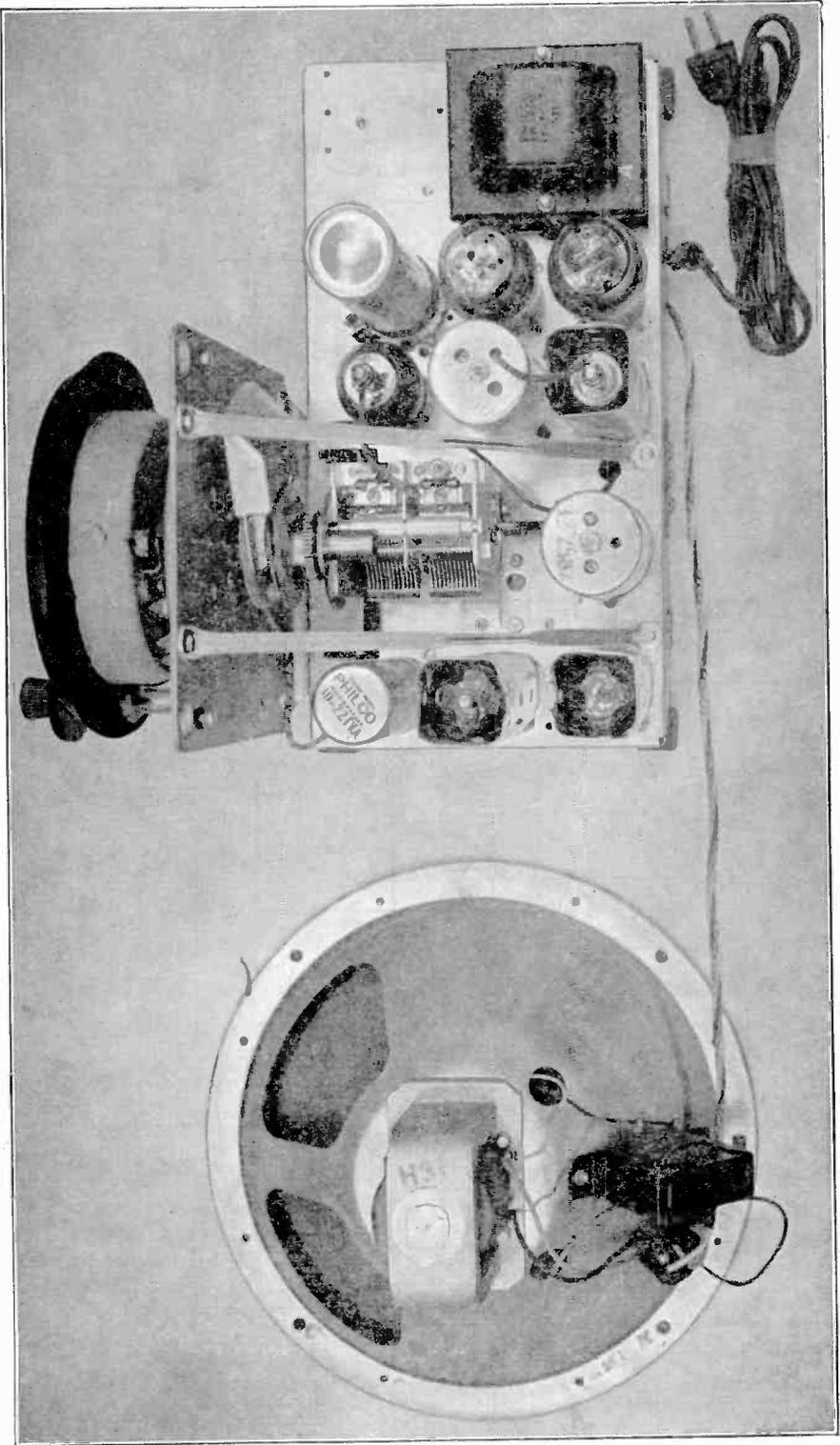


FIG. 5. Top of chassis of Philco model 38.7 receiver



Schem. No.	Description	Part No.
1	Antenna Transformer—Short Wave	32-2559
2	Antenna Transformer—Broadcast	32-2559
3	Condenser .05 mf.	30-4519
4	Tuning Condenser, Models 8 and 9	31-2026
5	Osc. Transformer—Short Wave	31-2030
6	Osc. Transformer—Broadcast	32-2560
7	Compensator Dual Models 8 and 9	31-6188
8	Compensator, 300 K.C. (Model 7)	31-6185
9	Compensator, 350,000 ohms mica	30-1084
10	Resistor 70,000 ohms (1/2 watt)	33-50330
11	Resistor 5000 ohms (1/2 watt)	33-25059
12	Resistor 10,000 ohms (3 watt) mid.	30-2217
13	1st I. F. Transformer	33-310639
14	Condenser .1 mf.	32-2580
15	Resistor 1.0 meg. (1/2 watt)	30-4455
16	Resistor 10,000 ohms (1 watt)	33-510339
17	Resistor .05 mf. (38-8 only)	33-310439
18	Shadower (38-8 only)	30-4474
19	2nd I. F. Transformer (mounted in 19)	33-510339
20	Resistor 51,000 ohms (1/2 watt)	33-49339
21	Resistor 490,000 ohms (1/2 watt)	33-49339
22	Resistor 51,000 ohms (1 watt)	33-351430
23	Condenser, mica, 110 mmf.	30-1031
24	Removed Prior to Production	30-4479
25	Volume Control	33-5216
26	Resistor 1 meg. (1/2 watt)	33-510339
27	Resistor 1 meg. (1/2 watt)	30-4358
28	Audio Shorting Switch (38-7 only)	Part of Selector Crank
29	Condenser .1 mf.	30-4400
30	Resistor 1.0 meg. (1/2 watt)	33-510339
31	Resistor 1.0 meg. (1/2 watt)	33-510339
32	Resistor 1.0 meg. (1/2 watt)	33-510339
33	Condenser .015 mf.	30-4515
34	Resistor 1.0 meg. (1/2 watt)	33-510339
35	Resistor 80,000 ohms (1/2 watt)	33-510339
36	Condenser .05 mf. (1/2 watt)	30-4417
37	Condenser .05 mf.	30-4497
38	Tone Control	30-1197
39	Output Transformer (Model 7)	30-1112
40	Output Transformer (Models 8 and 9)	32-7882
41	Cone and Voice Coil Assembly (H31)	32-7019
42	Cone and Voice Coil Assembly (K41)	36-3801
43	Cone and Voice Coil Assembly (HS)	36-3174
44	Cone and Voice Coil Assembly (S7)	36-3796
45	Field Coil Assembly (H31)	36-3316
46	Field Coil Assembly (K41)	36-3665
47	Field Coil Assembly (HS)	36-3801
48	Field Coil Assembly (S7)	36-3039
49	Electrolytic Condenser	30-2219
50	Power Transformer, 115V, 50/60 cycle	32-7833
51	Power Transformer, 110V, 25 to 40 cycle	32-7827
52	Power Transformer, 115/230V, 50/60 cycle	32-7835
53	Volume Control	3793DG
54	Wiring Diagram	42-1825
55	Pilot Lamp, Models 8 and 9	34-2064

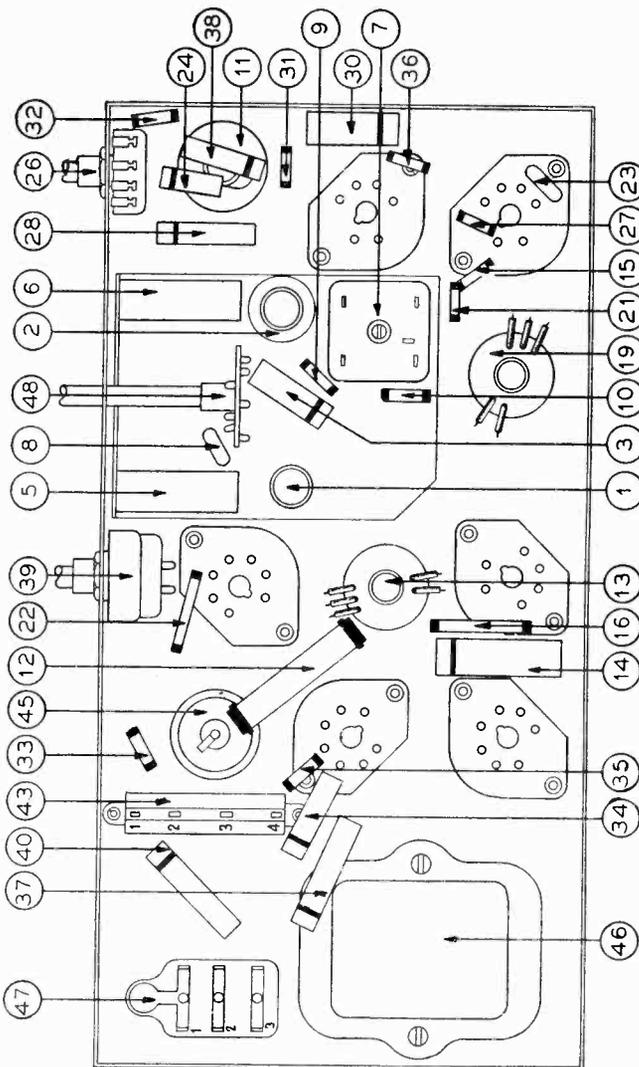


FIG. 2. Pictorial layout diagram for parts under chassis of Philco model 38-7 receiver. The parts list on this page serves as a guide when ordering replacement parts for this receiver.

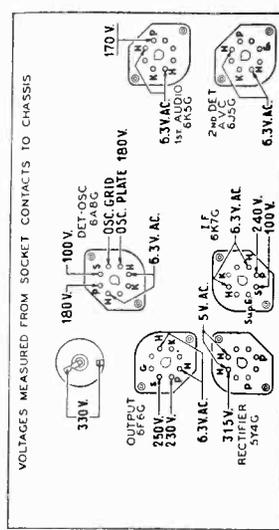


FIG. 4. Tube socket connection diagram and electrode voltages for Philco model 38-7 receiver.

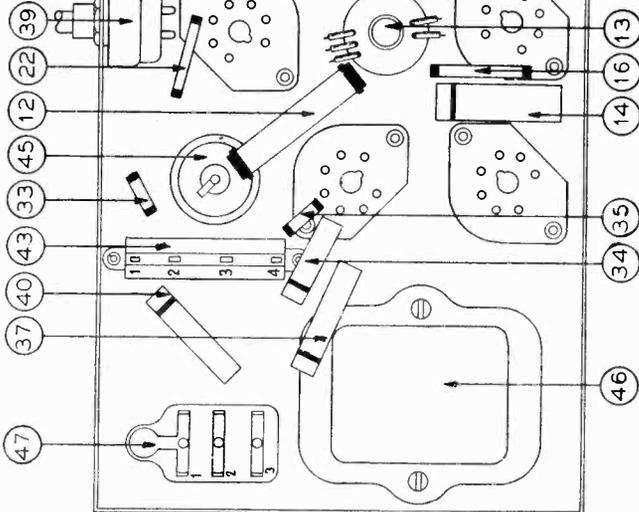


FIG. 3. Pictorial layout diagram for parts above chassis of Philco model 38-7 receiver.

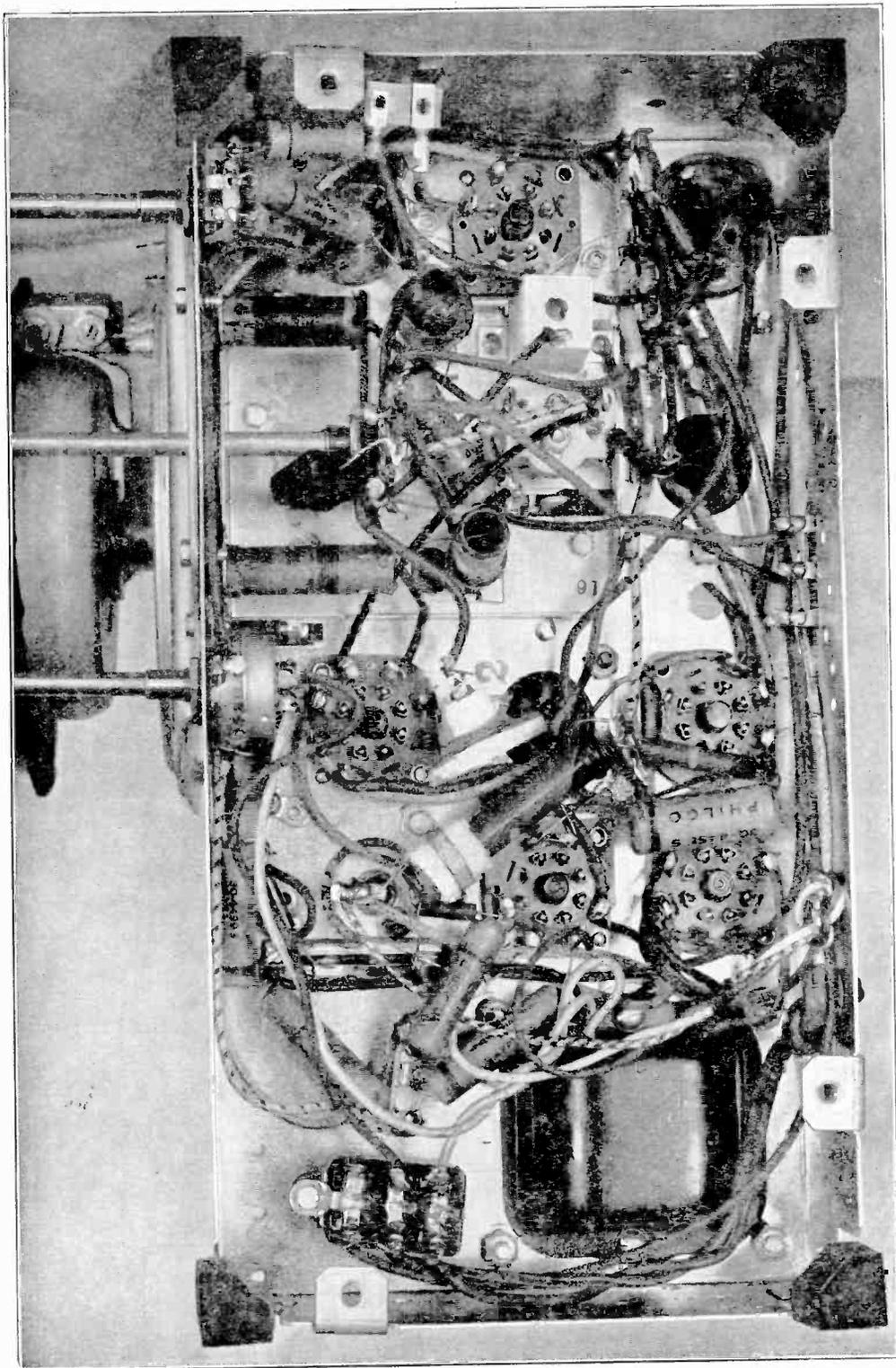


FIG. 6. Bottom of chassis of Philco model 38-7 receiver.

have is the chassis itself, which can be represented in this discussion by the photographs in Figs. 5 and 6. We will concentrate upon Fig. 5, which shows a top view of the chassis, since you will often want to identify the tubes and stages without removing the chassis from its cabinet.

The power transformer and the two-gang tuning condenser are, of course, easily identified. Likewise you can be pretty sure that the two cylindrical metal cans adjacent to the tuning condenser, each having two adjusting screws, are I.F. transformers. Their presence identifies the receiver as a superheterodyne. The fact that there is a lead coming out of one I.F. transformer to the top cap of a tube is further proof that we are dealing with a superheterodyne, for in tuned radio frequency receivers the top caps of tubes usually go to the stator sections of the gang tuning condenser.

The shapes of the completely enclosed metal cans, one to the right and below the power transformer, and the other at the lower right on the chassis, identify these parts as electrolytic filter condensers; with experience you will be able to spot these at a glance.

Once you have determined the type of circuit, you can proceed to identify the tubes. First of all, look for the rectifier tube, which will have two rectangular electrodes with a heavy filament wire inside each, but no other electrodes. We notice a tube like this, marked 5Y4G, at the upper right of the power transformer (see Fig. 5); reference to a tube chart verifies that this is a rectifier tube.

Below the rectifier tube and next to the power transformer in Fig. 5 is a tube which is larger than all other tubes except the rectifier. This has a rather large plate, with a cathode and several other elements inside. By counting the elements or by deter-

ining its number and referring to a tube chart, we identify this as a pentode; since there is no top cap connection for the control grid, we know definitely that it is not an R.F. pentode. It is logical to assume, then, that we have a power output pentode, a tube widely used in the output stages of receivers. A tube chart would verify this assumption.

To the left of the tuning condenser in Fig. 5 is a tube marked 6A8G; a tube chart identifies this as a pentagrid converter, the fourth grid of which connects to the top cap. We say, therefore, that this tube is for the oscillator-mixer-first detector stage. A lead from the top cap of this tube goes to the upper stator section of the tuning condenser, identifying it as the preselector tuning condenser. The other section of the tuning condenser must therefore tune the oscillator circuit.

At the right of the rectifier tube in Fig. 5 is a tube marked 6K7G; according to our tube chart, this is an R.F. pentode. Since there are only two sections in the gang tuning condenser, one for the first detector input and one for the oscillator, we know that there is no R.F. amplifier stage. This means that the 6K7G R.F. pentode must be used as an I.F. amplifier; the fact that there is a lead coming from the I.F. transformer to the top cap of this tube verifies our assumption. The tube chart also reveals that this is a super-control tube, and since a receiver designer ordinarily does not employ such a tube unless it is to be A.V.C.-controlled, we know that this receiver has A.V.C.

The two tubes at the upper right on the chassis in Fig. 5 still remain to be identified. The upper is marked 6J5G, and reference to a tube chart reveals it to be a triode which can have a number of different uses. The lower tube is marked 6K5G but here the tube chart tells us definitely that

it is a high-gain audio amplifier tube. Since we have not yet located the second detector-A.V.C. tube, with only these two tubes left to identify, we can say that the 6K5G is the first audio amplifier tube and the 6J5G is the second detector-A.V.C. tube. We could verify this by measuring the voltage between the plate and cathode of each tube; we would find that the tube selected as the audio amplifier had a reasonably high D.C. supply voltage, whereas the plate of the 6J5G tube was slightly negative with respect to the cathode. These measurements would tell us that a delay voltage was present in the A.V.C. system.

We have just seen how each tube in this receiver can be identified as to its function without removing the chassis from its cabinet. Naturally a person would not go through this procedure if a tube diagram were at hand, or if a schematic circuit diagram were available, for it is not always possible to determine the function of each tube on a chassis without tracing connections underneath the chassis. On the other hand, with experience you will be able to identify the various tube stages on most receivers almost at a glance, and will not have to refer to diagrams.

*General Suggestions.* The ability to recognize radio parts by their appearance alone is obviously quite desirable when it is necessary to identify stages on the chassis. For this reason it is important that you keep in touch with new developments as announced in radio magazines and in the latest catalogs of mail order radio supply houses or of your local radio parts distributor.

You can generally secure the latest tube charts from your local parts distributor, or from the manufacturer of the tubes which you handle. These charts will familiarize you with the

appearances and characteristics of new and old tubes alike.

The first thing you want to know, when working on a chassis without a service manual, is whether you have a T.R.F. or superheterodyne receiver. Look for I.F. transformers, as they are clues identifying a superheterodyne; if these I.F. transformers are under the chassis, as they often are, look for the screws of their adjusting trimmers on the top of the chassis. If one of the sections of the gang tuning condenser has rotor plates which are shaped differently from those in other sections, you have another clue toward a superheterodyne. A receiver having only two sections in the gang tuning condenser but with six or more tubes can reasonably be assumed to be a superheterodyne. Older radio receivers with three, four and five sections in the condenser gang will generally be T.R.F. sets, but make sure that there are no I. F. transformers before deciding this.

Next you will want to know how the receiver is powered. If there is a multi-lead cable in place of the line cord, it is a battery receiver; if there is a two-wire cord having at its end the familiar two-prong wall outlet plug, you can be sure that it is a socket-powered receiver. But do we have an A. C., D. C., or universal socket-powered receiver? If a power transformer is visible on the chassis, you can be sure it is an A. C. receiver. If there is a rectifier tube but no power transformer, it is a universal A.C.-D.C. receiver. If there is no power transformer and no rectifier tube, it is a D.C. receiver. These rules will enable you to identify practically all receivers, but be on the lookout for exceptions. A few transformerless A. C. receivers have been made; these will have the conventional full-wave rectifier tube, so you will have to check the rectifier con-

nections to make a positive identification. \*

Now let us concentrate our attention upon the stages ahead of the I.F. amplifier in a superheterodyne receiver. A two-section ganged variable condenser means that there is no R.F. amplifier stage ahead of the first detector, for one section serves for tuning the oscillator and the other for tuning the input to the first detector. Three sections ordinarily indicate that there is one stage of R.F. amplification ahead of the mixer-first detector, but occasionally you may find a band-pass preselector instead. You may find a pentagrid converter tube or a pentode tube being used as a combination mixer-first detector-oscillator, or there may be a pentode serving as mixer-first detector and a separate triode (or even a screen grid or pentode tube) serving as the oscillator tube.

Most superheterodynes use screen grid or pentode tubes in the R.F. first detector and I.F. stages; this means that the control grid connection for each of these tubes will be to a top cap. In the case of a pentagrid converter, the top cap will be connected to the fourth grid inside the tube, and there will be an external connection from this cap to the stator section of the detector input tuning condenser;

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\* There is seldom need to make a positive identification unless you isolate a defect to the power pack. In this case you will naturally remove the chassis, and an inspection of rectifier connections will reveal the type of circuit used. If it is an A.C. set with a voltage-doubling power pack circuit, you will usually find that the plate of one diode section is directly connected to the cathode of the other diode section. There will be two electrolytic condensers connected in series, with the negative lead going to the remaining diode plate and the positive lead to the remaining cathode. The common connection of these two condensers goes to one side of the power line, and the other side of the power line goes to the common plate-cathode connection. The circuit for this is given in an earlier lesson in your Course.

with a three-gang condenser the oscillator stator will be at one end, and will not have a connection to a tube cap if a pentagrid converter is used.

Because I.F. transformers require complete shielding, they will be found in metal shields or cans and will usually be mounted on the top of the chassis. The secondary of the first I.F. transformer will always be connected to the control grid of the I. F. amplifier tube; since this will be a screen grid or pentode tube, you can expect to find a flexible lead coming out of the top or side of the I.F. transformer and going to the top cap of the tube.

When there are two I.F. transformers, the second will feed into the second detector, usually by means of an under-chassis connection (only when the second detector is a screen grid or pentode tube will there be a connection from the second I.F. transformer to its top cap).

Since diodes or triodes are ordinarily used as second detectors, you can usually assume that any tube having a top cap connection into an I.F. transformer is an I.F. amplifier tube. If there are three I. F. transformers, look for two I.F. amplifier tubes; in all probability you will find them in line with the mixer-first detector and the second detector.

The second detector has a few peculiarities which permit easy identification. Look for a double diode, a double diode-triode, a double diode-pentode, or a triode following the last I.F. transformer. Occasionally you may run into a screen grid or pentode second detector which will have a lead from its top cap to the last I.F. transformer. When the R.F. and I.F. stages use pentode tubes and there is a pentagrid converter in the line-up, look for tubes such as the 56, 6C5, 6C6, 55, 85, 75, 6B8 and 6H6 serving as the second detector, the second de-

tector-A.V.C. tube, or as the second detector-A.V.C.-first A.F. amplifier.

Rectifier tubes in an A.C. receiver are easy to locate, for they are almost always *right next to the power transformer*. Obviously it would not be good design practice to run high voltage A.C. leads any distance through the chassis from the power transformer to the rectifier tube. Look for such tubes as the 80, 25Z5, 82, 83 5Z3, 5Z4 and 6X5, for these are all rectifier tubes. Most of these have glass envelopes, through which you can usually see two long black rectangular or cylindrical anodes, with a thick V-shaped filament inside each.

Output tubes can be recognized by their numbers and by their relatively large size in comparison to all other tubes except the rectifier. Double-triode power amplifier tubes such as the 19, 6N7, 6A6, 79 and 53 are instantly identified as output tubes connected for push-pull or push-push operation. Such tubes as the 45, 47, 2A3, 6L6, 2A5, 59, 43 and 42 are power output tubes which you will find used singly or in pairs for push-pull or push-push operation.

An additional tube, usually a triode, located between the second detector and the power tubes or near them is very likely an audio voltage amplifier tube.

Cathode ray tuning indicator tubes can, of course, be identified on sight. Automatic frequency control tubes, noise-limiting and noise-suppression tubes, A.V.C. tubes and A.V.C. amplifier tubes are not so easy to identify from the top of the chassis. When the circuit contains extra tubes which are not readily identified, it is wiser to secure the circuit diagram and identify these extra tubes by a process of elimination or by comparing tube numbers. With some receivers it is practically impossible to identify all tubes merely by studying the top of

the chassis; in these cases it is either necessary to refer to a service manual or remove the chassis and trace tube connections. When metal tubes are used, the tube numbers stamped on the sides of the tubes must serve as your guide.

### Locating Parts with a Pictorial Layout Diagram

Let us suppose that the receiver being used as our example in this lesson has developed an annoying squeal. We might logically suspect that a screen grid by-pass condenser is open somewhere; referring to the schematic circuit diagram in Fig. 1, we see that screen grid by-pass condenser 14 for the 6K7G I.F. amplifier tube is a likely offender. We can make a quick check by shunting this condenser with a good unit of the correct value (.1 mfd.), provided that we can locate condenser 14. An examination of the underside of the chassis reveals at least nine tubular paper condensers which could be 14; by referring to the parts layout diagram in Fig. 2, however, we learn that the condenser under suspicion is near the back edge of the chassis and is between two tube sockets. With this information, it is easily located on the chassis; turn to Fig. 6 and see how quickly you can locate it on the photo. If the squeal stops when we temporarily shunt this condenser with a good condenser, we know that condenser 14 requires replacement.

Now let us consider the case where the receiver has an annoying hum, and an inspection for surface defects reveals that the rectifier tube is gassy (a blue glow can be seen between its electrodes). Before inserting a new tube, we naturally want to check the electrolytic condensers with an ohmmeter to determine if excessive leakage through them was the cause of rectifier tube failure. To

locate these condensers, we first refer to the circuit diagram and determine what numbers are assigned to them. At the right of the rectifier tube in Fig. 1 are two electrolytic condensers, 45 and 11A, while above the power transformer is another, marked 11. This numbering indicates that 11 and 11A are in the same housing, and the fact that only two electrolytic condensers were found on top of the chassis is further proof of this assumption.

We locate condenser 11 on the pictorial layout diagram in Fig. 2, finding it in the upper right corner of the diagram, near the volume control. An inspection of the connections underneath this chassis would show three leads, colored black, green and red respectively. This means that the two condenser sections must have a common connection; an inspection of the schematic circuit diagram verifies this and shows that the common negative terminal is grounded. The black lead on the condenser itself is grounded, so this must be negative; the red and green leads are therefore the positive leads for these electrolytic condensers. A leakage test is now readily made with an ohmmeter, connecting the plus terminal of the ohmmeter (that terminal which goes to the plus terminal of the ohmmeter battery) to the plus terminal of the condenser. The practical man seldom bothers to figure out condenser polarity, however; he connects both ways and uses the highest-resistance reading, for he knows he will then have the correct polarity. One of the condenser leads is disconnected for this test, as there are usually other parts shunting the condenser.

Let us consider one more case, that where the receiver is dead and the defect is localized to the pentagrid converter by means of a stage-by-stage elimination test. Measurements

of electrode voltages on this tube show that there are no D.C. supply voltages; other measurements reveal that there is no screen grid voltage on the I.F. amplifier tube. A study of the schematic circuit diagram reveals that an open circuit through resistor 12 could be a cause of the trouble. The parts layout diagram in Fig. 2 shows that resistor 12 is located under the chassis, between electrolytic condenser 45 and I.F. transformer 13. Since it is a 10,000 ohm resistor, we have the additional clue that it will have a brown body, black end and orange dot in accordance with the RMA color code for resistors. You should now be able to locate this resistor in Fig. 6; it is a large carbon resistor, supported at one end by a metal clamp under which is white insulating paper. Naturally you would look for a defect in this resistor or for a break in its connection.

These two examples show the value of the parts layout diagram in bridging the gap between the schematic circuit diagram and the actual receiver chassis.

### **Locating Parts without a Pictorial Layout Diagram**

The procedure for locating on the chassis a part which is indicated on the schematic circuit diagram becomes considerably more involved when no pictorial layout diagram is available. Familiarity with the appearance of various radio parts will speed your search, as also will experience in tracing actual wiring on a chassis. A few examples will illustrate the procedure to be followed.

On the schematic circuit diagram, condenser 14 is shown connected between the screen grid of the 6K7G I.F. amplifier tube and ground; let us see how we would go about locating this condenser on the chassis. First of all we locate the 6K7G tube

on the top of the chassis and determine the position of its socket underneath the chassis. We would find this socket to be at the rear edge of the chassis; almost exactly midway between the sides of the chassis (you can locate it for yourself on Fig. 6 by referring to the socket connection diagram in Fig. 4).

The next step is the location of the screen grid terminal on this socket; a tube chart will give this information if a socket connection diagram like that in Fig. 4 is not available. We look for terminal *S*, for this letter (as well as the notations  $G_s$  or  $G_2$ ) are used to designate the screen grid terminal. We know that condenser 14 is connected between this terminal and ground, and a careful study of the schematic diagram shows that there is no other condenser connected to this screen grid terminal. We trace each lead in turn from this terminal of the socket until we locate one going to a tubular paper condenser marked .1 mfd.; we note that the other lead of this condenser goes to a soldering lug which is riveted to the chassis, and thus have definite proof that this is condenser 14.

Here is another example: suppose that the trouble has been isolated to the R.F. input circuit. We suspect an open circuit in one of the windings of R.F. transformers 1 and 2. A study of the schematic circuit diagram shows that if we connect an ohmmeter between the two antenna terminals marked *RED* and *BLK*, we can check continuity of the primary winding. If the ohmmeter indicates a resistance of about .1 ohms when the band change switch is in the broadcast position (shorting contacts *A1* and *A2*), we know that the primary of R.F. transformer 2 is good. A changeover to the short-wave setting of the switch removes the short across the primary of R.F. trans-

former 1, and if the ohmmeter reads about .2 ohm, we know that this primary winding also is good.

Now we study the circuit diagram again to see how we can check the continuity of the secondary coils. Observe that there is a complete conductive path from the fourth grid of the 6A8G tube to the secondary of the first I.F. transformer and control grid (top cap) of the 6K7G I.F. amplifier tube. The diagram further indicates that if we place an ohmmeter between the top caps of these two tubes, we should measure a D.C. resistance of  $.1+2.3+14$ , or a total of approximately 16.4 ohms. We do not even have to remove the chassis from the cabinet in order to make this test.

If, when measuring the resistance between the top caps of the first two tubes, we secured an infinite resistance reading, we would know that an open circuit existed somewhere along the path being checked. To make sure that the trouble is an R.F. coil, we could make an ohmmeter test between the top cap of the 6K7G tube and the chassis. The schematic circuit diagram indicates a resistance of a little over 2 megohms (the combined resistance of parts 15, 27 and a section of 43) between these two points. Any resistance reading differing greatly from this value, or an open circuit reading, would indicate trouble in the I.F. transformer. If the ohmmeter reads the correct value of 2 megohms, we make the same test between the top cap of the 6A8G tube and the chassis; let us assume that we get an open circuit reading.

To determine which of the R.F. coils is defective, we must locate them underneath the chassis and test each one individually. The sketches at the lower left in Fig. 1, with each coil connection numbered and the relative lengths of the windings shown, makes

identification of these coils on the chassis quite simple.

Oftentimes, however, these coil pictures are not provided on the circuit diagram. In this event we would trace from the top cap of the 6A8G tube to the input tuning condenser stator, and from there to point 3 on the secondary of R.F. transformer 1. (Each stator section has two terminals, one above and one below; the R.F. transformers are underneath the chassis, so we look underneath the chassis for a lead coming through it directly under the stator section in question.) We trace through the heavy wire forming the secondary winding to the other terminal, marked 4, and then in turn trace to point A6 on the band change switch, to point 3 of the secondary of R.F. transformer 2, and through this winding to terminal 4. We can then make a continuity test across this entire circuit or any part of it.

### Tracing Tube Electrode Circuits

Once the various tube stages have been identified, the defective stage isolated and the tube in that stage checked, the next logical step is a check-up of the circuits in the defective stage. These tests are simple and easy to make if you recognize that electrode circuits in any receiver can always be traced to certain definite points. It is then a simple matter to make continuity tests of the paths between the tube electrodes and these points. Let us trace the conductive paths from the electrodes in Fig. 1 before considering the general continuity-checking rules which apply to all circuits.

*Plate Circuits.* Suppose we start at the plate of the 6K7G I.F. amplifier tube and trace a conductive path as far as we can. From the plate we go through the primary of second I.F. transformer 19 in Fig. 1, down a lead

marked *RED* to loudspeaker field coil 44, and through this coil to the filament of the 5Y4G rectifier tube. Clearly the rectifier filament is one terminal in this path.

Now trace from the plate of the 6A8G pentagrid converter tube through the primary of first I.F. transformer 13 and down another lead marked *RED* through resistor 12 to the rectifier filament. The plate of the 6K5G first audio amplifier tube traces through resistor 36 and loudspeaker field coil 44 to the rectifier filament, and likewise the plate of the 6F6G output tube traces through the primary of the output transformer and through field coil winding 44 to the rectifier filament.

There is a very good reason why the plates of all these tubes should trace to the rectifier filament or to a rectifier cathode. In any rectifier type of power pack the filament is the highest positive voltage terminal, and consequently all electrodes which are supplied with a positive voltage must eventually trace to the rectifier filament.

*Screen Grid Circuits.* Since the screen grid of a tube is likewise supplied with a positive D.C. potential, you should also expect to trace a conductive path from it to the rectifier filament. We can check this very easily in Fig. 1. From the screen grid of the 6K7G tube (the middle grid, which is also connected directly to the screen grid of the 6A8G detector-oscillator tube), we trace through resistors 16 and 12 to the rectifier filament. In a similar manner we can trace from the screen grid of the 6F6G output tube through loudspeaker field coil 44 to the rectifier filament.

The only other electrode in this particular circuit which requires a positive D.C. potential is the second grid of the 6A8G pentagrid converter

tube, which serves as the anode of the oscillator. Observe that this traces through *oscillator plate coil 5* and then *through resistors 10* and *12* to the rectifier filament.

**Control Grid Circuits.** Since the control grid of an amplifier tube must be negatively biased with respect to its cathode, we know that there must be a conductive path between the control grid and the cathode of an amplifier tube. Let us verify this on the circuit in Fig. 1. The fourth grid of the 6A8G tube is serving as a control grid, so we trace from it through the secondary winding of R.F. transformer *1* and *2* and then through resistors *15*, *27* and *43* to the chassis and the grounded cathode. Clearly there is a continuous path here for direct current. The first grid of this tube, serving as the oscillator control grid, likewise traces to the cathode through oscillator grid coil *5*, through coil *6* and then through resistor *9* and the chassis. The control grid of the 6F6G output tube traces through resistors *35* and *43* to the chassis and cathode. A circuit can likewise be traced from the other control grids to their respective cathodes through the chassis.

As you know, the center tap of the high voltage secondary winding on the power transformer (feeding the two plates of the rectifier tube) is the lowest negative D.C. terminal in the power supply. A further study of Fig. 1 will show that the control grids can also be traced to this most negative point in the power pack. This means that in an A.C. receiver you can, after turning the set off, make a continuity test of each control grid circuit by connecting one ohmmeter lead to the control grid and the other to the cathode of the tube, to one of the plates of the rectifier tube or to the most negative point in the power pack. In a universal receiver, where

there is no power transformer, the most negative D.C. point will be the receiver side of the main power switch; be sure this switch is open when you make your test.

**Suppressor Grid Circuits.** When an external prong connection is provided for the suppressor grid of a tube, this will either trace to the cathode of the tube or to some negative supply terminal. For continuity checking purposes, then, you may treat the suppressor grid just as if it were a control grid.

**Diode Detector Circuits.** Since a diode detector is a rectifier and consequently passes direct current, we know that there must be a conductive path from the plate through the external circuit to the cathode. An ohmmeter connected between the cathode and plate of a diode detector should therefore indicate continuity. (There may be an exception to this rule in the case of a power pack rectifier tube used as a diode rectifier, for the load on this diode may be tubes in the receiver which are conductive only when their cathodes are heated.)

The second detector-A.V.C. tube in Fig. 1 employs two diodes, one for detection and one for A.V.C. purposes, but the plate of each can be traced to the cathode. For example, the actual plate of the tube traces through resistors *27* and *43* to the chassis and then to the cathode, while the grid (which serves as the other diode plate) traces through the secondary of I.F. transformer *19* and through resistors *20* and *21* to the chassis and cathode.

### **General Rules for Checking Continuity of Electrode Circuits in A.C. Receivers**

1. There should be a conductive path from the rectifier tube filament or cathode (the highest positive D.C.

terminal) to all tube electrodes which are supplied with a positive D.C. potential, such as plates and screen grids.

2. There should be a conductive path from the most negative D.C. terminal in the power pack to all control grids and suppressor grids which require zero or negative bias voltages. Likewise there should be a conductive path between the control grid and the cathode of a tube, and between the suppressor grid and cathode of a tube.
3. In diode detectors or diode A.V.C. tubes, there should be a conductive path between the plate and the cathode.

Failure to secure a conductive path in any of the cases mentioned, as evidenced by an infinite-resistance reading of the ohmmeter, indicates a break in the electrode circuit in question. It should then be a simple matter to check the circuit piece by piece to locate the break.

When the schematic circuit diagram and the parts layout diagram are available, all parts suspected of being open can be located first on these diagrams and then on the chassis. Each part is then checked for continuity, making sure that the test includes the connecting leads.

When neither the circuit diagram nor the pictorial layout diagram is available, the circuit must be traced on the chassis for the most likely path between the points in question before making ohmmeter measurements. For example, if you found that there was no continuity between the plate of the 6K5G first audio amplifier tube and the rectifier filament, you would first locate the plate terminal of the audio tube socket. There would be a wire from this terminal running around the front edge of the chassis to a terminal to which a resistor and a .015 mfd. condenser are soldered. Of course, no continuity should be expected through the condenser, so you trace through the resistor to the screen grid of the

power output tube and then to the field coil of the loudspeaker. You know that your circuit should trace through this field coil rather than through leads going to the electrodes of other tubes and so you arrive at the rectifier filament. Having traced the circuit, you can then proceed to test continuity of each part and section of it.

### How to Locate Trimmer Condensers

As you know, the location and identification of each trimmer condenser in a receiver is essential for alignment purposes. If the manufacturer supplies a circuit diagram like that in Fig. 1 and a trimmer layout diagram like that in Fig. 3, this task is simple. When no trimmer layout diagram is available, however, the problem becomes more complicated. Let us see how this would be done on the receiver being used as an example if the only service manual data available is the circuit diagram in Fig. 1.

First of all, we note that there are two holes in the top of each I.F. transformer through which can be seen adjusting screws. Those in the first I.F. transformer we can identify as I.F. trimmers 13A and 13B, while those in the second I.F. transformer must logically be I.F. trimmers 19A and 19B. We next note that there is a trimmer condenser mounted on each stator section of the gang tuning condenser. That one which is on the R.F. input stator section we identify as high-frequency trimmer 4A; the trimmer on the oscillator stator section must therefore be oscillator-high frequency trimmer 4B.

A glance over the circuit diagram in Fig. 1 shows that there are only two more trimmer condensers, 7 and 7A. Since manufacturers will make all trimmer condenser adjustments

available from the top of the chassis wherever possible, we look for these on the chassis and finally locate the two adjusting screws to the right and a little above the gang tuning condenser unit, near the last I.F. transformer. We must turn the chassis over and trace connections, however, in order to tell which is 7 and which is 7A. Figure 1 shows that these trimmers have one common connection through condenser 8 to ground, with the other connections going to opposite ends of oscillator coil 6. Trimmer 7A also connects to one terminal on oscillator coil 5, so we can positively identify as trimmer 7A on the chassis that trimmer which connects to two oscillator coils. The remaining trimmer, which connects to only one oscillator coil, will therefore be 7.

*General Trimmer-Locating Suggestions.* For each band in an all-wave superheterodyne receiver there will usually be the following separate trimmers: 1, one high-frequency trimmer for each preselector stage which is tuned to the incoming R.F. carrier frequency; 2, a high-frequency oscillator trimmer; 3, a low-frequency oscillator padder. (The receiver used as an example in this lesson is an exception to this rule, for there is one high-frequency oscillator trimmer for each band but only a low-frequency oscillator padder for the broadcast band. In the preselector, one high-frequency trimmer serves both bands.) In the case of a three-band receiver having one R.F. amplifier stage, there would be four trimmers in the preselector and oscillator sections for each band, making a total of twelve trimmers ahead of the I.F. amplifier. Usually there will be two trimmers for each I.F. transformer, but in the less expensive receivers which have no R.F. amplification and only one stage of I.F. amplification, a high gain I.F. transformer using

only one tuned circuit and consequently only one trimmer may be found.

Look for I.F. trimmer adjustment screws either at the top or on the side of the I.F. transformer shield; occasionally, however, you may find them at the bottom of the transformer, visible only from the bottom of the chassis. Some manufacturers mount these trimmers directly on the chassis rather than in the shield, but they will always be located close to the I.F. transformers they tune.

In a conventional broadcast band superheterodyne receiver, all of the high frequency trimmer condensers will be mounted on the stator plates of the condensers which they adjust. The low-frequency padder will be a separate unit, mounted somewhere near the oscillator coil or the oscillator tuning condenser section. If two or more trimmers are found on the chassis of a broadcast band receiver, trace the circuit to each in order to make sure that the one selected as the padder is not a wave trap adjustment or some other special circuit arrangement. The oscillator low-frequency padder will generally be in series with the oscillator coil; sometimes this padder will be shunted by a fixed condenser.

Ordinarily there will be no high-frequency trimmers mounted on the gang tuning condenser of an *all-wave receiver*; these will be located near the coils which they adjust, instead. (The Philco two-band receiver used as our example in the lesson is not considered an all-wave receiver.) Sometimes all of the oscillator circuit coils will be found in a separate aluminum shield can, along with the oscillator high-frequency trimmers, while another shield can will be used for the preselector coils feeding into the mixer-first detector, and one more shield for the antenna coils if an R.F.

amplifier stage is used. There will be one adjusting screw on each of these shields for each band in the receiver; for example, in the case of a three-band receiver there will be three high-frequency trimmer adjusting screws on the side or on top of each preselector and oscillator coil shield.

Sometimes the preselector and oscillator coils are placed below the chassis, completely shielded from each other and from other parts by metal partitions and a metal cover plate. The adjusting screws for the high-frequency trimmers will be accessible through holes in the cover plate.

Low-frequency padder condensers are ordinarily mounted on the chassis, and are usually ganged or grouped together on a common insulating strip. Identification of these in the case of an all-wave receiver can sometimes be made by noting the number of plates in each unit; that padder having the largest number of plates will be for the broadcast band, while the other with the fewest plates will be for the high-frequency band. It is best, however, to identify trimmers by referring to a circuit diagram and to a chassis layout when available. Experienced Radiotricians can often identify the bands controlled by preselector and oscillator trimmers simply by adjusting each in turn and noting the effects; if you attempt to do this, however, be sure to note the original setting of the trimmer so you can restore this setting in case the wrong trimmer is chosen.

### **Appraising Receiver Performance**

When servicing a receiver, it is obviously a waste of time to attempt to secure better performance than was originally intended by the manufacturer. Each receiver is designed

to sell for a definite price, and consequently, lower-priced receivers will lack many of the features which are incorporated in the more expensive sets.

Actual experience with the various types of receivers is, of course, the best guide for determining when a receiver is performing in a satisfactory manner, but there are a number of recognizable clues indicating how much can be expected from a particular receiver in the way of selectivity, sensitivity and fidelity.

First of all, the number of tuned stages in a receiver is an excellent guide as to its selectivity and sensitivity, for increasing the number of tuned stages improves both of these performance characteristics.

The presence of more than two tuned circuits between adjacent R.F. or I.F. amplifier tubes indicates improved selectivity, possibly with some sacrifice in gain (sensitivity). Plate and grid connections to taps on tuning coils are signs of improved selectivity; the use of these taps also indicates that the stages in the receiver provide more gain than is considered essential, for the introduction of a selectivity-improving scheme tends to cut down the gain.

The number of tubes in the entire R.F. and I.F. amplifier systems, not including the oscillator and oscillator control tubes, is another rough guide to the amount of selectivity and sensitivity in a receiver. A superheterodyne will give better selectivity and better gain than a T.R.F. receiver having an equal number of tubes, simply because the I.F. amplifier in a super utilizes higher Q factor circuits and more tuned circuits per stage than does a T.R.F. receiver. Furthermore the gain and selectivity will be more uniform over the entire band in a super than in a T.R.F. receiver.

The presence of a stage of R. F. amplification ahead of the mixer-first detector in a super indicates a highly sensitive receiver with a high ratio of signal level to converter noise level. In other words, with an amplifier stage in the preselector, weak stations should be heard with clarity and freedom from converter noise interference.

Do not expect the selectivity and gain of an all-wave receiver to be as good on the higher-frequency bands as on the broadcast band, for the Q factors of the tuning coils are considerably reduced at the higher frequencies.

When checking the selectivity of a receiver, pay no attention to the space on the tuning dial which is covered by a station, but rather, note whether stations which are received at your location with approximately equal intensity can be heard separately even though only 10 kc. apart.

To determine what you can expect from a particular receiver in the way of fidelity, study the loudspeaker and the compartment in back of it carefully. If the design of these indicates a wide range of frequency response, means for suppressing cavity resonance and means for utilizing bass reflection, you can be quite sure that high fidelity performance was intended.

The presence of band-pass circuits in the R.F. and I.F. amplifiers, especially in the preselector circuits, are equally important clues pointing to a high fidelity receiver. The mere fact that I.F. transformers have a tuned primary and tuned secondary is no indication that they are band-passed, but if these transformers are critically coupled or over-coupled (as indicated by your inability to secure a sharp peak response curve with a cathode ray oscilloscope), they are very likely designed for band-pass

use. If a receiver employs three or more I.F. transformers, and the design of the preselector appears to indicate a broad response, then band-passing is probably present.

An I.F. amplifier circuit employing variable coupling between the windings of the I.F. transformer is a fidelity control, permitting the customer to choose between high fidelity and high selectivity.

In receivers having automatic frequency control, the regular I.F. amplifier circuits are very likely intended to be band-passed for high fidelity if the discriminator is fed through one or more independent and highly selective tuned circuits.

### **Schematic and Chassis Wiring**

The fact that the actual wiring on a chassis may be considerably different from the connections indicated on the schematic circuit diagram cannot be emphasized too strongly even though the differences between schematic and actual chassis wiring have already been taken up in this Course. Remember that the lines on a schematic diagram merely indicate which terminals of the various parts are connected together *electrically*; these lines are not intended to show how wires actually run from point to point on the chassis, nor do they indicate the relative lengths of the connecting wires.

For example, referring to the first audio amplifier stage in Fig. 1, notice that parts 34 and 36 are both connected to the plate of this first audio tube. It might be possible to solder the condenser and resistor leads directly to the plate terminal of this tube socket. Actually, however, the resistor is soldered directly to this tube terminal and a long lead run from this terminal to condenser 34, which is soldered directly to the grid terminal of

the tube socket in the following stage. Experience with actual receivers will show you that other electrical connections indicated on a schematic circuit diagram can be made in many different ways on the actual chassis.

## TEST QUESTIONS

Be sure to number your Answer Sheet 32FR-2.

Place your Student Number on every Answer Sheet.

Never hold up one set of lesson answers until you have another ready to send in. Send each lesson in by itself before you start on the next lesson. In this way we will be able to work together much more closely, you'll get more out of your Course, and you will receive the best possible lesson service.

1. Is it necessary for a Radiotrician to analyze the complete circuit diagram of a receiver when repairing a defect? *No*
2. Into what two general trouble groups can all defective radio receivers be placed? *Dead, improperly operating*
3. What is the simplest and speediest test for isolating the defective stage in a dead receiver? *circuit disturbance*
4. Referring to the schematic circuit diagram in Fig. 1 what symbol identifies the circuit as that of an A.C. receiver. *Power line*
5. Give the numbers of the trimmer condensers in Fig. 1 which permit peak adjustments of the two I.F. transformers. *13 A, 13 B, 19 A, 19 B*
6. Near what easily identified part would you expect to find the power pack rectifier tube in an A.C. receiver? *Power pack*
7. Through what parts does the second grid (oscillator plate) of the 6A8G tube in Fig. 1 trace to the rectifier filament? *Screen*
8. What is the general rule for checking continuity in the circuits of all electrodes (in an A.C. receiver) which are supplied with a positive D.C. potential? *They trace to the positive end of the power pack*
9. Should there be a conductive path (continuity) between the plate and cathode of a diode detector? *yes*
10. Are high frequency trimmer condensers ordinarily mounted on the gang tuning condenser of an all-wave receiver? *No, a part*



## PERSONAL NEATNESS PAYS

Clothes may not make the man, but all too often they do determine the degree of his success in business. The radio man who keeps himself clean and neat gets the respect of his customers, and that one thing pays big dividends.

Good appearance is achieved by wearing appropriate conservative clothes for whatever kind of work you are doing, and keeping these clothes clean and well pressed at all times. Thus, inexpensive brown work shirts and trousers are fine for shop use and even for service calls, *provided they are clean and pressed*. But keep those shoes shined, keep your hair combed and cut, wash your hands immediately after handling a dusty or greasy chassis, and watch all those other little personal-neatness items which are being stressed so much in advertisements today.

To sum up, the cost of maintaining a neat personal appearance is negligible in comparison to the higher price a neat, well-dressed, business like radio man can demand and get for his services. Neatness inspires confidence, and confidence in turn makes customers accept your prices without question. Make a habit of neatness—it pays!

*J.E. Smith*

**BEHAVIOR OF RADIO WAVES  
AND RECEIVING ANTENNAS**

33FR-2



**NATIONAL RADIO INSTITUTE**

**WASHINGTON, D. C.**

**ESTABLISHED 1914**

# STUDY SCHEDULE No. 33

For each study step, read the assigned pages first at your usual speed, then reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind, then answer the Lesson Questions specified for that step. Study each other step in this same way.

1. Introduction; Radio Waves; Radiation Patterns of Transmitting Antennas; Vertical and Horizontal Polarization . . . . Pages 1-6  
Components of a Radio Wave; How Radio Waves Are Produced; Wavelength; Wavelength of an Antenna; Frequency-Wavelength Relationship; Field Intensity Surveys; How Radiation Patterns Are Secured; Action of a Receiving Antenna; Polarization of Radio Waves. Answer Lesson Questions 1, 2 and 3.
2. The Ground Wave; Television Receiving Antenna Problems; Sky Waves; The Kennelly-Heaviside Layer; Peculiarities of Sky Wave Reception . . . . . Pages 6-13  
Facts about ground waves; vertical and horizontal receiving antennas; multiple paths of television signals; Beam Characteristics of Ultra-High-Frequency Waves; the Kennelly-Heaviside Layer; Refraction of Radio Waves; Progressive Refraction; E, F<sub>1</sub> and F<sub>2</sub> Layers; Long-Distance Reception; Polarization of Sky Waves; Fading; Skip Distance; Selective Attenuation. Answer Lesson Questions 4 and 5.
3. Receiving Antenna Problems; Directivity of Receiving Antennas . . . . . Pages 13-18  
Three factors in selecting and installing a receiving antenna; four factors affecting directional characteristics of receiving antennas; Current and Voltage Distribution of Half-Wave Doublet Antennas; Current and Voltage Distribution of Grounded Vertical Antennas; Directional Characteristics; Radiation Patterns for Receiving Antennas; summary of receiving antenna installation rules. Answer Lesson Questions 6 and 7.
4. Coupling the Antenna to the Receiver; Transmission lines; Noise Reduction . . . . . Pages 18-24  
Signal-transferring problems; Voltage Feed For a Doublet Antenna; Current Feed; Antenna Resistance and Impedance; Surge Impedance of Transmission Lines; Typical Surge Impedance Values; noise problems; how noise-reducing antenna systems work; Noise-Reducing Loop Antenna. Answer Lesson Questions 8, 9 and 10.
5. All-Wave Antennas; Variable-Directivity Antennas . . . . Pages 24-29  
Types of All-Wave Antennas Commonly Used; Multiple Doublet Antennas; Single All-Wave Doublets; Use of Counterpoise for Noise Reduction; antennas with controllable directional characteristics.
6. Mail your Answers for this Lesson to N.R.I. for Grading.
7. Start Studying the Next Lesson.

# Behavior of Radio Waves and Receiving Antennas

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## Introduction

THE extremely high sensitivity of modern radio receivers has made many servicemen somewhat careless with the installation of receiving antennas. In addition to feeding strong signals from desired stations into a receiver, a good receiving antenna must also have a *high signal-to-noise ratio*, if station programs are to be heard with a minimum of interfering noises. The introduction and rapid acceptance of all-wave receivers brought added responsibilities to the serviceman, along with puzzling questions concerning the peculiarities of radio waves at various frequencies.

In order to answer these highly practical questions and solve the receiving antenna problems which occur in service work, you must first understand what radio waves are, how they behave, how a receiving antenna can be made to intercept a maximum of radiated energy, and how intercepted signals can be fed to a receiver with highest possible efficiency. All this information will be given you in this lesson.

## Radio Waves

### *Components of a Radio Wave.*

Whenever electrons flow through a conductor, *both an electric field and a magnetic field will always be produced.* (Of course, either field can exist by itself if no electron motion is involved; an electric field exists alone between two electrically charged bodies, while a magnetic field exists around a permanent magnet.) Furthermore, whenever electrons *change* their speed of travel through a conductor or oscillate back and forth, the result is a *moving* electric field and a *moving* magnetic field which together are known as an

*electromagnetic field.* If the electrons are oscillating back and forth at a high enough rate (at a high frequency), the electromagnetic field will travel off into space at the speed of light (186,000 miles per second), giving what is known as an *electromagnetic wave* or a *radio wave.* A radio wave, therefore, has two components: 1. *The electric field component;* 2. *The magnetic field component.*

*How Radio Waves Are Produced.* A study of the fields in the vicinity of a typical transmitting antenna such as that shown in Fig. 1A will give you a good idea of how radio waves are radiated into space. This is a *doublet antenna*, for it is broken at its center and the two sections are fed with an oscillating electron flow produced by a radio transmitter; it is also known as a *dipole antenna.*\* Suppose that the transmitter polarity at one instant of time is as indicated in Fig. 1A; electron flow will then be in the directions indicated by arrows *i*, making the upper end of the doublet negative and the lower end positive. We can consider the circuit between the two halves of the antenna to be completed through space, for

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\*Although you will find the terms "doublet antenna" and "di-pole antenna" used interchangeably in many cases, the following definitions are accepted by most radio engineers: A *doublet antenna* is fed at its center, and its two sections are essentially equal in length and are in the same straight line; a doublet antenna may or may not be at resonance for the frequency of the signal fed to it. A *di-pole antenna* is to the radio physicist a theoretically perfect antenna having uniform current throughout its length and with ends having opposite polarity; to the practical radio man a di-pole antenna has *essentially* uniform current throughout its length, is *not* at resonance with the signal frequency fed to it, has ends with opposite polarity, and may be fed anywhere along its length.

These halves form a condenser having the surrounding air as a dielectric. A *displacement current* (the equivalent of an electron movement) flows from the negative half of the antenna through space to the positive half.

To simplify our study, we can neglect the transmitter connection and consider this antenna as a single length of wire in space, having the polarity shown in Fig. 1B when electron flow  $i$  is upward as indicated; this antenna wire can be thought of as a source which is feeding power into space. The polarity will reverse from instant to instant of time in step with reversals in electron flow through the wire.

When electron flow through a doublet antenna is in the direction shown in Fig. 1B, a *magnetic* field will encircle the wire and will act in the direction shown in Fig. 1C; this magnetic field is usually designated by the letter  $H$ . At the same time an electric field having the direction shown in Fig. 1D will be set up in space by the oppositely charged halves of the antenna; this electric field is usually designated by the letter  $E$ . At any point in space, the electric and magnetic fields ( $E$  and  $H$ ) produced by an energized antenna *will always be at right angles to each other*.

A continuously oscillating (alternating) electron flow in a doublet antenna produces in the vicinity of the antenna an electromagnetic field; the relative intensities of the electric and magnetic components of this field at various distances away from the antenna at one instant of time are shown in Fig. 1E. At point 1, for example, there is an electric field component  $E_1$  which is acting upward parallel to the antenna (in the plane of this page), and a magnetic field component  $H_1$  which is at right angles to the electric field (towards you, at right angles to this page). The arrow lines  $E_1$  and  $H_1$  are intended to show the relative strengths of the electric and magnetic

fields at point 1, and the directions in which these fields act; these arrow lines *do not* represent the paths of electric and magnetic fields, for these paths are shown in Figs. 1C and 1D. Each other point in Fig. 1E, such as 2, 3, 4, 5, 6, 7 and 8, will likewise have its electromagnetic field with an electric component  $E$  and a magnetic component  $H$ ; note that between points 4 and 8 there is a reversal in the directions of the components, with  $E$  now acting downward in the plane of the paper and  $H$  acting away from you perpendicular to the paper. The electromagnetic field is traveling away from the transmitting antenna in all directions, at the speed of light; both the electric and the magnetic components of this field are always acting *at right angles to this direction of travel*.

A portion of the electromagnetic field produced by a transmitting antenna acts as if it were permanently associated with the antenna and were staying in its immediate vicinity; this portion is known as the *induction field*. The other part of the electromagnetic field, known as the *radiation field*, the *electromagnetic wave* or the *radio wave*, breaks away from the antenna and travels outward through space. Insofar as energy is concerned, the  $E$  and  $H$  components of the radiation field or radio wave are always equal to each other at a distance from a transmitting antenna (in the region of the radiation field); close to the antenna, in the region of the induction field, the  $E$  component may be considerably greater than the  $H$  component. We will consider only the radiation field (radio wave) for the remainder of this lesson unless otherwise indicated, since the induction field ordinarily exists only for a few hundred feet away from the antenna.

*Wavelength.* During the time required for one complete cycle of the alternating current flowing through the

antenna in Fig. 1E, the electromagnetic wave will travel from point *O* to point *S*, and both the electric field *E* and the magnetic field *H* at any point will also go through one complete sine wave cycle. The distance between point *O* and *S* is known as *one wavelength*.

When engineers study radio waves in space, they ordinarily consider only one component (usually the electric field component *E*), and consequently they think of a radio wave as having the simple sine wave form shown in Fig. 1F. The distance between the extremities of one complete cycle of this wave is therefore one wavelength. Fractions of a cycle are called fractions of a wavelength; half a cycle is thus

*Wavelength of an Antenna.* The characteristics of a receiving antenna depend to a great extent upon its length with respect to the wavelength of the signal being picked up. For this reason, it is common practice to describe an antenna as being "so many" wavelengths long for a particular signal frequency. When a radio signal produces a half-sine-wave distribution of current along a receiving antenna, we know that the antenna is half as long as one wavelength of that particular signal, and we say that the antenna is *one-half wavelength long*.

We could easily determine the wavelength of an antenna at a particular frequency by drawing the radio wave

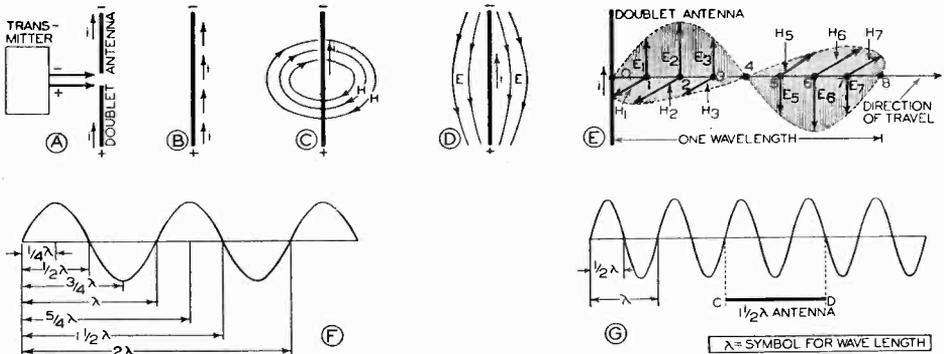


FIG. 1. These diagrams show the first steps in the formation of a radio wave by a simple doublet antenna which is fed by a transmitter. Electron flow *i* through the antenna wire in the direction shown produces a magnetic field *H* and an electric field *E*, both of which travel away from the antenna at the speed of light.

half a wavelength or simply a half-wave, and  $1\frac{1}{2}$  cycles are  $1\frac{1}{2}$  wavelengths or  $3/2$  waves.

The next important fact for you to recognize is that a *wavelength is a different value* (in feet or in meters) *at each frequency*. Doubling the frequency means that we get twice as many alternations or cycles of the radio wave into a given distance, and each cycle or wave will therefore be only one-half as long; Fig. 1G illustrates this clearly. The higher the frequency of a radio wave, the shorter will be the length of one complete wave or cycle.

and the antenna on the same line at the same scale, and simply counting the number of cycles covered by the antenna. Figure 1G illustrates this; line *CD*, which represents an antenna drawn to scale, covers  $1\frac{1}{2}$  cycles and consequently we have a  $3/2$ -wave antenna at this particular signal frequency. It is generally more convenient to figure out this information than to secure it graphically, however; simply divide the length of the antenna by the wavelength of the signal (using the same units of length), and the result will be the wavelength of the antenna. Example: Antenna length is 100 feet;

signal wavelength is 200 feet;  $100 \div 200 = \frac{1}{2}$ , and consequently we have a half-wave antenna in this example.

*Frequency - Wavelength Relationship.* Oftentimes the frequency of a radio wave rather than the wavelength is known. The preceding section indicates that there is a definite relationship between frequency and wavelength; this relationship is based upon the fact that an electromagnetic wave *in space* always travels at the same speed, approximately 300,000,000 meters per second or 186,000 miles per second (this is also the speed at which light travels through space). The following formula gives the relationship between the speed, frequency and wavelength of a radio wave in space:

**Wavelength in meters multiplied by frequency in cycles per second always equals 300,000,000.\***

Figure 1E shows only one of the many possible paths for radio waves away from the antenna. There will be many other paths in all directions. When a radio wave travels along the ground directly to the receiving antenna, it is known as a *ground wave*. When a radio wave travels up into the sky and is then bent back to the receiving antenna, it is known as a *sky*

*wave*. Both the ground wave and the sky wave must be considered when studying problems involving the transmission and reception of radio signals.

## Radiation Patterns of Transmitting Antennas

It is a well-known fact that radio stations which are intended to serve a local area, such as broadcast band stations in this country, use antennas which radiate signals having maximum intensity *along the ground*. Stations which must send messages or programs to far-distant receiving points use antennas which direct radio waves skyward, for it is now known that there is a layer in the upper atmosphere which will bend the sky wave back to earth again at a point considerably distant from the station. Commercial short-wave stations in transoceanic use, amateur radio stations, and short-wave stations which transmit entertainment programs to far-distant countries all use antennas which are carefully designed to produce the strongest possible sky wave.

*Field Intensity Surveys.* Careful measurements must be made of the amount of energy radiated in each direction (north, south, east and west) from a transmitting station in order to determine whether an antenna is giving adequate coverage of the desired local service area or is radiating satisfactory signals to desired distant points, as the case may be. It is customary to measure the energy in the electrical component *E* of the radio wave; the results are expressed in terms of microvolts per meter, millivolts per meter or volts per meter, with these units expressing the voltage existing between two points which are exactly one meter apart in space at the location where the signal intensity is measured. These field intensity units also correspond to the voltage which would be measured between the ends

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\*When dealing with antennas it is usually more convenient to speak in terms of wavelength than frequency. If either the frequency or the wavelength of a radio wave is known, you can easily determine the missing value. It is common practice to let *f* represent frequency in *cycles per second* and  $\lambda$  (Greek letter lambda) represent wavelength *in meters*; the above formula then becomes  $\lambda \times f = 300,000,000$ .

To find frequency when wavelength is known, use this variation:  $f = 300,000,000 \div \lambda$ . Example: If wavelength is 5 meters, what is the frequency? Answer:  $f = 300,000,000 \div 5 = 60,000,000$  cycles or 60 megacycles.

To find wavelength when frequency is known, use this variation:  $\lambda = 300,000,000 \div f$ . Example: If frequency is 1,500 kc., what is the wavelength? Answer:  $\lambda = 300,000,000 \div 1,500,000$ , or 200 meters.

To get wavelength in feet, multiply the wavelength in meters by 3.28 (roughly  $3\frac{1}{4}$ ).

of a true di-pole antenna exactly one meter long, erected at the receiving location being checked. Measurements are usually made with a loop receiving antenna, and the readings are converted by means of higher mathematics to equivalent values for a true di-pole antenna. The final values are plotted on a graph to give what is known as the *radiation pattern* of a transmitting antenna.

*How Radiation Patterns Are Secured.* If an engineer wants to know how well the ground wave is getting out in various directions from a broadcast station antenna, he might take suitable field intensity measuring equipment first to a position  $P_1$ , which is two miles due east of the antenna, place his pick-up antenna at the angle and direction which give maximum signal, and make the first measurement; he could then travel in a two-mile-radius circle around the antenna and make measurements at positions  $P_2, P_3, P_4$ , etc., in Fig. 2A, which are NE, N, NW, W, etc., from the antenna along this circle (any other convenient radius could just as well be used).

An alternative procedure involves selecting convenient positions each two miles away, regardless of their direction from the transmitting antenna, and measuring the angle  $\theta$  (see Fig. 2A) between a line to the antenna and a line running due east from the antenna. After measurements have been made for a sufficient number of points, the engineer draws a diagram similar to that in Fig. 2A, with radial lines running from the antenna location to each point at which a measurement was made. Along line  $A-P_1$ , starting from  $A$ , he plots the signal intensity measured at point  $P_1$ , using any convenient scale (he might let one inch represent 10 millivolts per meter). The length of line  $AE_1$  then represents the field intensity measured at point  $P_1$ , two miles due east of the antenna. The

same procedure is repeated for each other radial line, and a smooth curve is drawn through the various points. This curve is the *horizontal radiation pattern*; it indicates the approximate signal intensity in millivolts per meter which can be expected at a point two miles from the transmitting antenna at any angle with the due east line, eliminating the need for making measurements at all angles. The particular pattern shown in Fig. 2A indicates that practically no energy is being radiated along the ground in the NW, W and SW directions, and that almost uniform energy is being radiated in the N, NE, E, SE, and S directions.

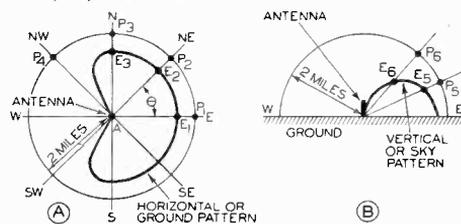


FIG. 2. Horizontal and vertical radiation patterns for a directional transmitting antenna.

Similar information concerning sky wave radiation can be obtained by making measurements from an airplane or blimp. An example of a pattern secured for a flight due east and west over an antenna is shown in Fig. 2B. This radiation pattern indicates that signal intensity is a maximum along the ground, but that considerable energy is being radiated into the sky in the form of a sky wave.

### Vertical and Horizontal Polarization

*Action of a Receiving Antenna.* A short straight antenna will pick up a maximum signal when it is *parallel* to the electric lines of force associated with a radio wave, as indicated in Fig. 3A. Free electrons in the antenna wire move back and forth along the wire in step with changes in the direction and intensity of the electric lines of force, with the result that an alternating cur-

rent is sent down to the receiver. When the antenna is at right angles to the electric lines of force, the electrons will oscillate from side to side across the wire, and no current will flow to the receiver.

You can also visualize the action of a receiving antenna by considering the magnetic lines of force, which are at right angles to the antenna. These magnetic lines, moving through space at the speed of light, cut across the receiving antenna and induce in it a radio frequency voltage which is proportional to the number of lines which cut the antenna per unit of time. This voltage in turn forces a current down the antenna lead-in wire to the receiver.

**Polarization of Radio Waves.** When the electric lines of force associated with a radio wave are perpendicular to

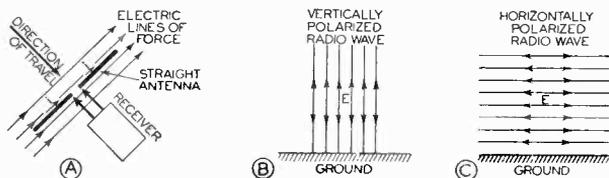


FIG. 3. The electric field  $E$  associated with a radio wave may have any angle whatsoever with respect to ground when it arrives at a receiving antenna, and this angle will have an important effect upon signal pick-up by a given antenna. When  $E$  is perpendicular to the ground, we say that the radio wave is vertically polarized; when  $E$  is parallel to the ground, the radio wave is horizontally polarized.

the ground, as indicated in Fig. 3B, they are said to be *vertically polarized*. When the electric lines of force associated with a radio wave are parallel to the ground, as indicated in Fig. 3C, they are said to be *horizontally polarized*. The magnetic lines of force associated with a radio wave are also polarized, but we can neglect them when considering the effectiveness of various receiving antennas.

The electric lines of force associated with a sky wave seldom have perfect vertical or horizontal polarization; they are usually at an angle with the ground, somewhat as shown in Fig. 4. Here the radio wave is traveling toward the ground after reflection from

the sky. The intensity and direction of the electric field at point  $P$  (above the ground on this path of travel) can be represented by the arrow line  $E$ . This arrow line can be considered to have two components; a vertical component  $E_V$  and a horizontal component  $E_H$ , as shown in Fig. 4. To secure a maximum signal at location  $P$ , you would have to place a straight doublet antenna in the direction indicated by  $E$ . A vertical antenna would only pick up a signal corresponding to the length of  $E_V$ , and a horizontal antenna would in this case give but slightly more signal pick-up corresponding to the length of  $E_H$ .

### The Ground Wave

The transmitting antennas used by broadcast stations for local coverage are generally of the vertical type, for

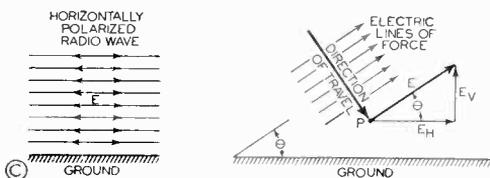


FIG. 4. An electric field arriving at the angle  $\theta$  to ground may be considered equal to a vertically polarized component  $E_V$ , acting with a horizontally polarized component  $E_H$ .

these can be designed to radiate considerable energy along the ground and at the same time keep the sky wave at a low enough intensity to avoid interference between ground and sky waves within the service area of the station. The electric lines of force produced by an energized vertical antenna are essentially parallel to the antenna when near the surface of the earth, and hence are very nearly vertically polarized. We might conclude that for maximum signal pick-up at the receiving location, the receiving antenna should be of the vertical type, but we know that the majority of receiving antennas are actually horizontal.

There are several reasons why a

horizontal antenna is more satisfactory for receiving purposes. First of all, high vertical antennas are expensive and difficult to erect, making them unsuited for the average home. Secondly, electric lines of force bend forward as they move away from the antenna, as indicated in Fig. 5B, due to the presence of the ground; one engineer describes this phenomenon by saying "the radio waves are pushed forward with their feet dragging on the ground." Another reason is this: Radio waves which travel over hilly or mountainous country or near tall steel buildings are reflected enough to change their angle of polarization and their direction of travel. At a reasonably distant receiving location such as that shown in Fig. 5A, there is invariably a large enough horizontal component  $E_H$  (Fig. 5B) to excite a horizontal antenna.

There is still another good reason for making receiving antennas horizontal. Electrical devices which cause man-made interference produce far more vertically polarized electric lines of force than horizontally polarized lines, and since these interference-producing devices are close to the receiver location, they would induce strong signals in a vertical receiving antenna. A horizontal antenna picks up a minimum of noise signals and still intercepts enough of the signal from a local transmitting antenna to give a satisfactorily high signal-to-noise ratio. In most cases, a horizontal antenna gives better reception of sky waves than does a vertical antenna.

Ground waves radiated by either a broadcast or short-wave station will follow the earth until they are totally absorbed or are too weak to affect receiving antennas. Only by increasing the amount of power radiated by the transmitting antenna can stronger ground waves and more distant reception of ground waves be secured. All reception at points outside the range of

ground waves will be due to sky waves reflected back to earth by a layer in the upper atmosphere. These sky waves are absorbed very little by the atmosphere, and consequently will often be reflected back to points thousands of miles away from the station with surprisingly high signal strength.

### Television Receiving Antenna Problems

At the carrier frequencies used in modern television systems, automobile ignition systems and electrical devices having sparking contacts can cause considerable interference. It is highly desirable to use a horizontal receiving

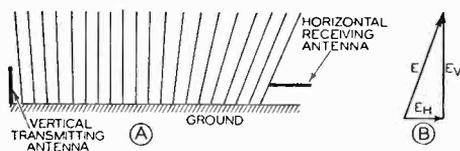


FIG. 5. Vertically polarized electric lines of force produced by a vertical transmitting antenna gradually change their angle with the ground as they travel away from the antenna.

antenna in order to keep out these vertically polarized interference signals. Television transmitting antennas are usually designed to radiate horizontally polarized radio waves, for these will be picked up with maximum efficiency by a horizontal receiving antenna and will therefore give a maximum signal-to-noise ratio.

Television signals may reach a receiving antenna in a city location over two or more different paths. This multiple transmission effect is due to reflection of waves from buildings, bridges and other steel structures, and may give two or more images superimposed on each other due to slight differences in the times of arrival of the signals over the various paths. This secondary or ghost image effect can be minimized by erecting the horizontal doublet receiving antenna as high as possible and in a direction which gives maximum reception of either the direct wave or

the reflected wave, but not both. When there are two television stations in a locality, a doublet antenna must necessarily be in a position which gives a satisfactory compromise between signals arriving from the two directions. An alternative procedure involves the use of two doublets at right angles to each other and crossed at their centers, with the system oriented to give best possible pick-up from both stations.

**Beam Characteristics of Ultra-High-Frequency Waves.** Ultra - high - frequency (u.h.f.) radio waves such as are used in modern television systems are above 40 megacycles. These u.h.f. waves behave like beams of light, in that they travel in essentially straight

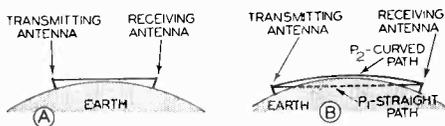


FIG. 6. Radio waves at ultra-high frequencies travel in essentially straight lines, limiting the reliable range of reception of television programs to line-of-sight distances.

lines from the transmitting antenna. This means that if the transmitting antenna is not visible from the top of the receiving antenna on a clear day, signals will in all probability not be received. With antennas at ordinary heights on level ground, the curvature of the earth definitely restricts the coverage of a given station. This is illustrated in exaggerated form in Fig. 6A; increasing the distance between the two antennas would make the radio waves pass over the top of the receiving antenna without affecting it, for these u.h.f. waves cannot bend around the earth. Television broadcasts from the tower of the highest building in the world, the Empire State Building in New York City, can normally be picked up only within a radius of about sixty miles of the transmitting station. Occasionally, reception at greater distances is possible because of a peculiar distribution of moisture in the air

which bends ultra-high-frequency sky waves back to the earth at more distant locations. In Fig. 6B, for example, reception of signals over the direct path marked  $P_1$  would be blocked by the earth, but under certain conditions the sky waves might be bent back to the lower atmosphere, taking path  $P_2$  to the receiving antenna. In general, however, sky wave reception at ultra-high frequencies is unreliable.

### Sky Waves; The Kennelly-Heaviside Layer

Before Marconi in 1901 succeeded in transmitting a radio message from England to the North American Continent, scientists generally believed that all radio reception was by means of ground waves, and that long-distance transmission was impossible because of the absorption of energy by the ground. The success of Marconi proved this theory false, and many new theories were advanced to explain the amazing long-distance transmission of signals. Kennelly in America and Heaviside in England simultaneously presented the explanation that there are electron layers in the sky which act like any conductive layer in reflecting and bending electromagnetic waves. These theories have been proved correct, and today we speak of this wave-bending layer in the atmosphere as the Kennelly-Heaviside layer. The reason for the existence of this layer is usually explained in the following manner: The surface of the earth is surrounded by air, which thins out (becomes rarefied) away from the earth. Under the action of ultra-violet rays of light from the sun, this rarefied air becomes ionized, with the result that a layer made up of free electrons, positive ions and negative ions is formed high above the earth.

A radio wave passing into the Kennelly-Heaviside layer will impart some of its energy to the free electrons in the

layer, setting them into vibration. The velocity of vibration will be the least at the highest frequencies, for even an electron has a certain amount of mass which prevents large amplitudes of vibration at high frequencies. The mass of the electron makes it act as a reactance, so its velocity is  $90^\circ$  out of phase with the velocity of the radio wave. The oscillating electrons in the layer produce a new radio wave which acts with the original wave. This combining of velocities which are  $90^\circ$  out of phase results in an apparent speeding up of the radio wave when it hits the electron layer.

Radio waves at low radio frequencies cause the electrons in the layer to vibrate with great amplitude. These electrons collide with air particles, giving up their energy and liberating ions and more electrons. In this way a portion of the energy in the original radio wave is lost in the Kennelly-Heaviside layer; the loss may amount to as much as 30% and occurs chiefly at the lower levels, where there are more ions of air to absorb energy.

**Refraction of Radio Waves.** Let us assume that the Kennelly-Heaviside layer has a uniform distribution of electrons and that the electric lines of force associated with a radio wave are moving toward it at the angle shown in Fig. 7A. As the radio wave enters the layer, portion *y* encounters electrons first and speeds up. The result is a bending of the electric lines of force and of the radio wave path itself; this bending of the path of travel is always down toward the earth. Scientists refer to this phenomenon as *refraction* rather than as bending.

**Progressive Refraction.** The electron density (number of electrons per unit volume) in the Kennelly-Heaviside layer is greater in the upper regions than in the lower regions. We would naturally expect this, since the upper portion of the layer is closer to the sun

and is consequently subject to greater ionization. A radio wave entering the Kennelly-Heaviside layer at an angle encounters regions of increasingly greater electron density, and consequently undergoes increasingly greater refraction or bending; this phenomenon is known as *progressive refraction*. It is quite possible for the bending to progress to the point where the radio wave travels horizontally with respect to the earth, then actually bends back toward the earth in the manner shown in Fig. 7B.

**E, F<sub>1</sub> and F<sub>2</sub> Layers.** Actually, the Kennelly-Heaviside layer is not as simple as is shown in Fig. 7B. The ac-

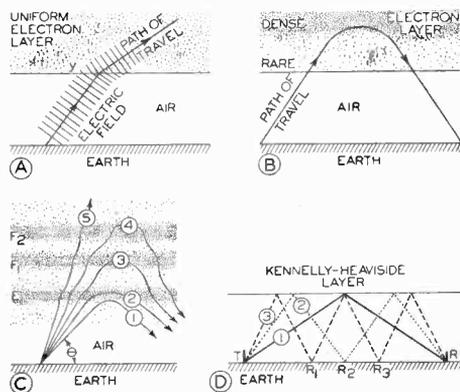


FIG. 7. These diagrams show how sky waves can be refracted (bent) back toward the earth by the Kennelly-Heaviside layer in the upper atmosphere.

tion of the sun upon the upper atmosphere is such as to produce three separate layers, one above the other. If we were able to measure electron densities, starting from the earth and going vertically upward, we would find that for the first thirty to forty-five miles there would be a negligible number of electrons. Continuing upward, the electron density would increase to the first maximum value at a height of about sixty miles, then gradually decrease. Higher up still, electron density would increase again and reach another maximum at about 125 miles. Farther up we would pass through another region

of gradually decreasing and then increasing electron density until we reached the third and final maximum-density level at about two hundred miles up. Above this level the electron density decreases to a negligible value, for at this height in the stratosphere there is almost a total vacuum, and little ionization can occur. These three maximum-density layers are known as the  $E$ ,  $F_1$  and  $F_2$  layers respectively.

The  $E$  layer is nearest the earth and is fairly constant in height; the  $F_2$  layer is highest and may vary in height from 150 miles to 225 miles. This highest layer is most affected by the sun and hence it varies greatly in height and electron density from day to night, from summer to winter, and with con-

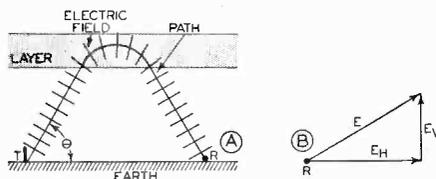


FIG. 8. Effect of the Kennelly-Heaviside layer upon the polarization of a sky wave.

ditions on the sun such as sun spots (solar eruptions). The  $F_2$  layer also varies with latitude, and may be higher or lower around the equator than it is in the north temperate zone or at either pole. Oftentimes the  $F_1$  (middle layer) and  $F_2$  layers will merge to form what is called an  $F$  layer. Less often, the  $E$  layer will separate into two layers, with one being known as the normal  $E$  layer and the other as the *sporadic E* layer.

A sky wave may be refracted back to earth by the lower portion of the  $E$  layer, as shown by path 1 in Fig. 7C; on the other hand, it may be refracted by the upper part of the  $E$  layer as in path 2, may pass completely through the  $E$  layer and be refracted by the  $F_1$  and  $F_2$  layers as shown by paths 3 and 4, or may pass completely through all three layers as in path 5 and be lost in interplanetary space. As a rule, the longer wavelengths (broadcast band

and longer wavelengths) are refracted by the  $E$  and  $F_1$  layers, and shorter wavelengths are refracted by the  $F_1$  and  $F_2$  layers. The extremely short waves (ultra-high frequencies) pass through all three layers, and hence are not ordinarily received.

The angle  $\theta$  in Fig. 7C (the angle between the sky wave and the earth) has a vital effect upon the path taken by the sky wave; the greater this angle, the more chance there is for the wave to get through the  $E$  layer and be refracted by the higher layers. A sky wave radiated at a high angle will be bent back to earth closer to the transmitting station than a wave radiated at a low angle and refracted by a lower layer. This statement is illustrated in Fig. 7D; radiation of the sky wave at the largest angle, along path 3, results in reception at point  $R_1$ , which is near to station T. Radiation at a low angle, along path 1, gives reception at point R, which is a considerable distance from the station. Of course, the ground wave will be received in the immediate vicinity of the station in all cases.

*Long-Distance Reception.* The Kennelly-Heaviside layer makes possible long-distance reception of signals from low-power radio stations, for this layer bends radio waves back to the earth at points far beyond the range of reliable ground wave reception. These sky waves will be reflected skyward again by the earth, and will oftentimes be refracted by the Kennelly-Heaviside layer and reflected by the earth one or more additional times. Each path from earth to sky and back again is called a *hop*. Signals from station T in Fig. 7D may reach receiving point R in 1, 2 or 3 hops, as illustrated by paths 1, 2 and 3 respectively in Fig. 7D. As a rule, signals on the longer wavelengths are sent with fewer hops than those on the shorter wavelengths. The best wavelength to use for communication between two given points depends con-

siderably upon the conditions in the electron layers. The number of hops is controlled by the angle at which the sky wave leaves the station; the ideal number of hops at any time depends upon conditions in the upper layers. Where reliable transmission is essential, the same message is oftentimes broadcast simultaneously on several different wavelengths, each reflecting from a different layer and giving a different number of hops.

### Peculiarities of Sky Wave Reception

*Polarization of Sky Waves.* If we analyze the electric field produced at receiving point *R* by the sky wave from station *T* in Fig. 8A, we find that it has

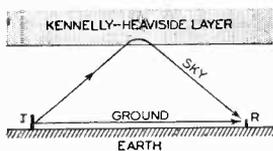


FIG. 9. This diagram shows why fading sometimes occurs within 50 miles of a broadcast station.

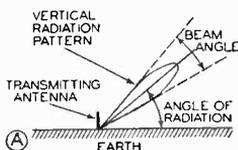
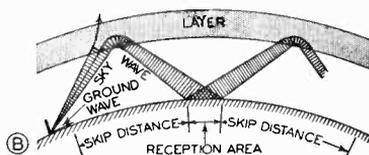


FIG. 10. Long-distance communication via short-wave radio depends entirely upon sky waves. The transmitter engineer chooses operating conditions which will make the sky wave come back to earth at the desired reception area.



a vertical component  $E_v$  and a horizontal component  $E_H$ , as shown in Fig. 8B. When  $\theta$  (the angle of radiation) is more than  $45^\circ$ , the  $E_H$  component will be the stronger, and a horizontal receiving antenna will give better signal pick-up than a vertical antenna. When a radio wave travels a long distance with a single refraction from the Kennelly-Heaviside layer,  $\theta$  may be less than  $45^\circ$ ; the vertical component of the electric field will then predominate, and a vertical antenna will give better signal pick-up provided there is no man-made interference near the receiver. When interference is present, a horizontal antenna is invariably more satisfactory, for the horizontal component  $E_H$  of a sky wave is usually strong enough to give a high signal-to-noise ratio.

For maximum signal pick-up with a

doublet antenna at *R* in Fig. 8A, the antenna wire should of course coincide with the electric lines of force nearest point *R* in this diagram; amateur and commercial antenna systems are sometimes tilted to take advantage of this fact, but for home entertainment purposes it is impractical and unnecessary to provide means for tilting the antenna to a new angle each time a new distant station is tuned in.

*Fading.* For signals below 3 megacycles in frequency, the sky wave is refracted and bent back to the ground almost immediately after its entrance into the lowest electron layer (the *E* layer). If there is an appreciable amount of high-angle radiation (sky waves) from the station, receiving an-

tennas about fifty miles away from the station will pick up the ground wave directly and the sky wave after refraction, as indicated in Fig. 9. The sky wave, traveling a considerably longer distance, may arrive *in phase with* the ground wave, in which case there is reinforcement and maximum signal pick-up; more likely, the two waves will be partially or completely *out of phase*, giving partial or complete cancellation of the signal. The phase relationship may vary from instant to instant, causing fading, for the Kennelly-Heaviside layer sometimes shifts up and down slightly. Briefly, then, sky waves cause fading at receiving locations near the limits of reliable ground wave reception from a given station because the phase relationship between the sky and ground waves changes continually due to refraction of the sky wave by the

rapidly - shifting Kennelly - Heaviside layer, giving partial or complete cancellation of the ground wave.

If the shifts which cause fading occur in less than one-tenth of a second, the a.v.c. system in the average receiver will not be able to compensate for them and the variations in signal strength will be annoyingly noticeable at the receiver loudspeaker. Even when fading takes place at a slow rate, the automatic changes in receiver gain may result in noticeable alternate periods of clear and noisy reception.

Fading is generally less severe during the daytime. The Kennelly-Heaviside layer is then so dense that radio waves at broadcast band frequencies are almost completely absorbed as they enter the lower regions of the layer. Fading at distances of from 40 to 60 miles away from a broadcast station can be kept at a minimum by designing the transmitting antenna to radiate most of its energy along the ground; vertical antennas which are about  $\frac{5}{8}$  wavelength high are best suited for this purpose.

*Skip Distance.* At frequencies of from 3 to 30 megacycles, the sky wave is used almost exclusively for communication purposes. The transmitting antenna is in this case designed to radiate most of its signal into the sky, and may have a vertical radiation pattern like that in Fig. 10A. Notice that maximum signal intensity occurs at a definite *angle of radiation* with the ground, and that radiation is concentrated over a definite angular distance which is known as the *beam angle*. There may be enough radiation along the ground in a case like this to provide adequate reception for distances of perhaps ten miles away from the station, but most of the energy is radiated into the sky, where it is either bent back to the earth or lost through absorption. This type of radiation pattern for a transmitting antenna provides re-

ception over a more or less restricted area at a great distance away from the station; the distance and the area both depend upon the width of the beam, the angle of radiation, the frequency of transmission and upon conditions in the Kennelly-Heaviside layer. The distance between the maximum limit of ground wave reception and the closest point at which the sky wave returns to earth is commonly known as the *skip distance* (see Fig. 10B). Different frequencies must be used for each distance covered, and frequency must be changed at night to compensate for the shift in the height of the electron layers and the resulting change in the skip distance, as each frequency is acted upon differently by the layers. As a general rule, high transmitting frequencies are best for daylight and lower frequencies are better for night transmission. Furthermore, radio transmission is better in winter than in summer, for in winter there is less absorption of signals by the electron layers.

*Selective Attenuation.* We must not overlook the fact that the transmission of voice and music is accomplished by sending a number of side-band frequencies along with an r.f. carrier. Each side frequency is acted upon by the Kennelly-Heaviside layer in a slightly different manner, and consequently the various frequencies may take various paths back to the earth or may be attenuated differing amounts. The result is that side frequencies in the received signal may have a different strength relationship to the carrier frequency than they had at the transmitting station, causing amplitude distortion in the demodulated audio signal; this cannot be corrected in the receiver. Since conditions in the Kennelly-Heaviside layer fluctuate quite rapidly, reception may be distorted severely at one moment and may become clear in the next instant, particularly when the receiver is located near

the boundary between a skip area and a reception area.

### Receiving Antenna Problems

Even though a transmitting station is designed to "lay down" a strong signal in a definite receiving area, part or all of this energy will be wasted unless the receiving antenna is properly designed. Three important factors are involved in the selection and installation of a receiving antenna:

1. That portion of the antenna which intercepts the electric and magnetic components of the radio wave must have *maximum possible signal pick-up*. This involves consideration of the directional properties of the antenna, the polarization of radio waves from favorite stations, and the frequency range over which signals are to be picked up.

2. The signal energy picked up by the antenna must be transferred to the receiver with a minimum of loss over the desired range of carrier frequencies. This involves matching the transmission line impedance to the impedance of the antenna and to the impedance of the receiver input, to secure *maximum transfer of energy from the antenna to the receiver*.

3. There should be a *maximum signal-to-noise ratio* at the receiver input. This is usually secured by designing the antenna system so it will accept a minimum of noise signals.

### Directivity of Receiving Antennas

The directional characteristics of a receiving antenna system depend upon a number of important factors: 1. The current and voltage distribution along the antenna; 2. The position of the antenna with respect to ground; 3. The position of the pick-up section with respect to other elements of the antenna, if the system is made up of more than one wire; 4. The electrical characteristics of the ground near the antenna.\*

\*Items 3 and 4 are not covered in this lesson since they are ordinarily neglected when designing and installing antennas for home radio receivers. These factors are extremely important in connection with amateur and commercial receiving antennas and with television receiving antennas, and are taken up in advanced lessons dealing with these subjects.

*Current and Voltage Distribution of Half-Wave Doublet Antennas.* A single straight half-wave doublet antenna like that in Fig. 11A, located in free space (away from the influence of the earth and other objects) and fed with energy either by a direct connection to a transmitter or by radiation of power through space from a distant transmitter, acts like a number of small elements of inductance and capacity connected together in the manner shown in Fig. 11B. If a.c. power is fed into the center in the case of a half-wave doublet transmitting antenna, the cur-

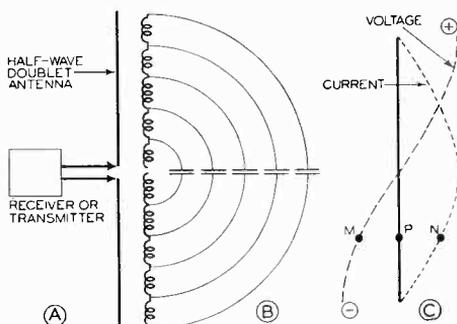


FIG. 11. Characteristics of a simple half-wave transmitting or receiving antenna in free space.

rent encounters small elements of inductance which serve to choke it and reduce its value, and for each inductance there is an elemental capacitive path which draws off a portion of the antenna current and brings it to a corresponding point on the other half-section of the antenna. The result is that the r.f. signal current is the greatest *at the center* of a half-wave receiving antenna, and gradually drops to zero as it approaches the extreme ends. Across each elemental inductance there will be a voltage drop due to current flow through it, and those elements nearest the center will naturally have the highest voltage drops. The result is that the r.f. voltage *with respect to the center of the antenna* increases rapidly as we move outward from the center, then increases more gradually to a maximum value at the ends; furthermore,

the ends will be opposite in polarity at any instant and will reverse in polarity *once for each cycle*. The curves in Fig. 11C show this current and voltage distribution; they tell, for example, that the effective (r.m.s.) value of the r.f. current at point  $P$  is proportional to the distance  $NP$ , and the r.m.s. value of voltage at point  $P$  is proportional to the distance  $MP$ . Notice that both curves are sinusoidal (having a sine wave shape), with maximum current at the center and maximum voltage at the ends. (Since the center of the antenna has zero voltage, it may be considered at ground potential and all voltages measured with respect to the center rather than to ground.)

This half-sine-wave distribution of current and voltage will be obtained for any doublet transmitting or receive-

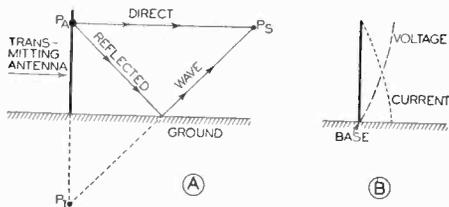


FIG. 12. Characteristics of a grounded vertical quarter-wave antenna. A perfectly conductive ground is assumed.

ing antenna in free space whenever the length of the antenna wire is equal to one-half the wavelength of the signal. This condition can occur at only one signal frequency; at that frequency the antenna radiates or picks up a maximum signal, and is said to be *at resonance*.

The essential requirement of a resonant receiving antenna is that it act as a pure resistive source. This requirement can be met by making the antenna one-half wavelength long or some multiple of a half wavelength (2, 3, 4, 5, etc., times as long). An antenna which is not a multiple of one-half wavelength for a particular radio wave will have either an inductive reactance or a capacitive reactance which raises

its impedance and reduces the amount of current which can be delivered to the receiver. It is possible to introduce either capacity or inductance in the antenna circuit in order to cancel out the existing reactance; this *tunes* the antenna to resonance, making it purely resistive and giving at its ends the zero-current, maximum-voltage condition which results in maximum signal pickup.

*Current and Voltage Distribution of Grounded Vertical Antennas.* If a vertical antenna is grounded at one end, it will be at resonance when only  $\frac{1}{4}$  wavelength long, for the earth serves to duplicate the effect of the other  $\frac{1}{4}$  wavelength. A quarter-wave grounded vertical antenna is shown in Fig. 12A; a receiving antenna at point  $P_S$  in space will receive a radio wave directly from point  $P_A$  on the transmitting antenna, and will receive another radio wave directly from point  $P_A$  by reflection from the ground. This reflected wave appears to come from a point  $P_I$  which is located the same distance below the surface of the earth as  $P_A$  is above the earth. In the language of radio engineers, we have an *image antenna* in the ground; although it is a purely imaginary structure, the observed effects are exactly the same as if it were present.

The current and voltage distribution curves for a grounded quarter-wave vertical antenna are shown in Fig. 12B; note that current is zero and voltage a maximum at the upper end of the antenna, just as with a half-wave antenna. (Remember that these curves indicate r.m.s. values; the current and voltage at each point along the antenna are varying in a sine wave manner at the signal frequency.) A grounded vertical antenna is at resonance whenever its height is some odd multiple of  $\frac{1}{4}$  wavelength; it can therefore be  $\lambda/4$  ( $\lambda$  is the symbol for one wavelength),  $3/4\lambda$ ,  $5/4\lambda$ , etc. An untuned vertical antenna (an antenna

whose height at a particular signal frequency is not a multiple of  $\lambda/4$ ) can be tuned to resonance, making it purely resistive and giving maximum signal pick-up at that signal frequency, by adding either a coil or a condenser in series.

Typical resonant receiving antennas are shown in Fig. 13; current distribution curves for these are shown in dotted lines. At *A*, *B* and *C* are grounded  $\lambda/4$  antennas, while at *D* and *E* are ungrounded  $\lambda/2$  antennas; in each case the free ends have zero current. A receiving antenna which is at resonance when straight will remain at resonance even though a part of it is bent at an angle. A resonant half-wave antenna for reception at 49 meters should be  $49 \div 2$ , or 24.5 meters (81 feet) long. A grounded quarter-wave antenna which is to be resonant at this same wavelength should be about 40.5 feet long.

A quarter-wave antenna which is resonant at 98 meters will be a half-wave antenna for signals at 49 meters and will be a full-wave antenna for signals at 24.5 meters. At wavelengths other than these values we can secure signal pick-up approaching that of a resonant antenna by introducing into the antenna system a condenser which will tune it to a lower wavelength, or by inserting a coil which will tune it to a longer wavelength. This practice is used quite often, especially when the antenna does not have the correct length for resonance at a desired frequency. For example, if an 81-foot length of wire is needed for a center-fed half-wave antenna and only 61 feet of room is available, the surplus 20 feet of wire can be wound in two coils of equal size, and one placed in series with each section of the antenna, near the center.

*Directional Characteristics.* Although the shape of a receiving antenna has little effect upon resonant

conditions, shape does affect the directional characteristics of an antenna. If an antenna is short in comparison to the wavelength of the signals which it picks up (most of the antennas used for 500 to 1,500 kc. broadcast band reception are in this class), best signal pick-up is secured when the electric lines of force associated with the radio wave arrive *parallel* to the antenna wire. Thus, the vertical portion of a short antenna will respond best to vertically polarized electric fields, and the horizontal portion of a short antenna will receive horizontally polarized waves best. Only short antennas (shorter than one-quarter wavelength of the radio wave which is intercepted) have these properties, for they are acted

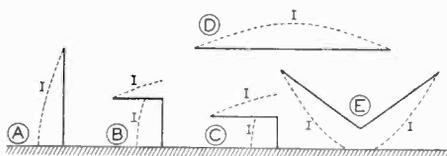


FIG. 13. These curves show that current distribution *I* along a quarter-wave or half-wave antenna is essentially independent of the shape of the antenna.

upon directly by the electric field and are not greatly influenced by surrounding objects or by the ground.

*Radiation Patterns for Receiving Antennas.* When the length of an antenna is one-fourth the wavelength of the signal picked up (or is longer than  $\frac{1}{4}$  wavelength), special patterns must be used to show the directional characteristics. These patterns are called *radiation patterns* because they are usually secured by making measurements while the antenna in question is connected to a transmitter and is radiating signals, but each pattern applies to a particular antenna *regardless of whether it is used for transmitting or receiving purposes*. A horizontal radiation pattern for a receiving antenna indicates the effectiveness of signal pick-up in any direction along the ground. A vertical radiation pattern for a re-

ceiving antenna indicates the effectiveness of signal pick-up at various angles to the ground in one vertical plane passing through the center of the antenna system.

In Fig. 14A is shown the radiation pattern in a vertical plane for a

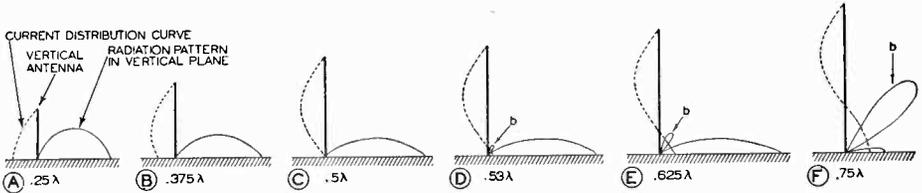


FIG. 14. Current distribution curves (dotted lines) and vertical radiation patterns (thin solid lines) at one particular frequency for grounded vertical antennas having various physical lengths.

grounded vertical  $\lambda/4$  antenna; this pattern will be the same for vertical planes in all directions (north, east, south or west) for this particular antenna and height. As antenna height is increased above  $\lambda/4$  the radiation pattern flattens out, indicating improved pick-up of the ground wave, as shown at B, C, D and E in Fig. 14. When antenna height becomes more than  $\lambda/2$  ( $.5\lambda$ ), a lobe or ear directed up into the sky begins to form, however; this indicates that the antenna is beginning to favor reception of sky waves. If the height of the antenna is

tern indicates that a vertical antenna receives ground waves equally well from all directions along the surface of the earth. When a grounded vertical antenna is shorter than  $\lambda/4$ , the center of a horizontal wire can be connected to the top of the vertical antenna to

form the T antenna shown in Fig. 15B; this does not affect the horizontal radiation pattern, but it does serve as a load which helps to tune the antenna to resonance. The current distribution curves show that currents are zero at the ends of the horizontal portion and increase gradually up to point a, where the two currents combine to flow down the vertical portion of the antenna. These currents are equal for a symmetrical T antenna, and since they flow in opposite directions toward point a in Fig. 15B they do not alter the non-directional characteristics of the verti-

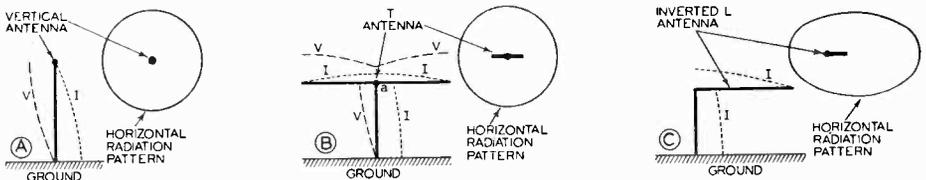


FIG. 15. Horizontal radiation patterns for grounded vertical, T and inverted L types of antennas.

increased beyond  $.75\lambda$ , only sky waves will be received effectively. Remember that these patterns are for grounded vertical antennas; when a vertical antenna is elevated above the ground, the patterns become quite different.

The horizontal radiation pattern for a quarter-wave grounded vertical antenna is shown in Fig. 15A. This pat-

tern indicates that a vertical antenna receives ground waves equally well from all directions along the surface of the earth. When a grounded vertical antenna is shorter than  $\lambda/4$ , the center of a horizontal wire can be connected to the top of the vertical antenna to form the T antenna shown in Fig. 15B; this does not affect the horizontal radiation pattern, but it does serve as a load which helps to tune the antenna to resonance. The current distribution curves show that currents are zero at the ends of the horizontal portion and increase gradually up to point a, where the two currents combine to flow down the vertical portion of the antenna. These currents are equal for a symmetrical T antenna, and since they flow in opposite directions toward point a in Fig. 15B they do not alter the non-directional characteristics of the verti-

cal portion. The T-type antenna is particularly desirable when it is impractical to erect a sufficiently high vertical antenna to give resonance. Resonant conditions produced by the antenna itself always give better signal pick-up than resonance produced by inserting coils or condensers.

A vertical antenna shorter than  $\lambda/4$

can be increased in effective length by connecting to its top a single horizontal wire as shown in Fig. 15C, giving an inverted L antenna. The horizontal portion serves to load the vertical portion; if its length is properly chosen, it will tune the antenna system to resonance at one desired signal frequency, giving maximum possible current at

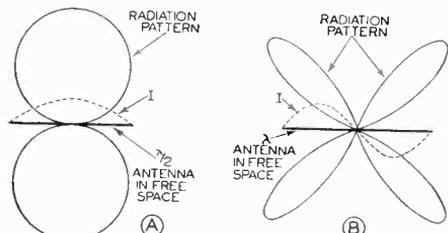


FIG. 16. Radiation patterns for  $\lambda/2$  and  $\lambda$  antennas in free space. The patterns are the same for all planes passing lengthwise through the antennas.

the receiver. Radio waves arriving from the direction in which the horizontal section is pointing are favored, giving a slightly directional radiation pattern in the horizontal plane, as shown in Fig. 15C. The natural wavelength of an L or T type antenna is determined by measuring from the ground to the end of the longest horizontal section.

The radiation pattern of a theoretically perfect half-wave antenna in free space will be the same in any plane

that it receives best from its sides, and has practically no signal pick-up from its ends. Practical doublet antennas have somewhat similar directional characteristics, and consequently should be kept broadside to a station from which maximum signal pick-up is desired.

The radiation pattern for a full-wave ( $\lambda$ ) antenna in free space, shown in Fig. 16B, resembles somewhat a four-leaf clover, indicating that best reception is restricted to four narrow beams, and that there is no reception either from the ends (in the plane of the antenna) or from the sides (at right angles to the antenna). More and more of these lobes appear in the radiation pattern as antenna length is increased in multiples of  $\lambda/2$ , with the lobes becoming longer and narrower; antennas longer than  $\lambda/2$  are therefore unsuited for general reception in all directions.

It is practically impossible to erect a horizontal doublet receiving antenna which is high enough above the ground so it will act as if it were in free space. The average radio receiver owner would have difficulty in erecting a horizontal antenna which is higher than about 50 feet above the ground (about 15 meters). This would make the antenna  $\lambda/4$  above the ground for a 60-

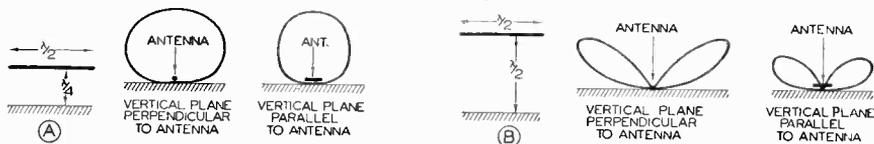


FIG. 17. Radiation patterns for a horizontal half-wave doublet antenna located one-fourth wavelength (A) and one-half wavelength (B) above a perfectly conducting ground.

passing through the antenna wire and will be a figure-of-eight pattern as shown in Fig. 16A. A three-dimensional radiation pattern, showing the pick-up characteristics in all directions, would appear like a round "doughnut," with the antenna wire running through the doughnut hole. The outstanding fact about a  $\lambda/2$  antenna in free space is

meter wave. The radiation patterns in two vertical planes at right angles to each other for a  $\lambda/2$  horizontal doublet antenna mounted a distance of  $\lambda/4$  above the ground are shown in Fig. 17A. These patterns tell us that sky wave reception is good and is essentially the same in all directions. The patterns might lead you to expect poor

ground wave reception, but ground waves are usually tilted sufficiently to induce ample voltage in an antenna of this type, giving quite good ground wave reception also. When the half-wave doublet antenna in Fig. 17A is raised another  $\frac{1}{4}$  wavelength, making it  $\frac{1}{2}$  wavelength above the ground, a decidedly directional radiation characteristic is obtained, as indicated in Fig. 17B. Sky waves coming in at a reasonable angle (the usual condition for these waves) are picked up quite well, but waves arriving at high or low angles of elevation are poorly received. An antenna such as this would not be good for reception of signals broadcast by an airplane flying overhead, nor for reception of ground wave signals unless they were well tilted.

The angle at which maximum energy reaches a receiving antenna is the same as the angle at which maximum energy leaves the transmitting antenna. In other words, if a given transmitting antenna radiates maximum energy into the sky at a definite angle of elevation, maximum pick-up of signals from that antenna will be obtained with a receiving antenna which has maximum pick-up at that same angle of elevation.

**Conclusion.** Keep the following facts in mind when installing receiving antennas:

1. A simple antenna having a pick-up portion which is  $\lambda/4$  meters long or less and in one straight line gives best reception of radio waves whose electrical lines of force are parallel to the pick-up wire.
2. Horizontal doublet antennas having pick-up sections with a total length of  $\lambda/2$  or less give best pick-up of signals arriving broadside to the antenna, but give fairly good pick-up from other directions as well at normal antenna heights above ground.

### Coupling The Antenna To The Receiver

Now that we have seen how the pick-up portion of an antenna can be made to produce maximum voltage and

maximum current for a given radio wave, we are ready to consider how maximum signal can be transferred from the antenna to the radio receiver. For antennas less than a quarter-wave long, the solution is quite simple, involving merely a direct connection as shown in Fig. 18. The entire length of the antenna system will pick up signals, and consequently length  $b-c$  must be added to length  $a-b$  when determining the operating wavelength of the system. For a  $\lambda/4$  system, the current will be a maximum at the ground end; if the receiver is connected close to ground, maximum coil current will flow through primary coil  $P$  and will induce

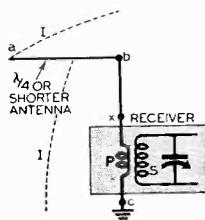


FIG. 18. Direct connection of an inverted L antenna to a receiver.

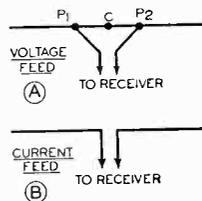


FIG. 19. Feed methods for a doublet receiving antenna.

the greatest possible voltage in secondary coil  $S$ . A long ground wire between receiver and ground moves the primary coil to a point where current is less, reducing the efficiency of the antenna; furthermore, an excessively long ground wire picks up noise and thus lowers the signal-to-noise ratio. Keep the ground wire as short as possible, and increase the length of the antenna itself up to  $\frac{1}{4}$  wavelength if maximum pick-up is required at a given wavelength.

A  $\lambda/4$  antenna for best reception of a 1,500 kc. (200-meter) signal should be 50 meters (165 feet) long. This length is rarely attained in the average receiving antenna, but a coil can be inserted in the lead-in wire (at point  $x$  in Fig. 18) to offset this lack of length. On short-wave bands, however,

even a 50-foot antenna is too long for  $\lambda/4$  operation. When an antenna is only slightly longer than  $\lambda/4$  (not more than about  $3/8$  wavelength long), the excessive length can be offset by inserting a variable condenser at point  $x$  and adjusting for maximum signal pick-up from the desired station (this tunes the antenna to resonance, making it equivalent to a  $\lambda/4$  antenna at that station frequency). Tuning of an antenna complicates the operation of a receiver, so it is customary to design the primary coil so it will make the average broadcast antenna tune approximately to the broadcast band wavelengths, and provide means for reducing the size of this coil on short wave bands.

**Voltage Feed For a Doublet Antenna.** With a doublet antenna, it is quite possible to connect two wires (a transmission line) to points which are equally spaced on each side of the center, as shown in Fig. 19A, in order to transfer to the receiver a portion of the r.f. voltage which is developed by the antenna. The voltages at  $P_1$  and  $P_2$  are opposite in polarity at any instant of time (see Fig. 11C), and hence the voltage between  $P_1$  and  $C$  will always add to that between  $P_2$  and  $C$ . This connection is commonly referred to as a *voltage feed* to the receiver.

**Current Feed.** When a doublet antenna is cut at its center and two wires are used to connect the cut ends to the receiver, we have what is known as *current feed* to the receiver. Antenna current flows along one half of the pick-up section, down one lead-in wire to the receiver, through the antenna coil of the receiver, up through the other lead-in wire, and finally out over the other half of the pick-up section. Since current in a  $\lambda/2$  or shorter antenna is a maximum at the center, this connection gives maximum current through the antenna coil of the receiver. Current feed is widely used with

antennas for all types of receivers.

A  $\lambda/2$  antenna which is designed for 80 meters will be a full-wave antenna at 40 meters, a  $3/2\lambda$  antenna at 30 meters, and a  $2\lambda$  antenna at 20 meters. Let us see if an antenna like this will work effectively at all three wavelengths with current feed. Figure 20 shows the current distribution for these four operating conditions; these curves show that current is a maximum at the center only for operation as a  $\lambda/2$  or  $3/2\lambda$  antenna. Current at the center is zero for  $\lambda$  or  $2\lambda$  operation, which means that a current feed at the center will theoretically give no signal transfer to the receiver under these conditions; it

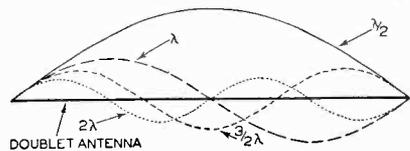


FIG. 20. Current distribution curves for  $\lambda/2$ ,  $\lambda$ ,  $3/2\lambda$ , and  $2\lambda$  operation of a doublet antenna.

would be necessary to move the tap to an off-center point of maximum current in order to secure efficient signal pick-up. This explains why a single  $\lambda/2$  antenna which is used for all-wave reception and has a current feed connection to the receiver will give excellent results at certain frequencies and will be very inefficient at other frequencies.

## Transmission Lines

### Antenna Resistance and Impedance.

Any antenna which is tuned to resonance will act as a resistance source of r.f. signals; at the center of a  $\lambda/2$  antenna this a.c. resistance will be about 72 ohms. For a grounded  $\lambda/4$  antenna, the a.c. resistance at the ground (measured between the antenna lead-in wire and the ground wire) will be about 36 ohms. A voltage-fed  $\lambda/2$  antenna will have a resistance considerably higher than 72 ohms; if the transmission line connections are made near the ends of

the antenna, this resistance may be as high as 5,000 ohms. At off-resonant conditions, reactive components are added to these resistance values, making the impedance of an untuned antenna quite high.

*Surge Impedance of Transmission Lines.* Any two-wire transmission line is a combination of elemental resistances, inductances and capacities arranged as shown in Fig. 21A. At low audio frequencies the inductances and capacities have a negligible reactive effect and the line simply has a d.c. resistance distributed along its length due to the resistance of the wire itself. At radio frequencies, however, the elemental inductances and capacities have much more effect upon the characteristics of the line than do the elemental resistances, and we can for practical purposes neglect entirely the presence of these resistances. The values of these elemental inductances and capacities control one extremely important characteristic rating of a transmission line—its *surge impedance*, usually designated as  $Z_0^*$ . The inductance values are determined by the diameter of the transmission line wires and the spacing between them; the capacity values are determined by the diameter of the wires, the spacing between the wires, and by the nature of the dielectric materials between the two wires. Neither the frequency of the radio signal nor the length of the line ordinarily has any effect whatsoever upon the inductance and capacity values. This means that the surge impedance of a transmission line depends only upon the construction of a unit length of a transmission line, and is

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\*This impedance is also known as the *characteristic impedance* of a transmission line; "characteristic impedance" is a more readily understood term, but "surge impedance" is the term more widely used. The value of this impedance depends upon the ratio of elemental inductance to elemental capacitance.

the same for any length of line and for any frequency. Increasing the distance between the two line wires will increase the surge impedance; reducing the diameter of the wires will increase the surge impedance; the use of a dielectric material which increases the capacity between the two wires will reduce the surge impedance.

If we were to measure the impedance between the input terminals of an extremely long (infinitely long) transmission line, we would get a value which is equal to the surge impedance of that line. Furthermore, this impedance would be purely resistive, for the large numbers of elemental inductances and capacities distributed over this infinitely long line make it act as a broadly tuned resonant circuit having the same impedance at all frequencies. This fact is illustrated in Fig. 21B.

If we were to measure the input impedance of a practical transmission line (seldom more than one or two wavelengths long in radio work), we would probably get a value considerably different from the surge impedance of the line; actually the value of impedance which we measure would depend upon the length of the line and upon how it was terminated (the nature of the load at the output end of the line). It is possible to compute the input impedance of a transmission line having any length and any type of load at its output end, but the mathematical procedure involved is highly complicated and for all practical receiving antenna purposes is quite unnecessary.

We are not concerned with this input impedance of a practical transmission line, because there is a simple way to make this input impedance equal to the surge impedance of the transmission line. If the output end of a transmission line is terminated in a load having an impedance exactly equal to the surge impedance of the transmission line, the input impedance of that

transmission line will then be equal to the surge impedance of that line regardless of its length; this is illustrated in Fig. 21C. If the input end of this line is now connected to a signal source having an impedance equal to the surge impedance of the transmission line, the impedance at each end of the transmission line will be equal to the surge impedance  $Z_0$  regardless of the length of the line. This phenomenon is widely used in the design of transmission lines for connecting receiving antennas to receivers, for it simplifies construction considerably and gives at

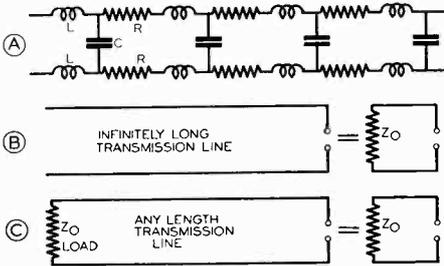


FIG. 21. Important facts concerning the behavior of transmission lines.

each end of the transmission line an impedance match which results in maximum transfer of power from the antenna to the receiver.

**Typical Surge Impedance Values.** A transmission line made up of two No. 18 wires spaced 2 inches apart will have a surge impedance of about 560 ohms. Two No. 10 wires separated by 2 inches have a surge impedance of about 440 ohms. No. 18 and No. 10 wires separated by 3 inches will have surge impedances of 610 ohms and 480 ohms respectively. These so-called "open" transmission lines are rarely used with receiving antennas, but are widely used with transmitter installations. When two No. 18 rubber-covered wires are twisted together or when one rubber-covered wire is placed inside a braided metallic covering, the surge impedance  $Z_0$  will range between 70 and 150 ohms; by careful design, this

impedance can be made equal to 72 ohms, making it possible to couple a current-fed  $\lambda/2$  antenna directly to the transmission line, securing a perfect impedance match and maximum transfer of power from the antenna to the transmission line. If the receiver is made to serve as a 72-ohm load on the other end of this line, we will have maximum transfer of power from the antenna to the receiver regardless of the length of the transmission line.

It is not essential that a 72-ohm transmission line be used for maximum transfer of power. An r.f. impedance-matching transformer, usually called an *antenna transformer*, can be used at the antenna to match the antenna impedance to the surge impedance of the transmission line regardless of what their respective values may be, thereby securing maximum transfer of energy. Another impedance-matching transformer can be used at the receiver end to match the receiver input impedance to the transmission line surge impedance. When a receiver is designed for use with a definite antenna system, the first r.f. transformer in the receiver is designed to match the surge impedance of the transmission line, eliminating the need for a separate matching transformer.

### Noise Reduction

It is useless to erect an antenna which will deliver a strong carrier signal to a receiver if this signal is "blanketed" by local man-made interference noise which is also picked up by the antenna. As a general rule, man-made interference produces electric fields which are *perpendicular* to the ground, just like the fields produced by a vertical transmitting antenna. If a horizontal antenna with a vertical down lead is used for reception, most of the interference noise will be picked up by the down lead; very little will be picked up by the horizontal section.

The chief problem in securing a noise-reducing antenna is that of preventing the down lead or transmission line from picking up noise signals.

With a doublet antenna, the transmission line inherently rejects noise signals. The two down leads of the transmission line are normally twisted together in the manner shown in Fig. 22A, so that very little electric field can exist between adjacent wires in the transmission line. Any noise voltages which are induced in these vertical wires cause currents to flow in the same direction in both wires at any instant of time. Noise currents therefore flow in opposite directions through the primary of the receiver transformer to the center tap and then to ground; as long as these two noise currents are balanced, they cannot produce any flux in the receiver transformer. Some noise signals may be picked up by the entire antenna system acting as a vertical mast; an electrostatic shield is usually placed between the primary and secondary windings of the receiver input transformer to prevent these noise signals from being transferred to the secondary winding.

The antenna and receiver transformers in Fig. 22A can give a perfect impedance match at only one frequency; the tuning action can, however, be made sufficiently broad by means of tight coupling in the transformers to give satisfactory reception over a wide range of frequencies.

Noise-reducing antenna systems for the 500 kc. to 1,500 kc. broadcast band are invariably of the single-wire horizontal type; a typical noise-reducing broadcast antenna is shown in Fig. 22B. Since limitations of space in the vicinity of the average home would normally prevent the use of a horizontal pick-up section which is  $\lambda/4$  meters in length, a more reasonable length of from 50 to 100 feet is usually employed, and the primary of the an-

tenna transformer is designed to bring this length up to  $\lambda/4$  for the highest frequency which the antenna is to pick up. The transmission line in this case is a single insulated wire surrounded by a flexible metal loom or shield; this shield is connected to the ground terminal of the receiver. A twisted 2-wire transmission line may be used if proper ground connections are made at the receiver, and the transmission line is properly connected to the antenna transformer. For maximum reduction of noise and the most favorable distribution of current in the antenna, the shielding metal loom must be grounded near the antenna, either to a vent pipe,

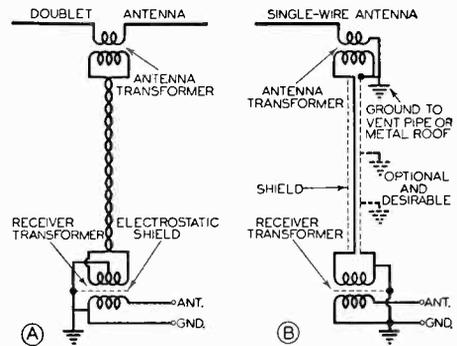


FIG. 22. Two types of noise-reducing antennas.

to the grounded metal roof of the building, or to a grounded gutter. Additional grounding of the shield at several points is optional, but should be done whenever the transmission line runs conveniently near a grounded object. This type of noise-reducing antenna may also be designed for all-wave reception.

Even though the vertical lead-in wires of an antenna system are carefully shielded against man-made noise fields, there is still a possibility that noise will be picked up by the antenna system. This is due to the fact that the electric fields produced by man-made interference may not always be exactly perpendicular to the earth; if they are slanted any appreciable amount and

extend an appreciable distance above the ground, they will induce noise voltages in the horizontal pick-up wires. When a noise-reducing antenna system fails to give a satisfactory signal-to-noise ratio, the usual procedure is to *change the position of the horizontal pick-up portion of the antenna*, such as by increasing its height, changing its direction, or moving it to a location farther away from the source of noise.

*Noise-Reducing Loop Antenna.* Another solution to the problem of noise

fact,  $E$  may be as much as 70 times stronger than  $H$ . Noise interference sources are generally so close to the receiver that the vertically-polarized electric component  $E$  of the noise is much stronger than the magnetic component of noise, whereas the  $E$  and  $H$  components of the desired radio signals are equal. If we can remove the electric component  $E$  of both the noise signals and the radio signals, depending upon magnetic components only for reception, we will then have eliminated the

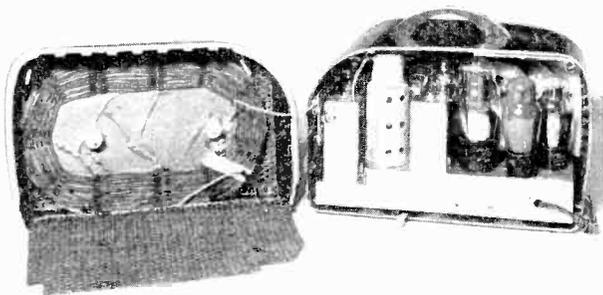
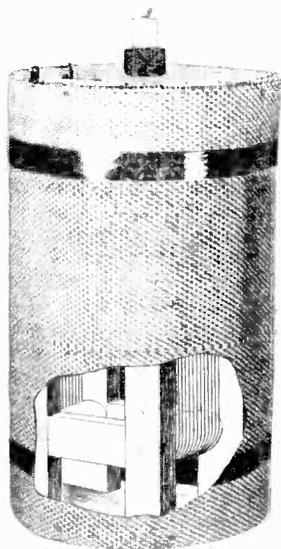


Fig. 23 (Left). Typical shielded loop antenna (General Electric Beamscope) designed for installation inside a console model home radio receiver. Part of the Faraday shield is cut away to show the loop winding and its wood frame inside. This unit is rotated to a position of minimum noise pick-up at the time the receiver is installed in a home.

Above: A compact shielded loop antenna mounted at the rear of this 5-tube superheterodyne table model receiver (Zenith Wavemagnet) eliminates the need for antenna and ground connections. The flat loop is wound on a spider-web type coil form, and is mounted between two flat shields woven with vertical copper wires and horizontal insulating cords. All shield wires are connected together at the bottom and grounded to the receiver chassis.

interference is the use of a shielded loop antenna which is built right into the cabinet of a radio receiver. To understand how this shielded loop can reduce noise signals, we must first consider the relative strengths of the electric and magnetic fields associated with radio waves. At points outside of the induction field produced by a transmitting antenna or source of noise interference, the electric and magnetic fields associated with the radiation field (the radio wave) are essentially equal. Close to the source, however, the electric component  $E$  is much greater than the magnetic component  $H$ ; in

noise almost completely without impairing reception of the desired radio signal. The chief purpose of the grounded Faraday shield which surrounds some types of built-in loop antennas is therefore to prevent *electric components of both station and noise signals from affecting the receiver*.

One commercial version of this shielded loop arrangement is shown in Fig. 23. The Faraday shield is in the form of a closed cylinder with sheet metal discs covering the top and bottom faces and with the sides covered with a coarse woven material in which the vertical threads are wire and the

horizontal threads are non-conducting fiber. Every vertical wire makes contact with the top metal disc, but only one of these vertical wires makes electrical contact with the bottom disc; this construction eliminates closed circuits in the vicinity of the loop, giving maximum pick-up by the loop of the magnetic component of radio signals. The shield is grounded, and consequently all signal and noise currents induced in its vertical wires by the  $E$  components are led off to ground without affecting the receiver. The well-known directional characteristics of a loop are utilized to give additional noise rejection; the loop is rotated for minimum noise, so its line of minimum pick-up is in the direction of the noise source. No external antenna or ground connections are necessary when one of these shielded loops is installed, but an outdoor antenna can be connected to the loop or in place of it if distant reception is desired. A loop of this type can be built right into the cabinet of a radio receiver, and can be tuned over the entire broadcast band by a main tuning condenser section which is adjusted to track with the oscillator; this tuning to resonance for each station increases the sensitivity of the loop very greatly.

### All-Wave Antennas

Either horizontal doublet antennas or single-wire horizontal antennas with end feed may be used for all-wave reception. The doublet is ordinarily preferred, for its balanced two-wire transmission line gives inherent rejection of vertically-polarized noise signals. Both types of all-wave antennas favor slightly the reception of signals from certain directions, but these directional characteristics can ordinarily be neglected when installing an all-wave antenna for the average radio listener. It is far more important to place the hori-

zontal pick-up section of the antenna at right angles to a nearby power line, in order to secure rejection of horizontally-polarized power line noise signals.

*Types of All-Wave Antennas Commonly Used.* Three distinct types of antenna systems are in common use for all-wave reception over the entire range from about 18 meters (16.5 megacycles) to 545 meters (550 kc.). These are:

1. An antenna system made up of two or more doublet antennas, each tuned to a different wave-length. With the customary broadness of tuning in doublet antennas, fairly uniform pick-up is obtained over the desired range. A typical system might employ three half-wave doublets, tuned respectively to 16, 25 and 49 meters.

2. A system employing a single horizontal doublet antenna cut at its exact center for current feed, and with a length which will keep it less than a full-wave antenna at any frequency in the range to be received. For example, if the receiver is to go down to 18 meters, a 50-foot horizontal doublet can be used, for it will be a full-wave antenna at about 15 meters and will have sufficient current at its center at 18 meters. At 30 meters this antenna would become a  $\lambda/2$  unit; at all longer wavelengths it will be shorter than  $\lambda/2$ , and will have maximum current at its center even though the current is gradually reduced due to off-resonant conditions as wavelength is increased.

3. A system employing a single horizontal wire having a properly grounded antenna transformer at one end for current feed, and with a length which will keep the effective pick-up portion of the system less than a half-wave long at any frequency in the range to be received. For example, if the shortest wavelength at which reception is desired is 18 meters, a 25-foot horizontal wire would very likely be used (this would be a half-wave antenna at 15 meters).

An all-wave doublet antenna gives maximum pick-up at a frequency which makes it  $\lambda/2$ . Pick-up decreases gradually above  $\lambda/2$ , for the current at the center of the antenna goes down. At  $\lambda$  operation the current at the center is zero, and there is practically no

signal pick-up. Above  $\lambda$ , pick-up increases again to a maximum at  $3/2\lambda$ , then drops to zero again at  $2\lambda$ . All this means that a doublet antenna will pick up stations having frequencies higher than that for  $\lambda$  operation, but reception will be erratic due to the highly directional and continually changing radiation patterns and to the dead spots in reception which occur at  $\lambda$ ,  $2\lambda$ ,  $3\lambda$ ,  $4\lambda$ , etc., operation.

A single-wire antenna acts in much the same way above its efficient operating length of  $\lambda/4$ ; there will be dead spots in reception at  $\lambda/2$ ,  $\lambda$ ,  $3/2\lambda$ ,  $2\lambda$ ,  $5/2\lambda$ , etc., for these operating lengths give zero current at the antenna transformer. Pick-up will be good at  $\lambda/4$ ,  $3/4\lambda$ ,  $5/4\lambda$ ,  $7/4\lambda$ , etc., for these operating lengths give high currents at the antenna transformer.

**Multiple Doublet Antennas.** A typical all-wave doublet antenna system employing three doublet antennas is shown in Fig. 24. Doublet  $A_1$  serves for the shortest wavelength,  $A_3$  for the longest wavelength and  $A_2$  for the intermediate wavelength. (Coils  $L_A$  serve merely to increase the effective length of doublet  $A_3$ , improving reception at about 50 meters.) Although these three antennas work best at their own resonant wavelengths, they are sufficiently broad to give adequate pick-up at in-between wavelengths. The connections between one or more of the doublets and the transmission line are transposed, as is done for  $A_1$  and  $A_2$  in Fig. 24, so that the signal voltages in the various doublets will add and give maximum transmission line current.

Whenever the wavelength of a desired radio signal is such as to give resonance for one of the doublet antennas in this system, the antenna impedance will be that of a normal doublet, which is 72 ohms. Since the transmission line is designed to have a surge impedance of about 72 ohms, a good

match between the antenna and the transmission line is secured at resonant frequencies. When the desired signal does not give resonance for any one of the doublets, the impedance of the antenna will be much higher than 72 ohms, but there is one interesting characteristic of a transmission line which can be used to counteract this mismatch: A practical transmission line (only a few wavelengths long) has a high input impedance whenever its length is some multiple of  $\lambda/2$  meters

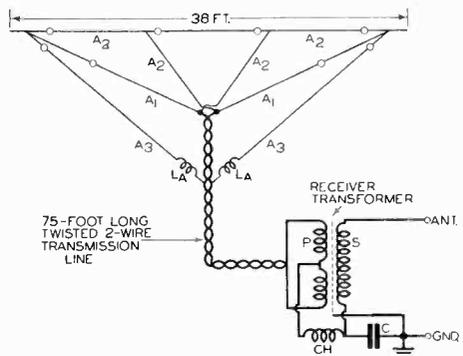


FIG. 24. All-wave noise-reducing antenna made up of three doublets. Because of its general resemblance to the web of a spider, this RCA-Victor product is commonly known as a spider-web antenna.

for a particular signal. By making the length of the transmission line a multiple of  $\lambda/2$  for wavelengths in between the natural resonant frequencies of the three doublets, efficient transfer of signals from the antenna to the transmission line is obtained at in-between wavelengths as well. Whenever the manufacturer of an all-wave antenna specifies that the transmission line supplied with the system must not be cut, and that if too short it should be lengthened in equal lengths of a definite number of feet, you can be sure that the purpose is improvement of match at off-resonant frequencies.

With proper design it is possible to build a multiple-doublet all-wave antenna which is reasonably effective at

all wavelengths between the lowest and highest natural wavelengths of the doublets employed. Response at wavelengths below this range can be obtained by adding shorter doublets. Fairly satisfactory response will be obtained at wavelengths above this range since all of the doublets will then be shorter than  $\lambda/2$ , and will have a maximum current at their centers. At broadcast band wavelengths, however, this maximum current becomes quite low even for the longest doublet; to offset this, the special receiver coupling transformer shown in Fig. 24 is employed to convert the entire antenna system into a T antenna for broadcast band reception.

Observe that the midpoint of primary winding  $P$  is connected to ground through choke coil  $CH$  and condenser  $C$ . At broadcast band frequencies, very little signal current is fed to the transmission line by the doublets; the entire system acts as a T antenna, and signal currents which are picked up by the transmission line flow down both transmission line wires in the same direction to the center tap of  $P$ , then through  $CH$  (which has a low reactance at broadcast frequencies), through secondary winding  $S$  to the antenna terminal of the receiver, and through the receiver input coil to ground. Condenser  $C$  has such a high reactance at low frequencies that it acts essentially as an open circuit. When an all-wave doublet antenna system is converted to a T antenna in this manner, the system of course loses the noise-reducing and directional properties of a doublet.

*Single All-Wave Doublets.* An antenna arrangement which gives satisfactory reception on short-wave bands and which automatically converts to a T type antenna at broadcast frequencies without losing its noise-reducing properties and without sacrificing an impedance match is shown in schema-

tic form in Fig. 25A. This system employs a single doublet antenna; it may be center-fed, in which case both sides of the horizontal doublet will be equal in length, or it may be fed from an off-center point, in which case the sides of the doublet will be unequal in length.

This unique system employs two transformers at the antenna and two at the receiver. At high frequencies, condensers  $C_1$  and  $C_2$  become so low in reactance that they short out coils  $L_9-L_{10}$  and  $L_{11}-L_{12}$ , making the two right-hand transformers ineffective at high frequencies; the transmission line circuit therefore takes the form shown

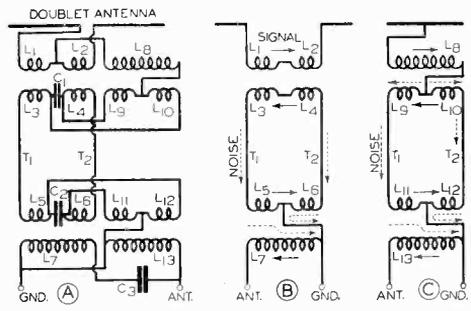


FIG. 25. Taco all-wave noise-reducing antenna system, made by Technical Appliance Corp. Actual circuit is shown at A; the effective circuit for short-wave bands is at B, while the effective circuit for the broadcast band is at C. Dotted arrows show directions of noise currents at one instant of time; solid arrows show directions of desired signal currents at one instant of time.

in Fig. 25B at high frequencies (at short wavelengths). As you can see, this is a conventional doublet arrangement; if its length is  $\lambda/2$  or less, current will flow along the entire antenna in the same direction at any instant of time. This current will flow through  $L_1$  and  $L_2$  in the same direction, inducing voltages in  $L_3$  and  $L_4$  which add together and cause current to flow down transmission line  $T_1$  and up transmission line  $T_2$  at one instant. The result is current flow through  $L_5$  and  $L_6$  in the same direction at any instant, inducing in  $L_7$  a voltage which is applied to the input of the receiver. The

transformers used in this system provide the desired broad impedance match for efficient transfer of power over a wide range of high frequencies. Noise currents flow through leads  $T_1$  and  $T_2$  in the same directions, then in opposite directions through  $L_5$  and  $L_6$  to their common terminal and to ground, so no noise signal voltages are induced in  $L_7$ .

At broadcast band frequencies, condensers  $C_1$  and  $C_2$  have high reactances, and hence have no shorting effect upon secondary windings  $L_9$ - $L_{10}$  and  $L_{11}$ - $L_{12}$ . Furthermore, at these low frequencies coil elements  $L_1, L_2, L_3, L_4, L_5, L_6$  and  $L_7$  have negligible inductive reactance and may be neglected, giving in effect the circuit shown in Fig. 25C. Condenser  $C_3$  in Fig. 25A has a high reactance at broadcast band frequencies, and serves to prevent  $L_7$  from shorting  $L_{13}$ . Again we find that noise signals picked up by the vertical down leads  $T_1$  and  $T_2$  are cancelled out at the receiver end of the transmission line (in primary windings  $L_{11}$  and  $L_{12}$ ). The entire down lead system acts as a vertical or T antenna, with the flat top section serving merely as a load which produces a larger r.f. current at the point along the antenna to which coil  $L_8$  is connected. This current flows down through  $L_9$ , then divides to flow through  $L_9$  and  $L_{10}$  and through  $L_{11}$  and  $L_{12}$  to the mid-point and then to ground, as shown by the dotted line arrows. This is normal current distribution for a T type antenna, but of course it does not feed the receiver. The flow of antenna current through winding  $L_8$  serves to induce voltages in  $L_9$  and  $L_{10}$  which act in the same direction and serve to circulate a signal current through the entire transmission line circuit. This current flows through windings  $L_{11}$  and  $L_{12}$  in the same direction, inducing a strong radio signal voltage in  $L_{13}$ . This winding in turn feeds the voltage to the receiver.

*Use of Counterpoise for Noise Reduction.* Another arrangement of a single doublet antenna which gives satisfactory signal pick-up along with noise reduction on all bands is that which employs a counterpoise for picking up strong noise signals. These noise signals are fed through the receiver input circuit in such a way that they "buck out" or cancel the usual noise signals, giving essentially noise-free reception on bands in which the doublet is acting as a T.

The circuit diagram of a typical antenna-counterpoise installation on a

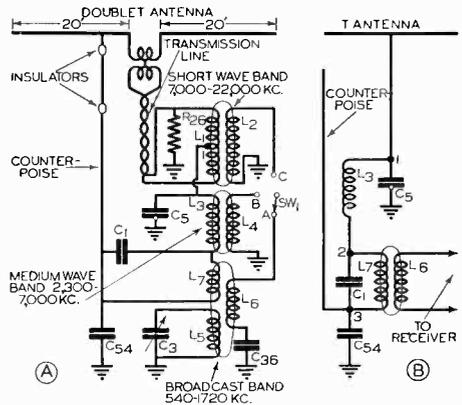


FIG. 26. All-wave noise-reducing antenna system employing a counterpoise (RCA-Victor Master Antenna as used with RCA-Victor model 99K receiver).

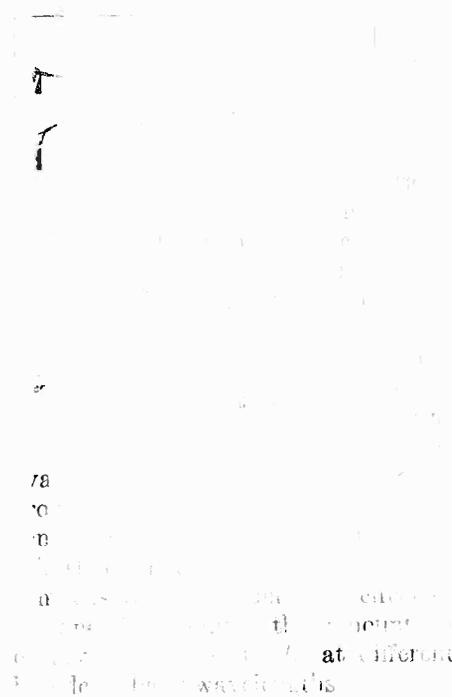
three-band receiver is given in Fig. 26A. Each band has a separate receiver matching transformer as a part of the receiver preselector (an antenna system like this can be used only with receivers designed especially for it). For short-wave reception, band-changing switch  $SW_1$  is at position C, and we have a normal noise-reducing doublet antenna with  $L_1$ - $L_2$  serving as the receiver matching transformer. Condenser  $C_5$  provides a path from the mid-point of  $L_1$  to ground for noise signals, and 1-megohm resistor  $R_{26}$  allows static charges picked up by the antenna to leak off to ground gradually without causing noise.

For broadcast band reception, switch  $SW_1$  is at position *A*, and we have T antenna action. To simplify an analysis of the antenna system under these conditions, we can redraw the circuit as in Fig. 26B to include only those parts which are effective during broadcast band reception. The length of the counterpoise is 10 feet longer than half the length of the transmission line, and consequently this lower portion of the transmission line (adjacent to the counterpoise) will pick up essentially the same amount of station and noise signals as the counterpoise. The upper portion of the transmission line is ordinarily quite high above the ground, out of the region of strong noise signals, but it will pick up and feed down through the transmission line a certain amount of noise along with the station signals. Let us trace the path taken by the noise signal currents, neglecting station signals for the time being.

Noise signals picked up by the T antenna will come down to point 1 in Fig. 26B (corresponding to point 1 at the center tap of  $L_1$  in Fig. 26A); some will go through  $C_5$  to ground, and the remainder will go through coil  $L_3$  to point 2. Again the noise currents divide, with some going through  $L_7$  in a downward direction to point 3, and others going through  $C_1$  directly to point 3. From here the noise currents either go to ground through condenser  $C_{54}$  or take a return path up the counterpoise to the T antenna again. Noise signal currents coming down the counterpoise divide at point 3, with some going to ground through  $C_{54}$ , some going directly to point 2 through  $C_1$ , and some going through  $L_7$  in the opposite direction to that taken by the T antenna noise currents. From point 2 these counterpoise noise currents go through  $L_3$  to point 1, from which some go to ground through  $C_5$  and others take the return path up the T antenna

and over to the counterpoise. By adjusting the value of trimmer condenser  $C_5$ , the division of the counterpoise and T antenna noise currents through out this circuit can be made such that the noise currents sent through each of the two pick-up systems will be approximately equal and opposite, and will therefore "knock out" or cancel each other.

In securing a directional antenna system in this manner, the antenna is a combination of a counterpoise and a T antenna. The T antenna is a doublet antenna, and the counterpoise is a doublet antenna.



### Directional Antenna System

A directional antenna system which is designed to have a controllable directional characteristic is shown in Fig. 27. A two doublet antennas, each 60 feet long, are located at right angles to each other. At 37 meters they become two doublets and have maximum pick-

up. At wavelengths longer than 37 meters they are less than  $\lambda/2$  but still have maximum current at the center and give quite satisfactory signal pick-up. For wavelengths less than 37 meters they may become  $\lambda$ ,  $3/2\lambda$ ,  $2\lambda$ , etc., antennas, with clover-shaped radiation patterns which make the directional adjustment more critical. Each doublet has a two-wire transmission line going to its own coil at the receiver end; the two coils are mounted at right angles to each other as shown in Fig. 27B, with

a third coil mounted in their center in such a way that it can be rotated for maximum coupling with either of the stationary coils or for any intermediate amount of coupling with both coils. In this way it is possible to control the direction from which maximum signal pick-up is secured. The performance is essentially the same as that of the rotatable beam antennas used by amateur radio enthusiasts and by commercial short-wave stations for long-distance radio communication.

### TEST QUESTIONS

Be sure to number your Answer Sheet 33FR-2.

Place your Student Number on every Answer Sheet.

Never hold up one set of lesson answers until you have another ready to send in. Send each lesson in by itself before you start on the next lesson. In this way we will be able to work together much more closely, you'll get more out of your Course, and you will receive the best possible lesson service.

1. In addition to feeding strong signals from desired stations into a receiver, what other characteristic must a good receiving antenna have?
2. What two components does a radio wave have?
3. When a radio signal produces a half-sine-wave distribution of current along a receiving antenna, how many wavelengths long is the antenna?
4. What natural condition in space makes long-distance radio communication possible?
5. Describe briefly how sky waves cause fading at receiving locations near the limits of reliable ground-wave reception from a given station.
6. At what point along a half-wave receiving antenna is r.f. signal current at maximum?
7. What information is given by the horizontal radiation pattern of a receiving antenna?
8. Why is an antenna transformer sometimes used to connect an antenna to a transmission line?
9. What change would you make in a noise-reducing antenna installation if it failed to give a satisfactory signal-to-noise ratio?
10. What is the chief purpose of the grounded Faraday shield which surrounds some types of built-in loop antennas?



## TOWERS IN THE SKY

Above the dust and clamor here below  
These sentinels of mind and spirit rise;  
Defly their fingers touch the vibrant skies  
Where winged words are passing to and fro.  
They dwell on high, where rarer currents  
flow—

Perhaps behold, in pity and surprise,  
The low estate of recompense we prize,  
The narrow round of things we seek to know.

O stalwart towers, unshakeable, serene,  
Make us more worthy of your office here:  
Refine our message, whatsoe'er it be;  
Attune our thoughts to listeners unseen.  
On far horizons may our words ring clear,  
Proclaiming there the truth which makes  
men free!

---

This poem, by George P. Conger, has long been one of my favorites, and I thought you might like it too.

A handwritten signature in cursive script that reads "J. E. Smith". The signature is written in dark ink on a light background.

**FREQUENCY-MODULATED SIGNALS**

**THE F.M. RECEIVER**

**34FR-2**



**NATIONAL RADIO INSTITUTE**

**WASHINGTON, D. C.**

**ESTABLISHED 1914**

# Study Schedule No. 34

For each study step, read the assigned pages first at your usual speed. Reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind, then answer the Lesson Questions for that step. Study each other step in this same way.

**1. Brief A.M. Review; Characteristics of a Frequency-Modulated Signal - - - - - Pages 1-9**

Spend at least twice as much time as usual on this first study step, because in it is the basis for the entire frequency modulation system of broadcasting. Once you get a clear picture in your mind of what an f.m. signal is like, you'll find that f.m. systems are no more difficult than the ordinary a.m. systems you have previously studied. Answer Lesson Questions 1 and 2.

**2. An Elementary F.M. System; Additional Requirements in an F.M. System - - - - - Pages 9-13**

Here is a brief survey of an entire f.m. system from the microphone through the f.m. transmitter and through the f.m. receiver. Simplified circuits and explanations are used so that you can concentrate upon getting a broad general picture first. Answer Lesson Question 3.

**3. The F.M. Receiver; The Preselector; The Frequency Converter Pages 13-18**

The block diagram of a combination f.m.-a.m. receiver in this section will prove a valuable reference guide for studying the remainder of this lesson, because it shows at a glance the relationships between the various stages of the receiver. Study each circuit slowly and carefully, referring to the circuit diagram each time a part is mentioned.

**4. The I.F. Amplifier - - - - - Pages 18-27**

Careful, slow study is required here also. You'll learn how the same i.f. tubes can be used both for f.m. and a.m. signals, and you'll learn how the limiter removes amplitude variations from f.m. signals. Answer Lesson Questions 4, 5, 6 and 7.

**5. The F.M. Detector - - - - - Pages 27-32**

Here's another section in which careful study will be well worth while if you intend to prepare yourself for any job whatsoever connected with f.m. systems. Answer Lesson Questions 8, 9 and 10.

**6. Modern F.M. Transmitters; The Future of F.M. - - - - - Pages 32-37**

This bird's-eye view of an f.m. transmitter completes your background knowledge of f.m. systems. With this knowledge, you should have no difficulty in mastering the details of any f.m. receiver maintenance job once you have completed the N.R.I. Course.

**7. Mail your Answers for this Lesson to N.R.I. for Grading.**

**8. Start Studying the Next Lesson.**

# Frequency-Modulated Signals

## Brief Review of Amplitude-Modulated Signals

A THOROUGH understanding of the characteristics of a conventional amplitude-modulated (a.m.) signal will help you to picture in your mind the characteristics of a frequency-modulated (f.m.) signal.

The signal peaks of an unmodulated r.f. carrier will all have the same amplitude, as shown at the left in Fig. 1A. When the r.f. carrier is amplitude-modulated with a single sine-wave audio frequency (such as that corresponding to a pure audio tone), the peaks of the r.f. signal will rise and fall in amplitude in accordance with variations in the audio signal, as shown at the right in Fig. 1A.

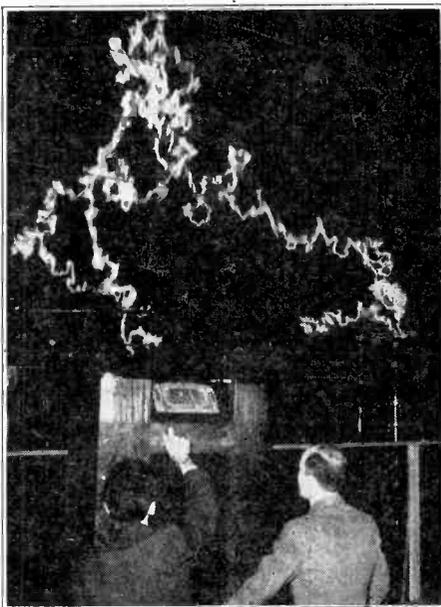
A dash-dash line drawn through the positive peaks of the modulated r.f. signal in Fig. 1A will give a curve known as the *envelope*, which has the same wave form as the original audio signal. (Another envelope having this same audio wave form is obtained by drawing a dash-dash line through the negative peaks, but at this time we need consider only the upper envelope. Both envelopes are shown for each signal in Fig. 1.)

Horizontal line XX at the level of the unmodulated carrier peaks will be the reference line for this envelope. Thus, we can say that the positive peak of the envelope goes above reference line XX, and the negative peak of the envelope goes below reference line XX. Furthermore, with 50% modulation as in Fig. 1A, the envelope peaks will have exactly one-half the amplitude of the unmodulated carrier peaks.

When the envelope peaks are equal in amplitude to the unmodulated car-

rier peaks, as in Fig. 1B, we have 100% modulation of the carrier.

When the envelope peaks are greater in amplitude than the unmodulated carrier peaks, the modulation percentage is higher than 100% and the envelope becomes distorted, as shown in Fig. 1C. When an over-



Courtesy General Electric Co.

A million-volt discharge of man-made lightning created a crashing roar which completely drowned out the musical program when this combination f.m.-a.m. receiver was tuned to a broadcast band a.m. station. When the set was tuned to an f.m. station carrying the same program, however, the music emerged from the loudspeaker clear and strong, with only a hardly noticeable static buzz in the background even though a million volts of electricity was dancing and sputtering a few feet away. Truly this is dramatic proof of the noise-reducing characteristics of an f.m. system.

modulated signal like this is demodulated in a receiver, the resulting audio signal will have the same distorted wave form as the dash-dash envelope in Fig. 1C.

When the frequency of the audio signal is varied, the frequency of the envelope changes in a corresponding

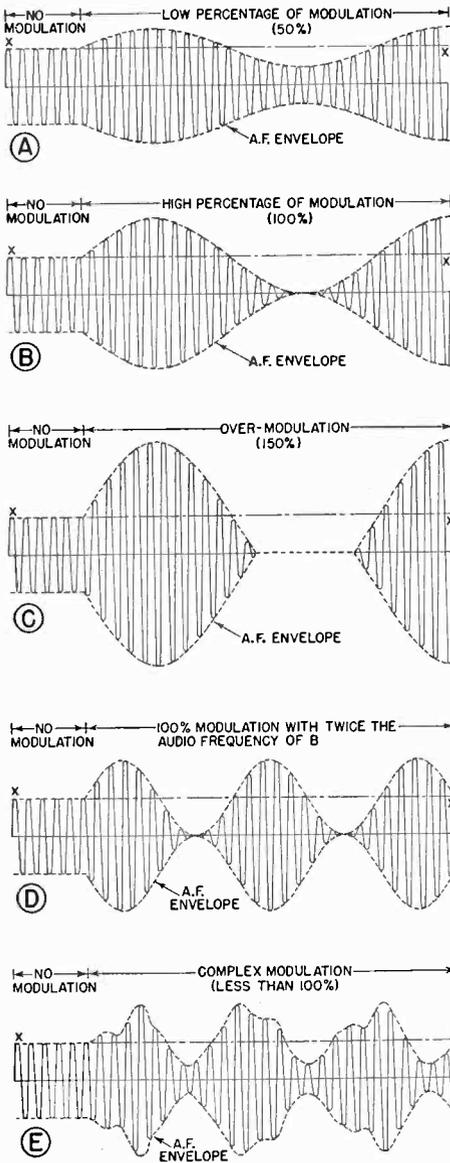


FIG. 1. Examples of amplitude-modulated signals.

manner. Thus, if we double the frequency of the sine-wave audio tone used for our first three examples in Fig. 1, we secure for 100% modulation the condition shown in Fig. 1D.

When voice or music is transmitted, the modulation envelope becomes far more complex, resembling that shown in Fig. 1E. When this envelope is

analyzed, it is found to have many different audio frequency components.

*Side Frequencies.* As you already know, an r.f. carrier signal which is modulated with a single fixed-frequency sine wave tone is equivalent to three different pure r.f. signals. One will have the assigned carrier frequency and constant amplitude. The other two, called *side frequencies*, will be respectively above and below the carrier frequency by the audio frequency value, and each one will vary in amplitude between zero and one-half the carrier amplitude as the percentage modulation varies between 0 and 100%.

Here is an example. If the highest frequency we wish to transmit is 5000 cycles (5 kc.), and the r.f. carrier frequency is 1000 kc., the three pure r.f. signals will be 1000 kc. (the carrier), 995 kc. and 1005 kc. (the side frequencies). If the lowest frequency to be transmitted is 100 cycles (.1 kc.), the two side frequencies going out with the 1000-kc. carrier will be 999.9 kc. and 1000.1 kc.

If a complex audio signal having many frequencies in the range from 100 cycles to 5000 cycles is being transmitted, there will be a 1000-kc. r.f. carrier signal traveling through space along with side frequencies ranging from 995 kc. to 999.9 kc. and from 1000.1 kc. to 1005 kc.

The percentage modulation for a complex signal varies from instant to instant; it is 100% when the sum of all the side frequency amplitudes at a particular instant of time is exactly equal to the carrier amplitude. Over-modulation occurs whenever the sum of all the side frequency amplitudes is greater than the carrier amplitude.

### Characteristics of a Frequency-Modulated Signal

Modulation of an r.f. signal can also be accomplished by varying the

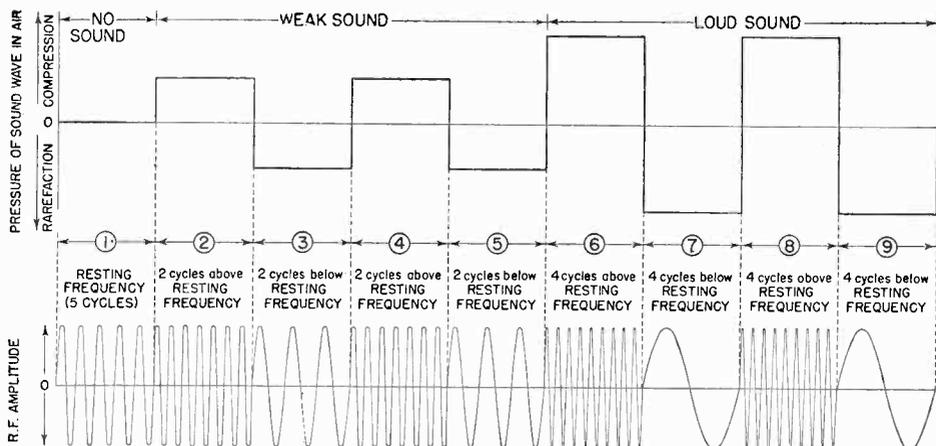


FIG. 2. Example showing how a sound wave having a square-wave form makes the r.f. signal frequency of an f.m. transmitter vary above and below the assigned resting frequency.

frequency of the r.f. signal in accordance with sound pressure, while keeping the r.f. amplitude essentially constant. This, basically, is *frequency modulation*.

Sound is an alternate compression and rarefaction of air particles. This means that air in the path of a sound wave is alternately being increased and decreased in pressure with respect to normal atmospheric pressure. When a sound wave in air is converted into its electrical equivalent by a microphone, we have a corresponding audio signal with positive and negative alternations, one alternation representing compression and the other rarefaction of air. The amplitude of an alternation depends upon the loudness of the original sound in air, and the frequency of the audio signal depends upon the pitch (frequency) of the original sound.

*The Resting Frequency.* With frequency modulation, the r.f. signal which is radiated by the transmitter is at an assigned r.f. value called the *resting frequency* or *center frequency* whenever no sound is being transmitted. The resting frequency thus corresponds to the instant when the air at the microphone diaphragm is at

*normal atmospheric pressure.* This is illustrated graphically by time interval 1 in Fig. 2. (The resting frequency of an actual f.m. transmitter would be some value above 88,000,000 cycles per second, but for illustrative purposes we arbitrarily choose 5 cycles to represent the resting frequency in this diagram.)

Although the compression half of a sound wave cycle in air could cause either an increase or decrease in the frequency of an actual f.m. transmitter (depending upon the number of a.f. stages and phase-reversing transformers which are between the microphone and the output of the modulator), it is a convenient and common practice among radio men to associate compression with an *increase* in f.m. transmitter frequency. We will follow this practice.

*Frequency Deviation.* When an a.f. signal is fed into an f.m. transmitter, the *frequency* of the r.f. signal thus swings *above* the resting value in proportion to increases in air pressure from the normal atmospheric value (compression). Likewise, the transmitter frequency swings *below* the resting value in proportion to decreases in air pressure from the nor-

mal atmospheric value (rarefaction). The amount of this swing above or below the resting frequency (the amount by which the instantaneous r.f. value differs from the resting value) is known as the *frequency deviation*. We illustrate these r.f. signal frequency changes in Fig. 2 for a sound having a square-wave form.

*Square-Wave Example.* Time intervals 2, 3, 4 and 5 in Fig. 2 represent two cycles of a weak sound having a square-wave form. During the first half cycle (interval 2), the air directly in front of the microphone is compressed, and the r.f. carrier frequency for this interval is therefore *higher* than the resting frequency. We have indicated this by showing 7 complete cycles for time interval 2 (two more than for the resting frequency in interval 1).

During the second half of the first sound cycle, we have rarefaction at the microphone, and the r.f. carrier frequency for this interval is *lower* than the resting frequency. We indicate this by showing 3 cycles for time interval 3 (two less than for interval 1).

In the second cycle of the weak sound, the process repeats itself, with the r.f. carrier frequency going above the resting value for time interval 4, and going below the resting value for interval 5.

In an f.m. system, doubling the sound pressure of the original sound doubles the deviation in transmitter frequency from the resting value. Time intervals 6, 7, 8 and 9 in Fig. 2 illustrate this for the square-wave sound under consideration.

During the first half cycle of this louder sound wave (interval 6), we have compression, and the r.f. signal frequency is twice as much above the resting frequency as it was for corresponding interval 2 of the weak sound. We indicate this in Fig. 2 by showing

9 complete cycles for time interval 6 (this is 4 cycles above the resting value).

For the rarefaction half of the loud sound cycle (interval 7), the r.f. signal frequency swings just as much below the resting frequency as it swung above the resting frequency for the compression half cycle. Thus, we show 1 cycle for time interval 7, this being 4 cycles below the 5-cycle value for the resting frequency.

The 4-cycle swing above and below the resting frequency is repeated for intervals 8 and 9 in Fig. 2, to give the second cycle of the loud sound.

*Sine-Wave Example.* The square-wave audio signal in Fig. 2 showed how sudden changes in the amplitude of an audio signal affect the output signal frequency of an f.m. transmitter. Now let us see how gradual changes in amplitude, such as those occurring in the sine-wave audio signal shown in Fig. 3, will affect an f.m. transmitter. This diagram can either represent variations in air pressure from a normal atmospheric value (as in Fig. 2) or positive and negative a.f. signal voltage swings.

First of all, we can say that the r.f. signal will be at its resting value whenever the audio signal passes through zero, such as at points *a*, *e* and *g*.

As the audio signal increases in amplitude from *a* to *b* to *c*, the frequency of the r.f. signal will rise above the resting value in a similar manner, to a maximum r.f. value corresponding to peak amplitude *c*. As the audio signal decreases gradually in amplitude to zero again at point *e*, the r.f. signal frequency will drop gradually back to the resting value in a similar manner. Note particularly that for the entire interval of time from *a* to *e* when the a.f. signal is positive, the transmitter frequency is *above* the resting value.

When the a.f. signal goes through its negative half cycle from *e* to *f* to *g*, the r.f. signal frequency will decrease in a correspondingly gradual manner from its resting value to a minimum value corresponding to point *f*, then rise gradually again to the resting value to complete the sine-wave cycle.

*Complex Voice or Music Signals.*

When a complex audio signal like that shown in Fig. *SB* is transmitted by an f.m. system, the transmitter frequency will vary above and below the resting value in accordance with the amplitude and polarity of the audio signal at each instant, even though this voice or music signal contains many different audio frequencies.

f.m. as they are used today.

Theoretically, the full audio frequency spectrum with a full range of volume can be handled satisfactorily by an f.m. system regardless of how small or how large the maximum deviation value may be. In actual practice, however, the added requirements of a high signal-to-noise ratio and minimum inter-station interference at receivers make necessary a *high* value for the *maximum frequency deviation*. The greater the maximum frequency deviation employed for desired signals, the less noticeable to the listener will be a given frequency deviation due to an interfering signal.

The channels which have been made available to f.m. broadcast stations in

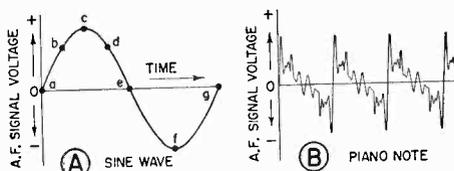


FIG. 3. A pure sine wave signal like that at *A* or a complex audio signal like that at *B* both make the frequency of an f.m. transmitter swing above and below its resting value exactly in proportion to the positive and negative swings of the audio signal.

Thus, we can say that the r.f. signal frequency will be *above* the resting value whenever the audio signal is *positive*, with the deviation from the resting value being proportional to the positive amplitude at each instant. Likewise, the r.f. signal frequency will be *below* the resting value whenever the audio signal is *negative*, with the deviation being proportional to the negative amplitude at each instant.

*Amount of Frequency Deviation.* Keeping in mind the fundamental f.m. fact that the instantaneous deviation in the frequency of the r.f. signal is proportional to the instantaneous amplitude of the audio signal, let us now consider actual frequency values for

the United States and its possessions by the Federal Communications Commission are .2 mc. apart in the very high frequency band between 88 and 108 mc., with the lower portion of the band from 88 to 92 mc. being reserved for non-commercial educational f.m. stations. Each channel assignment represents *the assigned resting frequency* of the station (the unmodulated signal frequency of the station).

A guard band at least 25 kc. wide is required by the F.C.C. beyond each extreme frequency swing of a station, which leaves a maximum of 75 kilocycles for the permissible frequency swing in either direction from the assigned resting frequency.

Percentage modulation, as we know it in connection with the a.m. system of broadcasting, does not exist in an f.m. system. Since the greatest permissible frequency deviation for a loud sound is 75 kc., this deviation corresponds to 100% modulation of an f.m. transmitter. Over-modulation (a deviation greater than 75 kc.) will not cause distortion in receivers, however, unless it makes the signal sweep beyond the linear portion of the S curve for discriminator action (to be considered later in this lesson).

The monitor engineer in an f.m. studio has meters before him which tell when the loudness level at the microphone is exceeding the value which gives the full 75-kc. deviation. Whenever necessary, he reduces the gain of the studio a.f. amplifier enough to prevent the transmitter from swinging more than 75 kc. off from the resting frequency. Likewise, the monitor increases the a.f. gain when the loudness level at the microphone drops way down for appreciable periods of time. Thus, the goal of the monitor engineer is essentially the same, for both f.m. and a.m. systems: To keep the audio level at receivers as high as is practical without making the program sound unnatural and without causing over-modulation at the transmitter.

*Actual Example.* Suppose that an f.m. station is assigned the 97.3-mc. channel, and its microphone is picking up a loud 1000-cycle sound. The resting frequency of this station will be 97.3 megacycles, and the maximum permissible deviation will be 75 kc. (.075 mc.) above and below this resting value. Thus, the r.f. signal frequency will go up to 97.375 mc. for the positive peak of each audio cycle, and will drop down to 97.225 mc. for the negative peak of each audio cycle. Since the audio signal passes through 1000 complete cycles in each second,

the f.m. transmitter will go through 1000 complete swings (from 97.3 to 97.375 to 97.3 to 97.225 to 97.3) in each second in order to follow the positive and negative amplitude variations of the audio signal.

If the lowest loudness level to be transmitted in this example is 1/100 of the loudest level, the deviation for this weakest audio signal will be .075 mc. divided by 100, or .00075 mc. For this weak signal, then, the r.f. signal frequency will vary from 97.30075 mc. to 97.29925 mc. and there will be 1000 complete swings like this in each second.

If the f.m. station in our example is transmitting a loud 100-cycle audio signal, the deviation will still be 75 kc. on each side of the resting frequency, but now there will be only 100 complete swings back and forth between the two extreme frequencies in each second.

*F.M. Side Frequencies.* Mathematical computations as well as actual measurements tell us that the continually varying r.f. signal frequency for an f.m. transmitter is equivalent to a carrier frequency and an infinitely large number of side frequencies. Fortunately, however, the essential side frequencies needed for high-fidelity reproduction are within the 150-kc. deviation range over which the r.f. signal sweeps. Receivers are designed for reception of this range, and consequently they receive the essential side frequencies.

Side frequencies more than 75 kc. off from the resting frequency might create interference with adjacent-channel stations if these stations were in the same locality, but the Federal Communications Commission prevents interference of this nature by keeping channel assignments at least 400 kc. (.4 mc.) apart for f.m. stations in the same service area.

### Technical Data on Side Frequencies.

When the carrier of an f.m. transmitter is modulated with an audio signal having a constant frequency of  $f$  cycles per second, the r.f. signal swinging above and below the resting frequency is equal to the following fixed-frequency signals: A signal having the resting frequency of the transmitter; two side frequencies, one above and one below the resting frequency by the audio value  $f$ ; two side frequencies, above and below the resting frequency by twice the audio value  $f$ ; two side frequencies, above and below the resting frequency by three times the audio value  $f$ ; etc.

Here is an example. If the resting frequency of an f.m. transmitter is 103.5 mc. and the audio modulation frequency is 10,000 cycles (.01 mc.), the equivalent individual frequencies will be: 103.5 mc., 103.49 mc. and 103.51 mc., 103.48 mc. and 103.52 mc., 103.47 mc. and 103.53 mc., etc., down to zero and up to infinity.

In your work with f.m. apparatus, however, you will find it more convenient to think of an f.m. signal as a single r.f. signal which is continually varying in frequency above and below its resting frequency. (We do this in amplitude modulation when we consider a modulated r.f. signal as a single r.f. signal which is varying in amplitude from instant to instant as in Fig. 1A.)

**Multiplex F.M. Operation.** Frequency modulation has the added advantage of permitting multiplexing, the sending of two or more programs or types of intelligence by the same transmitter, without increasing the required band width for the station and without interference between the programs being transmitted.

Here is an example: To transmit code messages along with a broadcast program, a 20-kc. oscillator signal (above the audio range) could be made to vary in amplitude in accordance with the code modulation, and this modulated 20-kc. secondary carrier then fed into the transmitter along with the regular voice or music program. At the receiver, the voice or music program would pass through the stages in the conventional manner, and the 20-kc. modulated code signal would be taken out ahead of

the audio amplifier, and demodulated by conventional means in a separate detector stage. Filters would be used to keep the 20-kc. signal out of the regular a.f. amplifier, to prevent overloading of a.f. stages.

**How F.M. Systems Reduce Interference.** At any instant of time, we can consider that an f.m. signal has a definite frequency, above or below the resting frequency of the transmitter. We can therefore represent a desired f.m. signal at the input of an f.m. receiver by a vector, as shown in Fig. 4A. The amplitude (length) of this desired-signal vector  $D$  will be constant, and the rate at which the

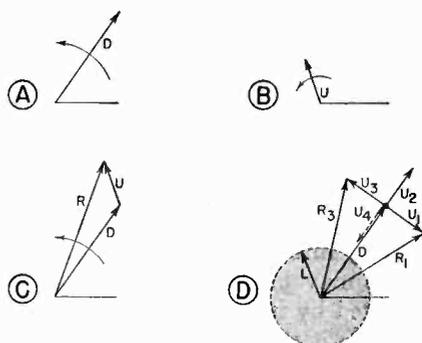


FIG. 4. Vector diagrams showing how an undesired signal  $U$  can affect the frequency and amplitude of a desired f.m. signal  $D$ .

vector rotates counter-clockwise will correspond to the frequency of the r.f. signal at the instant of time under consideration. The position of the vector with respect to the horizontal reference line in the diagram is unimportant.

Now, let us suppose that we also have at the input of our f.m. receiver an undesired signal. It may be due either to noise interference or to an undesired r.f. carrier signal from some other station, but nevertheless we know that this undesired signal has both frequency and amplitude. Let us say that at the instant of time under consideration for Fig. 4A, the

undesired signal has the amplitude and phase represented by vector  $U$  in Fig. 4B.

When we combine undesired-signal vector  $U$  with desired-signal vector  $D$ , as shown in Fig. 4C, we obtain resultant vector  $R$ . The amplitude of vector  $R$  is greater than that of desired-signal  $D$ , showing that an interfering signal can affect the amplitude of a desired incoming signal. Furthermore, vector  $R$  is ahead of vector  $D$ , indicating that the sudden arrival of this interfering signal makes desired-signal vector  $D$  increase its rotational speed suddenly; this means that during the time interval in which vector  $D$  is moving to its new position corresponding to vector  $R$ , the frequency of the desired signal is higher than it should be.

Interference vector  $U$  in Fig. 4B may have any phase relationship whatsoever throughout the entire range of  $360^\circ$ . It may be in phase with desired-signal vector  $D$ , as indicated by interference vector  $U_2$  in Fig. 4D; it may be  $180^\circ$  out of phase, as indicated by interference vector  $U_4$  in Fig. 4D; it may lead the desired vector by  $90^\circ$ , as indicated by  $U_3$ , or it may lag the desired vector by  $90^\circ$ , as indicated by  $U_1$ . These are the four extremes which could exist, and an analysis of them will take care of intermediate undesired-signal vector positions as well.

Careful study of the vector diagram in Fig. 4D reveals that when the interfering signal is *in phase* with the desired signal, it *increases* the amplitude of the resultant signal without changing its frequency. When the interfering signal is  *$180^\circ$  out of phase* with the desired signal, it *decreases* the amplitude of the resultant signal without changing its frequency. When the interfering signal leads the desired signal, the resultant will be speeded up in frequency, and the amplitude

will generally also be altered. Finally, when the interfering signal lags the desired signal, the resultant will be slowed down in frequency and will likewise be altered in amplitude. We thus come to the conclusion that an interfering signal in an f.m. system can change both the *frequency* and the *amplitude* of the desired f.m. signal.

You will learn later in this lesson that the limiter stage of an f.m. receiver reduces all incoming signals to a constant low amplitude corresponding to radius  $L$  in Fig. 4D. Since changes in amplitude due to noise vectors are all outside of this acceptance area, the limiter stage of a properly

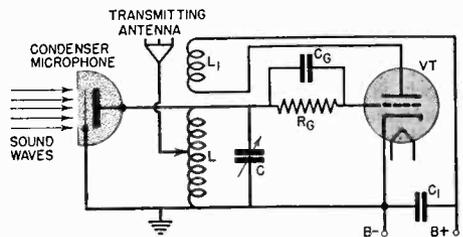


FIG. 5. Simple f.m. transmitter circuit, employing a condenser microphone shunted across the tank circuit of a conventional tuned-grid r.f. oscillator.

designed and adjusted f.m. receiver wipes out amplitude changes due to noise.

Any change in frequency due to noise will show up in the output of an f.m. receiver, for changes in frequency are not affected by the limiter section of the receiver, and are converted into corresponding changes in amplitude by the following frequency discriminator section. By using a high maximum deviation value, however, changes in the frequency of the desired signal due to an interfering signal in the receiver can be made negligibly small with respect to frequency changes due to the desired audio program.

Field tests have proved that the ratio of desired to undesired signals increases as the maximum frequency

deviation of the system is increased. Tests have also proved that with a maximum deviation of about 75 kc. (a total swing of 150 kc.), interfering signals become quite imperceptible at the loudspeaker when the desired signal is at least twice as strong as the undesired signal.

### An Elementary F.M. System

*The F.M. Transmitter.* In beginning our study of actual f.m. systems, let us first consider one of the simplest possible transmitter circuits capable of producing frequency-modulated signals.

A conventional vacuum tube oscillator circuit employing a coil-condenser resonant circuit can be tuned to frequencies in either direction from a reference frequency without appreciably changing the amplitude of the output signal. By taking a simple oscillator circuit like this and shunting its resonant circuit with a condenser microphone in the manner shown in Fig. 5, a simple but effective f.m. transmitter is obtained. Sound waves apply to the movable diaphragm of the microphone a varying pressure which makes this diaphragm move alternately towards and away from the fixed plate. This changes the capacity of the condenser microphone in accordance with variations in sound pressure, thereby alternately raising and lowering the frequency of the r.f. oscillator to give the desired frequency-modulated signal.

There are a number of drawbacks to this simplified f.m. transmitter circuit. The condenser microphone would have to be close to the oscillator, to prevent pick-up of stray signals by the microphone leads. Each condenser microphone would have to be designed to give a definite frequency deviation with its particular oscillator, and even with this precaution the amount of deviation would vary with

the particular performer using the microphone. Undoubtedly these problems could be solved by engineers if necessary, but fortunately there are more practical means for securing frequency modulation of a carrier signal. One such scheme will now be considered.

*Inductance-Tube Type of F.M. Transmitter.* A vacuum tube circuit can be made to act like an inductance, simply by adjusting circuit voltages so that the r.f. current drawn by the tube will lag the r.f. voltage applied to the tube. (This scheme is widely employed in the automatic frequency

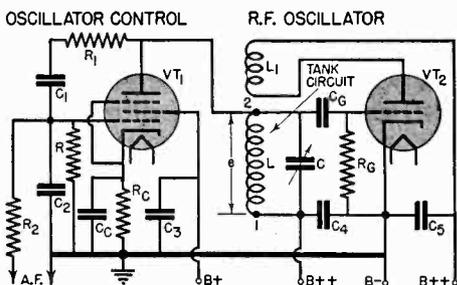


FIG. 6. This practical f.m. transmitter circuit employs an oscillator control circuit consisting of a vacuum tube which acts like an inductance shunted across the r.f. tank circuit. The effective inductance value varies from instant to instant in accordance with variations in the amplitude of the a.f. signal, thereby giving frequency modulation.

control circuits of some radio receivers.) By connecting across the coil-condenser resonant circuit of an r.f. oscillator the vacuum tube circuit which is acting like an inductance, and by making the inductance of this vacuum tube circuit vary at a desired audio rate, we secure frequency modulation.

An inductance-tube type of f.m. transmitter circuit is given in Fig. 6. Triode vacuum tube  $VT_2$  is connected into a conventional tuned-grid r.f. oscillator circuit, with  $L$  and  $C$  forming its tank circuit. Pentode tube  $VT_1$  serves as the oscillator control tube which acts like an inductance; its plate is connected directly to termi-

nal  $\mathcal{L}$  of tank inductance  $L$ , and its cathode is connected to terminal 1 of this coil through r.f. by-pass condensers  $C_3$  and  $C_4$  and the grounded chassis.

Now let us see how oscillator control tube  $VT_1$  can act as an inductance in shunt with tank circuit  $L-C$ . First of all, r.f. tank voltage  $e$  in Fig. 6 must be considered as the r.f. voltage source acting upon the oscillator control circuit. The two r.f. signal paths connected in parallel across r.f. voltage source  $e$  are the plate-cathode path of oscillator control tube  $VT_1$  and path  $R_1-C_1-C_2-C_4$ .

At radio frequencies, path  $R_1-C_1-C_2-C_4$  is essentially resistive (the reactances of all three condensers are low with respect to the resistance of  $R_1$ ), and hence the r.f. current flowing over this path is *in phase with its r.f. source voltage  $e$* . This r.f. current develops across condenser  $C_2$  an r.f. voltage which *lags* the r.f. current, and hence lags r.f. voltage  $e$ . (The a.c. voltage across a condenser always lags the condenser current.)

The r.f. voltage across  $C_2$  acts on the control grid of  $VT_1$ , making the tube pass an r.f. plate current which is *in phase with the applied r.f. grid voltage*. The r.f. plate current drawn from  $L$  by the oscillator control tube *therefore lags the r.f. voltage across  $L$* . This is exactly the same phase relationship as for an inductance load across  $L$ ; the oscillator control tube thus acts like an additional inductance shunting the tank circuit, and serves to *increase* the frequency of the r.f. oscillator.\*

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\* When two inductances are in parallel, the combined inductance is less than that of the smaller inductance. Lowering the total tank circuit inductance raises the oscillator frequency. If one of the parallel inductances is reduced in value, the oscillator frequency will increase; if one of the parallel inductances is increased in value, the oscillator frequency will decrease.

The a.f. modulation voltage which is applied to the control grid of  $VT_1$  through  $R_2$  varies the transconductance of the tube, and hence makes the a.c. plate current vary at an audio rate. Consequently, *the inductance of this tube also varies at an audio rate*. This in turn makes the frequency of the r.f. oscillator swing above and below its resting value at an a.f. rate, giving frequency modulation of the r.f. carrier without appreciable variation in the r.f. amplitude. When the a.f. signal is removed or drops to zero, the r.f. oscillator returns to its resting frequency, which is determined by the size of inductance  $L$ , the normal inductance of the oscillator control circuit, and the tank circuit capacity.

Resistor  $R_2$  prevents the a.f. source from shorting the input of the inductance tube; it is really an isolating resistor.

By adjusting the initial C bias on the oscillator control tube (varying the ohmic value of  $R_C$ ) and by monitoring properly the a.f. voltage fed into the oscillator control tube, the maximum deviation in frequency can be made any desired amount. The circuit of Fig. 6 is therefore a suitable signal source for an f.m. system which is to be employed in transmitting intelligence. The f.m. signal would be taken either from terminal 2 or from a link which is coupled to tank circuit inductance  $L$ .

*The F.M. Receiver.* Up to the point at which frequency-modulated signals are converted into amplitude-modulated signals, an f.m. receiver uses essentially the same circuits as a corresponding a.m. receiver. The a.f. amplifier and loudspeaker are likewise essentially the same for both systems. The important new action in an f.m. receiver is the conversion of the f.m. signal into the desired audio signal, so let us consider now the basic

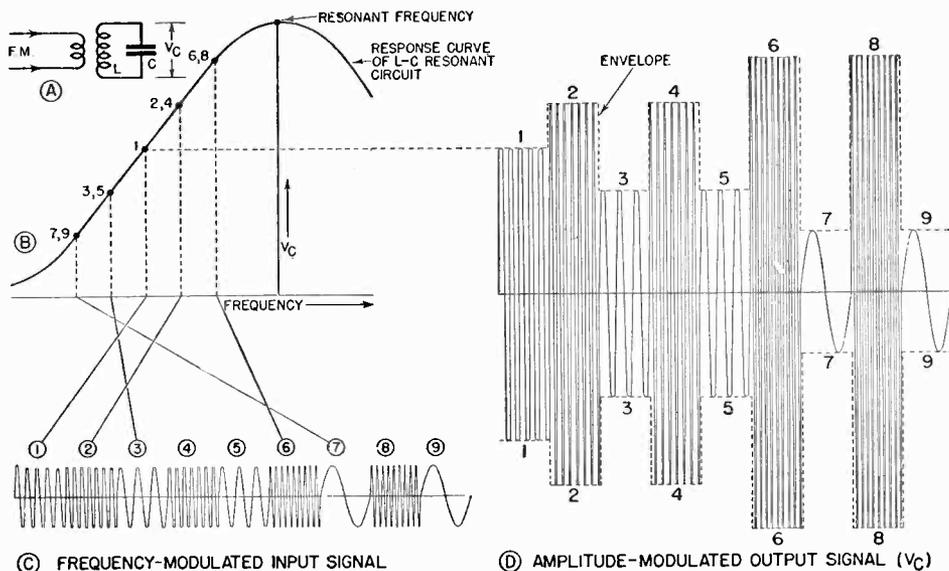


FIG. 7. These diagrams illustrate how a simple resonant circuit can convert an f.m. signal into an a.m. signal.

principles involved in this conversion.

If a frequency-modulated r.f. signal is introduced into an  $L-C$  resonant circuit which is tuned slightly above the highest deviation frequency, the r.f. voltage developed across the resonant circuit will vary with the frequency of the induced f.m. signal. An ordinary resonant circuit like the  $L-C$  circuit in Fig. 7A is thus a simple means for converting an f.m. signal into an a.m. signal.

A portion of the resonant response curve for this  $L-C$  circuit is shown in Fig. 7B. By applying to this curve in the proper manner the f.m. signal shown at the bottom of Fig. 7C (having a square-wave audio modulation), we can get a graphical picture of how the amplitude of r.f. output voltage  $V_C$  varies with the frequency of the incoming f.m. signal.

Let us assume that when the f.m. transmitter is at its resting frequency (time interval 1 in Figs. 7C and 7D), the operating point is at 1 in Fig. 7B. The vertical distance from 1 down to the horizontal reference line then de-

termines the amplitude of r.f. voltage  $V_C$  across the resonant circuit, so we show the r.f. output voltage for time interval 1 as an r.f. signal having this same amplitude and the same resting frequency value, as at 1 in Fig. 7D.

When the frequency of the f.m. signal increases to the value for time interval 2 in Fig. 7C, we move up to point 2 on the response curve, and thereby secure the output signal at 2 in Fig. 7D. Likewise, the input signal frequencies for time intervals 3, 4, 5, 6, 7, 8 and 9 in Fig. 7C give the output signal amplitudes shown at 3, 4, 5, 6, 7, 8 and 9 respectively in Fig. 7D.

Note that the signal frequency is unchanged by the resonant circuit, but the amplitude of the output signal varies in proportion to the frequency deviations of the input f.m. signal. A dash-dash line drawn through the positive peaks of the r.f. output voltage in Fig. 7D gives the original square-wave audio modulation of Fig. 7C, showing that we now have an amplitude-modulated r.f. signal which can be demodulated with a conventional am-

plitude detector circuit. The fact that our output signal also varies in frequency does not matter, for the amplitude detector circuit removes all r.f. components.

A complete circuit capable of converting an f.m. signal into an audio signal is shown in Fig. 8; this circuit can appropriately be called an *f.m. detector*. The final i.f. amplifier stage is also shown, and employs a 6SK7 pentode, with a double-tuned i.f. transformer ( $C_3-L_3-L_2-C_2$ ) in its input circuit. Resistor  $R_L$  across the primary of this i.f. transformer provides the loading required to flatten the response over the 150-kc. range through which

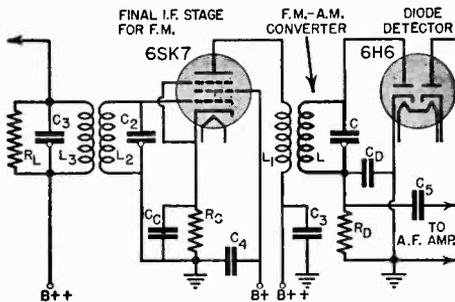


FIG. 8. This diagram shows how a resonant circuit can be used to convert the f.m. output signal of the i.f. amplifier into an a.m. signal which can be demodulated by a conventional diode detector.

the f.m. signal is varying during loud audio loudness levels.

Plate current of the final i.f. tube flows through coil  $L_1$ , inducing an f.m. signal voltage in coil  $L$  of the  $L-C$  resonant circuit in which frequency-to-amplitude conversion takes place. The r.f. voltage across this  $L-C$  circuit therefore varies in amplitude in accordance with variations in the original audio signal amplitude. This amplitude-modulated r.f. voltage is demodulated by one section of the 6H6 tube, and the resulting a.f. output voltage appears across diode load resistor  $R_D$ . Condenser  $C_D$  removes r.f. variations across  $R_D$ , so that only the desired a.f. voltage is fed into the a.f. amplifier.

## Additional Requirements in an F.M. System

The f.m. transmitter and receiver arrangements just described represent the absolute minimum required in an f.m. system. Transmitters must have some type of crystal control, so as to maintain the assigned resting frequency value within the limits prescribed by law, and receivers must also have additional features which insure high fidelity and operating stability, reduce noise and simplify tuning.

The remainder of this lesson will be devoted chiefly to practical f.m. receiver circuits, but basic principles of highly stable modern f.m. transmitters will be covered briefly at the end of this lesson text to make your f.m. training more complete.

Resonant circuits have proved unsatisfactory for f.m.-a.m. separating purposes, chiefly because the slopes of their response curves are not sufficiently linear for high-fidelity purposes. Instead, we find in modern f.m. receivers a variation of the discriminator circuit used in a.f.c. circuits. This discriminator circuit is simple to adjust and gives linear operation. It uses a push-pull circuit to convert frequency variation to amplitude variations at the i.f. frequency and then detects this signal.

In a practical f.m. receiver, an additional section called the limiter is required just ahead of the discriminator to restore constant amplitude, so that all amplitude rises due to noise or station interference will be removed.

The limiter is a vacuum tube amplifier which is so designed that all desired signals will cause plate current saturation. This action of the limiter stage also levels out the response of the r.f. system.

Since the limiter in an f.m. receiver delivers the same amplitude for both

weak and strong incoming signals, the loudspeaker volume level does not change when tuning from one f.m. station to another. This means that it is unnecessary to use an a.v.c. system in an f.m. receiver to prevent blasting or counteract fading. Most f.m. receivers do have a.v.c., but this is primarily to prevent extremely strong input signals from driving the grids of r.f. or i.f. amplifier tubes positive and thereby causing blocking or interference. If grid current were allowed to flow during positive grid voltage values, the

resulting distortion of the incoming f.m. signal would produce harmonics which could beat with oscillator harmonics and other station signals to produce interfering signals at the i.f. value for f.m.

Another requirement in a modern f.m. receiver is a tuning indicator (either a meter or a cathode ray tuning indicator tube). F.M. resonant circuits are so broad due to loading that the average person would have difficulty in tuning in an f.m. station by ear alone.

## The F.M. Receiver

*Types of F.M. Receivers.* Radio receivers which are capable of receiving f.m. signals can be divided into three groups, as follows:

1. Sets designed exclusively for f.m. reception. These can be table models or consoles, for listeners who already have good a.m. receivers.

2. Combination f.m.-a.m. receivers, providing all-wave a.m. reception along with f.m. reception. These will usually be large console sets and may also have automatic record changers, television, facsimile or sound-recording features.

3. F.M. converters, which are simply f.m. receivers without audio amplifiers or loudspeakers. They provide f.m. reception when connected

to the a.f. input terminals of an ordinary a.m. receiver.

All f.m. receivers should have high sensitivity in order to operate the limiter at saturation for the widest frequency swings of the weakest signal to be received, and hence superheterodyne circuits are invariably employed.

Typical circuits used in combination f.m. and a.m. receivers will now be analyzed.

### Combination F.M.-A.M. Receiver

A block diagram of a typical combination a.m. and f.m. superheterodyne receiver is given in Fig. 9. The circuits now to be considered will be

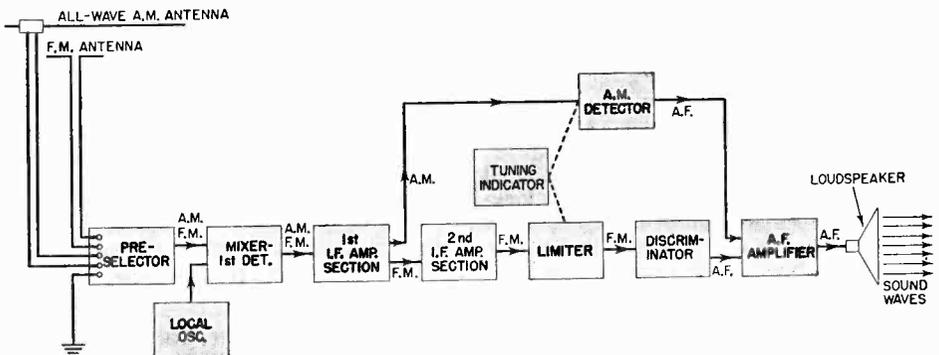


FIG. 9. Block diagram of a typical combination f.m. and a.m. receiver.

those employed in this particular receiver arrangement.

Note first of all that there are two antennas, one for regular all-wave a.m. signal pick-up, and the other for ultra-high-frequency f.m. signal pick-up. The band-selecting switch automatically connects the correct antenna to the receiver. (Fairly satisfactory results can be obtained with a single combination antenna on some sets, as you will later learn.)

For amplitude-modulated signals, the local oscillator will operate about 455 kc. above the incoming a.m. signal frequency, while for frequency-modulated signals, it will operate at 4.3 mc. below the incoming f.m. signal frequency.\* The i.f. signal produced by the mixer-first detector (about 455 kc. for a.m., or 4.3 mc. for f.m.) passes through the first i.f. amplifier section to a switching circuit.

For f.m. reception, the signals are fed into a second i.f. amplifier section, which raises the signal level the required amount for the limiter stage. The constant-amplitude output of the limiter is fed into the discriminator, and the resulting a.f. output signal is fed directly into the input of the a.f. amplifier.

For a.m. reception, the output of the first i.f. amplifier section is fed directly into a conventional a.m. detector, and its a.f. output is likewise fed into the input of the a.f. amplifier. The tuning indicator, if used, would

\*Although 10.7 mc. is the recommended and widely used i.f. value for f.m. receivers at the present time, you may also encounter other values. Some f.m. receivers are capable of operating not only on the 88-108-mc. f.m. band, but on the older 42-50-mc. band as well. This low-frequency band is no longer in use. These sets particularly may have a different value of i.f., generally lower in frequency. The higher value now in use minimizes chances for interference between the various signals which are picked up by the receiving antenna or produced in the receiver by superheterodyne action.

be connected through the band-changing switch to the a.v.c. source for either the f.m. or a.m. channel. A common power pack serves all sections.

### F.M. Receiving Antennas

The antenna for f.m. reception will ordinarily be a doublet of the correct length to give half-wave operation at an average frequency in the f.m. band. Since commercial f.m. stations are assigned to frequencies between 88 and 108 mc., a doublet which resonates at about 98 mc. will ordinarily be used. With proper design, its response will normally be broad enough to give satisfactory coverage of the entire f.m. band.

A frequency of 98 megacycles corresponds to a wavelength of 3.06 meters (wavelength in meters is equal to 300 divided by frequency in megacycles). A half-wave f.m. doublet antenna will therefore have a total length of about 1.5 meters; multiplying this value by 3.28 gives about 5 feet as the total length for an average f.m. doublet receiving antenna.

The simplest possible f.m. doublet is that shown in Fig. 10A, in which the transmission line is connected to the center of the 5-foot long doublet. Since a half-wave doublet has a resistance of about 72 ohms at its center, a transmission cable having a surge impedance of approximately this same value would ordinarily be used. Slight mismatches can be tolerated, however; in fact, a cable with a surge impedance of about 100 ohms will make the antenna response characteristic more uniform over the entire f.m. band. The transmission cable is connected to the receiver through a matching transformer.

At very-high frequencies, line-of-sight paths give the best transmission of radio waves, but, fortunately, reliable f.m. reception is possible con-

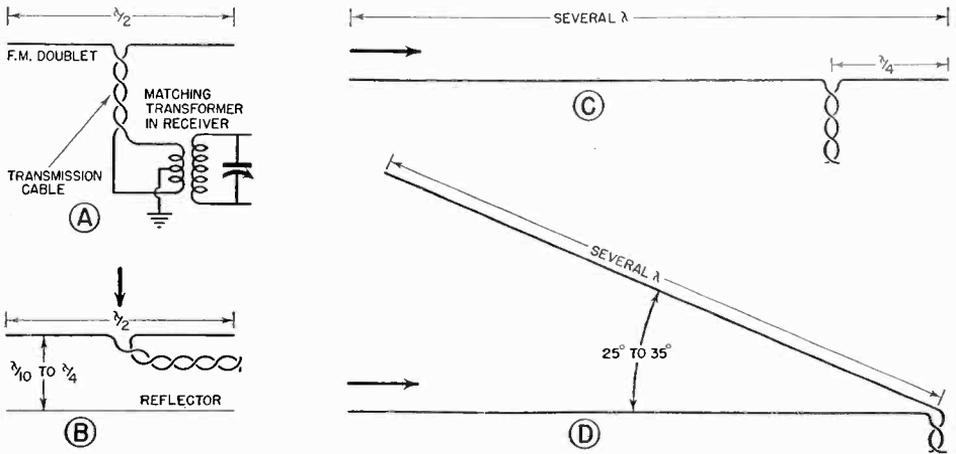


FIG. 10. Types of f.m. receiving antennas. The heavy arrow in each case indicates the direction (along the surface of the earth) from which signals are best received. The symbol  $\lambda$  represents "wavelength."

siderably beyond the line-of-sight range from a transmitter. (With a line-of-sight path, it should be possible to see the transmitting antenna with a telescope on a clear day from the position of the receiving antenna.)

A compact very-high frequency antenna built into the console cabinet of an f.m. receiver will often give adequate signal pick-up of stations up to about ten miles away. Even a quarter-wave vertical antenna can sometimes be used when signals are strong.

In locations considerably beyond the line-of-sight limit, the position of the receiving antenna becomes important. The half-wave horizontal f.m. doublet should then be as high as possible, and broadside (at right angles) to the path over which f.m. signals are arriving.

In remote locations where all of the desired f.m. signals are coming from essentially the same direction, such as from a single large city, a reflector like that shown in Fig. 10B can be used with a conventional doublet to increase the signal pick-up from the desired direction. The doublet is always placed between the reflector and the transmitter.

The antenna arrangements shown

in Figs. 10C and 10D give even better signal pick-up at remote locations. In Fig. 10C, one section of the doublet has the usual length of  $\lambda/4$  (one-quarter wavelength), while the other section is several wavelengths long. The long section is aimed directly at or slightly above the transmitter.

In the V doublet arrangement of Fig. 10D, both horizontal sections are equal, several wavelengths long, and aimed in the general direction of the transmitter.

Before mounting an f.m. doublet permanently, it is always a good idea to move the antenna toward or away from the transmitter a distance of about 5 feet, and tilt it at various angles while an assistant is checking signal strength at the receiver by watching the tuning indicator or using a meter, to find the position which gives maximum signal strength. Cancellation due to reflected waves arriving over different paths will then be a minimum.

### The Preselector

A preselector having at least one r.f. amplifier stage is just as desirable in an f.m. superheterodyne receiver as in an a.m. set, for r.f. amplification

ahead of the mixer-first detector in an f.m. receiver provides much needed extra sensitivity and minimizes converter noise. Just as with a.m. receivers, the higher-priced f.m. receivers will usually have an r.f. stage in the preselector. Less expensive f.m. sets will have a simple antenna transformer with tuned secondary, feeding directly into the mixer-first detector.

The circuit of a typical r.f. amplifier stage for a combination f.m.-a.m. receiver is shown in Fig. 11A. To simplify the diagram, only one all-wave a.m. range is shown; additional contacts on the three sections of the band-changing switch and additional input and output r.f. transformers would be provided for other a.m. ranges.

When the three switches ( $S_1$ ,  $S_2$  and  $S_3$ ) are set to position 1, as shown in Fig. 11A, the entire preselector is connected for f.m. reception. The half-wave f.m. doublet feeds its f.m. signal into the grid circuit of the 6SK7 r.f. amplifier tube through f.m. antenna matching transformer  $L_1$ - $L_2$ , with secondary winding  $L_2$  being tuned to the incoming f.m. signal frequency by gang condenser section  $C_1$ . The amplified r.f. plate current of the 6SK7 tube flows through winding  $L_3$  of the f.m. input transformer for the mixer, inducing in secondary winding  $L_4$  a corresponding r.f. voltage. Gang condenser section  $C_2$  tunes  $L_4$  to resonance at the desired station frequency, and the resulting f.m. signal voltage across this resonant circuit is applied to the grid and cathode of the mixer-first detector tube. Automatic C bias for the 6SK7 tube is provided by  $R_2$  and  $C_6$ .

Setting the three switches in the preselector circuit of Fig. 11A to position 2 automatically connects the circuit for a.m. reception on one band. Switch  $S_1$  disconnects the secondary

of the f.m. matching transformer from the r.f. amplifier tube grid, and connects the grid instead to the secondary of the a.m. antenna matching transformer. The all-wave a.m. doublet feeds its a.m. signal to the r.f. amplifier tube through a.m. matching transformer  $L_5$ - $L_6$ , with secondary winding  $L_6$  being tuned to resonance by gang condenser sections  $C_3$  and  $C_1$  in parallel. (The conventional condenser section for a.m. is split into two sections,  $C_3$  and  $C_1$ , so that one low-capacity section will be available for f.m. to give a favorable L-to-C ratio.) The r.f. output of the 6SK7 tube is fed to the mixer through a.m. input transformer  $L_7$ - $L_8$ , with secondary winding  $L_8$  being tuned by  $C_4$  and  $C_2$  in parallel.

When the switches are at position 2 for a.m. reception, a.v.c. voltage is applied to the r.f. and mixer tubes through  $L_6$  and  $L_8$  respectively. No a.v.c. is used on these two tubes during f.m. reception in this circuit, but engineers are now recommending that a.v.c. be provided for the r.f. amplifier tube of an f.m. receiver to prevent overloading of any tubes by extremely strong f.m. signals. The a.v.c. voltage for f.m. would be supplied by the *limiter* section.

A variation of the preselector circuit of Fig. 11A which makes possible the use of one doublet antenna for both f.m. and a.m. reception is shown in Fig. 11B. The remainder of the preselector circuit is not shown, since it is identical with that in Fig. 11A. The connections are such that doublet antenna action is secured for f.m. reception, but the antenna is automatically converted into a plain vertical antenna for a.m. reception. Resonant circuit L-C is tuned to the mid-frequency of the f.m. band (about 98 mc.); since it is a series resonant circuit, it has a low impedance at or near resonance, and has the

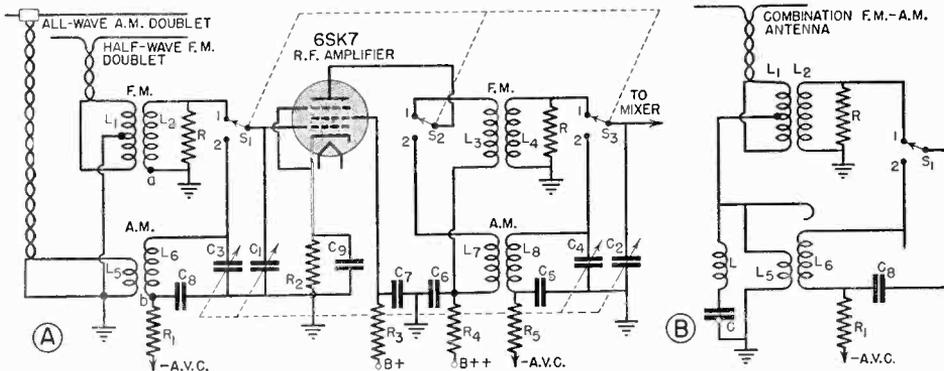


FIG. 11. Typical preselector circuit arrangements for a combination f.m.-a.m. receiver. If the r.f. amplifier tube is to have a.v.c., terminal *a* of f.m. coil  $L_2$  is connected to terminal *b* of a.m. coil  $L_6$  instead of to ground. A switch at the two a.v.c. sources (shown in Fig. 20) insures that the correct a.v.c. voltage (for f.m. or for a.m.) will be fed to this tube.

effect of grounding the center tap of winding  $L_1$ . Under this condition, the antenna acts as a very-high-frequency doublet for the f.m. band.

For a.m. bands, usually below 20 megacycles, the series  $L$ - $C$  circuit is considerably off resonance, and hence has a high capacitive reactance. Winding  $L_1$  is therefore ungrounded, and signal currents picked up by the two sections of the doublet flow through the two sections of  $L_1$  to the center tap, then through winding  $L_5$  to ground ( $L_5$  has much lower reactance than the  $L$ - $C$  circuit). The a.m. signal current through  $L_5$  induces a corresponding r.f. voltage in  $L_6$ , and this is fed through the r.f. amplifier tube to the mixer in exactly the same manner as for the circuit of Fig. 11A.

Observe that a resistor ( $R$ ) is shunted across each f.m. resonant circuit in Figs. 11A and 11B. These resistors serve to load the resonant circuit, providing broad resonance which gives essentially uniform response at all frequencies within the deviation range of an incoming f.m. signal. Ordinarily, resistor  $R$  has an ohmic value somewhere between 10,000 and 20,000 ohms, with the higher value being preferred in order that the preselector will have good selectivity. When no r.f. amplifier stage is

used, a band-passed preselector circuit ahead of the frequency converter is highly desirable, to broaden the response and at the same time improve selectivity.

### The Frequency Converter

In any superheterodyne receiver, the local oscillator and the mixer-first detector constitute a section commonly called the *frequency converter*. The local oscillator feeds into the mixer-first detector an unmodulated r.f. signal which beats with the incoming modulated signal to produce the correct i.f. signal.

A number of different circuit arrangements can be employed in the frequency converter of a combination f.m.-a.m. receiver. Some designers will use a pentagrid converter tube, some will use a combination triode-pentode tube, and some will use separate tubes for the oscillator and mixer-first detector.

It is customary to operate the local oscillator *above* the incoming signal frequency for f.m. reception, to secure greater stability and prevent local oscillator signals from creating interference in the television band below 88 mc. On a.m. bands also the oscillator is usually operated *above* the incoming signal frequency.

Switching sections must be provided on the band-changing switch of a combination f.m.-a.m. receiver to insert the correct inductance and variable capacity in the oscillator tuning circuit of the frequency converter. The oscillator would thus be tuned 10.7 mc. (or whatever other i.f. value is used for f.m.) *above* the incoming signal frequency for f.m. reception, and about 455 kc. *above* the incoming signal frequency for a.m. reception.

An example of a conventional frequency converter circuit employing a type 6K8 combination triode-pentode tube is given in Fig. 12. This circuit could be connected directly to the

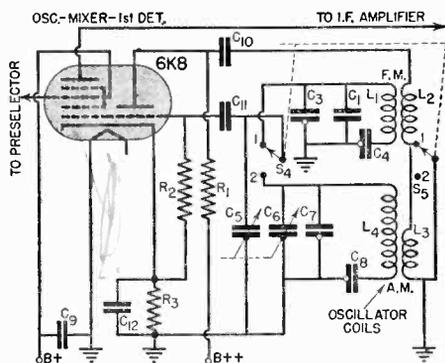


FIG. 12. Typical frequency converter circuit for a combination f.m.-a.m. receiver.

preselector circuit of Fig. 11, with all tuning condenser sections being controlled by one tuning knob and all switches being controlled by the band-changing knob.

When switches  $S_4$  and  $S_5$  in Fig. 12 are at position 1 for f.m. reception, oscillator tank circuit inductance  $L_1$  is tuned 10.7 mc. above the incoming f.m. signal frequency by tuning condenser section  $C_5$ . The high-frequency trimmer is  $C_3$ , and the low-frequency pad is  $C_4$ , while  $C_1$  serves as a temperature-compensating condenser. (Condenser  $C_1$  changes its capacity in a definite manner with temperature, thereby minimizing oscillator frequency drift by com-

pensating for other capacity changes which occur as the set comes up to normal operating temperature after being turned on.)

Feed-back from the oscillator plate circuit is provided by inductance  $L_2$ . Automatic C bias for the oscillator is provided by the flow of grid current through grid resistor  $R_2$  and grid condenser  $C_{11}$ . D.C. blocking condenser  $C_{10}$  prevents the plate supply voltage of the oscillator triode from sending direct current through oscillator feedback coil  $L_2$  to ground.

All leads in the f.m. oscillator section of the circuit should be as short as possible, so that resonance effects will be localized to the desired  $L_1$ - $C_5$  tuning circuit. All grounded elements in a section should be connected to a common point on the chassis. At very-high frequencies, long leads have appreciable inductance and capacity to the chassis, and hence might form additional resonant circuits which create trouble.

When switches  $S_4$  and  $S_5$  in Fig. 12 are at position 2 for reception on one of the a.m. bands, the oscillator tank circuit inductance is  $L_4$ . The tank circuit is tuned 455 kc. above the incoming a.m. signal frequency by tuning condenser sections  $C_5$  and  $C_6$  in parallel.  $C_7$  serves as the high-frequency trimmer, and  $C_8$  serves as the low-frequency padder. Feed-back energy from the plate circuit is now supplied by  $L_3$ , for removal of the short across  $L_3$  by  $S_5$  allows r.f. plate current to flow through both  $L_2$  and  $L_3$  to ground.

### The I.F. Amplifier

We cannot use a standard 455-kc. i.f. value for both f.m. and a.m. reception. The gain of a 455-kc. i.f. amplifier drops entirely too much when it is band-passed sufficiently to pass a 150-kc. wide f.m. signal. Fur-

thermore, the selectivity of this band-passed 455-kc. i.f. amplifier would be way too poor for standard a.m. reception. A 10.7-mc. i.f. amplifier is satisfactory for f.m., but its selectivity is likewise too poor for a.m. reception.

To meet these conflicting requirements in a combination f.m.-a.m. receiver, separate i.f. transformers can be used for f.m. and a.m., so that a.m. signals will pass through standard 455-kc. i.f. transformers, and f.m. signals will be served by 10.7-mc. i.f. transformers. With the exception of the first i.f. transformer primary, the connections can be such that signals will take the correct transformers in traveling through the i.f. amplifier. It is even possible to mount both the f.m. and a.m. transformers for a given stage in the same shield can.

When corresponding windings of the i.f. transformers for f.m. and a.m. are connected in series as shown in Fig. 13, signals will automatically take the correct path. Thus, when a 455-kc. a.m. signal is coming through, the transfer of signals will be accomplished by the 455-kc. a.m. transformer  $L_3-C_3-L_4-C_4$ , because  $L_1$  will be essentially a short-circuit path. When 10.7-mc. f.m. signals are coming through, signal transfer will be through f.m. transformer  $L_1-C_1-L_2-C_2$ , because  $C_3$  will be essentially a short-circuit path. In each case, the i.f. transformer not in use will be so far off resonance that its effect will be negligible.

**I.F. Switching.** During broadcast band a.m. reception, the local oscillator produces a signal somewhere between about 1000 and 2000 kc. (1 and 2 mc.) when the local i.f. value for a.m. is 455 kc. The sixth harmonic of the oscillator will then cover 6 to 12 mc., and there will be one tuning dial position at which this oscillator signal coincides with the 10.7-mc. i.f. value for f.m.

With the series arrangement of 455-kc. and 10.7-mc. i.f. windings, this oscillator harmonic (if strong enough) could get through the first i.f. transformer for f.m., and drive the self-bias so far beyond cut-off (due to increased plate current through the cathode resistor) that the desired 455-kc. i.f. signal would be blocked. The result would be a "dead spot" in the broadcast band at the setting which made this oscillator harmonic equal to 10.7 mc.

Other harmonics of the local oscillator can likewise create dead spots at different broadcast band dial settings,

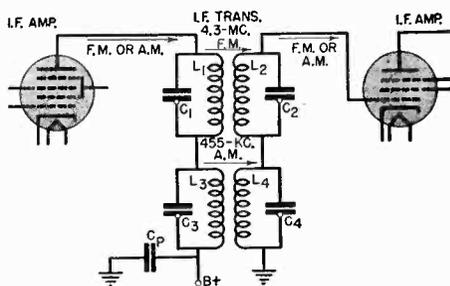


FIG. 13. I.F. transformer arrangement suitable for a combination f.m.-a.m. receiver. Separate i.f. transformers with corresponding windings connected in series are used for f.m. and a.m. No switches are employed, for desired signals automatically take the correct path due to circuit impedance values.

and hence it is necessary either to disconnect or short out the 10.7-mc. first i.f. transformer during broadcast band reception. When a switch is provided at the i.f. amplifier input for this purpose, only the desired i.f. signal passes through the first i.f. tube to the other i.f. transformers, and hence no additional i.f. switches are needed.

An example of a complete i.f. amplifier section which has provisions for disconnecting the unused first i.f. transformer primary is shown in Fig. 14. Section  $S_1$  of the band-changing switch connects to the converter the correct i.f. transformer primary winding ( $L_1$  or  $L_7$ ) for the particular signals being received. All other pairs

of i.f. windings are still connected in series just as in Fig. 13, however.

The i.f. input switch can just as well be between the grid of the first i.f. amplifier tube and the secondary windings of the first i.f. transformers, or can be used to short out the primary or secondary winding of the unused first i.f. transformer.

Note that the i.f. transformers for a.m. have fixed condensers, with tuning being accomplished with adjustable iron-core inductances. Pulverized iron cores are not effective at the 10.7-mc. i.f. value for f.m., but at a high frequency like this a low-resistance non-ferrous core (such as copper, aluminum or brass) can be used instead to vary the inductance. Non-ferrous cores of this type are inserted from the grounded end of the i.f. coil; they vary the inductance due to eddy currents induced in the core, and also add capacity to the tuning circuit. Non-ferrous cores can also be used for r.f. coils in f.m. circuits.

Sometimes the final i.f. stage is used only during f.m. reception. A simple method for eliminating this stage during a.m. reception involves grounding point *a* in Fig. 14 permanently, and connecting secondary  $L_{10}-C_{10}$  of the

second 455-kc. i.f. transformer directly to the a.m. second detector. Transformer  $L_{11}-C_{11}-L_{12}-C_{12}$  would be omitted.

Both i.f. amplifier tubes in Fig. 14 have resistors in their cathode leads to provide automatic C bias for both f.m. and a.m. reception. During a.m. reception, however, the diode a.m. detector supplies an a.v.c. voltage to the first i.f. amplifier tube, as well as to preceding tubes. (During f.m. reception, the a.v.c. voltage is provided by the limiter stage.)

*The Limiter.* The purpose of the limiter section in an f.m. receiver is to remove amplitude variations from the f.m. signal at the output of the i.f. amplifier, so that the signal fed into the discriminator will have constant amplitude. In serving this purpose, the limiter automatically corrects deficiencies in the frequency response of the preceding r.f. and i.f. stages. The limiter also provides an a.v.c. voltage for use during f.m. reception.

A typical single-tube limiter stage is shown in Fig 15. To understand how the 6SJ7 pentode in this circuit can function as a limiter, we must first consider the  $E_g-I_p$  characteristic curves for this tube under various

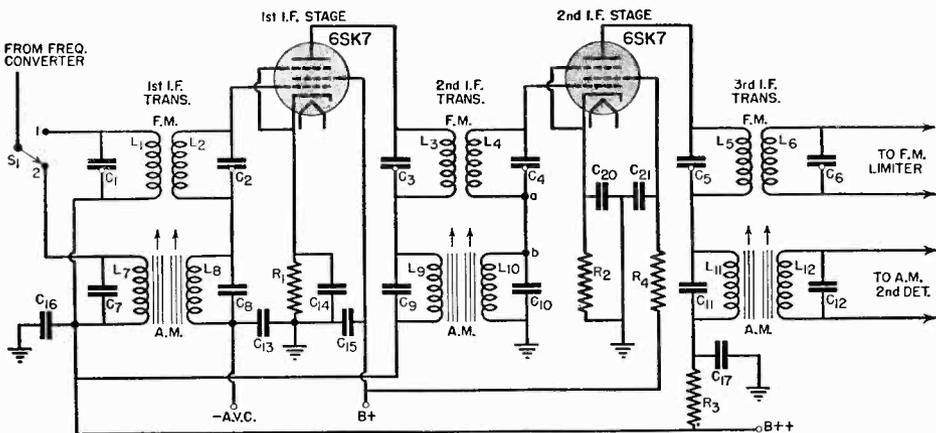


FIG. 14. Circuit diagram of a complete i.f. amplifier for a combination f.m.-a.m. receiver. Note the use of a switch for the primary windings of the first i.f. transformers. The groups of three parallel lines with arrows on a.m. transformers represent adjustable pulverized iron cores; these are adjusted in the same way as ordinary i.f. trimmers during i.f. alignment.

operating conditions.

When a 6SJ7 pentode is operated at normal voltages for amplifying purposes, such as with a 250-volt d.c. plate voltage and a 100-volt d.c. screen grid voltage, the static  $E_g-I_p$  characteristic of the tube will be like curve 1 in Fig. 16. This curve is essentially linear up to the highest plate current which can safely be passed by the

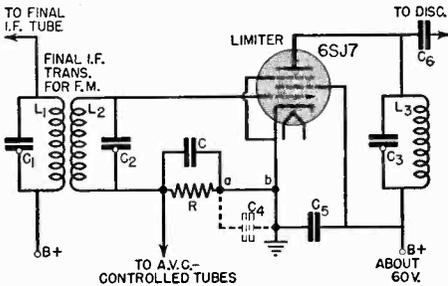


FIG. 15. Simplified one-tube limiter circuit for an f.m. receiver. Condenser  $C_4$  is provided for alignment purposes, as will be explained later.

tube, and consequently the plate current increases in proportion to positive grid voltage values. This is of no value for limiter action; we desire a characteristic which will make the plate current essentially constant regardless of how much the grid is driven positive, and hence the tube must be operated in such a way that plate current saturation occurs at a low positive grid voltage value.

When the d.c. plate and screen grid voltages of the 6SJ7 pentode are about 60 volts, we secure the desired condition whereby saturation begins at a fairly low positive grid voltage value, as indicated by curve 2 in Fig. 16. This curve is for the tube alone, and would be obtained from the circuit of Fig. 15 only if resistor  $R$  and condenser  $C$  were shorted out so as to give static conditions.

If a strong sine wave signal is applied to the grid of the 6SJ7 tube when it is operated with characteristic curve 2 in Fig. 16, the operating

point will be at  $o$ , negative peaks of the input signal will swing beyond cut-off (beyond  $c$ ), and positive peaks will swing into the saturation region of the curve (beyond point  $x$  at which saturation begins). As a result, both the positive and negative peaks of the plate current will be cut off, and we will secure a certain amount of the desired limiting action on strong signals.

Now let us return to the basic limiter circuit in Fig. 15, and consider its operation first with resistor  $R$  in the grid circuit but with condenser  $C$  removed. First of all, we can see that during half cycles which swing the grid negative, there is no grid current flowing through resistor  $R$ , and consequently the circuit acts essentially the same as if resistor  $R$  were shorted. In other words, our characteristic curve with  $R$  in the circuit will be the same for negative grid voltages as it was without  $R$ . For half cycles which make the grid swing positive, however, the flow of grid current through  $R$  will develop across it a voltage drop which opposes the applied positive grid voltage, and which consequently serves to provide a negative bias which

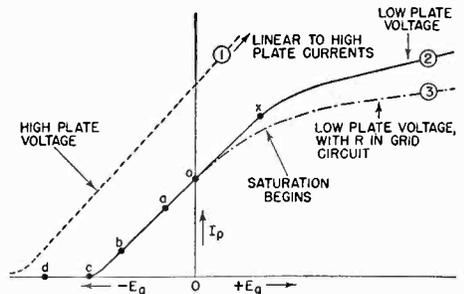


FIG. 16. Static characteristic curves of a vacuum tube for high (1) and low (2) plate voltages, and dynamic characteristic curve (3) for a limiter circuit using a low plate voltage and a resistor  $R$  (but no condenser  $C$ ) in the grid circuit. The plate load resistance for the static curves is assumed to be equal to the resistance of the plate load resonant circuit at resonance during dynamic operation.

is at each instant proportional to the positive applied voltage. The characteristic curve for this condition will

be like curve 3 in Fig. 16, which is flatter than curve 2. The saturation effect is considerably more pronounced now, and plate current during positive half cycles is limited by the negative bias across  $R$  as well as by the saturation characteristic of the tube at the low d.c. operating voltages used.

With  $R$  alone in the limiter circuit, the bias developed across it would follow individual r.f. positive peaks, and would consequently be varying continually. In a practical limiter circuit, resistor  $R$  in Fig. 15 would be shunted by condenser  $C$ , with the time constant of  $R$  and  $C$  being equal to the time of several r.f. cycles. Un-

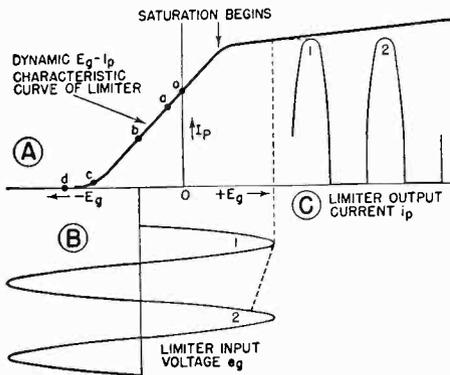


FIG. 17. These diagrams will help you to understand the action of the basic one-tube limiter circuit in Fig. 16.

der this condition, the  $R$ - $C$  circuit assumes a definite negative bias voltage which is maintained relatively constant over several cycles of the r.f. input voltage, and which hence exists for negative half cycles as well as for positive half cycles.

An automatic bias of this nature gives a characteristic curve which will be somewhere between curves 2 and 3 in Fig. 16; with  $R$  and  $C$  large in electrical size so as to give a long time constant, the dynamic characteristic curve will approach static curve 2 because we are approaching a fixed-bias condition, but with low electrical values for  $R$  and  $C$ , the dynamic char-

acteristic will approach curve 3 because the bias will now vary almost instantaneously with input signal strength.

A typical dynamic characteristic curve for a single-tube limiter stage is shown in Fig. 17. With this characteristic, the bias voltage developed across  $R$  and  $C$  will be proportional to the average signal strength over several r.f. cycles, and consequently the operating point might be at  $a$ ,  $b$ ,  $c$ , or even beyond cut-off at  $d$ , depending upon the r.f. signal strength. Thus, with a sine wave input voltage  $e_g$  which places the operating point at point  $b$ , the wave form of the resulting plate current  $i_p$  will be as shown at  $C$  in Fig. 17. This plate current is far from being sinusoidal like the input voltage, but we will still secure a sinusoidal output voltage across plate resonant circuit  $L_3$ - $C_3$  in Fig. 15 because this circuit is tuned to the 10.7 mc. i.f. value for f.m. It has a high  $Q$  factor, and therefore responds only to the desired fundamental frequency of the plate current pulses, rejecting the harmonics.

With the dynamic characteristic curve of Fig. 17 for a limiter stage, two things happen when the strength of the input signal increases: 1. The increased negative bias makes negative half cycles swing beyond cut-off for a longer period of time during each cycle, thereby reducing the operating angle during which plate current does flow; 2. The amplitude of the plate current pulses increases slightly, because an automatically produced bias can never completely counteract an increase in signal strength. The increased positive voltage on the grid produces a slight increase in the amplitude of the plate current pulses because the characteristic curve rises slightly in the saturation region, rather than being perfectly flat.

It is possible to design a limiter cir-

cuit so that for any reasonable increase in the r.f. input voltage to the limiter, the operating angle for plate current will decrease just enough to counteract the increase in the amplitude of the plate current pulses. The energy fed into plate resonant circuit  $L_3-C_3$  in Fig. 15 at the fundamental intermediate frequency for f.m. will then be constant, and the desired i.f. output voltage across this limiter resonant circuit will likewise be constant.

When this goal in limiter circuit design is realized, the over-all dynamic characteristic curve for the limiter will be like curve 1 in Fig. 18A, in which the r.f. output voltage of the limiter is plotted against the r.f. input

determine which over-all characteristic will be obtained, and hence these two parts in a limiter circuit are highly important.

Since the d.c. voltage produced across limiter grid resistor  $R$  is proportional to the strength of the f.m. signal at the limiter input, this d.c. voltage can be used for a.v.c. purposes during f.m. reception.

A limiter must be *fast-acting* (must have a *short time constant*) if it is to block out sudden noise surges. This assumes that there is sufficient r.f. amplification in the f.m. receiver so that the weakest desired signal will swing the limiter grid beyond the point at which saturation begins.

To show how the limiter can flatten

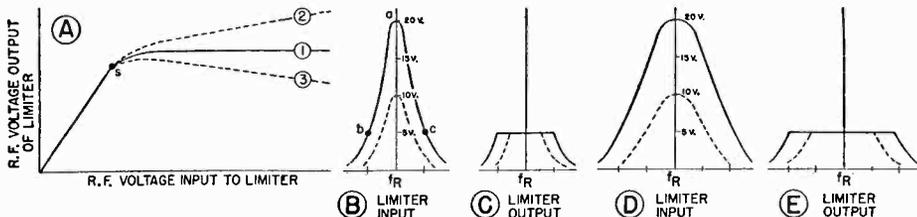


FIG. 18. These curves show how the limiter can correct the over-all response characteristic of the entire r.f. system (including the i.f. amplifier and the receiving antenna system) in an f.m. receiver.

voltage to the limiter. With this characteristic, the i.f. output voltage remains essentially constant regardless of input voltage, for all input voltage values which reach saturation (swing beyond point  $s$  in Fig. 18A).

If the design of the limiter circuit is such that the operating angle decreases faster than the plate current pulse amplitude increases, we have over-compensation and secure the over-all characteristic represented by curve 3 in Fig. 18A. Likewise, if the amplitude of the plate current pulses increases faster than the operating angle decreases, we have under-compensation and secure the over-all characteristic represented by curve 2 in Fig. 18A. The values employed for  $R$  and  $C$  in the limiter circuit of Fig. 15

the response characteristic of the r.f. system, let us assume that the limiter stage under consideration has a flat over-all grid voltage-plate voltage characteristic curve like 1 in Fig. 18A. Assume further that under this condition, all limiter input signals having peaks higher than 5 volts cause plate current saturation. (Positive grid swings will then go beyond point  $s$ , at which saturation begins.)

Now suppose that the r.f. system ahead of the limiter has the sharply peaked response characteristic shown in Fig. 18B. The vertical scale gives peak amplitude values; thus, if a strong input signal gave a 20-volt output peak amplitude at the resting frequency, this peak amplitude at the limiter input would drop down to 5

volts during the maximum deviation of 75 kc. from *b* to *a* to *c*). With the limiter characteristic of Fig. 18A, all signal peaks above 5 volts would be cut down to 5 volts by the limiter, and the solid-line curve in Fig. 18C would then represent the response characteristic of the r.f. system and limiter combined, for a strong f.m. input signal. Since this is flat over the desired band width of 150 kc., it is obvious that the limiter has flattened the highly-peaked response of the r.f. system.

Now suppose the receiver is tuned to a weaker f.m. signal, which gives a

input signal should now drop to a maximum peak value of 10 volts, the input and output response characteristics would be as shown by the dash-dash lines in Fig. 18D and 18E; even at this lower signal level, however, the combined r.f.-limiter response is still flat over a wide enough range to allow the entire frequency swing of 150 kc. to exist at the limiter output without changes in amplitude.

We thus arrive at the important conclusion that a broad over-all response for the r.f. and i.f. sections enables the limiter to handle weaker signals satisfactorily. Almost any sort of peak response is permissible if the r.f. and i.f. sections have sufficient gain, however, for the limiter will then flatten the over-all response of the preceding sections over the entire range of deviation frequencies.

*Time Constant of Limiter.* The time constant of the R-C circuit in the limiter is ordinarily made equal to the time of a few r.f. cycles, so the limiter will respond to general changes in the amplitude of the incoming signal without actually following individual r.f. cycles.

The importance of having a fast-acting limiter can be made clear by considering an f.m. signal which is varying in frequency from 75 kc. below to 75 kc. above its resting frequency (maximum deviation, corresponding to maximum loudness). With a receiver having a sharply resonant r.f. response like that in Fig. 18B, and with a peak limiter input of 20 volts at the resting frequency, the amplitude of the signal fed into the limiter will vary from 5 volts (at points *b* and *c*) to 20 volts (at *a*).

With the 20-volt input, the limiter will probably be operating near plate current cut-off, but this cut-off bias would be far too great to give the desired constant output amplitude when

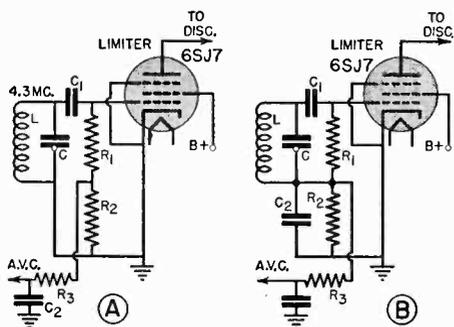


FIG. 19. Typical one-tube limiter circuits.

resting-frequency peak of only 10 volts at the output of the r.f. system. The dash-dash response curve in Fig. 18B would then give peak values for various deviations. When this weaker signal is fed into the limiter, the output response would be as shown by the dash-dash line in Fig. 18C. The band width over which we have uniform response is now obviously insufficient for standard f.m. signals, and distortion will occur during loud portions of the program.

On the other hand, if the r.f. system has the broadly-peaked response characteristic shown in Fig. 18D, a signal having a 20-volt peak would give a combined r.f.-limiter response corresponding to the solid line in Fig. 18E after passage through the limiter. This response is ideal, being flat for a frequency swing of about 250 kc. If the

the limiter input drops to 5 volts. In order to keep the limiter output amplitude constant over the entire deviation range of 150 kc., the C bias should automatically reduce itself as the limiter input signal amplitude drops.

Noise surges which enter an f.m. receiver are oftentimes strong but of extremely short duration. If these surges are to be blocked out, the limiter must be able to increase its own negative bias automatically for the duration of each surge. This action can occur only if a fast-acting limiter (with a short time constant for  $R$  and  $C$ ) is used.

The C bias produced by the R-C grid network of the limiter stage can change fast enough to flatten the r.f. response and squelch noise only if its time constant is kept very short (within the time of a few r.f. cycles).

**Fast-Acting Limiter Circuit.** In Fig. 19A is a practical limiter circuit employing a grid resistance ( $R_1 + R_2$ ) between the grid and cathode of the limiter, with condenser  $C_1$  and the grid-cathode capacity of the tube acting with the grid resistance to provide the required time constant for fast limiter action. This is attained by using a low capacity value for  $C_1$  and a low ohmic value for the grid resistance. A portion of the d.c. voltage developed across the grid resistance (the d.c. voltage across  $R_2$ ) is used for a.v.c. purposes during f.m. reception.  $R_3$  and  $C_2$  form the conventional a.v.c. filter which keeps r.f. components out of the a.v.c.-controlled tubes.

**Dual-Action Limiter Circuit.** In the limiter circuit arrangement shown in Fig. 19B, both  $C_1$  and  $C_2$  have low reactances at very-high frequencies.  $R_1$  and  $R_2$  in series form the grid return path.  $R_1$  and  $C_1$  together have a time constant equal to a few r.f. cycles, and hence provide a rapidly changing bias

equivalent to the fast limiter action of the circuit of Fig. 19A.  $R_2$  and  $C_2$  have a much longer time constant, and act to change the C bias voltage in accordance with the average strength of an incoming f.m. signal. In other words,  $R_2$  and  $C_2$  take care of the major changes in signal amplitude such as those occurring when tuning from a weak distant f.m. station to a strong local f.m. station, while  $R_1$  and  $C_1$  take care of changes in signal amplitude due to a peaked r.f. re-

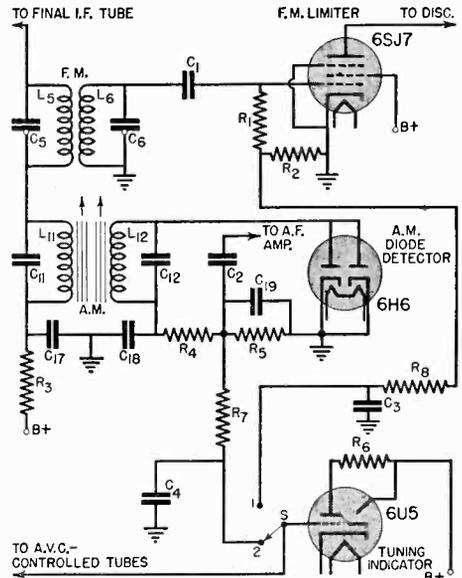


FIG. 20. One-tube limiter circuit with tuning indicator. I.F. amplifier connections to the conventional diode detector for a.m. reception are also shown.

sponse characteristic or to noise interference. This circuit arrangement thus provides independent control over two of the important factors which affect the design of a limiter circuit.

**Limiter with Tuning Indicator.** A more complete single-tube limiter circuit is shown in Fig. 20. This arrangement includes an electric eye for tuning purposes during both a.m. and f.m. reception. The output circuit of the i.f. amplifier is also shown, and is identical with that in Fig. 14. The

limiter input circuit is the same as that shown in Fig. 19A.

The a.v.c. voltage developed across  $R_2$  in Fig. 20 is fed to the tubes which are to be a.v.c.-controlled during f.m. reception through a.v.c. filter  $R_3-C_3$  and switch  $S$  (in position 1 for f.m. reception), and is also applied to the control grid of the type 6U5 cathode ray tuning indicator tube.

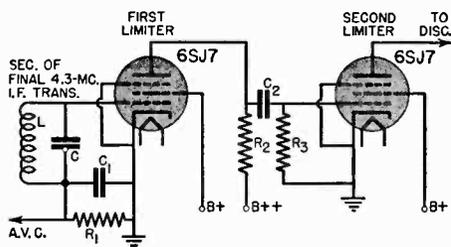


FIG. 21. Cascade limiter circuit employing two limiter tubes. The a.v.c. voltage is obtained from the first tube. The General Electric Model JFM-90 f.m. converter is one example of f.m. units employing this cascade limiter arrangement.

For a.m. reception, the output of the i.f. transformer for a.m. is fed into the diode detector (a 6H6 tube with both plates tied together), and the a.f. voltage component across  $R_5$  is fed to the a.f. amplifier through d.c. blocking condenser  $C_2$ . The d.c. voltage across  $R_5$  is fed to the a.v.c.-controlled tubes through a.v.c. filter  $R_7-C_4$  and switch  $S$  (in position 2 for a.m. reception), and is also applied to the tuning indicator tube.

*Cascade Limiter.* Improved limiter operation both for weak and strong signals, along with considerably higher gain, can be secured with two limiter tubes connected as shown in Fig. 21, in what is known as a *cascade limiter* circuit. The action of the first limiter tube is essentially like that of the limiter circuit in Fig. 19A, with the grid resistor and condenser being chosen to give an even shorter time constant so this first limiter will effectively reduce impulse noise. Whatever amplitude variations exist in the output of the first limiter tube are removed by the

second limiter tube, so that the f.m. signal which the limiter section feeds into the discriminator has essentially constant amplitude. Listening tests have shown that this cascade limiter reduces impulse noises far more than any single-tube limiter circuit.

One engineer recommends that the r.m.s. value of limiter output voltage be at least 20 volts for effective noise reduction. The minimum voltage gain for the entire r.f. system including the limiter should be at least 20,000,000 times to permit reception of weak desired f.m. signals without noise interference. To provide this amount of amplification, an r.f. amplifier stage ahead of the frequency converter is highly desirable.

### The F.M. Detector

The f.m. detector or discriminator used in an f.m. receiver to convert f.m. signals into audio signals is much like the frequency discriminator circuit employed in the a.f.c. system of a conventional a.m. superheterodyne receiver. Technically speaking, the discriminator converts deviations in frequency from an assigned resting value into positive and negative d.c. voltages corresponding to the original audio signal.

Although the action of a.f.c. discriminator circuits has been covered elsewhere in the course in connection with a.m. receivers, the action will be reviewed briefly here in connection with the typical f.m. discriminator circuit given in Fig. 22.

Discriminator transformer  $T_1$  is of special design, with its primary winding  $L_2$  tuned to the 10.7-mc. i.f. value by  $C_2$ , and with a center-tapped secondary winding  $L_3$  tuned to the same i.f. value by  $C_3$ . The limiter output current flowing through  $L_2$  induces a corresponding f.m. voltage in  $L_3$ , and resonant step-up produces across the two sections of  $L_3$  the voltages  $e_1$  and

$e_2$ , which are always equal in magnitude and are  $180^\circ$  out of phase with each other when considered with respect to center-tap  $p$ .

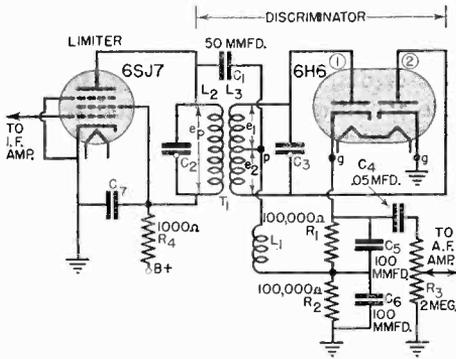


FIG. 22. Typical discriminator circuit for an f.m. receiver. The discriminator load resistors are the 100,000-ohm resistors  $R_1$  and  $R_2$ .

Limiter output voltage  $e_p$  in series with  $e_1$  is applied to diode section 1 of the 6H6 double-diode tube. (D.C. blocking condenser  $C_1$  provides an r.f. path from the upper end of  $L_2$  to  $p$ , while the chassis and r.f. by-pass condensers  $C_5$ ,  $C_6$  and  $C_7$  complete the path for r.f. signals from the lower end of  $L_2$  to the cathode of diode section 1. Thus,  $e_p$  exists across choke coil  $L_1$ , and acts in series with  $e_1$  or  $e_2$  on either diode.) Likewise, limiter output voltage  $e_p$  in series with  $e_2$  is applied to diode section 2.

The net voltage applied to each diode section is, therefore, the vector sum of the two individual voltages acting on that section. Each diode section rectifies its net applied r.f. voltage and produces a proportional d.c. output voltage across its load resistor.

The load resistor for diode section 1 is  $R_1$ , and the load resistor for diode section 2 is  $R_2$ . The chassis provides the connecting path between the lower end of  $R_2$  and the cathode of diode section 2.

Electrons flow in opposite directions through  $R_1$  and  $R_2$ , as you can readily

verify by tracing the diode circuits. This means that the combined voltage across both  $R_1$  and  $R_2$ , which is the output voltage of the discriminator, will at each instant be the difference between the individual voltages. If the individual voltages are equal, the discriminator output voltage will be zero; if the voltages across  $R_1$  and  $R_2$  are different, the combined voltage will have the polarity of the larger of the two individual voltages, and will be equal in magnitude to their numerical difference.

Let us consider now the factors which make the output voltage of one diode higher than that of the other. First of all, we must choose some voltage or current for reference purposes. Since  $e_p$  is common to all circuits under study, we can use it as our reference voltage.

Phase relationship in this discriminator circuit must be considered for three different conditions: 1. When the limiter output signal frequency is equal to the i.f. resting frequency to which the discriminator resonant circuits are tuned; 2. When the limiter output frequency is less than the i.f.

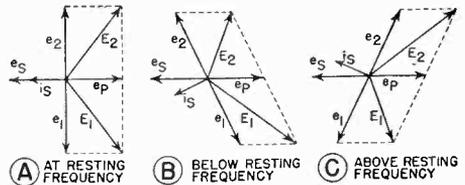


FIG. 23. These vector diagrams will help you to understand the action of the discriminator in an f.m. receiver.

resting frequency; 3. When the limiter output frequency is higher than the i.f. resting frequency. The vector diagrams for these three conditions are shown at A, B and C respectively in Fig. 23, with primary voltage  $e_p$  serving as the reference vector in each case.

The r.f. voltage  $e_s$  which is induced in secondary winding  $L_3$  is  $180^\circ$  out

of phase with the primary r.f. voltage  $e_P$ , hence  $e_S$  is shown  $180^\circ$  out of phase with reference vector  $e_P$  in each of the vector diagrams in Fig. 23.

When the limiter output signal is *exactly* at the i.f. resting value of 10.7 mc. to which the discriminator circuits are tuned (when no sound is being transmitted), secondary tuned circuit  $L_3-C_3$  is at resonance, and secondary current  $i_S$  flowing through  $L_3$  will be in phase with  $e_S$ , as indicated in Fig. 23A. The voltage produced across the entire secondary winding by this secondary current will therefore *lead* both  $i_S$  and  $e_S$  by  $90^\circ$ . The resulting secondary voltage ( $e_1+e_2$ ) across  $L_3$  is utilized only in connection with the center tap of  $L_3$ , however, so if we show  $e_1$  leading  $i_S$  by  $90^\circ$ , we must show  $e_2$  as lagging  $e_1$  by  $180^\circ$ , just as in Fig. 23A.

Adding  $e_P$  and  $e_2$  vectorially gives  $E_2$  as the resultant voltage acting upon diode section 2. Likewise, adding  $e_P$  and  $e_1$  vectorially gives  $E_1$  as the resultant voltage acting upon diode section 1. The vector diagram in Fig. 23A shows that these two voltages are equal for the no-modulation condition, and hence the d.c. voltages developed across  $R_1$  and  $R_2$  by the two diode sections are equal in magnitude. The resultant voltage across both  $R_1$  and  $R_2$  is therefore zero, just as it should be, since no a.f. signal should be obtained when there is no a.f. modulation at the transmitter.

When the limiter output signal frequency is *lower* than the i.f. resting value to which resonant circuit  $L_3-C_3$  is tuned, this circuit becomes *capacitive*, and  $i_S$  leads  $e_S$ , as shown in Fig. 23B. Since voltages  $e_1$  and  $e_2$  must be  $90^\circ$  out of phase with  $i_S$ , we have the unequal resultant voltages  $E_2$  and  $E_1$ , as in Fig. 23B. With diode section 1 getting the higher r.f. voltage  $E_1$ , we secure a higher d.c. voltage across  $R_1$  than across  $R_2$ , and the combined

voltage across  $R_1$  and  $R_2$  is therefore *positive* with respect to ground. The more the limiter output frequency swings *below* the i.f. resting frequency, the greater will be this positive voltage applied to the a.f. amplifier input.

By a similar analysis, we obtain the vector diagram shown in Fig. 23C for the condition wherein the limiter output frequency is *higher* than the i.f. resting frequency. The net voltage applied to the input of the a.f. amplifier by  $R_1$  and  $R_2$  combined is now *negative* with respect to ground.

The frequency discriminator circuit shown in Fig. 22 thus produces at its output a d.c. voltage which is at each instant proportional to the deviation in the incoming signal frequency from its resting value, and having a polarity determined by the direction in which this frequency deviation occurs. The discriminator thus converts an f.m. signal directly into the original audio signal voltage used for modulation purposes at the f.m. transmitter.

R.F. by-pass condensers  $C_5$  and  $C_6$  in Fig. 22 must have a low reactance at 10.7 mc., and yet must have a high reactance at audio frequencies so there will be no serious shunting effect upon the a.f. voltage developed across  $R_1$  and  $R_2$ . This a.f. voltage is fed to volume control  $R_3$  through d.c. blocking condenser  $C_4$ , which prevents the C bias voltage of the first a.f. amplifier tube from entering the discriminator circuit and prevents the d.c. discriminator output voltage from acting on the grid of the first a.f. tube.

*S Curve for Discriminator Action.* The solid-line curve in Fig. 24 shows the relationship between the incoming r.f. signal frequency of an f.m. receiver (with respect to the resting value) and the d.c. output voltage of the discriminator. Because of the similarity of this curve to the letter S, it is commonly known as an S curve.

This curve, representing the characteristics of the f.m. receiver up to the input of the audio amplifier, should be linear over the entire deviation range of the incoming f.m. signal, for otherwise amplitude distortion would be present in the a.f. output of the discriminator.

The linearity of the S curve in Fig. 24 depends upon two things, the design of the discriminator transformer, and the over-all response characteristic of all the stages ahead of the discriminator. The over-all response is determined chiefly by the dynamic characteristic of the limiter; the response of stages ahead of the limiter affects the limiter only when signals are too weak to cause saturation of the limiter for the entire deviation range.

In designing the discriminator transformer, the Q factor of each resonant circuit and the coupling between the two windings are particularly important; these factors must make the discriminator characteristic combine with the r.f.-limiter characteristic in such a way that the individual resultant voltages,  $E_1$  and  $E_2$ , will at each instant be proportional to the frequency deviation. When this condition is attained, the combined discriminator output voltage across  $R_1$  and  $R_2$  will be proportional to the frequency deviation, thereby giving across these resistors the desired a.f. voltage without amplitude distortion.

It is desirable to have the S curve linear over a somewhat wider range than the maximum deviation, to compensate for inaccuracies in tuning, frequency drift in the local oscillator, or misalignment of the discriminator resonant circuits.

A falling off in the amplitude of the limiter output near the deviation limits, due to too weak a signal at the limiter input, will reduce the frequency range over which the S curve

is linear. The dash-dash S curve in Fig. 24 illustrates this condition.

When the S curve for discriminator action has too short a linear region for weak signals, reproduction will still be satisfactory at medium and low program loudness levels, but *amplitude distortion will occur during*

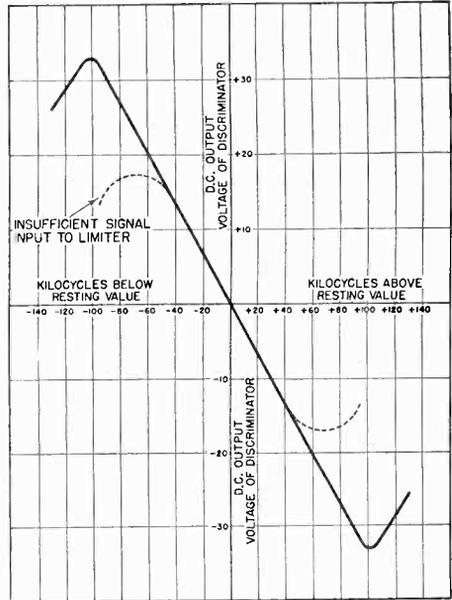


FIG. 24. S curves for an f.m. receiver, showing how the discriminator output voltage varies with frequency deviation. Curves like this are obtainable only if the over-all grid voltage-plate voltage characteristic of the limiter is flat beyond the point at which saturation begins (like curve 1 in Fig. 18A). When this over-all limiter characteristic is not flat (is like 2 or 3 in Fig. 18A), the angles which the curves make with the horizontal axis will be different (giving either lower or higher discriminator output voltages), and the curves will no longer be linear (resulting in distortion of the audio output signal). You will often find an S curve like this drawn from the lower left to the upper right. Reversal of connections to the discriminator diode plates gives this result; either connection is satisfactory, for the polarity of an a.f. signal cannot be detected by human ears.

loud sounds. Thus, all sounds loud enough to cause a deviation of more than 40 kc. will swing the signal frequency beyond the linear region of the dash-dash curve in Fig. 24. This is why it is so important that the limiter in an f.m. receiver deliver a constant-amplitude output signal over the entire deviation range. The goal of the f.m. receiver designer is to provide

sufficient voltage gain ahead of the limiter so that the limiter can maintain constant output amplitude for the weakest desired incoming signal.

*Alignment of Limiter and Discriminator Stages.* During servicing of an f.m. receiver, the limiter and discriminator can only be aligned for best possible operation. Little can be done to alter the linearity of the discrim-

point, and connect across the condenser a 0-200-microampere d.c. meter.\* (In circuit diagrams of f.m. receivers, you will often find in the grid return lead of the limiter stage a condenser, about .01 mfd., shorted out by a lead. This condenser is inserted for measuring the grid current during alignment of the limiter; the service technician simply unsolders his microammeter, then connects his microammeter across the condenser. This is illustrated in Fig. 16, where  $C_4$  would be the .01-mfd. condenser provided for measuring purposes. The microammeter would be connected to points *a* and *b*, after first opening the lead between these two points.)

With these preliminary connections made, turn on all apparatus, adjust the limiter input trimmers ( $C_1$  and  $C_2$  in Fig. 16) for maximum grid current as indicated by the meter, adjust the signal generator output voltage so that the meter reading is at least 50 microamperes, then readjust  $C_1$  and  $C_2$  a final time for a maximum meter reading. This 50-microampere or larger current is necessary so the limiter will provide normal loading on the preceding resonant circuit.

Without changing the s.g. settings, remove the microammeter (or vacuum tube voltmeter) from the limiter input circuit and restore the connection between points *a* and *b*. Now connect a high-resistance voltmeter across one of the diode load resistors (across either  $R_1$  or  $R_2$  in Fig. 22), and adjust the discriminator transformer primary trimmer ( $C_2$  in Fig. 22) for maximum output voltage.

Next, connect the high-resistance voltmeter across both  $R_1$  and  $R_2$  (a

\* With limiter circuits which also provide an a.v.c. voltage, a vacuum tube voltmeter connected to the a.v.c. source can be used in place of the microammeter. The adjustment would be made for maximum a.v.c. voltage, as this would then correspond to maximum grid current.

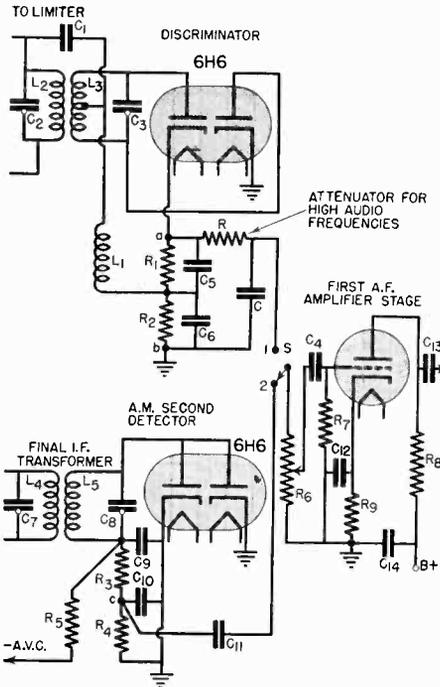


FIG. 25. This diagram shows how section S of the band-changing switch in an f.m.-a.m. receiver can be used to connect either the discriminator or the a.m. second detector to the input of the audio amplifier. Note that volume control  $R_8$  is effective at both switch settings.

inator, for this linearity is controlled chiefly by the limiter characteristics and the design and construction of the discriminator transformer.

*Alignment Procedure.* Connect a signal generator to the input of the final i.f. amplifier stage for f.m., set this s.g. to the i.f. resting frequency (usually 10.7 mc.), then open the grid return lead of the limiter circuit, insert a .01-mfd. condenser at this

connection between the cathodes of the diode sections in Fig. 22 will do this), and adjust the discriminator transformer secondary trimmer ( $C_3$  in Fig. 22) for zero d.c. output voltage. This last adjustment is usually quite critical.

This completes the alignment of the limiter and discriminator stages. The remainder of the tuned circuits in an f.m. receiver are then aligned, usually for peak response, by adjusting each alignment trimmer in turn either for maximum a.v.c. voltage or for maximum grid current in the limiter.

*The Audio System.* The over-all fidelity of an f.m. receiver is determined to a great extent by its audio system. If the receiver is intended only for f.m. reception with a minimum of noise, and high fidelity is not required, an ordinary a.f. amplifier and loudspeaker will probably be employed. When high-fidelity f.m. reception is desired, however, the audio amplifier will be designed to handle a frequency range from about 30 cycles to about 15,000 cycles, and a specially designed reproducing system will be used to handle this extremely wide range of audio frequencies.

The sound-reproducing system in a high-fidelity f.m. receiver will usually employ at least two loudspeakers, one to handle low and intermediate frequencies, and the other to handle the higher frequencies. These loudspeakers will be coupled to the audio amplifier output stage through special networks, in order to secure proper division of output energy to the two loads. The entire sound-reproducing system will be housed in an acoustically corrected cabinet. The normal rating of the audio system will ordinarily be at least 15 watts, to secure sufficient power for proper reproduction of loud bass notes.

In the transmission of f.m. signals, it is customary to accentuate or pre-

emphasize the higher-frequency notes at the transmitter, so that these will over-ride noise. To cut these high frequencies back down to normal, high-fidelity f.m. receivers have a high-frequency attenuator in the audio amplifier. This high-frequency attenuator (usually a simple resistor-condenser circuit) compensates for the pre-emphasis at the transmitter and thus restores the normal balance between highs and lows.

In a combination f.m.-a.m. receiver, a single audio system will serve for both types of reception. A switch like that shown in Fig. 25 will ordinarily be provided to connect the audio amplifier input circuit to the detector stage being employed for a particular program. This switch will be ganged to the main band-changing switch in the r.f. system, so as to make detector switching automatic.

In the circuit of Fig. 25, 50,000-ohm resistor  $R$  and .001-mfd. condenser  $C$  in the f.m. channel form the attenuator which counteracts f.m. transmitter pre-emphasis of high audio frequencies. All commercial f.m. transmitters will have the same amount of pre-emphasis, and this will be such that an attenuator having a time constant of 75 micro-seconds will restore normal receiver response for high audio notes. Normal attenuation of the a.f. amplifier at high audio frequencies, due to stray shunt capacities between circuit leads and ground, must be taken into account when designing the attenuator. Thus, the values used for  $R$  and  $C$  in Fig. 25 give a time constant of only 50 micro-seconds, but the a.f. amplifier has stray shunt capacities which make the total attenuation equivalent to that of a 75-micro-second attenuator.

If no switch were employed at the a.f. amplifier input (if both points 1 and 2 in Fig. 25 were connected to the a.f. amplifier input permanently), the

diode detector not in use would place an additional load on the other diode during positive peaks, causing amplitude distortion of the audio signal. For example, if an f.m. signal were being received, point *a* in Fig. 25 would be made alternately positive and negative with respect to point *b* and the chassis by the audio signal. This in turn would make point *c* in the unused a.m. detector alternately positive and negative with respect to the chassis. Since point *c* is connected to the plates of the unused diode through resistor  $R_3$ , the plates would likewise be positive for half of each cycle, and would be conductive. The path through  $R_3$  and the diode would

limiter. Comparing the circuit in Fig. 26 to the discriminator, you will find that one diode is reversed, so that we have a half-wave rectifier rather than a push-pull type. Now the voltage across  $C_2$  and  $C_3$  is the sum of the two rather than their difference. Hence, the audio must be taken off in a bridge arrangement from terminals *m* and *n*.

When the signal input is such that *k* is positive with respect to *l*, the diodes both conduct, causing  $C_4$  to be charged (in parallel with  $C_2$  and  $C_3$  in series) to a value corresponding to the average carrier level of the signal. With no modulation, the voltages  $e_1$  and  $e_2$ , added vectorially to  $e_3$ , are equal, so the  $C_2$  and  $C_3$  drops are

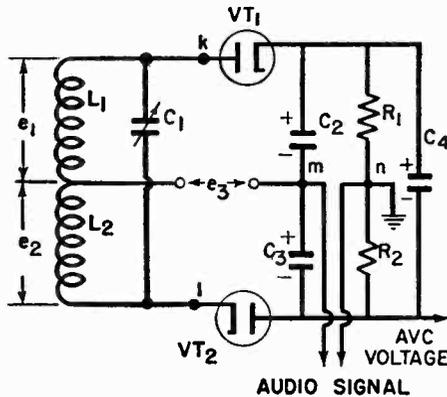


FIG. 26. A ratio detector.

then be a load on the f.m. detector during each positive a.f. peak, thus reducing the positive peaks. To prevent this condition, the detector switch is used.

*Ratio Detectors.* Recently, there have been introduced several hybrid circuits that perform both the functions of limiting and discrimination in a single tube. One example of this is the "ratio" detector, also known as the Seeley detector. Basically, it is like the discriminator you have been studying, except that it is insensitive to amplitude changes and hence acts as its own

equal. Hence *m* is at the same potential as *n*, and there is no audio. When the  $e_1 - e_3$  vector is longer than the  $e_2 - e_1$  one, the drop across  $C_2$  is greater than that across  $C_3$ , and vice versa.

If we assume that the initial  $C_4$  voltage is 10 volts, then the  $C_2$  and  $C_3$  sum must equal this, or be 5 volts each at the resting frequency. Then, should the signal make the  $C_2$  voltage 8 volts, and the  $C_3$  voltage 2 volts, point *m* will be 3 volts more negative than formerly. (Point *n* remains fixed by the  $C_4$  drop.) When the signal swings the other way, then point *m* moves positive. Hence, we get our audio out-

put from the frequency swings, as in the discriminator circuit.

The  $C_4$ - $R_1$ - $R_2$  time constant is made purposely long, so that the voltage across it follows the average amplitude of the incoming signal. This fixes the potential of terminal  $n$ . Also, should there be any sudden amplitude change, such as may be produced by noise, there can be no quick change in the  $C_4$  voltage; hence the  $C_2$ - $C_3$  sum remains relatively fixed. The noise pulse is dropped across the diodes because they pass more current trying to charge  $C_4$  to a new value. As long as the  $C_2$ - $C_3$  sum is constant, only frequency variations can cause an audio output by shifting the  $C_2$  and  $C_3$  volt-

tion at the transmitter, and that the frequency deviation from this resting frequency be at all times proportional to the sound level at that instant.

Two different methods for securing frequency modulation with the required stability and linearity are widely used. One involves the use of an inductance tube and an automatic crystal monitoring control, while the other, suggested by Major Armstrong, starts with a crystal oscillator and utilizes a phase-shifting circuit. Let us analyze each method in turn.

*Inductance Tube Transmitter.* A block diagram of a commercial f.m. transmitter of the inductance tube

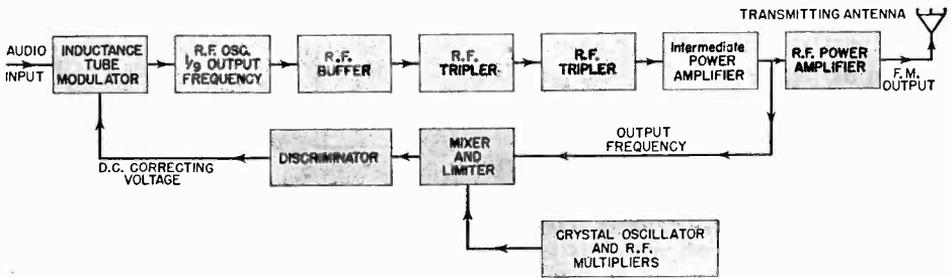


FIG. 27. Block diagram of an f.m. transmitter employing an inductance tube in the modulator stage. The main oscillator stage is of the ordinary L-C type, but its frequency is corrected automatically by the auxiliary crystal oscillator acting through an automatic frequency control system.

ages individually. Hence, this is a combination limiter-discriminator that is popular in the less expensive sets particularly.

### Modern F.M. Transmitters

Although the method shown in Fig. 6 for producing frequency modulation of a carrier signal is simple and effective, it lacks frequency stability; an r.f. oscillator shunted by an inductance tube in this manner will drift in frequency with variations in temperature and with variations in the d.c. supply voltage.

In commercial f.m. broadcast stations, it is essential that the transmitter return to its assigned resting frequency whenever there is no modula-

tion is given in Fig. 27. In this system, the oscillator is of the conventional L-C type, with temperature-compensated parts in its tank circuit (no crystal). This r.f. oscillator is tuned to some fraction of the assigned resting frequency of the f.m. station, and frequency-multiplying stages are employed ahead of the final amplifier to bring the frequency up to the assigned resting value. Thus, in the f.m. transmitter chosen as an example for Fig. 27, the r.f. oscillator is operated at one-ninth the assigned resting frequency, and its output is fed into two frequency tripler stages which bring the signal up to the correct frequency.

The inductance tube in the modu-

lator stage is connected across the tank circuit of the L-C oscillator, and hence any change in the grid voltage of the inductance tube will change the frequency of the oscillator. Two voltages act upon the grid of this inductance tube; one is the audio signal voltage applied by the speech input amplifier, and the other is a d.c. voltage produced by the discriminator in the following manner.

Let us assume that the L-C oscillator is at exactly one-ninth of the assigned resting value, and that there is no audio modulation. By means of weak capacitive coupling to some point in the r.f. amplifier, the signal at the assigned resting value is fed into a mixer stage, where it beats with a lower fixed r.f. signal produced by a temperature-controlled crystal oscillator acting through frequency-multiplying stages. The result is a signal at the difference frequency, corresponding to the i.f. signal in a superheterodyne receiver.

The mixer output signal is fed into a discriminator circuit which is tuned exactly to this difference frequency, and which therefore produces a d.c. voltage which is proportional to deviations from this difference frequency. With the L-C oscillator exactly correct, there is negligible deviation, and the d.c. output voltage which the discriminator applies to the grid of the inductance tube is therefore zero.

When the L-C oscillator drifts in frequency, the discriminator produces a d.c. voltage which is proportional to the magnitude of the frequency drift and has a polarity determined by the direction of the drift. This d.c. voltage changes the inductance of the inductance tube exactly enough to correct the frequency drift.

The action of this automatic frequency-correcting circuit is the same when audio modulation is present, for a condenser filter in the discriminator

prevents audio signals from traveling from the discriminator to the inductance tube. Only the d.c. voltage actually due to drift from the assigned resting frequency can act upon the inductance tube.

Sometimes a portion of the audio output of the discriminator is utilized to provide inverse feed-back which compensates for distortion occurring in the transmitter. In this case, a special coupling circuit would be used to apply both the d.c. and a.f. components of the discriminator output to the inductance tube grid.

The frequency-tripling stage is simply an r.f. stage operated as a class C amplifier, with its plate resonant

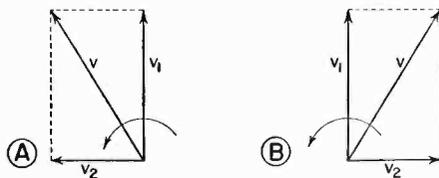


FIG. 28. Vector diagrams showing how a frequency-modulated signal can be produced by the Armstrong phase-shift method.

circuit tuned to the third harmonic of the input frequency.

Any tendency for the f.m. transmitter to drift from its assigned resting frequency thus produces a d.c. voltage which acts upon the grid of the inductance tube in such a manner as to correct the tendency toward drift. A zero-center type of voltmeter connected across the discriminator output indicates when the drift is getting dangerously great, thus warning the operator to retune the L-C oscillator manually before the discriminator loses control.

*Armstrong Phase-Shift Transmitter.* The basic principle underlying the Armstrong method is: When a signal having a definite frequency is flowing through a circuit, and another signal having this same frequency but  $90^\circ$  out of phase is suddenly sent into the circuit, there is an instantaneous phase

shift in the original signal, with the result that the frequency of the combination either increases or decreases. We can illustrate this principle by means of the vector diagrams in Fig. 28.

Vector  $V_1$  in Fig. 28A represents the original signal, having a definite frequency and a constant amplitude. As this vector rotates in the conventional counter-clockwise direction at a rate corresponding to its frequency, let us suddenly introduce into the circuit a voltage  $V_2$  which has the same frequency as voltage  $V_1$  but leads  $V_1$  by exactly  $90^\circ$ . The sudden introduction of this new voltage  $V_2$  causes the net voltage  $V$  of the combination to shift in a counter-clockwise direction, corresponding to a momentary increase in frequency. If voltage  $V_2$  is suddenly increased in amplitude after it is introduced into the circuit, resultant vector  $V$  speeds up again, producing another momentary increase in frequency.

On the other hand, if the suddenly-introduced  $V_2$  lags  $V_1$  by  $90^\circ$ , the resultant vector  $V$  will move clockwise, corresponding to a momentary decrease in frequency, as indicated by the vector diagram in Fig. 27B. If vector  $V_2$  is varying continually in both amplitude and polarity in accordance with a sine-wave audio signal, resultant voltage vector  $V$  will swing above and below its original frequency in a corresponding manner.

All this means that we can produce frequency modulation of a carrier signal by combining it with another signal which has the same carrier frequency but is alternately  $90^\circ$  leading and  $90^\circ$  lagging the original signal in accordance with positive and negative swings of an audio signal, and varies in amplitude in accordance with variations in the amplitude of the audio signal.

In order to make the deviation in frequency dependent only upon the amplitude and polarity of  $V_2$ , an attenuator is introduced into the audio amplifier of the transmitter so as to make the audio output voltage decrease gradually as the audio frequency increases. This insures that a 1-volt audio signal at 10,000 cycles will produce the same frequency deviation as a 1-volt audio signal at 1000 or 100 cycles.

As a rule, both  $V_1$  and  $V_2$  in an f.m. transmitter of this type will have relatively low frequency values, and fre-

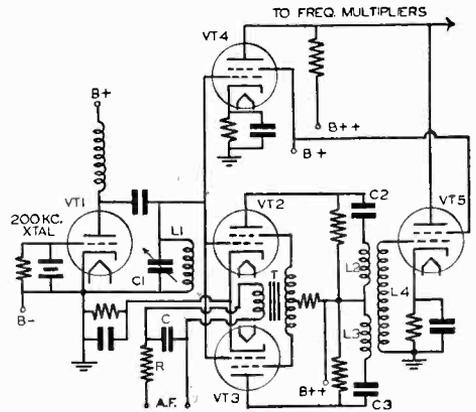


FIG. 29. Basic circuit used in the Armstrong phase-shift type of f.m. transmitter.

quency multipliers will be used to bring the f.m. signal up to the correct frequency for the transmitting antenna.

The basic Armstrong circuit for securing frequency modulation by means of this phase shift method is given in Fig. 29. Vacuum tube  $VT_1$  in a crystal oscillator circuit produces the initial low r.f. carrier value (200 kc. would be a typical value) with negligible frequency drift. The output of this crystal oscillator is fed into r.f. amplifier tube  $VT_4$  and also into the balanced r.f. stage including tubes  $VT_2$  and  $VT_3$ .

Note that the grids of this balanced stage are fed in phase, while the output of this stage goes to an untuned

r.f. transformer having a center-tapped primary  $L_2$ - $L_3$ . With in-phase plate currents flowing in opposite directions through the primary to the center tap, and with the tubes balanced so that the currents in the two sections of the primary are normally equal in magnitude, the resultant flux linked with secondary winding  $L_4$  is zero, and no voltage is induced in this winding for transfer to the frequency multipliers through amplifier stage  $VT_5$ .

The values of the parts in the plate circuits of the two balanced amplifier tubes are such that the r.f. current in each half of the primary lags the a.c. grid voltage by approximately  $90^\circ$ ; in other words, the plate loads for  $VT_2$  and  $VT_3$  are essentially pure inductances.

The operation of this circuit depends upon the basic fact that when the a.c. resistance of a tube is varied its a.c. plate current will vary correspondingly. We can change the a.c. plate resistance of a tube by varying the screen grid voltage.

In the circuit of Fig. 29, the screen grid voltages for both  $VT_2$  and  $VT_3$  in the balanced amplifier stage are applied through the secondary winding of audio transformer  $T$ . With this arrangement, an a.f. voltage applied

to the primary winding will make the screen grid voltage on one tube increase, and make the screen grid voltage on the other tube decrease. This causes an unbalance in the r.f. plate currents flowing through  $L_2$  and  $L_3$ , and consequently we secure a resultant flux which links with  $L_4$ .

The resultant current flowing through both  $L_2$  and  $L_3$  will either lag or lead the a.c. grid voltage by  $90^\circ$ , depending upon which coil current is greatest. This means that the r.f. voltage induced in secondary winding  $L_4$  will either lead or lag the a.c. grid voltage by  $90^\circ$ , depending upon whether the a.f. input voltage is swinging positive or negative, and the amplitude of this r.f. voltage in  $L_4$  will vary in accordance with the amplitude of the a.f. input signal. The voltage across  $L_4$  thus corresponds to vector  $V_2$  in the diagrams of Fig. 28.

Amplifier stage  $VT_5$  merely provides a means for securing the proper relationship between the amplitudes of r.f. voltages  $V_1$  and  $V_2$ . These two voltages, from tubes  $VT_4$  and  $VT_5$ , are fed in parallel into the frequency multiplier system. They combine to give the desired frequency-modulated signal, as was explained in connection with the vector diagrams in Fig. 28.

## TEST QUESTIONS

Be sure to number your Answer Sheet 34FR-2.

Place your Student Number on every Answer Sheet.

Send in your answers for this lesson immediately after you finish them. Doing this insures that the graded answers will reach you while the subject matter is still fresh in your mind, and you will get the greatest possible benefit from our speedy personal grading service. Never hold up a set of lesson answers.

1. To secure a high signal-to-noise ratio with minimum inter-station interference at receivers, should the maximum frequency deviation in an f.m. system be *high* or *low*? *high*
2. In what two ways can an interfering signal affect a desired f.m. signal?
3. What is the purpose of the discriminator in an f.m. receiver? *to convert F.M. to A.M. variation*
4. What section in an f.m. receiver produces the a.v.c. voltage? *limiter*
5. What section in an f.m. receiver removes amplitude variations from the f.m. signal? *limiter*
6. What characteristic should a limiter have in order to block out sudden noise surges? *very short time constant of grid R-C*
7. Why is it permissible to have a peaked over-all response for the r.f. and i.f. sections in a properly designed f.m. receiver? *Because limiter action keeps off am. & essential low's constant*
8. Describe the type of distortion which occurs when the S curve for discriminator action has too short a linear region for weak signals.
9. Why is it essential that grid current be drawn by the limiter tube during alignment of the resonant circuit ahead of the limiter?
10. Why is an attenuator for high audio frequencies used in the audio system of an f.m. receiver?



## SHALL I CHANGE JOBS?

Before you can answer this question, you must set your goal in life. With a goal in mind, apply these three questions to your present job: 1. Does it give reasonable pay for the present? 2. Does it give knowledge, training or experience which will be worth money to you in the future? 3. Does it give you prestige, or bring you in touch with men who can help you to attain your goal?

Judge each new job opportunity also by these three questions. Hold each job only long enough to learn what is needed for the next job. Then, if a vacancy up ahead is unlikely, you are justified in changing to a corresponding job somewhere else.

The important thing is to keep going ahead. Learn something every day. When you reach the point where you are no longer learning, no longer progressing ahead, then you are already moving backward, and it's time for you to change jobs.

First, though, be sure you really are prepared for a better job. Be sure your training is complete, and be sure you are able to apply it.

*J. E. Smith*

**AUTOMATIC TUNING  
CONTROL SYSTEMS**

35FR-1



**NATIONAL RADIO INSTITUTE**

**WASHINGTON, D. C.**

**ESTABLISHED 1914**

# Study Schedule No. 35

For each study step, read the assigned pages first at your usual speed, then reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind, then answer the Lesson Questions specified for that step. Study each other step in this same way.

**1. Introduction; Automatic Tuning Systems; Problems of Automatic Tuning; Mechanical Automatic Tuning Systems - - - Pages 1-9**

This lesson, which completes the foundation of your Radio career, contains a wealth of thoroughly practical information on push-button tuning systems. Only fundamental facts are given—basic knowledge which will enable you to master in a jiffy the intricacies of any specific system you may encounter in your work. Answer Lesson Questions 1, 2 and 3.

**2. Electrical Automatic Tuning Systems - - - - - Pages 9-14**

The first time you work on a set employing this system of tuning, you'll see the importance of the fundamental facts in this section. Pay particular attention to the material dealing with adjustment of trimmer condensers. Answer Lesson Question 4.

**3. Electro-Mechanical Automatic Tuning Systems - - - - Pages 14-18**

The various types of motor-driven tuning systems are classified here in such a way that you can readily recognize each type in an actual receiver and quickly figure out how it should work. Some of the most fascinating mechanisms in Radio are described here. Answer Lesson Questions 5 and 6.

**4. Audio-Silencing and A.F.C.-Releasing Switches for Electro-mechanical Systems - - - - - Pages 18-23**

When you tune a receiver manually, one station after another blasts from the loudspeaker. When a motor does the job, however, a simple switch keeps the set silent until the desired station is tuned in. You learn how this is done, and also study two complete electro-mechanical systems which are presented as examples. Answer Lesson Question 7.

**5. Tuning Motors - - - - - Pages 23-28**

Although you will invariably replace defective tuning motors rather than attempt to repair them, a knowledge of how they work is interesting and useful background information for a Radio man. Pay particular attention to the methods used for reversing these motors, because trouble often develops in the reversing switches. Answer Lesson Questions 8, 9 and 10.

**6. Mail your Answers for this Lesson to N.R.I. for Grading.**

**7. Typical Receiver Diagrams and How to Analyze Them - - Reference Text 35X**

Study carefully now the explanation of the a.c.-battery portable receiver on pages 1-6, so as to familiarize yourself with these recently-developed sets which operate from a single self-contained storage battery.

Plan to study the other circuits in this text along with the first five lessons of your Advanced Course, one circuit per lesson.

**8. Start Studying the Next Lesson.**

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# Automatic Tuning Control Systems

## Introduction

**A**UTOMATIC tuning, which permits tuning in a desired station accurately and more or less instantly simply by pressing a lever or button, is rapidly reaching the point where it is being considered essential in a receiver. The average listener soon learns that he can obtain practically all desired programs from a few nearby broadcasting stations, with a clarity and freedom from interference which cannot be obtained in the case of distant or foreign stations. This listener then wants to be able to change quickly from one station to another without having to bother with accurate tuning, so he can choose the most interesting program. In making these things possible, automatic tuning has fulfilled a real need among radio listeners.

## Automatic Tuning Systems

Although manufacturers are using many different schemes for providing automatic tuning, we can divide these into three groups according to the operating principle employed, as follows:

### 1. Mechanical Automatic Tuning Systems.

By pressing a button or rotating a telephone-type dial, the listener himself provides the force required to rotate the gang tuning condenser to the setting for a desired station. This is a purely mechanical action, with no electrical switching whatsoever; tuning is essentially instantaneous.

### 2. Electrical Automatic Tuning Systems.

Pressing a button switches an entirely new set of condensers, pre-adjusted to a particular station, into the tuning circuit of the receiver in place of the gang tuning condenser. The action here is entirely electrical, hence tuning is instantaneous.

3. *Electro-Mechanical Automatic Tuning Systems.* Pressing a button closes the circuit to a small electric motor, which then rotates

the gang tuning condenser to a desired station. Electrical switching here causes a mechanical force to be applied to the gang tuning condenser. A certain amount of time is required, once a button is pressed, for the motor to complete the tuning process.

In all three systems, the initial adjustments which insure accurate automatic tuning to desired stations must be made by the radio dealer at the time of the installation. Printed tabs having the call letters of the desired stations are attached to the push-buttons themselves or to the escutcheon surrounding the buttons, to identify the station selected by each button.

## Problems of Automatic Tuning

Proper design and installation of an automatic tuning device does not insure accurate automatic tuning indefinitely; many other factors must be considered in the tuning device itself and in the circuits which are controlled.

*Mechanical Vibration.* All three automatic tuning systems are subject to vibrations and jars, especially at the time the buttons are pressed, which may alter the initial adjustments. The inaccuracies of tuning introduced by vibration may not be noticeable when local stations are tuned in on a receiver having a reasonably broad response, but when automatic tuning is employed on a highly selective receiver for the reception of semi-local or distant stations we really have a serious problem. Automatic frequency control (A.F.C.) is an effective solution, though somewhat expensive; improvements in the mechanical design of the automatic tuning system can greatly reduce undesirable effects of vibra-

tion, making tuning sufficiently accurate without A.F.C. provided that automatic tuning is restricted to local stations.

*Frequency Drift.* Changes in the output frequency of the local oscillator due to changes in temperature present another serious automatic tuning system problem, particularly when A.F.C. is not used. Oscillator frequency drift is most noticeable in the high-frequency bands of an all-wave receiver, but there can be appreciable frequency drift in the broadcast band as well. In a conventional manually tuned receiver, this frequency drift can be compensated for by shifting the tuning dial setting slightly after the receiver has warmed up and reached a constant internal temperature, but with automatic tuning systems the listener has no means of correcting for this drift.

Investigation has revealed that the chief cause of oscillator frequency drift in an ordinary superheterodyne receiver is *the fact that a rise in temperature will increase the dielectric constants of insulating materials which are present in the oscillator circuit.*\* Some dielectric materials are affected more than others by changes in temperature; when materials such as molded plastics, bakelite, low-grade rubber insulation and waxed paper insulation are present in the oscillator circuit, the result is an increase in the effective capacity in the oscillator tuning circuit as temperature goes up, producing a reduction in the oscillator output frequency

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\* As you will recall from earlier lessons, the dielectric constant of a material is equal to the capacity of a condenser with that material between its plates divided by the capacity of the same condenser when only air is between the plates. The higher the dielectric constant of the material used in a condenser, the higher will be its capacity.

and consequently lowering the I.F. signal frequency.

By using only dielectric materials like mica and ceramics (which are little affected by temperature) for oscillator circuit parts, engineers have succeeded in reducing frequency drift. In addition, special temperature-compensating condensers are often shunted across the regular condensers in the oscillator tuning circuit. These condensers are each made up of a fixed metal plate and a movable plate to which is connected a bi-metallic strip. Air forms the dielectric between the two plates of this compensating condenser. Increases in temperature cause corresponding changes in the shape of the bi-metallic strip, and these in turn cause the movable plate to move farther away from the fixed plate, reducing the capacity. The amount and nature of compensation required to offset frequency drift due to temperature changes varies with each receiver design.

*Muting.* In some mechanical and electro-mechanical tuning systems the receiver is tuned quite slowly through many stations before the desired one is reached, creating an annoying situation if the receiver is not silenced in some way during the actual tuning process. It is for this reason that the audio system is often silenced or muted during the time the tuning mechanism is being rotated.

*Eliminating A.F.C. While Tuning.* An A.F.C. circuit tends to hold on to stations while passing through them, so it is perfectly possible for a receiver to "hang on" to a station adjacent to that corresponding to a particular push button. For this reason a switch is built into the automatic tuning mechanism to disconnect the A.F.C. system temporarily while the station is being tuned in.

## Mechanical Automatic Tuning Systems

Mechanical automatic tuning systems may be divided into two general groups according to the manner in which they are operated by the listener:

1. *Rotary or telephone dial types*, in which the listener himself provides the rotary motion which turns the tuning mechanism to the correct setting for a desired station. Automatic stops prevent him from moving beyond the correct setting.

2. *Direct push types*, in which the listener applies a direct push or force to a button or lever. Either a gear, cam or lever arrangement is used to convert this force into the rotary motion required to turn the tuning condenser to the correct setting for a desired station.

*Telephone Dial Type.* The basic principles involved in a telephone dial type of mechanical automatic tuning system are illustrated in Fig. 1. An analysis of the problems involved in this purely imaginary system will make it easier for you to understand the more complicated systems actually used.

In Fig. 1A the rotor and stator of a conventional tuning condenser are shown in the minimum-capacity or highest-frequency position. As the rotor meshes more and more with the stator plates, the capacity increases and the receiver tunes to a lower frequency. A circular metal disc, on the rear side of which are several metal pins or pegs, is attached to the shaft of this tuning condenser. Suppose that a 1300 kc. station can be tuned in by rotating the rotor clockwise through the angle marked  $\theta_1$ ; if the pin on this circular metal disc makes contact with a stop stud the instant this tuning condenser setting is reached, we can readily see that this 1300 kc. station will be tuned in automatically.

Now suppose that turning the tun-

ing condenser rotor through an angle  $\theta_2$  tunes in a 1100 kc. signal. A pin placed at the position marked 1100 on this circular disc will automatically stop the rotor at the correct position for this station. As many additional pins as are desired can be placed upon the circular metal disc, one for each station.

At once we see a mechanical difficulty. The pin for the 1300 kc. station will prevent the rotor from turn-

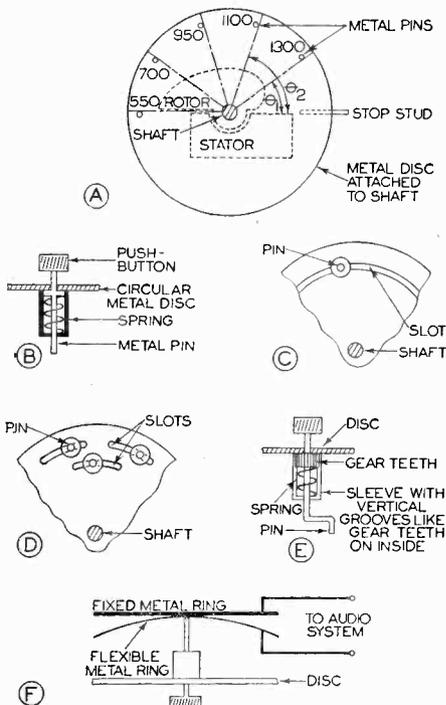


FIG. 1. Basic principles of the telephone-dial type mechanical automatic tuning system.

ing to the positions required for lower-frequency signals. Suppose, however, that the pin is made in the manner shown in Fig. 1B, with a spring which normally holds the pin outward so it can pass over the stop stud. Pressing in the button and rotating the entire mechanism clockwise causes this pin to strike the stop stud, stopping the rotor at exactly the correct position for a desired station.

Still another mechanical difficulty arises. With the construction shown in Fig. 1A, the positions of the pins cannot be changed and consequently the choice of stations is limited to those selected at the time of construction. We could eliminate this objection by mounting the pins in a circular slot like that shown in Fig. 1C, so they could be moved to the correct position for a desired station. This arrangement prevents selecting adjacent stations, however (each push-

pin is made in the form of a crank arm which can be rotated to any desired position when the button assembly with its gear teeth is pulled out of the correspondingly grooved sleeve. In this way the effective end portion of the pin can be set at any position within the swing area of this crank arm, eliminating entirely the need for slots in the circular metal disc.

To silence or mute the receiver when a button is pressed and the mechanism rotated, the pin could well

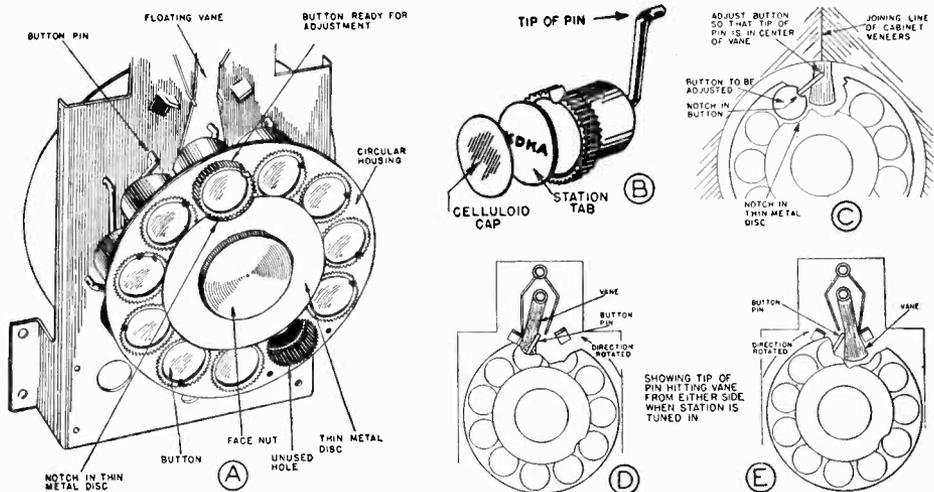


FIG. 2. Constructional details of the Emerson automatic dial mechanism as used in models AT-170, AT-181 and other Emerson receivers. When a station is tuned in, the button selected for that station will be near the top of the dial. One of the eleven holes shown at A is not used, and is covered by a blank space on the bakelite cover plate. When operating this dial mechanism, the direction in which the dial is rotated must be such that the blank space does not have to pass across the top of the dial, for the rotor plates are then either completely meshed or completely unmeshed, and cannot be rotated farther. Since this dial rotates through  $360^\circ$ , and ordinary tuning condensers rotate only through  $180^\circ$ , it is obvious that a gear drive is used between the dial shaft and the condenser shaft.

button has a certain minimum thickness), but we could overcome this difficulty with a staggered arrangement of slots, as illustrated in Fig. 1D.

Obviously, a staggered slot arrangement would not give a pleasing arrangement of push-buttons on the receiver dial. A symmetrical arrangement can be secured, however, by utilizing the construction shown in Fig. 1E; here the position of the button is fixed on the metal disc, and the

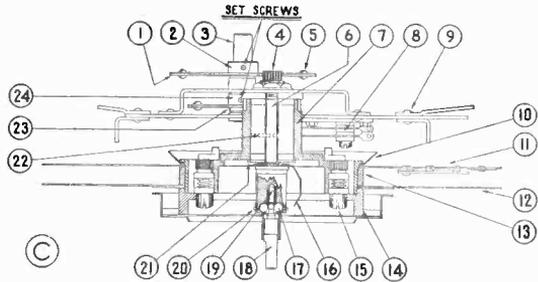
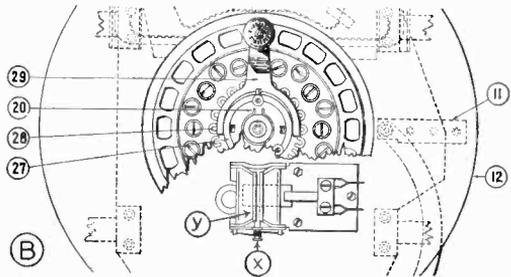
be made to depress a flexible metal ring as it moves, causing this ring to contact a fixed metal ring behind and thus short the input of an audio amplifier stage, as illustrated in Fig. 1F. Releasing of the push-button after the station setting has been reached would open the audio circuit muting switch.

*Emerson Automatic Dial.* A mechanical automatic tuning system of the telephone dial type is used on a number of Emerson receivers. The

operating principle is essentially the same as that illustrated in Fig. 1E.

Constructional details of this Emerson automatic dial mechanism are shown in Fig. 2A. A number of push-buttons, each mounted on a pin which is shaped like a crank arm (see Fig. 2B), are arranged symmetrically on the circular dial. A station call letter tab, protected by a transparent celluloid cap, fits into a circular recess in the top of each button. The

genious pivoted or floating vane serves to stop the tuning condenser assembly at exactly the same point regardless of the direction in which the dial mechanism is rotated. A spring normally holds the vane in the middle position shown in Fig. 2C, and serves also to absorb the shock when the button pin is suddenly forced against the vane. Fixed stops, shown on each side of the vane in Figs. 2D and 2E, prevent the vane from swing-



Courtesy Philco Radio & Television Corp.

FIG. 3. Three views of an automatic tuning system used on a number of Philco receivers which also have A.F.C. The numbered parts on these diagrams are all identified in the Philco service bulletin on automatic tuning; the important parts are also identified in this lesson.

sides of each button are grooved in much the same manner as a gear, as also are the holes in which the buttons slide; this procedure prevents a button from turning in its hole yet allows it to be moved in or out.

To operate this Emerson automatic tuning dial, the button corresponding to the desired station is pressed and the entire dial then rotated until the pin hits a stop which prevents further rotation; this tunes in the desired station correctly. In this unit an in-

ing beyond a definitely limited angle.

Since this particular mechanism is used chiefly on inexpensive receivers which do not employ A.F.C., no provisions are made for blocking the audio system or shorting the A.F.C. system during the tuning process.

*Philco Automatic Tuning Dial.* The general appearance of the Philco telephone type automatic tuning dial with cover plate removed is shown in Fig. 3A. In using this unit, the lever arm is swung to the position corre-



repeated for each other station selected.

*Direct Push Types of Mechanical Automatic Tuning Systems.* A number of different mechanisms are being used to convert an ordinary direct push on a button into rotation of a tuning condenser to the correct setting for a desired station; let us look over a few of them.

First of all, keep in mind that a gang tuning condenser must be rotated through an angle of  $180^\circ$  (one-half of a complete revolution) to cover an entire tuning band. Because of the mechanical difficulties involved in converting a direct push into a full  $180^\circ$  of rotation, it is customary to use an auxiliary shaft which can be rotated through a maximum angle of  $90^\circ$  or less, with the gears between this shaft and the tuning condenser shaft to step up this rotation to the required amount.

*Heart-Shaped Cam with Lever and Roller.* In receivers employing this

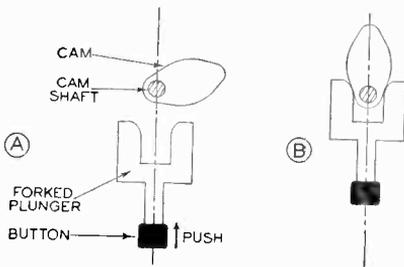


FIG. 5. Egg-shaped cam and plunger mechanism used in the mechanical automatic tuning systems of some Philco receivers.

particular tuning unit, there will be one complete set of parts like those in Fig. 4A for each station which is to be tuned automatically. A downward pressure on the station-selecting button is applied to the heart-shaped cam through a roller at the other end of the lever arm, forcing the cam to rotate to a position which brings the roller to point G, closest to the center of the shaft. For example, with the

cam in the position shown, pressure on the button will cause the cam and shaft to rotate clockwise. The cams for the different stations are mounted side by side on a cam shaft which is geared to the tuning condenser shaft,

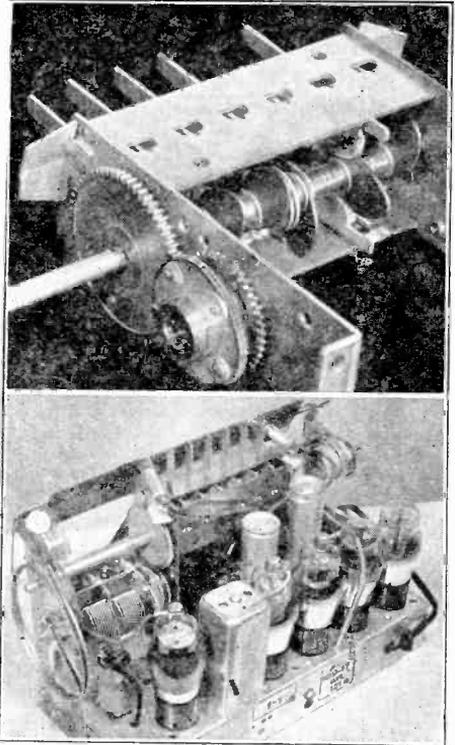


FIG. 6A (above). Mechanical automatic tuning unit used in a number of Philco receivers. One plunger is in its depressed position, and its forked end is holding the corresponding cam in the position shown in Fig. 5B.

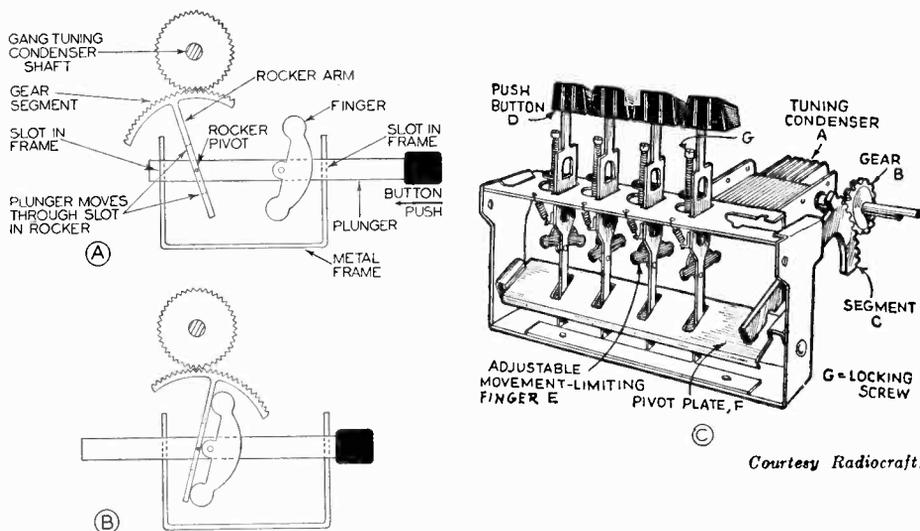
FIG. 6B (below). Chassis of Philco model 39-17 superheterodyne receiver, with automatic tuning mechanism visible at the right of the gang tuning condenser. Bakelite buttons are pushed over the ends of the six plungers after the chassis is mounted in its cabinet. Note the gear drive between the cam shaft and the tuning condenser shaft. The tuning control knob is placed on the cam shaft; the locking screw which holds all cams rigidly in position (by friction with spacing washers) fits into the center of this knob.

the cams being separated by spacing washers and held in position by friction. A mechanical locking device is provided for locking each cam rigidly in position once it is adjusted for a station.

**Cam with Plunger.** In some systems a straight plunger, with or without a roller, is used in place of the lever, as indicated by the dotted lines at the left in Fig. 4A. The push-button tuner shown in Fig. 4B is an example of this particular mechanism; a  $\frac{5}{8}$ -inch movement of the button here provides a  $60^\circ$  maximum rotation of the cam shaft, and this can be stepped up to  $180^\circ$  with a 3-to-1 gear ratio.

tween the cam shaft and the tuning condenser shaft is clearly visible in this photo.

**Finger and Rocker.** A finger and rocker mechanism which provides mechanical tuning in still another manner is illustrated in Figs. 7A and 7B, "a sketch of this unit" is shown in Fig. 7C. For each station there is a plunger (flat metal strip) sliding freely through two slots in opposite sides of a metal frame. At



Courtesy Radiocraft.

FIG. 7. The diagrams at A and B show one of the finger and rocker units used in the mechanical automatic tuning systems of some Crosley receivers (including the Crosley Safety-Tune auto radio), while the sketch at C shows the complete tuning unit with a gear drive to a gang tuning condenser. Tightening the screw on the plunger locks the finger rigidly in position. To set up a button for a station, this screw is loosened so the finger can rotate, the button is pushed all the way in, the station is tuned in manually, and the screw is tightened to lock the finger at the correct angle. A spring returns the button to its normal position when pressure is released, leaving the tuning condenser at the correct setting for the desired station.

**Egg-Shaped Cam with Forked Plunger.** In another system of the direct push type, illustrated in Fig. 5A, the cam is somewhat egg-shaped and the roller is replaced by a U-shaped or forked metal piece. Pressing the button makes the forked plunger take the position shown in Fig. 5B, holding the cam in a definite position. Photographs of the Philco version of this arrangement appear in Figs. 6A and 6B. The gear drive be-

one end of this plunger is the push-button; clamped to one face of the plunger is a metal "finger" which can be set at any desired angle to the plunger and held in position by a locking screw and clamp arrangement (omitted from Figs. 7A and 7B to simplify the diagrams, but shown in Fig. 7C). Pressing in a button makes the rocker rotate to the same angle as the finger; on the rocker is a gear segment which meshes with a gear on

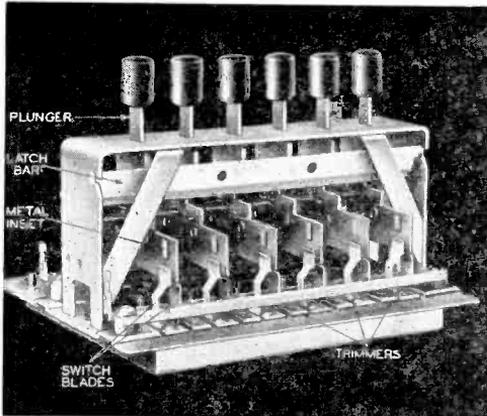
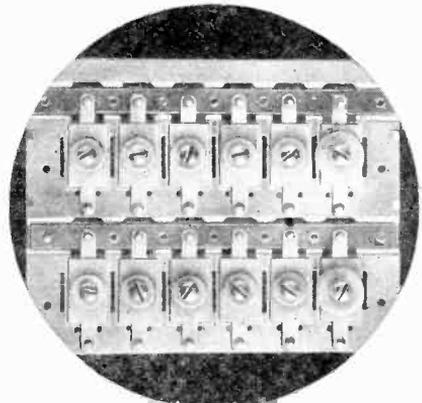


FIG. 8A. Sprague electrical automatic tuning unit with built-in trimmer condensers. The switch blades in the foreground are normally held apart by the strip of insulating material mounted on each plunger; when a button is pressed, the metal inset moves down between the blades, shorting them and closing the circuit to one set of trimmer shaft condensers.



Courtesy Sprague Products Co.

FIG. 8B. Bottom view of the Sprague unit, showing the trimmer condensers and their adjusting screws. Each push-button controls one upper and one lower trimmer on this gang assembly. One terminal of each trimmer is grounded to the frame of the unit, and this frame is in turn grounded to the receiver chassis.

the tuning condenser shaft and thus provides the correct tuning condenser setting for the station assigned to that button.

In many of the receivers which incorporate a mechanical automatic tuning system, some form of compensating condenser is employed in the oscillator circuit to compensate for temperature changes which might otherwise alter the accuracy of the automatic tuning system. The tuning action is so rapid that audio silencing switches are usually omitted.

### Electrical Automatic Tuning Systems

Instead of rotating the tuning condenser when a new station is desired,

electric automatic tuning actually removes the variable condenser in the tuned circuits and replaces them with new condensers which were previously adjusted to the correct values for that particular desired station.

Push-button switching mechanisms like that shown in Fig. 8A are used in electrical automatic tuning systems. When one of the buttons on this unit is pressed down, the button which formerly was down is released, removing that set of condensers, and an entirely new set of condensers is switched in. The entire process of switching is practically instantaneous. It is common practice to mount the set of pre-adjusted condensers right on the switching mechanism. In Fig.

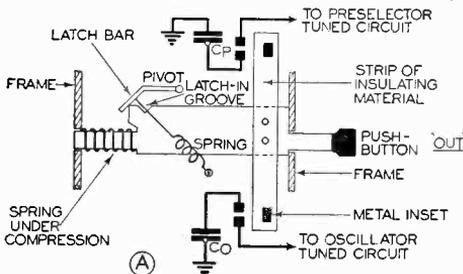


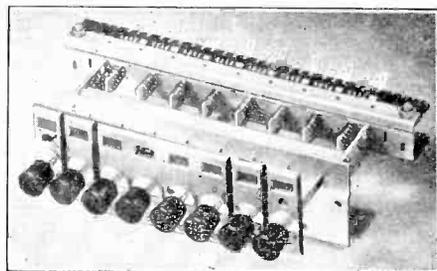
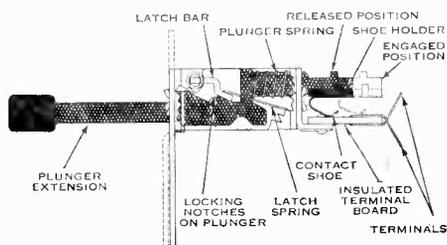
FIG. 9. These two diagrams show the construction of one of the push-button assemblies in the Sprague unit (Fig. 8) and also show the relative positions of the parts when the button is in its normal or "out" position (A) and when the button is pushed "in" (B).

8B is a bottom view of the unit in Fig. 8A; as you can see, there are two trimmer condensers, each adjusted by a screw, for each of the buttons on the unit.

Now let us see exactly what happens when one of the push-buttons in the unit shown in Fig. 8 is pressed. The diagram in Fig. 9A gives a side view of the button assembly in its *out* and *open* position, while that in Fig. 9B shows the same assembly with the button pushed in. First of all, observe that the button is on the end of a sheet metal plunger which slides freely in and out through slots in the frame of the switching unit. A spring

tuned circuit of the receiver. The fact that one terminal of each tuning condenser section in a receiver is grounded makes it possible to use this simple switching system.

A separate push-button and plunger assembly like that shown in Fig. 9A is required for each station which is to be automatically tuned, but only one latch bar is needed for the complete switching mechanism. Pressing of any button lifts up the latch bar; this serves to release any other plunger which may have been previously pushed in and held by the latch bar. Since stations can be tuned in just as fast as the buttons can be



Courtesy P. R. Mallory & Co., Inc.

FIG. 10. Photo and cross-section diagram of Mallory-Yaxley multiple push-button switch. Trimmer condensers are not built into this unit, but can be mounted either on the sides or at the rear of the unit; each trimmer must then be connected to one of the soldering lugs on the insulated terminal board. This unit is available with many different combinations of contacts and terminals, and can be used to switch in three trimmer condensers at a time in receivers having three tuned circuits.

under compression, acting between the plunger and the left-hand frame in Fig. 9A, normally holds the plunger entirely out. When the button is pushed in, a pivoted flapper or latch bar falls into a groove in the plunger, as indicated in Fig. 9B, preventing the plunger from returning to its normal *out* position.

A strip of insulating material, with a U-shaped metal piece set into each end, is attached to the plunger. Pressing in the button causes each metal piece to move between a pair of switch contacts, shorting the contacts and thus inserting a set of pre-adjusted trimmer condensers in the

pressed, there is no need for audio silencing switches or A.F.C. releasing switches.

Although many different forms of switching mechanisms are being used for condenser substitution purposes in electrical automatic tuning systems, all have essentially the same construction as that shown in Fig. 9. There will be some form of plunger for each push-button, with switching contacts or blades mounted on the plunger but usually insulated from it. Likewise there will always be a latch bar to hold a plunger in when a button is pressed. A photograph of one other switching mechanism employing

these principles, together with diagrams showing its construction, are given in Fig. 10.

*Typical Circuits.* In Fig. 11 is shown the circuit for the preselector and oscillator sections of a typical superheterodyne receiver employing two tuned sections. Tube *VT* is a conventional pentagrid converter. Condenser  $C_P$  of the gang tuning condenser tunes the preselector coil  $L_2$  to resonance, while  $C_O$  tunes the oscillator coil  $L_3$ . Only three connections need be made to this circuit to incorporate electrical automatic tuning; one lead from the push-button switching mechanism goes to point *x*, which is the stator terminal for preselector tuning condenser  $C_P$ . Another lead goes to point *y*, which is the stator terminal of the oscillator tuning condenser  $C_O$ , and the third lead is grounded to the tuning condenser frame or to the chassis.

*Semi-Automatic Connection.* Observe that the gang tuning condenser section in Fig. 11 is still present in the circuit after the push-button pushing mechanism is connected; this is known as a semi-automatic connection, because the gang tuning condenser must be turned to its minimum-capacity (highest frequency) setting before automatic tuning can be used. When this is done, pressing of any one of the four station buttons serves to tune in a station. For example, pressing button 1 places trimmer condenser  $C_{P1}$  across the preselector tuned circuit and places trimmer condenser  $C_{O1}$  across the oscillator tuned circuit.

Since an automatic tuning system provides for reception of only a limited number of stations, it is obvious that some means for restoring manual tuning must be incorporated in the receiver. (Manual tuning cannot be used when the automatic tun-

ing system is connected, because of the fact that one set of trimmer condensers will always be in shunt with the gang tuning condenser section.) An extra button is often provided on a push-button tuning for the sole purpose of restoring manual tuning. This button is designated as *M* in Fig. 11, and serves to open the two ungrounded leads to the push-button switching mechanism. Pressing button *M*, which is often marked

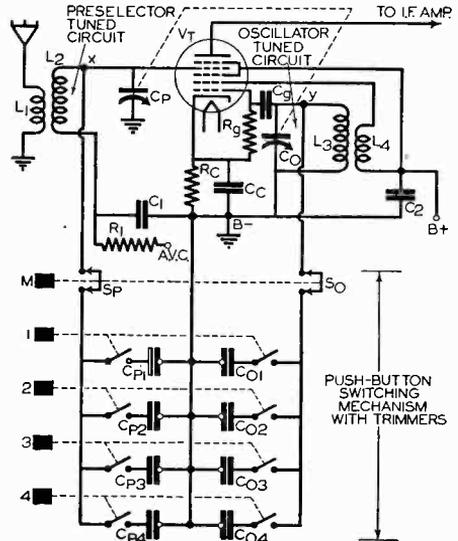


FIG. 11. This circuit shows a semi-automatic connection of an electrical automatic tuning unit to a superheterodyne receiver employing a pentagrid converter tube. Pressing button *M* restores manual tuning; pressing any other button inserts a set of preadjusted trimmer condensers (one trimmer for the oscillator tuning circuit and one for the preselector) in the tuning circuits of the receiver, after first breaking previous trimmer connections. Each set of trimmers is adjusted initially to exact resonance for one desired station.

MANUAL, opens switches  $S_P$  and  $S_O$  and at the same time releases any other button which may have been pressed in.

*Full-Automatic Connection.* Naturally, it is somewhat of a nuisance to turn the tuning dial to its highest frequency setting whenever automatic tuning is desired. This undesirable characteristic of the semi-automatic connection can be eliminated by

modifying the switching mechanism so it will disconnect both sections of the gang tuning condenser from the receiver circuit whenever automatic tuning is used. Fig. 12A illustrates one way in which this can be accomplished, and Fig. 12B shows how the MANUAL button could provide the required double-pole, double-throw switch action.

Oftentimes it is undesirable to have long leads running from the receiver chassis to the push-button tuning mechanism (because of the inductance of these leads); for this reason

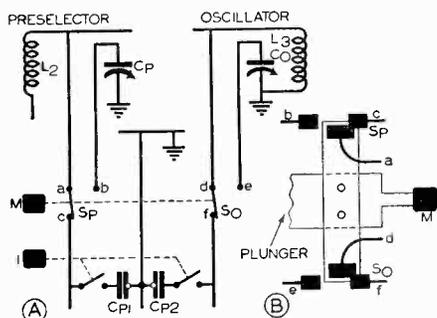


FIG. 12. At A is a simplified circuit showing a full-automatic connection of an electrical automatic tuning unit to a superheterodyne receiver having two tuned stages. Pressing button M restores the gang tuning condenser connections and breaks the two connections to the tuning unit; pressing any other button releases M, disconnecting the gang tuning condenser sections, and inserts a set of trimmers in the tuned circuits. The sketch at B shows how this switching could be accomplished with a plunger like that in Fig. 9.

the manual-tuning switch is sometimes incorporated in the band-changing switch. In this case one setting of the switch might be labeled BROADCAST MANUAL and another setting labeled BROADCAST AUTOMATIC. In one receiver the change-over from manual to automatic operation is accomplished by pushing the wave band switch knob in or out; this particular switch also operates dial lights which illuminate the words *automatic* or *manual* to indicate which particular tuning system is being used.

**Tuning by Coil Substitution.** Just as with mechanical automatic tuning systems, the oscillator tuned circuit can be the cause of frequency drift in an electrical automatic tuning system. To secure better frequency stability, some manufacturers are using a push-button switching system to substitute adjustable coils instead of trimmer condensers in the oscillator tuned circuit. Special coils employing pulverized iron cores which can be moved by means of an adjusting screw to change the inductance of the coil are used for this purpose. A fixed condenser, usually of the temperature-compensating type, provides the necessary capacity for the oscillator circuit. Because of the higher cost of variable-permeability iron core coils, they are usually used only in the oscillator circuit. A slight change in trimmer condenser capacity will have far more de-tuning effect in the oscillator tuned circuit than in a preselector tuned circuit.

**Initial Adjustments.** It is neither advisable nor necessary to make each adjustable part in an electrical automatic tuning system cover the entire 540 to 1500 kc. broadcast band. A more economical and stable construction is secured by limiting the tuning range of each adjustable coil or trimmer condenser to a definite section of the broadcast band; for example, one set of adjustable parts may be designed to tune from 540 to 900 kc., another set may cover the range from 700 to 1300 kc., and the third and final set might cover the range from 1000 to 1500 kc. Notice that there is enough overlapping between these three groups so that a station near the limit of one group may also be tuned in by another group.

Receivers employing electrical automatic tuning are usually adjusted by the servicemen as a part of the

installation job, rather than by the listener. First of all, it is wise to prepare a list of the stations which have been selected by the listener for automatic reception. Assign one push-button to each of these stations, making sure that each button is in the proper frequency group, as specified in the instruction manual supplied with the set. The station call letter tabs, also supplied by the manufacturer, can now be affixed to the buttons or the escutcheon. The receiver should be turned on during this preliminary work, so it can warm up for twenty to thirty minutes and reach its stable operating temperature. If you suspect that the I.F. amplifier may be out of alignment, be sure to check this and realign if necessary before making any preselector and oscillator adjustments.

To set up a button for a station, tune in the station manually and note the nature of its program at that time. Now push in the button assigned to that station (if a semi-automatic connection is used, be sure to turn the tuning condenser to its minimum-capacity position before adjusting any trimmer condensers); locate the oscillator trimmer condenser or variable inductance controlled by that button, and adjust until the desired station is heard with maximum audio output. For best results do not depend upon your ears, but use an output indicator or the tuning indicator in the receiver (if available). With this done, locate the preselector trimmer condenser which is controlled by this button and adjust for maximum output in the same manner; you will note that this adjustment is quite broad, whereas the setting of the oscillator trimmer was quite critical. Repeat this procedure for each other push-button.

Radiotricians sometimes prefer to

use a signal generator which is set to the station frequency and fed into the receiver input as a guide for initial adjustments of the trimmer condensers; if the signal generator is quite accurately calibrated, only slight additional adjustments of the trimmers will be needed after the signal generator is removed.

Oftentimes the initial adjustments will change slightly during the first

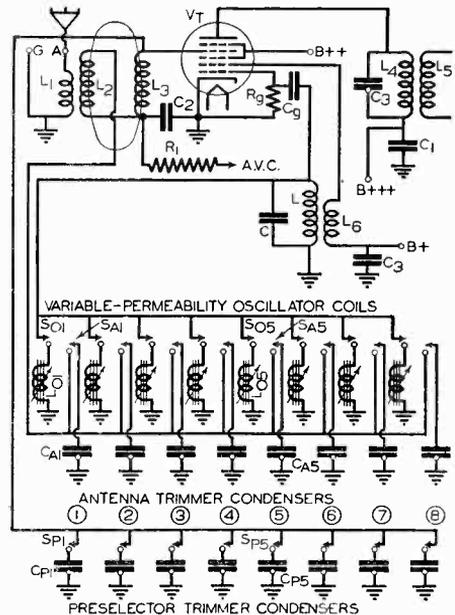


FIG. 13. Circuit diagram of the tuned sections of the RCA-Victor model HF-1 high-fidelity broadcast band superheterodyne receiver, designed for automatic reception of eight different local stations. Eight push-buttons control the electrical automatic tuning system; there is no tuning dial or knob.

few days after they have been made. When the receiver does not have A.F.C., it is a good idea to return in about a week and make final adjustments which will correct for any changes which may have occurred.

*RCA-VICTOR Model HF-1.* This is an excellent example of a receiver employing a push-button switching mechanism for substituting a pre-adjusted variable inductance in the

## Electro-Mechanical Automatic Tuning Systems

oscillator tuned circuit and substituting a pre-adjusted trimmer condenser in each of the two preselector tuned circuits. No manual tuning control is provided. The circuit diagram for the tuned sections of this receiver is given in Fig. 13; vacuum tube *VT* is a conventional pentagrid converter tube, with the first two grids acting as the oscillator grid and plate respectively. The oscillator tuned circuit is made up of condenser *C*, fixed inductance *L*, and one of the adjustable-core oscillator coils shown just below in the diagram. Coil *L* acting alone with condenser *C* resonates below the correct frequency, and the pre-adjusted coil which is shunted across *L* by the switching mechanism lowers the net inductance in the circuit exactly the correct amount for a particular station. Coil *L*<sub>6</sub> serves as the oscillator feed-back coil.

Two tuned circuits in the preselector make up a band pass tuner. In one of these circuits is fixed inductance *L*<sub>2</sub> and one of the pre-adjusted antenna trimmer condensers which is controlled by the push-button switching mechanism; the other preselector tuned circuit is made up of coil *L*<sub>3</sub> and one of the trimmer condensers. All three input circuit coils, *L*<sub>1</sub>, *L*<sub>2</sub> and *L*<sub>3</sub>, are mutually coupled inductively.

Provisions are made for automatic tuning of eight different stations. The switching action is as follows: Pressing button 1 closes switch *S*<sub>01</sub>, shunting *L*<sub>01</sub> across oscillator coil *L*, and simultaneously closes switches *S*<sub>A1</sub> and *S*<sub>P1</sub>, inserting pre-adjusted trimmer condensers *C*<sub>A1</sub> and *C*<sub>P1</sub> in their respective preselector tuned circuits. Pushing in button 5 would first release all switches controlled by button 1, then close switches *S*<sub>05</sub>, *S*<sub>A5</sub> and *S*<sub>P5</sub>, which would place parts *L*<sub>05</sub>, *C*<sub>A5</sub> and *C*<sub>P5</sub> in the tuned circuit.

Electro-mechanical automatic tuning systems can be divided into two general types according to the method of operation involved: 1, *the non-homing system*; 2, *the self-homing system*. Each type will generally include the following sections:

1. A small electric motor which drives the gang tuning condenser through speed-reducing gears and which can be reversed by means of a switch.

2. A switching mechanism which can be adjusted to stop the driving motor at predetermined positions which correspond to the gang tuning condenser settings for desired stations.

3. A group of push-button-controlled switches, each of which starts the motor and connects into the motor circuit the proper switch mechanism for stopping the motor at the correct point (these may be located at any reasonable distance away from the receiver, making remote control tuning possible).

4. A means for silencing the audio system of the receiver during the interval when the motor is driving the tuning condenser, *in order to prevent annoying blasts of sound as the receiver is tuned past strong undesired stations*; a means for releasing temporarily the A.F.C. system while the tuning motor is in motion or just after it stops, *in order to allow the desired station to "take hold" of the A.F.C. system*.

In the non-homing system the motor always drives the variable condenser in the same direction in which it was rotated by the previous automatic tuning action. If this direction is toward the desired station, the motor stops when it has driven the tuning condenser to its correct setting. If the direction is away from the desired station, the motor is automatically reversed when the tuning condenser reaches the limit of its rotation, and the condenser is then driven past its original setting to the desired new setting. In this non-homing system it is important that the motor stop instantly when its circuit is

opened by the switching mechanism, for otherwise it might coast long enough while power is off to drive the tuning condenser beyond the correct setting. This is called a non-homing system because the motor does not always "go home" to the correct new setting immediately.

In the self-homing system, the electric motor always travels in the correct direction to bring the tuning condenser to the desired new setting in the quickest possible time; if the

Variable condensers can be made which will rotate through a full revolution, but it then becomes difficult and costly to make the rotor plates sufficiently rigid.

*Non-Homing Electro-Mechanical Tuning Systems.* The essential features of a non-homing system are illustrated in Fig. 14. First of all, observe that a small reversible electric motor drives the shaft of the main tuning condenser through a chain of speed-reducing gears. These gears

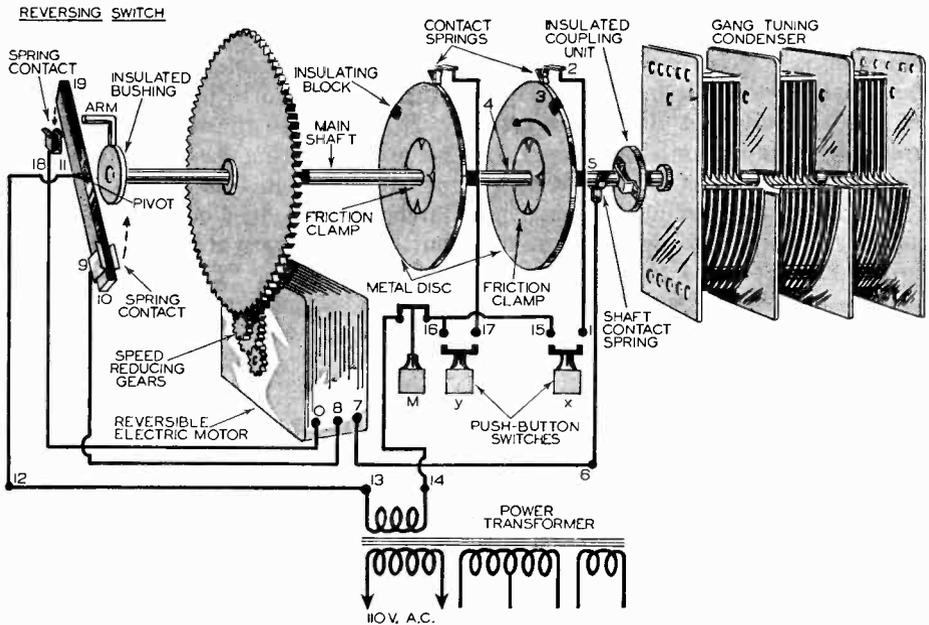


FIG. 14. Essential features of a non-homing electro-mechanical automatic tuning system.

motor overruns the proper stopping position because of inertia after power has been cut off, it will instantly reverse and correct its error automatically.

In both the self-homing and non-homing systems the variable tuning condenser cannot swing through more than 180° (one-half of a complete revolution), because the rotor plates of the tuning condensers used in modern receivers are given added rigidity by a metal strip at one end.

are necessary because the midget motors used in these systems rotate at a speed higher than one thousand revolutions per minute, a speed which is far too great for tuning purposes.

For simplicity we will consider a system having only two station-selecting buttons, designed as *x* and *y* in Fig. 14. For each button there is one metal disc on the extended tuning condenser shaft, held rigidly in place by a friction clamp. A small block of insulating material is set into the circumfer-

ence of each disc, as indicated. Above each disc is a contact spring which normally makes contact with the disc but which is insulated from the disc whenever the insulating block is directly under the spring. Each contact spring is connected to the switch controlled by its push-button; the push-button switching mechanism is the same as that used in electrical automatic tuning systems, with a latch bar serving to release a previously pushed button before holding in the button for a desired station.

Now let us see what happens in this non-homing electro-mechanical system when button  $x$  is pushed in. We see immediately that the switch controlled by this button shorts together points  $1$  and  $15$ , so let us start at point  $1$  and trace the circuit through the motor to point  $15$  again. Follow through from point  $1$  to point  $2$ , then through the contact spring to point  $3$  and the metal disc provided for this button. From the disc we trace through the tuning condenser shaft from point  $4$  to point  $5$  and then through a shaft contact spring to points  $6$  and  $7$ . From point  $7$  we know that current must flow through the motor and come out either at point  $O$  or point  $8$ . Tracing from point  $O$ , we find an open circuit at point  $18$  and know that current cannot flow over this path at this time. We therefore trace from point  $8$  to point  $9$  and then to point  $10$ , the other terminal of the reversing switch. From  $10$  we go through the metal arm of the reversing switch to point  $11$ , to point  $12$ , to point  $13$ , through a secondary winding on the power transformer (which provides the required voltage for the motor) to point  $14$ , through manual tuning switch  $M$ , and finally we are back at points  $15$  and  $1$  again.

With this complete circuit for the

motor between its terminals and the source of power, the motor begins to rotate, and the direction will be such that the tuning condenser shaft and the metal disc will rotate in a counter-clockwise direction (as indicated by the arrow on the metal disc) until the insulating block in this disc comes directly under the contact spring. This breaks the circuit between points  $2$  and  $3$ , opening the motor circuit and stopping the motor. If the position of this metal disc on the condenser shaft is properly chosen, the tuning condenser setting will now be exactly correct for receiving the station assigned to button  $x$ .

Now suppose that we desire to receive the station assigned to button  $y$ . We push in this button, and the latch bar lifts just enough to release button  $x$  and open the circuit between points  $1$  and  $15$ . The latch bar then holds in button  $y$ , and the corresponding switch shorts points  $16$  and  $17$ . Since the arm at the left end of the condenser shaft has not yet touched the reversing switch, contacts  $9$  and  $10$  are still together and the circuit traces from point  $17$  through the left-hand metal disc, through the shaft, through the shaft contact spring to point  $7$ , through the motor to point  $8$ , through contacts  $9$  and  $10$  and the reversing switch to point  $11$ , and then through the power transformer winding and switch  $M$  to points  $16$  and  $17$ . The motor rotates in the same direction as it did for button  $x$ , and the insulating block on this metal disc moves away from its contact spring. The motor continues rotating in this counter-clockwise direction until the arm at the end of the shaft flips the reversing switch over to a position which opens the circuit between points  $9$  and  $10$  and closes the circuit between points  $18$  and  $19$ . Now the circuit through the motor is from point

7 to point *O* and the internal connections of the motor are such that this reverses the direction of rotation, making the tuning condenser rotate in a clockwise direction. This continues until the insulating block has moved under the contact spring and opened the motor circuit.

The speed of the motor is so high that this tuning action takes place in a few seconds. During the tuning process, the station-selecting knob on the panel of the receiver is rotating and the dial pointer is moving, since both are driven by the tuning condensershaft. (For simplicity, the manual tuning mechanism has been omitted from the diagram in Fig. 14.)

*Self-Homing Electro-Mechanical Automatic Tuning Systems.* The basic features of a self-homing system are shown in Fig. 15. First of all, we note that there is no separate reversing switch for the motor; an ingenious switching mechanism automatically starts the motor in the correct direction, eliminating the need for reversing the motor during the tuning process.

In this mechanism, a single circular disc of insulating material is permanently and rigidly mounted on the tuning condenser shaft. To the edges of this disc are fastened two semi-circular metal segments, with their ends separated by insulating segments *J* and *K* so there is no electrical circuit between them. The lead marked 6 connects metal segment *v* to one metal slip ring which is mounted on but insulated from the tuning condenser shaft, and the lead marked 9 connects the other metal segment (*w*) to the other metal slip ring. Surrounding and resting upon the two semi-circular metal segments are a number of sliding contacts or brushes, one for each push-button, which can be set at any required position along

the circumference of this disc.

Now let us push in button *x* and trace through the motor circuit to see what will happen. The switch controlled by button *x* has only two terminals, 11 and 1, which are shorted together when the button is held in by the latch bar. Starting at terminal 1, we trace to point 2 and through the manual tuning switch *M*, through power transformer winding  $S_M$  and then to terminal 3 on the motor. If terminal 4 is the other motor terminal in the circuit, we get a clockwise rotation of the tuning condenser shaft, and if terminal *O* is in the circuit we

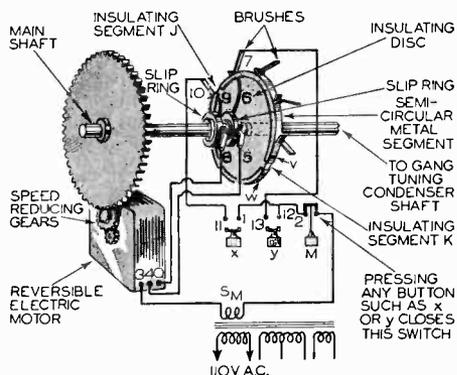


FIG. 15. Essential features of a self-homing electro-mechanical automatic tuning system.

get a counter-clockwise rotation of the shaft. Let us trace from point 4 first; we go to slip ring contact 5, then through the slip ring and through connecting lead 6 to metal segment *v* on the disc. None of the brushes which are resting on this disc provides a closed circuit to point 1 again, so we know that motor current does not take this path.

Returning to the motor, we trace from point *O* to slip ring contact 8 and through this slip ring and connecting lead 9 to metal segment *w*. Brush 10, resting on this metal segment, connects directly to point 11, which is shorted to point 1 by the

switch, and hence we have a closed circuit for the motor. The motor rotates the shaft in a counter-clockwise direction until brush 10 is directly over insulating segment *J*. This breaks the circuit, and the motor stops. If the motor coasts far enough after the circuit is broken so that brush 10 touches metal segment *v*, the motor circuit is instantly closed through its other path (4, 5, 6, 10, and 11 to 1) and the motor starts up in the reverse direction rotating the shaft in a clockwise direction. This quickly returns brush 10 to its position above the insulating segment, and since the motor has not had a chance to build up speed, it stops instantly. If brush 10 is properly located, a station will now be tuned in correctly.

Now suppose we wanted to receive the station assigned to button *y*. We press in this button, releasing button *x* and shorting points 12 and 13. Current now flows over the path 12-2-3-4-5-6-7-13-12, and the motor turns the tuning condenser shaft in a clockwise direction until brush 7 is directly over insulating segment *J*. If the motor "over-runs" the insulating segment, brush 7 will contact segment *w* and the motor will be reversed just long enough to correct the error. In this way a self-homing or self-correcting automatic tuning action is obtained.

Although only two station-selecting buttons have been shown in Figs. 14 and 15, as many additional buttons can be used in both these systems as are desired. In each case, one additional manual tuning button *M* must be pressed to open the main motor supply lead when manual tuning is desired. Since the common latch bar acts upon all buttons, the mere act of pressing a station button releases the manual tuning button, closing its

switch and restoring automatic tuning.

With non-homing systems, one metal disc with its insulating block is required for each station which is to be automatically tuned; with self-homing systems only one large disc is required, with one brush resting on the metal segments of this disc for each station which is to be automatically tuned. The metal segments may be made wide enough so that two brushes can rest side by side or very close together on it, permitting automatic tuning of stations which are only 10 kc. apart.

### **Audio-Silencing and A.F.C.-Releasing Switches for Electro-Mechanical Systems**

There are five methods in general use for silencing the audio system, of which the first is the most common:

1. Grounding the grid of an audio tube.
2. Shorting the terminals of the loud-speaker voice coil or shorting the primary winding of the output transformer.
3. Shorting the diode load resistor in the second detector circuit.
4. Applying a large negative C bias to an audio tube in order to block the tube.
5. Applying a large negative C bias to the tubes in the frequency converter and the I.F. stages of a superheterodyne in order to reduce the sensitivity of the R.F. system.

Two methods are in general use for removing A.F.C. action while the tuning mechanism is in operation or for temporarily releasing the A.F.C. system when tuning is completed:

1. Shorting the output of the A.F.C. discriminator sections.
2. Applying a large negative C bias to the tubes in the frequency converter and I.F. stages. (Reducing the input voltage to the A.F.C. system in this manner prevents the system from holding a station which is even a small amount off resonance.) This method also provides audio silencing.

Regardless of which method is used for silencing the audio system and releasing A.F.C. action, the switches

used must be interlocked with the automatic tuning mechanism. The switches can be actuated either by mechanical movement of some part of the tuning system or by the flow of current through the motor circuit.

*Switch Control by Motor Shaft Movement.* An electric motor consists essentially of a rotating part, called the armature or rotor, and a stationary part called the stator. The interaction of the magnetic fields produced by these two parts causes rotation. For greatest rotating force, the rotor must be exactly in the center

receiver operation and A.F.C. action.

When voltage is applied to the motor, the rotor instantly centers itself by moving to the right, as indicated in Fig. 16B. This causes a collar on the shaft of the motor to press against a lever arm which actuates a multi-contact spring switch, shorting all contacts to ground and thus grounding the audio grid and shorting the discriminator output resistor. When current through the motor is interrupted, the rotor-centering force disappears and the spring again pushes the rotor to the position

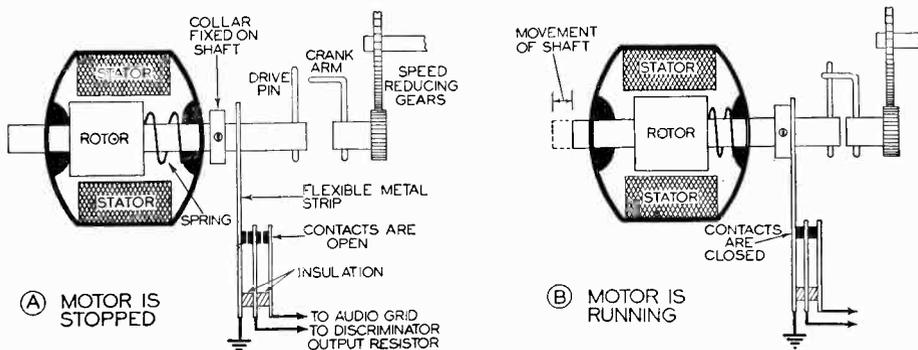


FIG. 16. Details of audio-silencing and A.F.C.-releasing switches which are actuated by thrust (end-wise movement) of the motor shaft when voltage is applied to the motor of an electro-mechanical automatic tuning system.

of the stator; if the rotor is off center to one end or the other, it will be magnetically pulled into the center whenever voltage is applied to the motor. This endwise movement of the rotor is used in some electromechanical automatic tuning systems to actuate the audio silencing and A.F.C. releasing switches.

The arrangements of parts for a switch control action of this type is shown in Fig. 16A. A spring on the rotor shaft, pressing between the rotor and the frame of the motor, serves to push the rotor off center when the motor is stopped. Under this condition the contacts of the switches are open, permitting normal

shown in Fig. 16A, opening the audio and A.F.C. circuits.

The end movement of the rotor shaft is also used to engage the motor with the speed-reducing gears which drive the gang tuning condenser shaft. The movement of the rotor shaft to the right when voltage is first applied causes a drive pin on the motor shaft to engage with a crank arm on the shaft of the smaller gear, as indicated in Fig. 16B. When current to the motor is interrupted by the station-selecting system, the drive pin moves away from the crank arm, allowing the motor to coast to a stop without causing undesirable rotation of the gang tuning condenser. The

friction load on the gang tuning condenser itself is sufficient to make it stop instantly when the driving force is removed. The switches shown in Fig. 16 may, of course, be used for any of the other audio silencing and

of any station-selecting button sends current through the relay coil as well as the motor. The relay armature is attracted to the relay core by the magnetic flux created by the relay coil, and this movement of the armature causes the audio grid and A.F.C. load contacts to be grounded. When the station-selecting mechanism interrupts the current to the motor, the current flow through the relay likewise stops, allowing the spring to move the armature away from the core and open the audio and A.F.C. circuits.

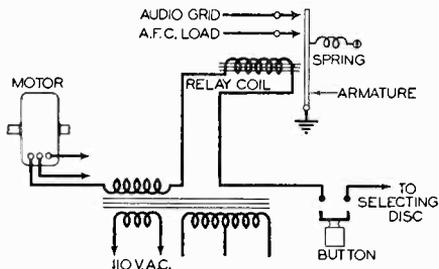


FIG. 17. Method of using a relay in the motor circuit to actuate audio-silencing and A.F.C.-releasing switches in an electro-mechanical automatic tuning system.

A.F.C. releasing methods which are employed by receiver designers.

*Switch Control with Electro-Magnetic Relays.* The flow of current in the main lead to the motor of an electro-mechanical automatic tuning

*Electrical Circuit (without Switches) for Audio Silencing and A.F.C. Releasing.* In this circuit, the A.C. voltage for the A.C. motor is also applied to a twin rectifier stage, and the resulting D.C. voltages are made to increase the negative C bias on an audio amplifier tube enough to silence the audio system, and to increase the negative C bias on the

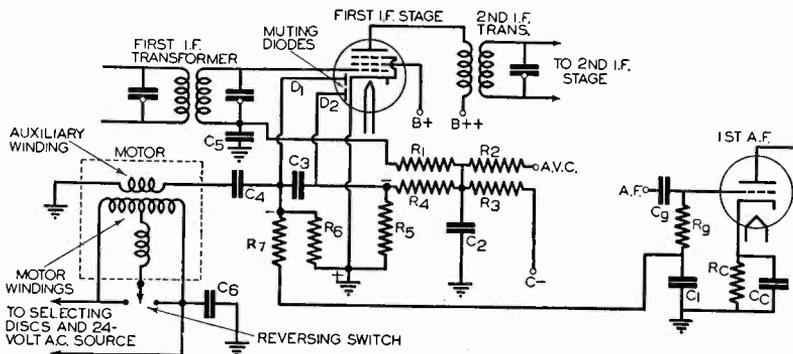


FIG. 18. Circuit diagram showing connections for an electrical system which silences the audio amplifier and releases the A.F.C. system during the operation of the electro-mechanical automatic tuning system. This scheme is used in some Motorola receivers. The use of a duo-diode-pentode in the first I.F. stage eliminates the need for an extra muting tube.

system can be used to actuate an ordinary A.C. relay having contacts which are connected to silence the audio system and release the A.F.C. system, since this motor current flows only during the tuning operation. The circuit arrangement is quite simple and is shown in Fig. 17; the pressing

I.F. stages in order to prevent the A.F.C. system from acting during the tuning operation.

A typical circuit arrangement is shown in Fig. 18; the A.C. voltage developed by an auxiliary winding on the motor core is applied to the two diode plates of the first I.F. tube

through D.C. blocking condensers  $C_4$  and  $C_3$ , the return path being from the grounded cathode to the grounded end of the auxiliary winding. On alternate half cycles the diode sections act as shorts across resistors  $R_5$  and  $R_6$ , and the current pulses flowing through these resistors on the remaining half cycles produce rectified voltage drops across them.

The D.C. voltage across  $R_6$  acts in series with the normal C bias on

the I.F. tube grids. Condenser  $C_6$  places the motor circuit at R.F. ground potential.

*RCA Electro-Mechanical Automatic Tuning System.* This electro-mechanical system is of the non-homing type, and consequently the motor will sometimes drive the gang tuning condenser away from a desired setting until the limit of condenser rotation is reached and a reversing switch is tripped. A top view

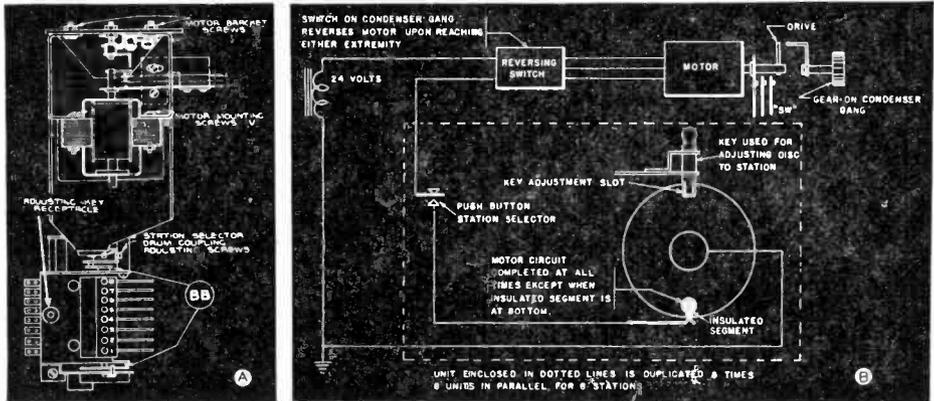


FIG. 19. RCA electro-mechanical automatic tuning system of the non-homing type. Shaft coupling units are designated as *BB*.

the first A.F. tube, driving this tube sufficiently negative to prevent audio amplification. Condenser  $C_1$  and resistor  $R_7$  make up a filter which smooths out ripples in the rectified D.C. voltage across  $R_6$ .

The D.C. voltage developed across  $R_5$  when the motor is operating acts upon the grid of the I.F. tube through resistors  $R_4$  and  $R_1$ , and drives this grid sufficiently negative to reduce the strength of the I.F. carrier signal and thus release the A.F.C. system. Note that this A.F.C. releasing voltage acts in shunt with the A.V.C. voltage and the normal C bias voltage for the I.F. tube. Condensers  $C_2$  and  $C_5$  are filter condensers, while resistors  $R_2$ ,  $R_3$  and  $R_4$  serve to isolate the various negative voltage sources acting upon

of the tuning mechanism is shown in Fig 19A; the motor, the speed-reducing gears, the A.F.C. and A.F. shorting switches, and the pin and crank arm drive are all essentially the same as for the system shown in Fig. 16, while the reversing switch is like that shown in Fig. 14.

The simplified diagram in Fig. 19B shows how this system operates. The metal disc with an insulated segment and a slot on opposite sides is one of eight discs which are mounted on a common shaft and held in position by friction; beneath each disc is a spring contact. During manual tuning the metal discs turn with the gang tuning condenser but the electric motor remains motionless since the pin on its shaft is not engaged with

the crank arm on the speed-reducing gear.

To make the preliminary station-setting adjustment for this RCA mechanism, one of the buttons is pressed, and after the motor has stopped, an adjusting key (provided with the unit, and kept in a special adjusting key receptacle when not in use) is inserted in the adjusting hole corresponding to this button; this places the key in the slot on the metal disc, exactly as shown in Fig. 19B. The A.F.C. system is turned off by means of a switch on the receiver panel, and the receiver is now tuned manually to the station desired for that button. Removal of the adjusting key com-

shorting switches are closed. The motor stops when the metal disc has turned to the position where the contact spring is over the insulated segment; if the gang tuning condenser reaches the limit of its travel before this occurs, the reversing switch changes the direction of motor rotation.

*General Electric Electro-Mechanical Automatic Tuning System.* Diagrams for this non-homing system are given in Fig. 20. As you can see, no metal station-selecting discs are used; instead, a metal arm with a knife-edge spring contact is mounted on the tuning condenser shaft, and swings over a number of metal contact but-

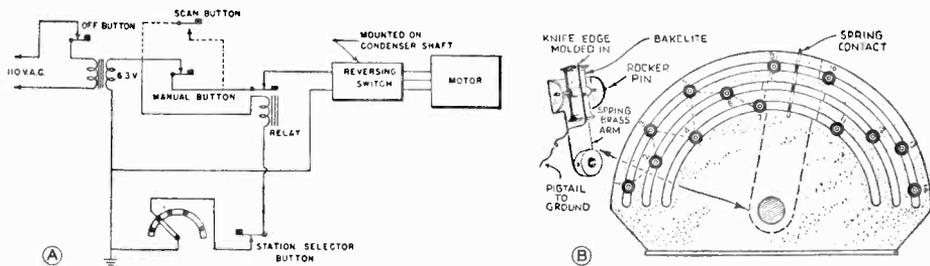


FIG. 20. General Electric non-homing electro-mechanical automatic tuning system.

pletes the adjustment for this station; the process is then repeated for each other button. It is not necessary to lock the metal discs in position, since there is sufficient friction to prevent them from slipping during normal receiver operation.

The operation of this automatic tuning system is as follows: The pressing of the push-button station selector in Fig. 19B closes the circuit to the motor through the reversing switch and through the contact spring and metal disc; the motor starts up in the direction of its previous rotation, with its rotor automatically centering in the stator so that the drive pin engages with the crank arm on the gear and the A.F.C. and A.F.

tons which are set into slots in an insulating panel. The tuning motor drives the tuning condenser shaft through reducing gears; when the limit of rotation is reached, a reversing switch changes the motor direction and starts the contact arm moving back over the contacts again. All this is shown in Fig. 20A.

The circuit connections for one contact button are indicated in Fig. 20B; here is how the system works. When the *MANUAL BUTTON* is pressed, it opens the motor supply circuit and raises the latch bar enough to release all other push buttons. The latch bar also releases the *OFF BUTTON* at this time, thus closing the primary circuit of the power transformer.

Under this condition neither the relay nor the motor gets power (because all station selector button switches as well as the manual button switch are open), and the receiver can be tuned manually.

When the *STATION SELECTOR BUTTON* is pressed, the latch bar releases the *MANUAL BUTTON*, closing its circuit. Power is thus applied to the motor, and it rotates the gang tuning condenser and the rotating contact arm until this arm is directly over the contact button corresponding to the station selector button which has been pressed. This closes the relay circuit, and its armature pulls in, opening the motor circuit and stopping the motor. The relay remains closed until the pressing of another station selector button switch opens its circuit. The A.F. system is silenced and the A.F.C. system is temporarily released by means of a conventional switching arrangement operated by the end-thrust action of the motor (this is not shown in Fig. 20).

Another interesting feature of this General Electric system is the non-latching *SCAN BUTTON*. When manual tuning is in use, pushing of the scan button will close the motor circuit, and it will continue to rotate the gang tuning condenser back and forth through its entire 180° of rotation as long as the button is pressed. In this way stations can be tuned in roughly without the trouble of turning a manual tuning knob, and when a desired station is heard the scan button can be released. Accurate final tuning can then be done manually if necessary, but usually the A.F.C. system will be able to take hold instantly and tune in the desired station when the scan button is released.

To set up any station selector but-

ton, the desired station is tuned in manually while the manual button is depressed. A contact button near the contact arm is now loosened and moved directly under this arm. Labeling the station selector button with the call letters of the station tuned in completes the setting up process; this is repeated for each other button.

### Tuning Motors

Despite their small size, tuning motors require careful design and construction. They must operate

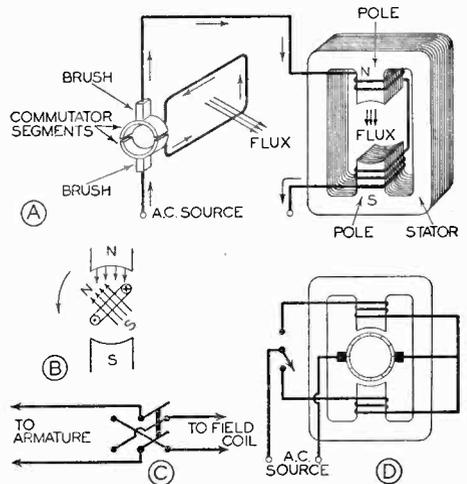


FIG. 21. Details of a series-wound tuning motor. Short arrows indicate direction of current flow for one half of cycle. Plus sign in wire indicates current flow into paper; dot in wire indicates current flow out of paper.

with as little noise as possible, must operate with little or no oiling, must not heat up excessively while in use, and must operate from an A.C. source. Usually they are designed for 6.3 volt operation, in which case they can be connected in shunt with the filament supply of the receiver; some motors are designed for higher voltages, however, and for these an extra secondary winding must be provided on the power transformer.

There are two types of small A.C. motors in general use in automatic

tuning systems: 1, series motors; 2, induction motors. The fundamental operating principles of each will be discussed.

**Series Motors.** In Fig. 21 is a simplified diagram illustrating the operating principles of a series motor. For convenience, the armature is shown as a single loop of wire whose ends are connected to two semi-circular commutator segments; in an actual motor, there will be many loops of wire and many more commutator segments, and the entire armature will be mounted in the center of the stator, between the pole pieces. Two fixed brushes make contact with the commutator segments as indicated. The armature loop and the field coils are connected *in series* across the A.C. source, as you can easily see by tracing the current from one source terminal through the circuit as indicated by arrows to the other source terminal; all motors having the armature and field coils connected in series in this way are known as *series motors*. In an actual series motor the wire loop is wound around a cylindrical soft iron core whose shaft is free to revolve in bearings at each end. The stator (the stationary part of the motor) is usually made from sheet metal laminations to reduce eddy current losses. Field coils are wound over the poles of the stator, and are connected together in such a way that both will produce flux in the same direction at any instant. When current flows through this motor in the direction indicated by arrows, the loop of wire on the armature will set up a magnetic flux in the direction indicated, and the field coils will likewise set up a magnetic flux which makes one pole north and the other pole south in polarity, as indicated.

Naturally the magnetic fields produced by the armature and stator will

interact with each other when the armature is inside the stator. For the armature position and direction of current flow shown in Fig. 21B, the armature will act as a pivoted magnet having the indicated polarity. The armature *N* pole will be repelled by the stator *N* pole and attracted by the stator *S* pole, and likewise the armature *S* pole will be repelled by the stator *S* pole and attracted by the stator *N* pole, causing counter-clockwise rotation. Since the like poles are closest together now in Fig. 21B, the repelling action will be strongest; when the armature has rotated a bit farther, the unlike poles will be closer together and the attractive force will be the strongest, reach-

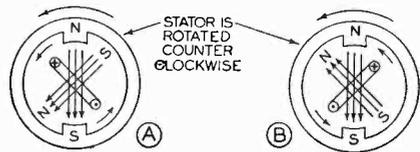


FIG. 22. Simplified diagram showing the operating principles of an induction motor.

ing a maximum when the armature *N* pole is next to the stator *S* pole. At this time the current through the armature loop is reversed in direction by the commutator, and the entire process of repulsion and attraction repeats itself, causing the armature to continue rotating in the same direction until it is again horizontal and current flow is again reversed by the commutator.

**Reversing a Series Motor.** With a given connection of armature and field coils, as indicated in Fig. 21A, the armature will revolve in a certain direction; to reverse this direction of rotation it is necessary to reverse the direction of current flow through *either* the field coil *or* the armature (*but not through both*). This can be done by inserting a double-pole,

double-throw switch in the circuit as shown in Fig. 21C. If, however, a center-tap field coil arrangement like that shown in Fig. 21D is used, a single-pole, double-throw switch will serve for reversing purposes. Note that with this arrangement only one field coil is carrying current and producing magnetic flux at any one time.

A series motor operates on either an A.C. or D.C. source, since the reversal of current flow through both the field coils and the armature does not change the direction or magnitude of the forces acting on the armature.

**Induction Motors.** All induction motors operate on the principle that when a closed or shorted loop of wire is pivoted in the center of a rotating magnetic field, the loop will rotate in the same direction as the field. Let us see why this statement is true.

Suppose we have a magnetic field like that shown in Fig. 22A, made up of an N pole and an S pole, which is being rotated by some mechanical means in the counter-clockwise direction indicated. Between these two poles is a shorted copper loop or ring which is pivoted at its center and is therefore free to turn. If the initial positions of the loop and magnetic field are as shown in Fig. 22A, a certain amount of magnetic flux will pass through the loop. When the magnetic field begins to rotate counter-clockwise, however, the amount of flux through the ring will be reduced and consequently the flux linkage in this coil or loop will be reduced. From Lenz's law we know that whenever there is a change in flux linkage in a coil, a voltage is induced in the coil in such a direction as to oppose the original change in flux linkage. This means that the loop in Fig. 22A will set up a flux to aid the original flux, and the coil

will act as a small magnet having the polarity indicated. Since like magnetic poles repel and unlike poles attract, we can readily see that the coil will rotate in the same direction as the magnetic field, as indicated by the arrows.

The shorted loop cannot rotate at exactly the same speed as the magnetic field, for then there would be no change in flux through the loop. The loop is continually slipping or lagging behind the magnetic field, but even when it slips past the pole pieces it is still forced in the same direction of rotation. Suppose that the relative positions of the coil and magnetic field

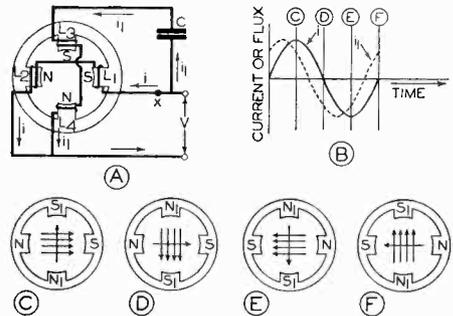


FIG. 23. These diagrams show how a condenser can be made to produce a clockwise-rotating magnetic field for a split-phase induction motor. Short arrows indicate current flow.

are as indicated in Fig. 22B at one instant, with the field traveling faster than the loop. The number of flux linkages in the loop will now be increasing, and consequently the loop will set up a flux to oppose the original flux, in an attempt to maintain a constant number of flux linkages. The polarity of the loop will be as indicated, and since like poles repel we still have the loop rotating in the direction of the magnetic field. In a practical induction motor there are a large number of shorted loops mounted on the armature, each contributing a rotational force.

Electric motor design engineers

have devised a number of simple ways for producing a rotating magnetic field by electrical means in a single-phase A.C. induction motor (the induction motors used for radio tuning purposes will operate on a two-wire, single-phase, A.C. power line, and hence are often called single-phase motors).

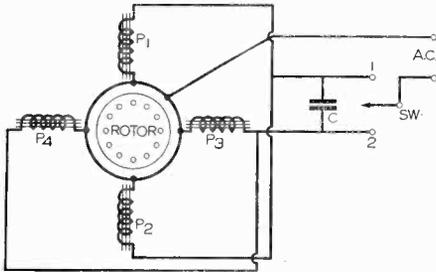


FIG. 24. Circuit of General Electric condenser split-phase induction motor as used in a number of automatic tuning systems. When switch *SW* is set at contact 1, current to poles *P*<sub>3</sub> and *P*<sub>4</sub> (which are connected together in parallel rather than in series) must flow through *C*; when *SW* is at 2, current to *P*<sub>1</sub> and *P*<sub>2</sub> flows through *C*, reversing the direction of the rotating magnetic field and thus reversing the direction of motor rotation.

*Capacitor Split-Phase Induction Motor.* In Fig. 23 is shown a method of producing a rotating magnetic field by means of a four-pole stator, with a coil wound on each pole and a condenser connected in series with two opposite coils to make them draw a current which is approximately 90° out of phase with the current through the other coils.

A.C. voltage *V*, applied to the motor terminals, sends a current *i* through field coils *L*<sub>1</sub> and *L*<sub>2</sub>, and the flux produced by this current is *in phase* with the current. The current which voltage source *V* sends through field coils *L*<sub>3</sub> and *L*<sub>4</sub> must pass through high-capacity condenser *C* and the capacitive reactance of this condenser is so high with respect to the inductive reactance of these coils that current *i*<sub>1</sub> leads the A.C. voltage *V* by almost 90°. The curves in Fig. 23*B* give the phase relationship of the two currents, *i* and *i*<sub>1</sub>, and show that

*i*<sub>1</sub> leads *i* by almost 90°; since a flux is always in phase with the current which produces it, these curves can also represent the flux produced by each pair of poles.

You can readily see that sine wave alternating current *i*, flowing through the horizontal pair of poles, will cause the flux to reach a maximum in one direction (Fig. 23*C*), drop almost to zero (Fig. 23*D*), reach a maximum in the opposite direction (Fig. 23*E*), and again drop almost to zero (Fig. 23*F*). The vertical pair of poles is going through this same complete cycle of changes in flux, but the flux is here 90° ahead of that for the horizontal poles. At each instant, the fluxes produced by the two pairs of poles combine, and it is the resultant flux which acts upon the shorted loops of wire on the armature. Referring to Figs. 23*C*, *D*, *E*, and *F*, you can see that the weaker flux in each case (shown by a single arrow line), will have little effect upon the stronger flux. Furthermore, the position of the stronger flux at each quarter-cycle is such that we have the same effect as was produced by the rotating magnetic field in Fig. 22,

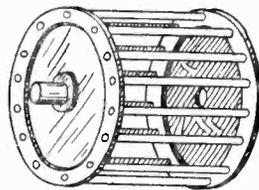


FIG. 25. Typical rotor of an induction motor, made up of copper rods which are riveted and soldered at each end to circular copper discs mounted on the rotor shaft. Before assembly, laminated sheet steel discs having slots for the copper rods are placed on the shaft inside this "squirrel-cage"; this steel core serves to reduce the reluctance of the path for the rotating magnetic flux and for the fluxes set up by the many shorted loops in the rotor.

with our magnetic field rotating in a clockwise direction in this particular case.

A counter-clockwise rotation of the magnetic field could be secured in

Fig. 23 simply by inserting the condenser in series with the other pair of coils, such as at point  $x$ . Since in tuning motors it is usually necessary to have a reversing switch, this particular type of induction motor will usually be found connected as shown in Fig. 24, with a single-pole, double-throw switch arranged to change the condenser from one circuit to the other. The condenser could just as well be replaced by a high-inductance coil, making current  $i_1$  in Fig. 23A lag rather than lead current  $i$ , for it is the difference in phase which creates the rotating magnetic field. The rotors used in this and all other types of induction motors are much the same in construction and appearance, resembling that shown in Fig. 25.

**Shaded-Pole Induction Motor.** Another method of producing a revolving magnetic field is shown in Fig. 26A. Alternating current is fed through coil  $L_1$ , which is wound on one of the poles of the motor, with the result that an alternating flux  $\theta_1$  is produced at the pole face. A short-circuited coil  $L_2$  is wound around one portion of the pole; when flux  $\theta_1$  is increasing, that portion of

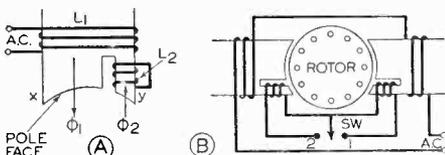
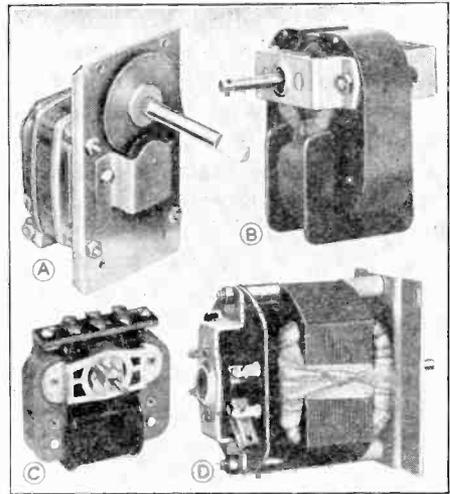


FIG. 26. Diagrams showing principles of a shaded-pole type induction motor.

$\theta_1$  which passes through coil  $L_2$  will likewise be increasing. The result is that a voltage will be induced in  $L_2$ , and this will set up a flux  $\theta_2$  which tends to oppose any change in the flux through this coil. In other words, when  $\theta_1$  is increasing,  $\theta_2$  will oppose the main flux  $\theta_1$  and the flux near point  $y$  on the pole face will be quite weak. When  $\theta_1$  is decreasing,  $\theta_2$

will attempt to prevent this decrease by aiding the main flux, and under this condition there will be a stronger flux at point  $y$  than at point  $x$ . During one-half of a cycle the region of



Typical examples of motors used in electro-mechanical automatic tuning systems.

*Courtesy Alliance Mfg. Co.*  
A—Alliance Model R shaded-pole induction motor. The shading coils are connected according to the diagram in Fig. 26B, and motor is therefore reversible. Note the speed-reducing gear assembly mounted directly on the motor.

*Courtesy Delco Appliance Div., General Motors Sales Corp.*  
B—Delco split-phase (reversible) induction motor; this unit employs the auxiliary high-inductance winding shown in Fig. 27 to produce the out-of-phase flux required for a rotating magnetic field. These motors are made for 5, 20 or 115 volt A.C. operation.

*Courtesy Barber-Colman Co.*  
C—Reversible single-phase, shaded-pole Barcol induction motor designed for 110-volt A.C. operation. A single main coil serves for both poles; separate shading coils are wound around a portion of each pole and connected as in Fig. 26B to permit reversal of the motor.

*Courtesy Delco Appliance Division*  
D—Delco reversible series motor. Two brushes resting on the commutator feed current to the armature windings in the correct direction. Field coil connections are as shown in Fig. 21D, permitting reversal of the motor with a simple single-pole, double-throw switch.

maximum flux travels from  $x$  to  $y$ ; during the next half cycle this process repeats itself, with the point of maximum flux again moving from point  $x$  to point  $y$ . We thus have a maximum flux rotating always in the same direction across this pole face,

from the main pole toward the shorted coil, which is known as a *shading coil*. The rotor will follow this rotating flux, and consequently *the rotor of an induction motor will always rotate toward a shading coil*.

The direction of rotation can be reversed by placing a shading coil on each pole of the motor, with the various coils connected together in the manner shown in Fig. 26B. When switch *SW* is at contact 1, only the right-hand shading coil will be shorted and effective, and the rotor will rotate in a clockwise direction, toward this shading coil. When switch *SW* is at contact 2, the left-hand shading coil will be shorted and the direction of rotation will be counter-clockwise.

Instead of short-circuiting the shading coils, as was done in Fig. 26B, they may be connected together and to the main voltage supply. Since their reactances will be different from the reactances of the main field coils, the shading coils will produce fluxes which are out of phase with the main fluxes, thus giving a rotating magnetic field.

*Split-Phase Induction Motor with Reactive Winding.* Another way of producing a rotating magnetic field is that shown in Fig. 27, which involves using an extra pole on which is wound a coil ( $L_3$ ) having a high inductive reactance. The A.C. line current flowing through the regular field coils  $L_1$  and  $L_2$  produces a pulsating A.C. magnetic flux, which alone is not sufficient to cause rotation. When switch *SW* is set at contact 2, placing  $L_3$  across  $L_2$ , the current through high-inductance coil  $L_3$  will lag the current through low-inductance coil  $L_2$  by almost  $90^\circ$ , and consequently the flux produced by  $L_3$  will lag the flux from  $L_2$  by almost  $90^\circ$ . This phase relationship will be maintained as each flux increases and decreases in a sine wave

manner, with the result that we have the required rotating magnetic field. When switch *SW* is set at contact 1, shunting  $L_3$  across  $L_1$ , we secure this same phase relationship between the fluxes but now the resultant or combined flux will rotate in the opposite

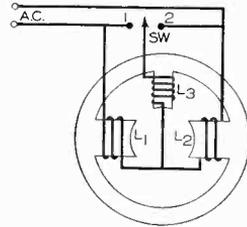


FIG. 27. Simplified diagram of a split-phase induction motor employing an auxiliary pole with high-inductance winding  $L_3$  for producing the required rotating magnetic field.

direction, causing reversed rotation of the rotor.

*Speed.* The maximum speed at which an induction motor can operate is of course the rotational speed of the rotating magnetic field. The motor can approach this speed only under no load conditions, and will slow down (due to increased slippage) as a load is applied. The maximum speed can easily be computed; the speed in revolutions per second is equal to the frequency of the alternating current in cycles per second divided by the *number of pairs of poles*.

The tuning motors used in radio receivers ordinarily have two poles, forming a single pair, since this construction has been found efficient as well as economical in cost. For 60-cycle current, then, the speed will be 60 divided by 1, or 60 revolutions per second. To obtain the speed in revolutions per minute (r.p.m.) we multiply by 60, getting 3600 r.p.m. as the maximum speed of an induction tuning motor. If a gang tuning condenser were driven directly by such a motor, the required half-revolution would take only a little more than

1/200 of a second; obviously this is far too fast. Speed-reducing gears must therefore be used with induction motors; ordinarily they are designed

to give a reduction in speed of at least 100 to 1, which means that the longest time required for a motor to tune in a station will be only a few seconds.

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## TEST QUESTIONS

Be sure to number your Answer Sheet 35FR-1.

Place your Student Number on every Answer Sheet.

Never hold up one set of lesson answers until you have another ready to send in. Send each lesson in by itself before you start on the next lesson. In this way we will be able to work together much more closely, you'll get more out of your Course, and you will receive the best possible lesson service.

1. Into what three groups (each having a different operating principle) can automatic tuning systems be divided? *mechanical, electrical, electro-mech*
2. What is the chief cause of oscillator frequency drift in an ordinary superheterodyne receiver? *temperature rise*
3. Into what two general groups can mechanical automatic tuning systems be divided according to the manner in which they are operated by the listener? *Direct push & rotary*
4. Where will a slight change in trimmer condenser capacity have more detuning effect—in a preselector tuned circuit or in the oscillator tuned circuit? *oscillator*
5. Name the two general types of electro-mechanical automatic tuning systems. *Homing & non-homing*
6. In an electro-mechanical automatic tuning system, why is the A.F.C. system released temporarily while the tuning motor is in motion or just after it stops? *To allow new station A.F.C. to drop*
7. What method of removing A.F.C. action during a tuning operation also provides audio silencing? *Placing a large negative bias on A.F. stage*
8. What two types of small A.C. motors are in general use in automatic tuning systems? .
9. How can the direction of rotation be reversed in a series motor?
10. In what direction will the rotor of a shaded-pole induction motor rotate?



## UNDERSTANDING

“Happy is the man that findeth wisdom,  
and the man that getteth understanding.  
For the gaining of it is better than the gaining of  
silver.

And the profit thereof than fine gold.  
Understanding is more precious than rubies:  
And none of the things thou canst desire are to be  
compared unto it.

Its ways are ways of pleasantness,  
And all its paths are peace.  
It is a tree of life to them that lay hold upon it.  
And happy is every one that retaineth it.”

—Adapted from Proverbs.

The world is yours. Stored in your subconscious mind is knowledge acquired through your own efforts, plus abilities bequeathed by all the centuries of the past. Once you truly understand the tremendous capacities which are within you—once you realize your responsibility to others in the world—once you acquire the knowledge needed for success in your own chosen field—you can be as great as anyone who has ever lived before you!

*J. E. Smith*

**PROFESSIONAL  
RADIO SERVICING  
TECHNIQUES**

36RH-1



**NATIONAL RADIO INSTITUTE**

**WASHINGTON, D. C.**

**ESTABLISHED 1914**

# ★ THIS IS YOUR FIRST SPECIALIZING LESSON ★

## STUDY SCHEDULE NO. 36

For each study step, read the assigned pages first at your usual speed, then reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind. Study each other step in this same way.

1. The Customer's Complaint ..... Pages 1-10

Let the customer talk, because he can often give you valuable clues to the location of the trouble, and because *his* complaints are the ones you are expected to clear up. To help you to recognize various complaints when described by the customer or when you are checking receiver performance yourself, we take up here in detail the identifying characteristics of each of the ten common complaints encountered in radio work.

2. Basic Defects in Radio Receivers ..... Pages 10-19

A radio receiver generally fails because of a simple defect in just one part. Locating this defective part takes at least 90 percent of a Radiotrician's time on the average job. The actual repair or replacement is usually a simple matter once the trouble is found. Here we review just about all of the defects which could possibly occur in each type of radio part, so you will know what to expect in the way of defects when looking for the trouble in a receiver.

3. Complete Professional Servicing Procedure ..... Pages 20-28

The ten steps in repairing a radio receiver the professional way are outlined in this section. Remember them—use them according to the instructions—and you'll save many hours of time in locating troubles.

4. Answer Lesson Questions, and Mail your Answers to N.R.I.

5. Start Studying the Next Lesson.

# PROFESSIONAL RADIO SERVICING TECHNIQUES

## The Customer's Complaint

WHEN you as a Radiotrician are called to service a radio receiver which has been in use for several months, it is reasonable to assume that some defects have developed in that receiver. The customer's complaint will be based upon a definite change in the performance of the receiver, and his description of this change in performance can be an important clue to the actual technical defect.

In rare cases, you may be called to service a receiver which has been in use only a few days. Here it is entirely possible that the customer is expecting too much of that particular set. Your job then involves explaining the limitations of various types of receivers.

One thing a Radiotrician must recognize right from the start is that no radio receiver is perfect. Modern receivers are the result of an engineering compromise between desirable technical characteristics, cost and sales-getting features. Thus, good selectivity and exceptionally high fidelity cannot both be obtained at the same time. A communication receiver has good selectivity at a sacrifice in fidelity; a high-fidelity broadcast-band receiver has poor selectivity; the average home radio has a compromise between the two; the large, high-priced home radio may have a special circuit and an extra control which permits changing the selectivity and fidelity characteristics at will.

A radio engineer or a well-trained musician might object to a receiver having harmonic distortion greater than 5%, yet the average customer would not ordinarily notice a gradual

rise in harmonic distortion to 10%. It is only when a circuit defect occurs which garbles speech and music or makes reproduction raspy or unintelligible that the average customer calls a serviceman to eliminate distortion.

► Poor sensitivity is a typical example of a complaint which is some-



A portable tester like this is excellent for a new servicing business because it serves equally well for bench work and for testing tubes in homes of customers. Here's a practical tube-testing tip—remove only one tube at a time from the receiver for testing, so you can replace tubes correctly without looking for chassis markings or tube layout diagrams. This rule is particularly important when the set has no markings on tube sockets.

times unjustified. A particular receiver may be entirely satisfactory to a person desiring only reception from nearby powerful stations, yet this same set might be inadequate for a rural listener who must depend upon distant stations for his programs. Your job as a Radiotrician is to restore the original performance of the receiver, not to

change or improve the factory design. If the receiver itself is inadequate for the customer's requirements, by all means recommend that he obtain a more suitable set for his purpose.

By checking the performance of each receiver which passes through your hands, you will quickly acquire the ability to predict the type of performance each receiver can be expected to give, and will have no difficulty in determining whether or not a customer's complaint is justified.

## LET THE CUSTOMER TALK

Ordinarily, about all you will know after a customer phones is that his receiver is not working satisfactorily and that he would like you to fix it. You can often secure additional helpful information from the customer, however, either over the phone or at his home.

First of all, let the customer describe in detail exactly what he is complaining about in the way of receiver performance. The customer may have several complaints, so be sure to let him tell you about all of them. You must remember that people's tastes differ greatly as to what is and is not good in radio reception, and you cannot do a satisfactory repair job unless you know exactly what the customer expects.

Don't ask the customer outright, "What's wrong with the set?" Use questions like these: "How is the set acting now? When did the trouble first start? Did it come suddenly or gradually? Did you notice any other trouble before this really serious one started?"

After securing this information from the customer, turn on the set and check its performance yourself to verify the customer's complaint and to see if there are other clues to the location of the trouble.

If the receiver is dead when you first try it, always check for removed tubes,

a disconnected power cord plug, a disconnected antenna or other disconnected leads. Oftentimes people will disconnect wires or remove tubes as soon as trouble develops, in the hope of preventing further damage to the receiver.

Defects like howling, noise, hum or severe distortion are easy to recognize, but they may mask other troubles which existed previously. Furthermore, defects like occasional noise, inability to pick up certain stations, fading, blasting or intermittent troubles are not so obvious and sometimes would not even be noticed by a Radio-trician during an initial check of performance. Considerable time can be saved by allowing the customer to tell how his receiver is misbehaving, hence listening to the customer is an important part of every professional servicing technique. You'll find that most people are willing and anxious to talk, if only you give them a chance and listen respectfully to what they say.

## TYPES OF COMPLAINTS

No matter how a customer describes the complaints, you can usually recognize them instantly and place them in one or more of the following groups for purpose of analysis:

1. *Dead receiver.*
2. *Noisy reception.*
3. *Annoying hum.*
4. *Squealing or howling.*
5. *Distortion.*
6. *Low volume.*
7. *Poor sensitivity.*
8. *Poor selectivity.*
9. *Station interference.*
10. *Intermittent reception.*

We will now take up in turn each of these major complaints, and learn what clues to look for in each case in order to recognize and verify the customer's complaint while checking receiver performance. In many in-

stances, we will also take up effect-to-cause reasoning techniques for localizing the trouble in a general way. It should be pointed out, however, that this lesson is just a preview of the entire field of professional servicing. Later lessons will give you more detailed instructions.

**1. Dead Receiver.** The customer may simply say that the set does not play, but even in a dead receiver there may be clues indicating the nature of the trouble. If the pilot lamps light up, you know that the set is receiving power from its source. See if all the tubes light up or get warm. Listen for the low-level hum and weak rushing sounds which are heard on almost any receiver when no stations are tuned in. Hearing these means that the power supply and loudspeaker are very likely good. Also listen for the normal rushing sound usually heard when the volume is turned up with no station tuned in (proof that the stages between the volume control and loudspeaker are good). A popping or clicking sound should be heard from the loudspeaker when the ON-OFF switch is turned on and off quickly after the receiver has been operating for a few minutes; failure to get it can mean a loudspeaker defect, power pack defect, or an a.f. output stage defect.

In the case of an all-wave receiver, try to tune in stations on two or more of the ranges. If reception occurs on any band, then the trouble is in the preselector or oscillator section for the dead ranges.

Even simple listening tests like this can yield valuable information from a dead receiver. For example, a low-level hum heard all the time the set is on, associated with a popping sound when the power switch is manipulated, means that the main power pack is functioning. Absence of all of these symptoms is an indication that the power pack is dead, the last a.f. stage

is dead, or the loudspeaker is defective.

► Other combinations of symptoms will give other conclusions as to the source of trouble, once you have learned to analyze the customer's complaint and verify it by checking the performance of the receiver yourself. Truly, a thorough check of receiver performance is an important part of professional servicing techniques.

**2. Noisy Reception.** A certain amount of noise is always present both in amplitude and frequency-modulated receivers, in phonograph amplifiers and in public address amplifiers, even when the units are operating properly. The same phenomenon of noise is present in television receivers, though here it appears as white spots on the screen rather than as sound. The constant quest of design engineers for a high signal-to-noise ratio is evidence that noise is a real problem in any system of radio reception or sound amplification.

If it were not for noise, distant broadcast band reception would be limited only by the gain of the receiver, and it would be unnecessary to have high-power transmitters.

Noise is a Radiotrician's problem only when it becomes greater than is normal for the particular system in question. Here again, considerable judgment is required to determine whether or not the amount of noise is normal. By checking performance of good receivers whenever you have an opportunity, you will quickly learn how much noise to expect from various types of sets with various types of antenna systems.

Noise which is due to something entirely outside of the receiver is called *external noise*, to distinguish it from *internal noise* which is due to a receiver defect or to limitations in receiver design.

*External Noise.* Spark-producing devices like motors, generators, vibrators, diathermy apparatus, household electrical appliances with moving contacts, electro-medical devices, ignition systems in automobiles, and oil burner electrical systems all produce a type of interfering noise which is often called *man-made static*. Manufacturers of electrical equipment are today turning out units which produce a minimum of noise interference. Special noise filters are available for apparatus which is producing excessive noise interference, and their installation is covered elsewhere in the Course.

Noise-reducing antennas on receivers will reduce the effects of external noise signals, so whenever the source of noise is something beyond your control (streetcar motors, elevator motors in buildings, and power stations are examples), you should consider noise-reducing antenna systems.

In addition to man-made interference, we also have nature's source of noise, consisting of electrical disturbances produced by local or distant electrical storms and by lightning discharges. Natural noise is known as atmospheric disturbance or static. In a broadcasting system employing amplitude modulation, little can be done to reduce the effects of atmospheric disturbances. With a good frequency modulation receiver, however, radio reception can usually be enjoyed right through the strongest local electrical storms.

*Internal Noise.* Noise originating inside a radio receiver can be due either to an actual circuit defect or to certain unavoidable and natural characteristics of radio tubes and circuits.

A poor connection in a receiver can produce a crackling noise which interferes with reception and is hence a justifiable customer complaint. Jarring of the receiver or tapping on certain parts will usually increase the in-

tensity of this noise. Your job as a Radiotrician is to locate and correct the faulty connections or locate the faulty part which is the cause of this type of internal noise.

Circuit noise exists because of the erratic motions of free electrons in conductors. These movements increase with temperature, hence the effect is called *thermal agitation*. The electron movements produce small voltages in conductors. When these voltages are in the input stage of a receiver, they are amplified thousands of times by the high-gain tubes in modern receivers, and the result is a characteristic rushing noise.

In addition, we have in every radio tube another type of noise which is known as the "shot effect." It is due to the irregular movement of electrons from the cathode to the plate inside a vacuum tube, a movement which is often compared to the falling of sand or raindrops on a tin roof.

Tube and circuit noises together produce a noise signal which is only a few microvolts at the most, except possibly in the frequency converter tube where it is much higher. This internal noise is a sort of hissing sound. It is most evident when the set is tuned off a station and the volume control advanced for maximum loudness, for the a.v.c. system is then set for maximum r.f. gain, and the a.f. system is getting the entire output of preceding stages.

To make sure that signals will override noise originating in the frequency converter tube, receiver design engineers place an r.f. amplifier stage ahead of the converter whenever cost and design considerations permit. This extra stage makes distant reception more enjoyable. Without an r.f. stage in a super, the noise in the frequency converter would be the chief factor limiting distant reception.

When confronted with a customer's complaint of noisy reception, deter-

mine first whether you have an internal or external noise problem. The simple procedure for determining this quickly is given elsewhere in the Course.

**3. Annoying Hum.** If you listen intently, you will ordinarily be able to hear hum from the loudspeaker of any receiver operating from an a.c. power line, from a vibrator-type power supply, or from a motor-generator set. Keeping this normal hum below the level at which it becomes objectionable is an important design problem.

In high-fidelity receivers which reproduce sounds as low as 30 cycles, considerably more hum reduction is required than in midget or table model receivers which do not reproduce much below 150 cycles. In any receiver, the hum should be only barely audible, and listeners in the same room should not be conscious of its presence during a program.

There are a number of receiver defects which can cause the amount of hum to increase considerably. Thus, the hum level may increase gradually as the electrolytic filter condensers in the power pack dry out, or when one half of the rectifier tube has considerably lower emission than the other half. On the other hand, the hum level may rise suddenly due to failure of a filter or by-pass condenser, or development of cathode leakage in a tube. Hum may become so intense that it overrides broadcast programs, or may be annoying only when tuning between stations.

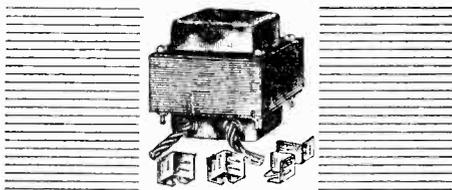
Hum is one trouble which definitely irritates the listener. The longer he listens, the more offensive the hum becomes and the more critical he becomes of hum.

**4. Squealing or Howling.** In the days of regenerative receivers, it was normal for a set to squeal and to cause squeals in nearby receivers during tuning. This type of set is fortunately al-

most extinct now, but modern receivers may still squeal or howl when certain defects occur in circuits or parts.

Squealing due to external sources is rather rare today. There is little chance that the carrier signals of two broadcast stations will beat with each other to produce a squeal, because all stations are assigned frequencies not likely to produce interference and are required to maintain operation on assigned frequencies with a high degree of accuracy.

Occasionally, an experimental regenerative receiver or the oscillator circuit of a superheterodyne will radi-



Example of a universal-mounting power transformer for radio receivers.

ate a signal and produce an audible squeal in nearby receivers. You can recognize this external squeal source by its intermittent nature and by the fact that its frequency varies as the offending set is tuned.

Internal sources of squealing are more numerous, and are covered in detail elsewhere in the Course. You may encounter squeals due to troubles inherent in the process of frequency conversion in a superheterodyne receiver. As a general rule, the number of tuned circuits in a receiver, its basic design and its i.f. value determine just how much of this trouble will occur. Also, squealing may be due to oscillation caused by an open by-pass condenser or misplaced connecting lead. The defect may be in the r.f., i.f. or a.f. system of the receiver, or can even be due to an open output filter condenser in the power pack.

Defects in the supply circuit filters of an a.f. amplifier, or defective by-pass condensers which are common to two or more screen grids of r.f. or i.f. tubes, can cause a put-put noise which is commonly called *motorboating*.

Loose or flexible elements in tubes or other radio parts, or even a thin, flexible chassis itself can give rise to a variable-pitched howl under the influence of powerful sound waves from the loudspeaker, giving an effect called *microphonics*.

The ability to recognize, localize and remove the cause of squeals, howls, motorboating and microphonics is an important part of the Radiotrician's work if he is engaged in radio servicing.

**5. Distortion.** When a customer complains that voice or music is muffled, harsh, raspy or unintelligible, you have a definite and justifiable complaint of distortion due to an internal receiver defect.

Equally important cases of distortion, however, are those situations where the customer's complaint is rather indefinite. He may say, "*It just doesn't sound right.*" Careful questioning in cases like this will probably reveal that the customer quickly gets tired of listening to the receiver, or reveals that listening for an hour or so gives him a headache or makes him so irritable that he gets up and turns off the set in disgust. Certain amounts of distortion are not noticeable as such by the average person, but an hour or more of listening to distorted reproduction produces a definite reaction.

In the case of distortion, it is important to find out whether it occurs on local or distant stations or both, whether it occurs on all bands of an all-wave receiver, and whether it occurs only at high or low settings of the volume control or at all settings. Combining a thorough check of receiver performance with this addi-

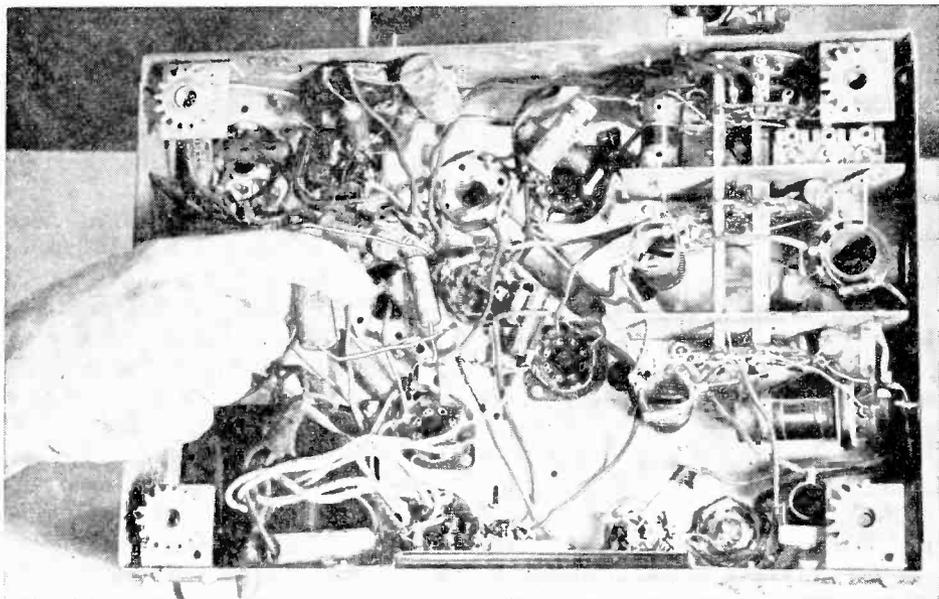
tional analysis of the customer's complaint is an important feature of radio service work, because it helps to isolate the probable location of the defect.

There will be cases of distortion in which there is nothing you can correct. Thus, selective attenuation of the side-bands from sky-wave signals, or phase cancellation of signals at a location near the skip distance limits, can give distortion.

Distortion may be due to low emission in tubes, gas in tubes, defects in supply circuits which change operating voltages, or opens or shorts in some parts or in wiring.

Distortion can be due to improper alignment, and this in turn may be either the result of normal aging of parts or due to a breakdown in some part. Distortion may exist in the loudspeaker due to a weak field, to complete absence of field excitation, to a broken spider, to a rubbing voice coil, or to a damaged or old cone. It is your job as a radio serviceman to recognize and correct abnormal distortion. In fact, it would be rare indeed, except perhaps in the case of a trained musician, that you would be called at all for distortion which was not definitely abnormal.

**6. Low Volume.** The average radio receiver is so designed that reception of a fairly powerful local station with the volume control fully advanced and with an antenna of average effectiveness will overload the loudspeaker and perhaps produce rattling. Fortunately, however, the a.v.c. system in a receiver levels out variations in signal strength sufficiently to prevent excessive overloading of the loudspeaker or other stages at normal volume control settings. The average listener rarely advances his volume control fully; he is satisfied to know that the set has sufficient reserve "power" to blast out if he does advance the control.



The quickest and simplest way to check a by-pass condenser suspected of being open is by shunting it temporarily with a good condenser while the receiver is in operation, as illustrated here. It is the general practice to discharge the good condenser after each test, either by holding both its leads against the edge of the chassis for a moment or by touching the leads together.

With the possible exception of certain inexpensive midget sets, modern receivers will give more than sufficient volume for the average room in a home, even for weak local stations and for not-too-distant stations. It takes a little experience to be able to tell whether or not a receiver is delivering a normal amount of volume, but you probably know already what the acceptable volume is for ordinary types of sets.

When the volume is definitely lower than normal both for local and distant stations, but the tuning indicator reacts normally during tuning of stations or you can still tune in the distant stations, you can be quite sure that a defect exists in the audio system. It may be a low-emission tube, low electrode voltages on a.f. tubes, or weak field excitation for the loudspeaker.

When low volume is associated with

poor sensitivity (the next complaint to be discussed), so that distant stations cannot be tuned in and the tuning indicator (if present) just barely works when stations are tuned in, look for a defect in the r.f. system.

When local stations come in with adequate volume but both volume and sensitivity are poor for distant stations, you can suspect a poor antenna system. It is easy enough to recognize the defect of low volume, but in applying professional servicing techniques you must go further and investigate every symptom which may help you to isolate the probable causes of the trouble.

**7. Poor Sensitivity.** A customer would probably describe the condition of poor sensitivity as inability to pick up his favorite distant station. Your first step is to determine whether the trouble is actually poor sensitivity, or whether it is a natural condition.

Ask if the customer has picked up that distant station regularly before, both summer and winter, with that *same receiver* and *same antenna system* in exactly that *same location*. For example, a New Yorker moving to Washington may complain of inability to get New York stations, yet his set was never capable of giving good distant-station reception.

If questioning indicates the possibility that the receiver is defective, check its pick-up performance by trying to tune in one or more distant stations on the receiver. Judge the sensitivity according to the manner in which the stations come in, making due allowance for the time of day and season of the year. Memorize the frequencies of several near-distant stations which can be picked up regularly in your locality with a good receiver, so you can tune them in quickly when checking the sensitivity of a receiver.

If a receiver fails to give adequate volume on distant stations when used with an antenna known to be in good condition, then poor sensitivity is definitely the complaint, and is probably due to a circuit defect. It should be noted that the customer will generally complain of low volume rather than poor sensitivity, although both troubles are present when the defect is in the r.f. system.

Complaints of poor sensitivity are often not justified. For example, a customer may complain that a station 500 or more miles away came in clear and loud one day, but the next day was weak or could not be heard at all. This may be an entirely normal condition which is not the fault of the receiver; the trouble can be due to variations in conditions up in the stratosphere.

Complaints of poor sensitivity may also be due to normal variations in radio reception with changes in season. Reception is generally far better in

winter than in summer, and better at night than during the day for broadcast band stations. This is a situation which most people recognize, but you will still encounter customers who will welcome an explanation of these peculiarities of radio.

You thus see that a radio serviceman is more than a trouble shooter and repairman. He must have a broad general knowledge of radio, covering the broadcasting system as a whole and covering basic design features. He should know, for instance, that a receiver with limited r.f. gain but high audio gain produces extremely loud signals on local and semi-local stations, but does not bring in distant stations.

**8. Poor Selectivity.** When the customer complains that a powerful local station prevents him from receiving a distant station on a nearby frequency, you have the condition of poor selectivity. It may be a condition inherent in receiver design, or it may be due to a circuit defect.

A leaky tuning condenser, a damp coil, conductive dirt on tube sockets, or dirt elsewhere in r.f. circuits can definitely produce poor selectivity. These defects lower the Q factors of resonant circuits, thereby lowering the sensitivity and selectivity of the receiver. In extreme cases the symptom of low volume may also be present.

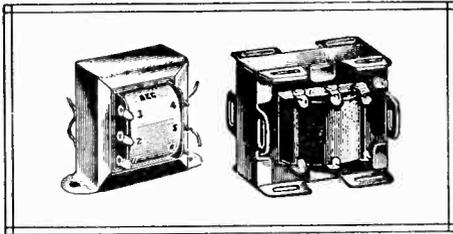
On the other hand, poor selectivity may be due to the natural characteristics of the superheterodyne circuit employed, coupled with some local receiving condition. It may be the result of several stations operating on nearly the same frequency, with atmospheric conditions particularly favorable for reception of the more distant of the two stations, so that both came in at the same time.

Whenever a customer's complaint of poor selectivity is found to be due to atmospheric conditions, an explanation

in non-technical language will satisfy and please the customer in most cases.

It is obvious that in this brief presentation of the complaints of low volume, low sensitivity and poor selectivity, we can merely mention only a few of the causes. Remember that all these troubles are taken up in greater detail elsewhere in your Course. They are mentioned here only to stress the need for more than casual recognition of the customer's complaint. The value of a thorough check of receiver performance cannot be stressed too highly in this study of professional servicing techniques.

**9. Station Interference.** There may be as many as 40 or 50 low-power stations operating on some frequencies. Under certain favorable atmos-



Examples of universal output transformers. Order one designed for use with the output tube or tubes in the set, and other specifications will usually take care of themselves.

pheric conditions, some are bound to interfere with each other, so that two stations having the same frequency are heard at the same time. This type of interference usually occurs at the high-frequency end of the broadcast band, because most of the low-power stations are crowded together here. You can recognize troubles of this sort by the fact that they occur only for a few stations, not for all signals received. Nothing can be done about it.

When interference is due to a code station operating at or near the i.f. value of the receiver, or is due to image interference or to harmonics of the oscillator beating with a high-frequency

station, a wave trap which is tuned to the frequency of the interfering station will give a solution. Many sets have a built-in wave-trap circuit.

**10. Intermittent Reception.** When a complaint exists for a short period of time and then corrects itself, you have the trouble known as *intermittent reception*. As one example, a tube filament can break in such a way that it opens after being heated for a short period of time, causing a dead receiver, but makes contact after the filament has cooled off. Reception is thus alternately good and bad, with the cycle repeating itself at regular intervals.

A break in a connecting wire inside or outside a part can cause intermittent reception. Thus, the internal connection between the pigtail leads and the metal foil of a paper condenser may break due to a strain during assembly of the condenser or during the wiring of the receiver. The break may be such that any vibration or heat in the chassis will cause a temporary open circuit. A loud portion of a program from the loudspeaker may break the connection, and jarring of the cabinet may restore the connection. Heat may likewise intermittently make and break such a defective contact.

In addition to being intermittently dead, the set may intermittently have any of the other receiver complaints, depending upon the location of the defect. Thus, if the condenser in question happens to be a screen grid by-pass condenser, the break will normally cause intermittent *oscillation*. If the condenser is a plate-grid coupling condenser in an audio amplifier, the complaint could be intermittent *weak reception* or *dead reception* during those intervals when the condenser is open.

Just imagine how many open connections you could have in the various coils, condensers, resistors, transformers, loudspeakers and tubes in a

radio receiver. Imagine further the number of possible shorts, partial shorts and leaks which could occur. Combine these with the condition whereby the defect lasts only for a short period of time, perhaps vanishing as soon as you touch the set, and you have intermittent reception—a type of complaint which taxes the ingenuity of radio servicemen to the utmost.

In the customer's home, it should suffice to confirm the customer's complaint that an intermittent exists, and secure as detailed a description as possible of the exact nature of the trouble during the duration of the intermittent defect. It is well to do this, because the intermittent defect may not appear immediately when you turn on the receiver and check its performance. Find out whether the trouble is intermittent dead reception, in-

termittent hum, intermittent oscillation, etc. Knowing which one it is, you can localize your search to those defects which will produce the type of complaint observed.

Another valuable clue to an intermittent defect is the rate at which the set becomes intermittent and the regularity of the trouble, so be sure to ask the customer how often the trouble occurs. Intermittent troubles which recur at a more or less regular rate are due to heat, whereas troubles which occur at irregular intervals are due to vibration of poor connections.

In general, if an intermittent trouble does not reveal itself in ten or fifteen minutes while you are in the customer's home, it is best to check over the antenna, ground and power cord systems to clear them of suspicion, then remove the chassis and take it to your shop for further observation.

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## Basic Defects in Radio Receivers

A radio receiver is essentially an electrical assembly of tubes, resistors, condensers, coils, transformers, a loudspeaker, switches and connecting wires.

To emphasize the large number of parts which make up an average radio receiver, we made a census of the parts and connections in a typical modern seven-tube, two-band a.c. superheterodyne. (We used the Philco Model 41-230.) We found:

- 7 tubes
- 19 resistors
- 31 condensers
- 20 coil windings
- 42 socket connections
- 136 connections of parts

**255 items total.**

A defect in any one of these parts can produce one or more of the receiver

complaints just studied. But this total is by no means a true indication of the number of defects we can have in an average receiver. Take tubes, for instance—in a typical pentode tube there can be over two dozen defects. Resistors and condensers can also cause a wide variety of troubles.

In order to get a better over-all picture of the problems encountered in radio servicing, let us now consider in detail the various kinds of defects which can exist in each kind of radio part. We need not consider mechanical defects like broken or crushed parts, since they can be spotted readily during the initial surface inspection of the receiver for obvious defects.

**1. Vacuum Tube Defects.** A tube tester is an indispensable instrument

for any radio serviceman. The following defects occur in tubes and are revealed by tube testers:

*Low Emission.* Inability of the cathode or filament in a tube to emit the normal number of electrons when heated is revealed by a low (BAD) reading on the meter of the tube tester.

*Open Elements.* Some types of tube testers will reveal open elements, but this defect can also be identified readily by the action of the circuit. An open filament is readily detected because the tube will feel cold when touched, and no filament glow will be visible. It is always wise, however, to check the tube in a tube tester, because an open in the socket or in a filament lead can cause the same symptoms.

*Shorted Electrodes.* These are readily detected either with a tube tester, by circuit action, by continuity tests with an ohmmeter, or by voltmeter tests.

*Leakage between Electrodes.* Cathode-to-filament leakage often occurs in heater-type tubes. The trouble is revealed by most models of tube testers, but is not in itself sufficient cause for discarding a tube. Cathode leakage does no harm when the cathode and filament are both grounded or at the same potential with respect to ground. Leakage is important only when the cathode is ungrounded.

*Excessive Gas.* A certain amount of gas is present in all tubes, and causes grid current to flow. Only in high-resistance grid circuits is gas objectionable, however. A voltage test across the grid resistor constitutes a satisfactory test for gas, assuming that the preceding grid-plate coupling condenser is not leaky. Gas is one defect not ordinarily revealed by tube testers.

**2. Socket Defects.** Defects in tube sockets are sometimes visible on inspection, but more often a professional servicing technique is required to iso-

late the defect. It is quite important to realize that professional techniques are required to locate even simple socket defects such as the following:

*Open Prong Contacts.* Repeated insertion and removal of a tube from its socket may spread apart the prong contacts so much that they no longer grip the tube prongs. Faulty material used in the construction of a socket will cause the same trouble. When the defect cannot be detected visually, make a continuity check between the bottom end of the suspected tube prong and its socket contact.

*Shorted Prong Contacts.* Shorts can occur between adjacent contacts on a tube socket, particularly if a number of wires are grouped together on the contact lugs or there is excessive solder. This trouble can be suspected if noise occurs when the tube is wiggled in its socket. The remedy is rearranging of connections to the socket terminals.

*Leakage.* Dust or a conductive greasy film on the surface of a tube socket will provide a leakage path between socket terminals, with the trouble being most serious in the case of leakage between grid and plate terminals. Brushing the socket with a small, stiff round paint brush or a toothbrush will clear up this trouble and also identify it by restoring receiver operation. Charring of the insulating material of a socket between the high-voltage plate terminal and other terminals may also cause leakage paths. Charred sockets should be replaced.

**3. Condenser Defects (Paper and Mica).** The method of testing a condenser depends upon the nature of its defect and upon its capacity value.

*Shorted Condensers.* A short can occur in a condenser if a surge voltage punctures the dielectric, allowing the metal foil on each side of the dielectric to make contact. An ohmmeter will

always reveal shorts in condensers. It is usually best to unsolder one condenser lead when making an ohmmeter test for a short in a part.

*Leakage.* This is the radio man's term for the condition whereby current "leaks" through a condenser due to a greatly lowered resistance between the condenser terminals. This lowered resistance may be the result of internal deterioration of the dielectric, or to an accumulation of conductive dirt on the surface of the condenser between the terminal leads. One condenser lead must usually be disconnected in order to make a leakage resistance check with an ohmmeter, because the leakage resistance value will be comparable to that of parts usually shunted across condensers.

There are many places in receivers where a small amount of condenser leakage is unimportant. An alert Radiotrician will recognize these and not waste time checking leakage in such locations. As an example, leakage in a condenser shunted across a cathode resistor is relatively unimportant. In other positions even a small amount of leakage is bad; thus, leakage in a grid-plate coupling condenser can cause distortion and other troubles. An alert, properly trained radio serviceman never makes unnecessary tests.

*Opens.* These are common both in paper and mica condensers, and occur particularly at the point where the pig-tail leads are bonded to the metal foil inside the condenser housing. The most practical test for open by-pass condensers is simply to shunt each suspected condenser in turn with a good one of approximately the same capacity while the set is in operation.

An ohmmeter test for open condensers is not very satisfactory. A large condenser (disconnected from the circuit) may take a charge from the ohmmeter battery, causing a momen-

tary deflection of the meter pointer, if it is not open. The amount of deflection depends on the ohmmeter range and on the size of the battery in the ohmmeter, as well as the capacity of the condenser. Only condensers above about .05-mfd. give a noticeable deflection, so smaller condensers cannot be checked this way at all. Shunting a suspected condenser with one known to be in good condition is the best test for open condensers.

An open can often be detected by wiggling each condenser in turn while the receiver is operating; noise will occur when the defective condenser is touched. Of course, the capacity test in a condenser tester will reveal an open.

**4. Condenser Defects (Electrolytics).** Electrolytic condensers perhaps require replacement more often than any other type of condenser used in radio equipment.

Wet electrolytic condensers become ineffective when not used for long periods of time. Modern dry electrolytics are much better in this respect, and will ordinarily give long life if not overloaded by excessive voltage and not dried out by excessive heat. Nevertheless, voltage surges and unusual climatic conditions will cause dry electrolytics to become defective.

*Opens.* These are rare in electrolytics, but high-resistance joints can sometimes occur internally at the junctions between the foil strips and the contact lugs or terminal leads, due to corrosion. Substitution of a good condenser is perhaps the most practical way to check this.

*Shorts.* A short will occur in an electrolytic condenser if an excessive voltage of the correct polarity is applied, or if voltage of incorrect polarity is applied for any length of time. In a wet electrolytic, the short will probably heal, but in dry electrolytics it is invariably permanent. An ohmmeter will reveal shorts.

*Leakage.* All electrolytic condensers have a certain amount of leakage, which is equivalent to a resistance shunted across the condenser. The leakage resistance can be measured with an ohmmeter, and will have different values depending upon the polarity of the ohmmeter connection. The leakage resistance will be larger when the positive terminal of the condenser is connected to the positive terminal of the voltage source in the ohmmeter. This is the correct connection for a check of leakage. Be sure to discharge the condenser by shorting its terminals before reversing ohmmeter leads.

An ohmmeter test gives only a general check-up of the condition of an electrolytic condenser. Better information can be obtained with a capacity tester, or by a simple substitution test. If connecting a good electrolytic in place of the suspected one clears up the trouble, you can be sure the defect has been located.

*Poor Power Factor.* A perfect condenser would theoretically have a power factor rating of zero, as compared to a power factor rating of one for a perfect resistor. Resistance in series with condenser capacity internally raises its power factor. A condenser with high power factor (approaching the characteristics of a resistor) dissipates energy just as a resistor does, and this produces heat in the condenser. This heat causes evaporation of the solution or chemical paste in the electrolytic condenser, raising the power factor still more and eventually drying out the unit entirely.

A condenser which feels hot to the touch is definitely drying out and has a high power factor. It should be replaced, because it will break down very soon due to the heat.

When you substitute a good electrolytic condenser and this clears up

the trouble, you have identified the original condenser as defective.

### 5. Tuning Condenser Defects.

Gang tuning condensers in receivers are particularly susceptible to mechanical damage because they are usually entirely exposed and have moving parts. Their troubles are as follows:

*Opens.* The mechanical construction of a tuning condenser is such that an open in its circuit is extremely unlikely; if it does occur, it will be at the terminals and can readily be repaired by resoldering. Continuity tests between the connecting leads and the rotor and stator will reveal the trouble.

*Shorts.* Mechanical strain applied to the chassis during handling, warping of the frame of the gang tuning condenser, warping of tuning condenser plates, loosened mounting screws, accidental dropping of the chassis, or tampering with the gang tuning condenser can cause a short between the rotor and stator sections. You can usually identify such a condition by a scraping sound heard when the unit is rotated. To test for shorts electrically, the coil associated with the gang tuning condenser must be disconnected temporarily if it is connected between rotor and stator plates. Shorts usually occur over limited portions of the movable range. Flaky conductive material, such as the metal plating applied to some condensers, can lodge between rotor and stator plates and cause shorts.

*Leakage.* The high-resistance range of an ohmmeter can be used to detect leakage in gang tuning condensers. The chief cause of this leakage is dust between the rotor and stator plates and on the insulating sections in the unit. The dust can be blown out or wiped out with a pipe cleaner of the type obtainable at any tobacco store.

*Poor Contacts.* Any resistance at the wiping contacts in a gang tuning

condenser or in the stator-mounting screws is rather difficult to measure with ordinary test equipment; as just an ohm or so is appreciable here. Watch for poor wiping contacts serving the rotor section. This is a frequent cause of low sensitivity and poor selectivity, and also of r.f. oscillation. These three symptoms are thus a direct clue to poor rotor contacts in many cases.

#### 6. Trimmer Condenser Defects.

Both the air dielectric and mica dielectric types of trimmer condensers are fortunately reasonably free from trouble. Only in rare cases will they become open, shorted or leaky, and they then require testing like any variable tuning condenser. The mica type is subject to capacity changes as a result of changes in temperature or normal aging, but this is usually easy to recognize because it affects the alignment of the receiver. The mica dielectric can crack or flake, causing changes in capacity and eventual shorting of adjacent plates.

A change in capacity is one defect which cannot be located by simple meter tests; you must be able to recognize the effects of capacity changes on receiver performance.

**7. Fixed Resistor Defects.** As a general rule, low-wattage units (ranging from .1 watt to 3 watts) will be of the carbon or metallized type, and higher-wattage units will have wire-wound construction, oftentimes covered with a ceramic cement. You will occasionally encounter small 1- and 2-watt wire-wound resistors molded in a bakelite housing which resembles that of some carbon resistors, but these wire-wound units will rarely have more than about 5000 ohms resistance.

Resistors which crack or break in any way can usually be spotted visually, so we will concentrate here upon defects which can be located only by tests.

*Opens.* Overloading of a resistor by sending excessive current through it can burn out the resistance material or cause an open at the point where the wire lead makes contact with the resistance material. You can check for opens in resistors with an ohmmeter.

*Shorts.* A direct short between the two terminals of a resistor is not at all common. However, it is entirely possible for resistor leads to touch each other, to touch the chassis or touch other parts and give the same shorting effect. Also, resistors encased in metal can short to the metal case anywhere. When not visible to the eye, a short in a resistor can be located with an ohmmeter.

*Changes in Resistance.* Carbon resistors are particularly susceptible to changes in resistance, whereas wire-wound resistors rarely change. Overloading of a carbon resistor or even continued use at normal temperatures will often cause a marked decrease in resistance, which increases the resistor current and overloads it still more.



Volume control potentiometers should be replaced when they cause noisy operation.

The resistance value can be checked with an ohmmeter in the usual manner. Remember, however, that carbon resistors are generally used at points where a great deal of variation in resistance value is tolerated or where the resistor is operated well under its rated wattage. Normally, variations as great as 20% in resistance value are entirely permissible.

**8. Variable Resistor Defects.** Variable resistors and potentiometers are far more subject to trouble than

fixed resistors. Since they are mechanical in operation, we have wear in moving parts to consider. As a rule, the defect will be readily apparent because rotating of the control knob while the set is in operation will cause noise or intermittent operation. An open volume control will not provide proper control of volume even though it may permit partial transfer of the signal.

*Opens.* In both carbon and wire-wound controls, movement of the contact arm over the resistance element may eventually wear away the metallic or carbon deposit, or wear down the nichrome resistance wire, creating an open. Loss of spring tension in the movable arm may also give an intermittent or full open.

Wearing away of the resistance element reduces its heat-dissipating capabilities, so that a current-carrying control unit may be overloaded by normal current or momentary excessive current after it has worn down. This causes an open by burning out the resistance element. When the defect is not visible, an ohmmeter check will isolate the trouble.

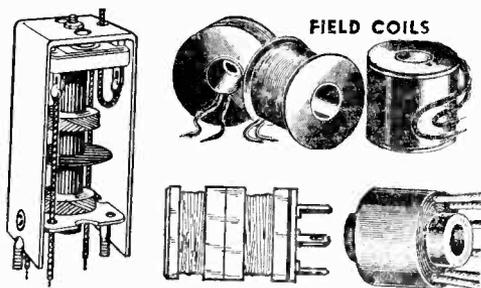
*Shorts.* As with fixed resistors, shorts are not common. In units where the metal case is "hot" and an insulating bushing is used between the chassis and the mounting bushing of the control, a defective insulating washer or bushing will often create a short to the chassis. To locate a trouble of this sort, you usually have to unsolder all leads, then test between each terminal of the control and the chassis.

*Change in Resistance.* Wearing off of the carbon or metallized material in a variable resistor or potentiometer will cause the total resistance to increase, but this is not ordinarily of importance. The chief symptom in trouble of this nature will be noise. Manipulation of the control during the initial performance check should reveal this trouble, either by noise com-

ing from the loudspeaker or by failure of the potentiometer to control volume or tone in a normal manner.

**9. Air-Core Coil Defects.** Here we are concerned with air-core r.f. and i.f. transformers as well as with r.f. chokes. Coils consist simply of copper wire and insulation, but several types of trouble can develop.

*Lowered Q Factor.* A coil which becomes damp or coated with conductive dirt will develop high r.f. resistance,



A number of different manufacturers produce exact replacement coils of all types.

which has the effect of lowering the Q factor. This in turn affects the operating characteristics of the stage in which the coil is used. For example, lowered Q factor in an r.f. coil can cause poor sensitivity or poor selectivity even in a properly aligned stage, and can cause low output in an oscillator stage. The Q factor of a coil could be measured with a Q meter, but this information would be of no value unless the normal Q factor and the permissible tolerance in Q factor were known.

Inspection of the coil and temporary adjusting of resonant circuit trimmers are the usual techniques for isolating the trouble to a coil having lowered Q factor. Trimmer adjustments will be sharp for a high Q coil, but quite broad for a low Q coil. Experience in evaluating receiver performance and adjusting tuned circuits will help you to decide when a coil should be baked

out or replaced because of lowered Q factor

*Shorted Turns.* These may develop in all coils, particularly those employing special diamond, basket or bank windings. If a great many turns are shorted out, and the normal resistance of the coil is indicated on the circuit diagram, the defect may be detected with an ohmmeter as an appreciable reduction in coil resistance. Otherwise, the action of the circuit is the only clue to the trouble. For example, in the case of an r.f. coil in a resonant circuit, more capacity will be needed to align the receiver, and both sensitivity and selectivity will be poor even after



Several mail order radio supply firms offer a coil repair service which involves rewinding of defective coils when exact duplicate replacements are not available. This service is also available for the coils of magnetic loudspeakers.

alignment. We also have the possibility that adjacent coils may touch each other, particularly when wound close together or one over the other. Here an ohmmeter test from one coil to the other will verify the trouble.

*Opens.* The windings in a tightly wound coil may break due to expansion with temperature, especially at points where the wire passes through the coil form or connects to a terminal. More often, however, corrosion at a terminal will create a high-resistance joint. In a primary coil which carries the plate current of a tube, the direct current may start electrolysis which causes corrosion and an open. Noise will be the first symptom that this condition exists. An ohmmeter test

will indicate above-normal coil resistance and high resistance due to corrosion at joints.

**10. Iron-Core Coil Defects.** Since iron-core coils are used chiefly in audio frequency and power frequency circuits, we are not concerned with changes in Q factor. Actual mechanical defects in the coil windings or failure of insulation between windings are the two chief problems here.

*Opens.* These are readily detected with an ohmmeter. They can be due to electrolysis, particularly at the terminals of filter chokes where fairly high currents are flowing through joints formed by dissimilar metals. Of course, a sudden voltage surge or continued overloading of a coil with excessive current may melt the wire and cause an open.

*Shorts.* Shorts between turns due to failure of insulation are not readily detected with an ohmmeter, but it is usually possible to detect shorts between layers of windings because this creates a greater change in resistance.

Shorts in coils are best located by their effect upon receiver operation. For example, if there is a short in a choke coil, normal current will give a lower a.c. voltage drop than normal across the coil, with hum as the symptom. A short in a power transformer will cause overheating, with eventual production of smoke and charring of the insulation, and with lowered output voltage. These same effects will appear when turns are shorted to a grounded iron core and the transformer circuit is grounded at some other point, but the short to the core can also be detected with an ohmmeter.

An odor of burned insulation and a charred appearance is always an indication of a defect in an iron-core transformer. A hot unit without these symptoms does not necessarily mean a defective unit, however; a certain amount of heat is normal in trans-

formers which are handling power. The exact amount of heat varies with different manufacturers and with different transformer uses; some transformers will operate quite cool, while others are designed to operate at the heat limit set up by the Board of Fire Underwriters.

Removal of all of the tubes is an easy way to check a power transformer for shorts; if the smoking and overheating stops when all loads are removed by doing this, you either have a broken-down filter condenser which is drawing excessive current, or you have a shorted tube or some other short to ground in a plate or screen grid supply circuit. If the smoking continues, the transformer itself is defective, or there is a short in the rectifier tube socket or in the transformer leads.

**II. Loudspeaker Defects.** In addition to becoming defective, loudspeakers are subject to normal wear and aging due to the fact that the voice coil-cone assembly is continually moving during operation. Once a defect has been isolated to the loudspeaker, you have the following possibilities to contend with:

*Weak Flux.* An open can occur in the field coil of an electrodynamic loudspeaker due to corrosion, electrolysis or over-loading. If the field coil is also serving as a filter choke, as is usually the case, the receiver will be dead. If there is a separate filter choke, there may be enough residual magnetism in the iron core of the loudspeaker to permit operation, but sounds will be weak and greatly distorted. A simple ohmmeter check will reveal an open field coil.

Weak and distorted reception will also occur when the permanent magnet in a p.m. dynamic loudspeaker loses its magnetism. You must remember, however, that there are many other possible defects in a receiver which can produce these same symp-

oms. The defect should definitely be isolated to the loudspeaker before making extensive tests on the loudspeaker.

*Open Voice Coil.* The voice coil may open at one of its joints or in the flexible leads which connect it to the output transformer secondary. The voice coil leads are subject to breakage even though extremely flexible, because these leads must connect between fixed terminals on the loudspeaker and rapidly moving terminals on the voice coil. An ohmmeter will detect an open in the voice coil if you first disconnect one voice coil lead from the output transformer.

*Grounds.* You can have a ground in the field coil, the voice coil or in a hum-bucking coil. Grounds can be found by a simple ohmmeter test between the suspected coil and the frame of the loudspeaker, if inspection of the circuit diagram shows that no ground should exist.

*Defective Spider.* The flexible material of the spider may become brittle and crack, or may lose its elasticity. When the condition is serious, the symptoms will be a peculiar type of distortion. Visual inspection of the spider will reveal the trouble in most cases.

Sometimes the spider will get loose at the points where it is glued to the cone, causing fuzzy tones. Regluing with cone cement will fix this. Partly loosened dust caps in the center of the cone will cause the same trouble.

*Off-Center Voice Coil.* As a loudspeaker ages, there is usually a certain amount of entirely normal warping and shifting of parts. Rough handling can cause this same warping and shifting, and the condition becomes serious when the voice coil rubs against the pole pieces.

Iron filings, bits of metal, or hard particles of dirt lodged in the voice coil can cause the same trouble as a shifted spider or other shifted parts.

You can detect a rubbing voice coil by pushing the cone in and out with your fingers when the receiver is turned off, for the vibration due to rubbing will be transmitted to your fingers, and you can hear the grating or scraping sound of the voice coil rubbing against the pole piece. The symptoms of an off-center voice coil are distortion of low notes and buzzing sounds. There is much more voice coil movement at low frequencies than at high frequencies, hence rubbing of the voice coil may distort men's voices without affecting women's voices.

*Cone Defects.* A cone may become hard and inflexible due to aging and drying out; the result is a rattle and failure to give normal fidelity. The outer edge of the cone, which is glued to the frame, may become loose and cause raspy, buzzing sounds and distortion at low audio frequencies. A cone may become softer than normal due to absorption of moisture, causing distortion. In a few cases the cone may actually crack or tear, causing distortion and giving a defect you can readily see on inspection. Yes, even cone defects require a certain amount of analysis for detection.

**12. Defective Connections.** Under this classification we have hook-up wire used for connecting together the various parts and tube sockets, and the tie-down terminals used for supporting small parts by their leads. A defective connection is an excellent example of a trouble which may require many minutes to locate even with professional servicing techniques, but which ordinarily can be repaired in a few seconds.

*Opens.* An open at a joint may be due to poor soldering at the factory in the first place. Remember that soldering, even with most modern production methods, is still a manual operation in which the human element is the predominating factor affecting quality.

As a rule, those entrusted with soldering operations in factories have had considerably more training in soldering techniques than the average serviceman, so opens at soldered joints are more likely to be present in equipment which was previously serviced. Oftentimes a soldered joint which looks good may be held together only by rosin, which is an insulator and hence causes an open. Corrosion and electrolysis at joints may cause opens, partial opens or even intermittent opens.

The procedure normally used to locate opens in wiring involves checking continuity from each tube electrode to the chassis, the rectifier plate or the rectifier cathode. Opens will then be indicated by comparison of ohmmeter readings with resistance values indicated on the circuit diagram.

Once the defective circuit has been isolated, the parts and connecting wires which make up the circuit can be checked with a step-by-step ohmmeter procedure, but more often the serviceman will simply push or pull on each joint in the suspected circuit while the receiver is in operation. The defective joint will cause noise when moved or jarred. Another favored procedure is application of a hot soldering iron to each joint in the circuit, to make the solder flow freely over the entire joint.

*Shorts.* Connections and bare leads sometimes short to the chassis or to an adjacent connection, particularly when terminal lugs are close together as on tube sockets and terminal strips. Where wires go through holes in the chassis, frayed insulation may allow the wires to touch the chassis and cause a short. In the case of a high-voltage lead such as a plate supply lead, sparks will be observed at the point of contact when the lead is wiggled.

Shorts may occur at joints, where the wire is bare and contact with the

chassis or some other terminal is more likely. Oftentimes terminal lugs become loose or bent, or excessive solder is used in making connections at the lugs, with the result that a short exists.

When the circuit diagram indicates that a complete circuit is isolated from the chassis, a simple ohmmeter test between that circuit and the chassis will indicate whether or not a chassis short exists.

Leakage between connections is common and may cause trouble in r.f. circuits, but drying out of a moist or damp chassis will usually clear up the trouble. If it does not, new leads or parts will be required.

**13. Switch Defects.** The contacts of the various switches used in radio equipment may corrode, causing opens or noisy partial opens. Loss of springiness in the movable contact arm can cause the same symptoms and troubles. In addition, connections to the switch terminals may be open, shorted or partially open.

Circuit continuity tests will usually reveal switch defects quickly. It is not at all uncommon for a movable contact arm or for a terminal lug of a switch to break off and cause an open, and here also an ohmmeter continuity test will reveal the defect.

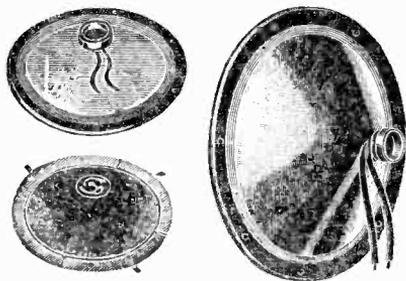
## LOCATING DEFECTS

Once the defective part in a receiver is located, any one with a little mechanical ability and a knowledge of soldering can make the necessary replacement or repair in a few minutes. The real work of a Radiotrician is locating the defective part or connection, or locating the cause of the trouble in such cases as improper alignment of tuned circuits.

When the trouble in a receiver is

simply a defective part or connection, and unlimited time is available, a person with a little training in the use of a tube tester, an ohmmeter, and perhaps a condenser tester can test each part and connection in the receiver until he locates the defective one.

Of course, the chances are good that the defective part will be located long



Exact duplicate replacement cones are made for practically all loudspeakers.

before running through the entire test procedure for all parts. But even if you were able to average 50 tests per chassis, you can readily imagine what an enormous amount of time is required when using a part-testing technique of this nature.

As a matter of fact, there are many servicemen, without the type of professional training you are now acquiring, who actually do test various parts one after another until they find the defect. With experience they learn that certain parts or connections should be checked first for each type of complaint. They become good guessers unconsciously, without knowing why, but with professional servicing techniques you can locate troubles faster than these men right from the start of your servicing career. Your techniques will work on all sets, whereas testing of parts will tell nothing at all about many kinds of receiver troubles.

# Complete Professional Servicing Procedure

The preceding analysis of the many possible defects which can occur in a radio receiver indicates that there are a great many defects which can be detected by checking voltages with a voltmeter or by checking continuity with an ohmmeter. Nevertheless, the time required for such random tests leads to great inefficiency in servicing, and particularly leads to occasional jobs which take way too much time or cannot be fixed at all.

This means that the man who relies only upon a guess-and-try method of testing each and every part in turn will frequently find himself confronted with troubles which defy detection by his testing techniques. After wasting a great deal of time in testing parts, he will be forced to resort to such simple versions of professional servicing techniques as he is able to carry out. If given sufficient time, he may eventually be able to localize the defect, for as a last resort he can begin putting in new parts one by one until he eventually does find the trouble, but what a waste of time such a procedure is!

The professional servicing procedure presented here has as its goal the correction of the receiver defect in the least possible time. This goal eliminates immediately the guess-and-try test methods which are the chief technique of the untrained mechanic. To become a professional radio serviceman—a true Radiotrician—you must master a carefully planned series of professional servicing techniques. These utilize your training, ability and experience to ferret out the trouble in the least possible time.

**Ten Steps.** A *complete* professional servicing procedure for repairing radio receivers involves a maximum of ten distinct steps, as indicated in Fig.

1. Success in carrying out Step 3 can, however, make possible the omission of up to five of these steps, and success in Step 4 may permit omission of the next four steps.

These steps are so planned that you will be led directly to the trouble in a logical, step-by-step manner, even when you do not have enough clues to lead more directly to the trouble. As your training continues, however, and as you gain more practical experience, you will soon find Step 3 assuming more and more importance. Thus, you will get a logical, direct approach to the trouble with a minimum of wasted time and effort. As you become more like the thoroughly trained professional, you will learn the short cuts which can frequently be used to save even more time.

Since the professional technique you want to develop involves use of all of these steps, let us now look into these steps in detail, to see just how they are related to each other.

## 1. Determine the Complaint.

This has already been discussed in detail, and can be summed up here in just one sentence: *Find out exactly what the customer expects you to do.*

## 2. Confirm the Complaint.

This involves trying out the set—checking its performance. The things to watch for while checking performance vary with the nature of the complaint, as you learned earlier in this lesson. In general, however, this step involves turning on the receiver, tuning to the dial settings of a few local stations and noting how they come in, then trying out the volume control, tone control, band-changing switch and any other controls with which are present, and noting how the tuning indicator reacts if one is present.

### 3. Effect-to-Cause Reasoning.

When a receiver becomes dead, hums, distorts, howls, oscillates, has inadequate volume, has poor selectivity, has poor sensitivity, or has some other symptom which you have verified by checking receiver performance, there is definitely a *cause* for the observed condition. If you can figure out what that cause is by means of reasoning, it is only logical that you check the likely cause first.

Usually, however, there are a number of possible causes for each type of complaint encountered. To utilize effect-to-cause reasoning efficiently, you must know when to give up checking probable causes and start the next step in the procedure. Once you begin checking every possible cause for an observed symptom, you are really resorting to the guess-and-try methods of the untrained radio man.

► Proper use of effect-to-cause reasoning can be best explained by means of a few examples. Let us take first the case of a dead receiver. You know that a burned-out filament in a tube will ordinarily cause a dead receiver, so isn't it logical to suspect tubes when the symptom is no reception? Since you are eventually going to test the tubes in the receiver anyway, it is quite logical and permissible to test the tubes after you have made sure that the receiver is obtaining power and its antenna-ground system is normal. When tubes are cleared of suspicion, however, you have before you dozens of possibilities because a break anywhere in a signal circuit can cause a dead receiver. This is where you should give up effect-to-cause reasoning and proceed to the next step.

► How about oscillation or howling? You know that an open screen grid-to-cathode (or chassis) by-pass condenser will usually cause a stage to go into oscillation, so it is entirely logical to suspect and check the various

screen grid by-pass condensers when oscillation is the symptom. This can be done quickly by the substitution method, placing a good condenser of approximately the correct value across each suspected screen grid by-pass condenser in turn while the receiver is in operation. If none of these tests restore normal performance, however, and there are no associated symptoms to limit the trouble to a particular section, your next move should be an inspection for surface defects, followed by the isolation procedures.

► A leaky input filter condenser will cause distortion, hum, an overheated power transformer and possibly red-hot plates in the rectifier tube. Therefore, if you observe all these symptoms, you would correctly question the input filter condenser. If it is leaking sufficiently to draw excessive current from the high-voltage secondary winding of the power transformer, it will overload this transformer and cause it to heat up excessively. Furthermore, the rectifier tube will be required to pass excessively high plate current, bombarding the plates and making them red hot. This high plate current may draw gases out from the plate and produce a glow discharge inside the tube, giving even greater plate current due to gas ionization. The excessive drain on the power transformer and the d.c. voltage drop in the rectifier tube will lower the d.c. output voltages for the signal circuit tubes, and this may cause distortion. The defective filter condenser and the greater load on the transformer will together increase the amount of a.c. ripple in the filter system, giving excessive hum.

► A thorough knowledge of circuit actions thus makes it possible at times to analyze a number of observed symptoms and reason out the exact location of the defect. It is unnecessary to memorize long lists of symptoms and possible defects when you are able

to figure things out for yourself like this.

► In general, when there is only a single symptom, it is good practice to check first the more common causes of that particular trouble. For example, if the symptom is hum, it would be wise to check the rectifier tube, test the power pack filter condensers, and check cathode-heater leakage in those signal circuit tubes in which the cathode is not tied to ground.

► If the only complaint is distortion, and you are convinced that it is not due to a loudspeaker defect, check for leakage in a grid-plate coupling condenser in an audio stage before proceeding to the isolation technique, because a leaky coupling condenser produces a positive bias on an audio amplifier stage having R-C coupling, thereby causing distortion.

#### **4. Inspect for Surface Defects.**

As there may be some surface trouble like a tube out of the socket or dead, plug out of the wall socket, grid cap off of a tube, etc., an inspection for surface defects should be made before going into the chassis.

This inspection may precede effect-to-cause reasoning, become a part of this step or follow it, depending on the complaint. It may even become a part of the second step, as you may confirm the complaint by seeing if the tubes light or get warm, sniff for odors indicating overloaded parts, and listen for noises as you rotate controls while trying the set.

The nature of the trouble, together with effect-to-cause reasoning, will suggest possible surface troubles to look for. If the set plays but is noisy, tubes are not dead; there is probably a loose connection or intermittent short somewhere, and thus this step may be a part of the effect-to-cause reasoning process.

Finally, if effect-to-cause reasoning

does not suggest anything, it is still desirable to spend a minute or two looking over the surface of the set. However, don't waste time; learn to go over the surface of the set with a quick glance. If you don't see anything wrong, go on to the isolating steps immediately.

#### **5. Isolate the Defective Section.**

A single test, oftentimes without instruments, will in many cases enable you to localize the trouble to a defective section.

Thus, if the receiver is dead but all tubes heat up, the logical next step is an isolating technique to determine which section of the receiver (power pack, a.f. amplifier, r.f. amplifier, i.f. amplifier, or local oscillator) is defective. In later lessons, you'll learn exactly how to do this.

#### **6. Isolate the Defective Stage.**

By using appropriate techniques given later, you localize the trouble to the defective stage which is in the defective section.

#### **7. Isolate the Defective Circuit.**

Once the defective stage is isolated, you can usually make a few simple circuit continuity tests or d.c. electrode voltage measurements to localize the trouble to a particular tube circuit in the defective stage (to the grid circuit, plate circuit, screen grid circuit or suppressor grid circuit). This eliminates quite a few parts from your list of suspects for the next step.

#### **8. Isolate the Defective Part.**

Using either an ohmmeter or d.c. voltmeter as you prefer, you test the suspected defective circuit (or the suspected stage if circuit-isolating techniques cannot be applied) until you have tracked down the offending part. This should not be a haphazard task, however, for a rational step-by-step procedure will save time.

A logical general procedure for locating the defective part in the defec-

tive stage or circuit involves checking the tube first of all, if it has not already been checked. Next comes a check of the tube electrode supply circuits, either with a d.c. voltmeter or with an ohmmeter used with all power turned off.

► Electrode voltages are checked with respect to the cathode or chassis, after making sure that the main plate supply voltage to the stage is correct. If normal d.c. voltage is absent at some electrode, one probe is anchored to the cathode or chassis of that tube and the other is moved back along the supply circuit, part by part, until a point is reached at which normal d.c. voltage is obtained. By proper interpretation of the voltage readings, shorts or grounds can be detected. By watching the meter pointer, noisy connections and noisy parts can be localized.

► It is not always possible to locate a defect by means of d.c. voltage measurements. For instance, a defect in a signal circuit which is not carrying direct current would not be revealed by a d.c. voltage measurement. Coil defects, tuning condenser defects and certain grid circuit defects are examples. Grid bias voltages in high-resistance grid circuits can be measured accurately only with extremely high-resistance voltmeters, and these are not always available. For these reasons, some technicians prefer to use continuity tests for locating the defect in a stage. Continuity tests with an ohmmeter can be used to supplement d.c. voltmeter tests or can be used exclusively. It is a matter of personal preference in some cases, and a necessity forced by instrument limitations in others.

Continuity tests with a multi-range ohmmeter are made with the receiver turned off and with the power plug pulled out of the wall outlet. Continuity is checked first from each tube electrode (at the tube socket ter-

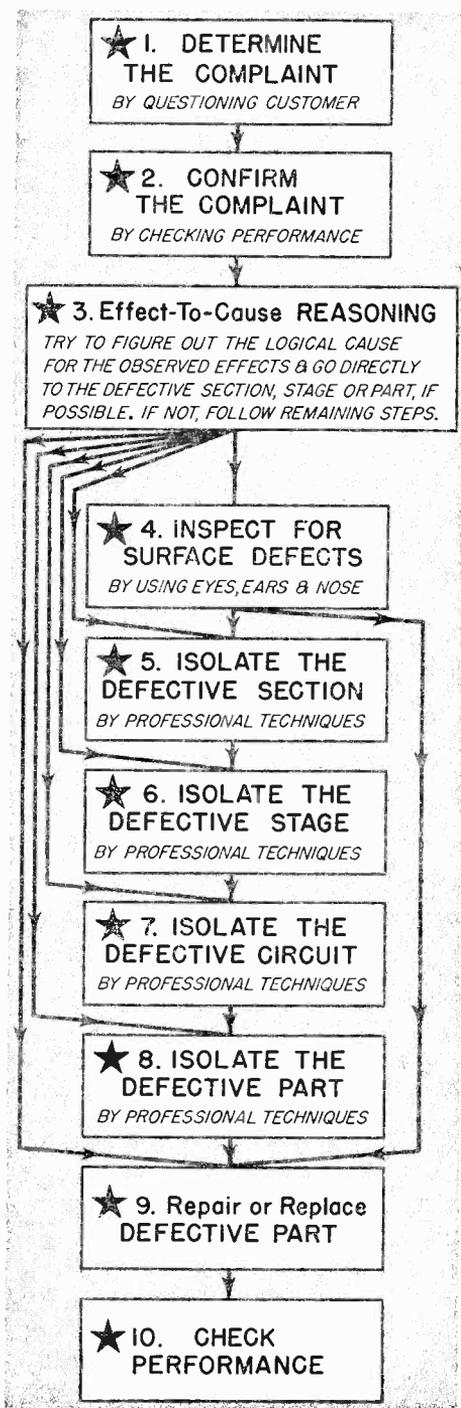


FIG. 1. Complete professional servicing procedure for radio receivers.

minals) to some reference point, such as the rectifier cathode or the rectifier plate, to establish existence of continuity in complete supply circuits.

If an infinitely high ohmmeter reading is obtained for any particular step, this indicates there is an open circuit. To find which part in the circuit is actually open, you can move the ohmmeter probes along and check each part, or you can check small groups of the parts, or you can leave one ohmmeter probe at one end of the circuit and move the other one back along the circuit, including fewer and fewer parts between the probes until the open is located.

This procedure will also detect shorts when the change in resistance due to the short is appreciable, in relation to the total resistance being measured. If you have a 20-ohm coil in series with a 100,000-ohm resistor, and check across both together, you could not tell whether the coil is shorted or not, as the reading would be so little different from 100,000 ohms that you would be unable to detect the presence of the coil. Values of low resistance parts must be checked directly across the part itself, to determine shorts.

Of course, we are just giving the barest description of the methods used—we'll go into them in detail later.

**9. Repair or Replace the Defective Part.** If the defect is traced to a poor connection, the repair is a simple mechanical procedure. If the trouble is traced to alignment, the correction of the trouble is likewise a mechanical procedure, but one requiring considerable technical knowledge and involving the use of test instruments.

When the trouble is traced to a defective part, it would appear that the simplest procedure would be to secure an exact duplicate replacement from the manufacturer of the receiver or one of his distributors. But remember

that more than 10,000 different receivers and amplifiers have been built by hundreds of different manufacturers, many of whom are no longer in business, and that each of these sets contains a minimum of perhaps 50 different parts. If you insisted on making exact duplicate replacements and wanted to keep on hand a stock of all parts which you might need, your stock would be tremendous. On the other hand, if you carried no parts at all in your own stock, you would be ordering parts every day for service work.

Fortunately, radio parts are standardized to a great extent, thus making exact duplicate replacements unnecessary in the majority of cases. Thus, such parts as tubes, pilot lamps, single fixed resistors, fixed condensers of all types and batteries are replaceable with products of any make, provided size and electrical characteristics are suitable. When space permits even size can be ignored. Furthermore, electrical ratings can be overlooked in many cases. For example, an r.f. by-pass condenser originally specified as .01 mfd. can be replaced with a .1-mfd. condenser without a noticeable change in performance. Also, if the original condenser is rated at 400 volts, you can use a condenser having a higher voltage rating, such as 600 volts.

By stocking only basic parts and by making wise substitutions instead of exact replacements, the Radiotrician can conduct his business with a minimum of overhead expense and a minimum of wasted time. It is a matter of good business to keep the stock of parts on hand as small as possible, because parts require a capital investment and there is always the risk that purchased parts may never be used.

The ability to determine whether or not an exact duplicate replacement is required for a particular part can be acquired by experience, or can be fig-

ured out from your knowledge of how the part works in its circuit. A combination of experience and knowledge is the ideal situation and is the one you will soon possess.

Only in a few cases will it be necessary to order an exact duplicate replacement, and in these few cases the replacement part will usually be available from your local radio parts distributor or from a mail order radio supply house.

With such items as variable tuning condensers, tuning dial mechanisms, r.f., i.f. and oscillator coils, and loudspeaker parts, the need for exact duplicate replacements becomes considerably more important. In some cases the mechanical construction and styling make it impossible to use universal replacement parts. In the case of air-core coils, the entire receiver design is usually based upon these coils, and few receivers have identical circuit design. When replacing loudspeaker cones, it is again necessary to have an exact duplicate new assembly of voice coil, spider and outer rings, so as to duplicate the original response of the loudspeaker. With the loudspeaker field coil a similar proposition exists; the physical construction, number of ampere-turns, resistance and current-handling ability must all correspond to original specifications.

When exact duplicate replacement parts are not available from distributors, parts which will work equally as well as the original designs can generally be obtained from local or mail order radio distributors. This is particularly true of loudspeaker assemblies, loudspeaker field coils and air-core coils. Some firms will even rewind defective coils or make a suitable cone assembly to order if the exact duplicate part is not carried in stock.

► There will be a few conditions in which it is impossible to secure an exact duplicate replacement part. An

alert and ingenious serviceman can still "carry on," by making a repair himself or having a repair made by a competent machinist or mechanic.

Thus, any broken mechanical part can be duplicated by a machine shop. A missing or broken knob of an unobtainable type presents a problem which is solved by replacing all of the knobs with a new design. If a replacement field coil for a loudspeaker is not obtainable, replace the entire loudspeaker. Power transformers, iron-core choke coils and audio transformers can be replaced with universal types having necessary adjustments to fit in most receivers.

If necessary, r.f. coils can be re-



Exact duplicate replacement vibrators are available for practically any auto radio set.

placed with standard types; the only coil which may give real trouble is the oscillator coil, for it controls the receiver dial readings. Naturally, greater care and more thought is necessary when selecting equivalent replacement parts; thorough mastery of radio fundamentals is invaluable when situations requiring this arise.

**10. Check Performance.** Try the receiver out, to make sure that all of the customer's complaints have been cleared up. It is often a good idea to let the set run for several hours (volume can be turned down for this) before making this final check-up, to see if any new troubles develop after the set is thoroughly warmed up. This is particularly desirable for sets which have not been in operation for a month or more, or when dealing with a cus-

tomer who may be unreasonable regarding future troubles which are not your fault.

**Discussion.** In a sense, locating trouble in radio receivers is similar to expert detective work. The technique to use in each case depends upon the nature of the job, your analysis of it, and the test equipment you have at hand. Each complaint calls for its own careful selection of service steps.

Noise, for instance, is a complaint for which only a limited use of effect-to-cause reasoning is justified. It is usually best to start right away on the isolating techniques which reveal in turn the section, stage and circuit containing the defect.

You can isolate a noisy stage without any test equipment, simply by silencing each stage in turn while working from the loudspeaker towards the antenna. You can silence a stage by removing its tube; if the noise stops when this is done, you know that the defect is either in that tube or in the stages ahead of the tube (towards the antenna). If the noise continues with the tube removed, you know that the trouble is between that tube stage and the loudspeaker, or is in the power pack.

Of course, a simple isolation procedure such as this has its limitations, and requires further tests for final verification. If you have an expensive signal-tracing instrument, you can listen to the signal at each stage while working from the antenna to the loudspeaker; the first stage at which noise is heard is the one which is likely to contain the defect.

In this Course you will learn still other methods of isolating the defect when the complaint is noise; some will be better for shop work where full equipment is available, while others will be more suited for work which has to be done in the customer's home with minimum equipment.

## SUGGESTIONS FOR MASTERING PROFESSIONAL SERVICING TECHNIQUES

Servicing does not follow an absolutely fixed pattern. You cannot say that hum is only caused by an open filter condenser or that distortion is only caused by a leaky coupling condenser. These are common causes for the particular complaint, and may even be the usual cause in some particular receiver. However, there are always exceptions to the general rule. To be a successful radio serviceman you must have a general approach which will localize the trouble regardless of the cause and must then fill in by effect-to-cause reasoning and from practical experience, for those special cases where short cuts are possible. You should never have to resort to the guess and try procedure of the inefficient, untrained serviceman. Unlike many other skilled occupations, that of a radio serviceman is a one-man job. Once a serviceman starts a job, he is in the best position for completing the work. No one sitting on the sidelines can give definite advice unless the serviceman can state clearly the exact trouble, all the symptoms, and all the results of his measurements. Even then, an adviser can only suggest more likely sources of trouble, and suggest more efficient isolating techniques.

In order to become an expert professional Radiotrician, you must know the various professional servicing techniques, you must acquire experience, and you must be able to figure out technical problems. As you proceed to master the various techniques, as given in the following lessons, you will appreciate more and more the importance of knowing the fundamentals of radio circuits.

**Value of Reviewing Early Lessons.** A review of previous lessons on radio fundamentals at the rate of two

or three lessons a month will prove highly beneficial during your study of these advanced lessons. Early lessons will have an entirely new meaning, and fundamentals of circuits will seem far more practical now that you really recognize their value in radio servicing work.

**Following the N.R.I. Plan.** Elsewhere in the Course, a plan for acquiring practical radio servicing experience in your own home is given. We strongly recommend that you give this phase of your training immediate attention, if you want to get the most out of these advanced lessons on radio servicing. Obtain a suitable radio receiver chassis as soon as you can, and begin to carry out on it the N.R.I. plan for securing the experience so necessary for efficient radio servicing.

You can also use this chassis in connection with your advanced servicing lessons; whenever you study a particular professional technique or learn about locating a particular defect, try it out on the chassis and demonstrate for yourself the relative effectiveness of the recommended procedures. Your studies will take on a new meaning when you do this, and you will have little trouble in remembering what you studied. You may not be able to test everything which you study, but those tests which you do carry out will definitely prove worth while.

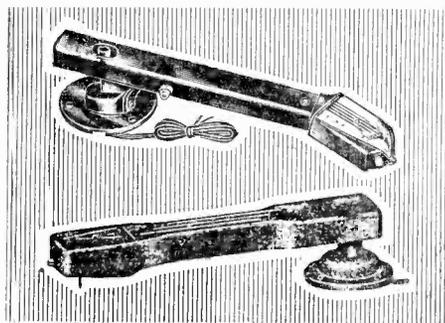
### PRACTICAL ADVICE ON REMODELING RADIO EQUIPMENT

Radio servicemen eventually learn by sad experience that it is by far the better policy to stick to their own work, and avoid remodeling sets. Any attempts at remodeling always involves changes in the design, which is a far more intricate problem than might be imagined.

For example, it would seem to be a

simple matter just to change sockets, so newer type tubes can be used. Trouble starts, however, when you begin trying to get the same output level, the same selectivity and the same sensitivity from the remodeled unit. The newer tubes may be capable of even better performance than the originals but the circuits must usually be redesigned to utilize this superior capability.

Similarly, changing from an i.f. value of 175 kc. to 456 kc. will give greater freedom from image interfer-



### MAGNETIC Phono Pick-ups

These pick-ups generally require such repairs as replacement of rubber parts or replacement of a defective coil.

ence. However, just swapping transformers introduces problems, for the tuning dial range depends on the i.f. value. You will then be involved in the necessity of changing the oscillator coil and, even if you get the same tuning range, you will still have the problem of getting the same gain and selectivity at the new i.f. value.

The customer will quickly recognize any deficiency in performance in a remodeled set because of his familiarity with the performance of the original receiver. Not only will you have a "headache" on your hands in attempting to improve the original perform-

ance, but you will also run the serious risk of having a dissatisfied customer in the end.

The cost of the material required, plus a fair rate for the time spent in remodeling, will usually total up to so much that you could not possibly charge the customer for the whole job. You cannot expect to make a ten year old receiver have all the qualities and performance features of a modern unit unless you replace practically all the parts in it, and redesign the unit completely.

This means you take a loss on most remodeling jobs. It is far better to recommend that the customer purchase a new and modern receiver. For comparable performance, the new receiver

will be cheaper, and you will win a satisfied customer for future work.

Another thought—even though you do redesign a receiver successfully, you will very likely be expected to keep that receiver operating satisfactorily for the rest of its life. In other words, you become “married” to that receiver and will be blamed for all troubles which develop in it.

As a general policy, a radiotrician should turn down all opportunities to remodel receivers or amplifiers. It is the job of a radiotrician to recognize, isolate and repair defects in radio equipment, align receivers, and do certain other things that are necessary to restore the original performance of the receiver.

# Lesson Questions

Be sure to number your Answer Sheet 36RH-1.

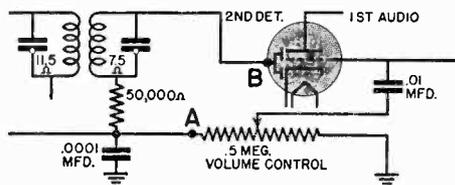
Place your Student Number on every Answer Sheet.

Send in your set of answers for this lesson immediately after you finish them, as instructed in the Study Schedule. This will give you the greatest possible benefit from our speedy personal grading service.

- When the customer's complaint is fading on distant stations, with local reception being entirely satisfactory, which of the following procedures should be used: (a) *Look for trouble in the receiver*; (b) *Explain to the customer that fading is due to natural causes over which you have no control*; (c) *Look for some electrical apparatus nearby which is causing the fading.*
- A customer says that both distant and local stations are very faint. You notice that the tuning indicator operates normally when stations are tuned in. In what section of the receiver would you first look for trouble?
- When a stock of new condensers is at hand, what is the simplest and best way of finding out whether a particular suspected condenser is open?

- Suppose ohmmeter measurements in the circuit at the right show an open circuit between B and ground. Which part is defective if further tests give readings of:

B to A: 50,000 ohms.  
A to Ground; OPEN.



- A raspy, buzzing sound is heard along with distortion of low-frequency sounds in programs, such as men's voices and drum beats, but high-frequency sounds come through clearly. Which part in the receiver would you suspect?
- What complaint is associated with a leaky coupling condenser in an R-C coupled audio stage?
- Name, in correct order, the ten steps in the complete professional servicing procedure.
- A 410-ohm, 1-watt cathode resistor in the output stage is burned out. You are unable to get a new exact duplicate resistor, nor can you get a 400-ohm resistor of any kind. Would two 800-ohm, 1-watt resistors in parallel work satisfactorily?
- Suppose that you trace the trouble to a defective 90,000-ohm carbon resistor having standard 20% tolerance. The distributor does not have this value in stock, but says a 100,000-ohm resistor will work just as well. Is he right?
- Would it be practical to modernize a 15-year old t.r.f. receiver so it would pick up foreign short-wave stations and could be tuned just by pressing buttons?



## BE BUSINESS-LIKE

No matter how much you know about the technical side of radio servicing, you will attain success much faster once you acquire a business-like attitude.

The first essential of a success-getting attitude is nothing more or less than *self-confidence*. You must know what to do and when to do it, and must be able to go ahead and do it.

Confidence in your own technical knowledge will come automatically soon after you complete the Course and engage in radio work, because your N.R.I. radio training ranks among the finest in the world.

Equally important, however, is confidence in your ability to deal with people. This type of self-assurance is developed by planning your moves carefully *before* tackling a job, and by being methodical, accurate and careful. Before each interview with a customer, spend a few minutes figuring out what questions or problems might arise, and figure out your answer for each one. You will then be able to answer questions promptly *and confidently*, in a business-like manner which commands respect and inspires confidence in others. Be business-like—it pays!

*J.E. Smith*

**HOW TO ISOLATE  
THE DEFECTIVE  
SECTION AND STAGE**

37RH-2



**NATIONAL RADIO INSTITUTE**

**WASHINGTON, D. C.**

**ESTABLISHED 1914**

# Study Schedule No. 37

For each study step, read the assigned pages first at your usual speed. Reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind. Study each other step in this same way.

1. **Inspection for Surface Defects** - - - - - **Pages 1-6**

Simple defects outside a receiver chassis occur often enough to make a quick inspection for surface defects highly desirable. It is important to realize that the actual inspection takes only a few minutes, far less than the time it takes to read this section. Once you've worked on a few receivers, you'll find that this quick external inspection of a radio set becomes second nature to you.

2. **Effect-to-Cause Reasoning** - - - - - **Pages 6-12**

Here are practical examples to illustrate exactly how effect-to-cause reasoning can bring you directly to the defective stage or part, thereby saving considerable time. Reasoning admittedly will not always work; in fact, the important thing for you to realize in connection with this section is that you should be ready to give up reasoning and start isolation procedures unless some very definite clues exist.

3. **Defective Section and Stage Isolation** - - - - - **Pages 12-19**

This section is a general survey of the different isolation techniques which may be used to isolate a defective section or stage. Therefore, it is not intended to teach you exactly how to carry out isolation techniques for particular complaints. These instructions will come later.

4. **Test Equipment for Signal Tracing** - - - - - **Pages 19-23**

Basic circuits and general operating principles of four different kinds of special servicing instruments are given in this section. Here you learn enough about signal tracing instruments to be able to quickly put to use any of this equipment you may encounter.

5. **Isolation Techniques for a Dead Receiver** - - - - - **Pages 24-28**

Here you receive detailed instructions for applying different isolation techniques to a dead receiver. The choice of techniques depends upon personal preferences as well as on the instruments you have available.

6. **Answer the Lesson Questions, and Mail your Answers to N.R.I.**

7. **Start Studying the Next Lesson.**

# HOW TO ISOLATE THE DEFECTIVE SECTION AND STAGE

## Inspection for Surface Defects

**A** PROFESSIONAL serviceman is concerned with two vital points: first, to find the defect; and second, to find the defect as quickly as possible. There are many ways of locating a defect, but only a carefully planned procedure will lead to the cause of the trouble in a direct, quick manner.

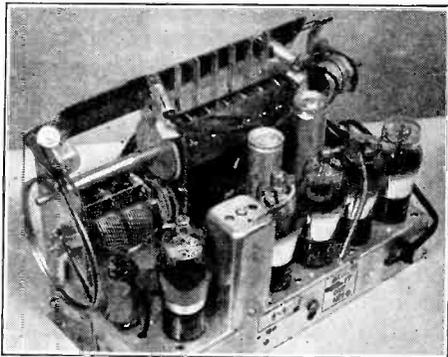
Never lose sight of the fact that a serviceman is selling time. If you can shorten the time consumed on any repair job, you can handle more jobs and have more time to plan your business and to advance yourself. It is important, therefore, that you learn the basic procedure for localizing trouble. This straightforward procedure of isolating the defective section, stage, circuit and part can be followed even by a serviceman with little experience and will save time, compared to guess-and-try methods.

Before studying the defective section and stage isolation procedures, we should complete the four preliminary steps of the technique shown in Fig. 1. These preliminary steps may lead you directly to the trouble. If they fail to do so, they will at least indicate the correct isolating method for the particular complaint, thus making other tests unnecessary.

We have already covered the first two steps, so we can now concentrate on steps 3 and 4. In this lesson we are going to study step 4 before step 3, because the inspection for surface defects will, in many instances, follow the confirmation of the complaint or may become a part of the effect-to-

cause reasoning process, as you will learn shortly.

Logically, we should go over the external connections to the radio and as much of the top of the chassis as possible before removing the chassis from the cabinet. However, in the case of midget receivers, it may be necessary to remove the chassis from the cabinet



*Courtesy Philco*

A surprising number of complaints are caused by defects on the surface of a radio. Watch for such items as defective tubes, tubes in the wrong socket, top cap connections loose or broken, defective speaker or antenna and ground leads, etc. Each complaint will have its own list of possible defects.

before a complete inspection can be made.

Remember that we are trying to save time. Therefore, concentrate on those surface defects which could cause the complaint, instead of making a minute examination of everything. We have divided the following discussion into steps, according to the ten common complaints, so that you can become familiar with the symp-

toms caused by different surface defects.

If you do not find the trouble in a few moments, when checking for surface defects, go on to the next step in the service procedure.

**1. Dead Receiver.** If the pilot lamp does not glow and if none of the tubes heat up, first make sure that the receiver is getting power. See if the power cord plug is in the wall outlet, and inspect the plug. Feel along the length of the power cord to see if there is a break. The wall outlet may be defective or may not be getting power; if you suspect this, check voltage at the outlet with a meter or by plugging a floor lamp into the outlet. Wiggle the plug in the outlet, as it may not be making contact.

A burned-out tube is a common trouble which can cause a dead receiver, so look for a cold tube or one in which the filament does not glow. If all tubes are cold and the set is of the universal a.c.-d.c. type, check all tubes in a tube tester, because the tube filaments are connected in series and an open in any one of them will block filament current to all tubes.

If the tubes light or heat up, look for a disconnected tube top-cap connection, inspect the loudspeaker leads, the loudspeaker plug or terminal connections, and the antenna and ground connections. Look particularly for an open voice coil lead if the set is completely dead (no hissing, hum or other sound from the loudspeaker). If there are any special switches at the back of the chassis, see if they are set correctly.

Be sure the power source is of the correct type for the receiver. There are still a number of 110-volt d.c. homes in large cities, and a.c. receivers will not work in these; in fact, the power transformer will burn out if an a.c. receiver is plugged into a d.c. outlet. Furthermore, with universal

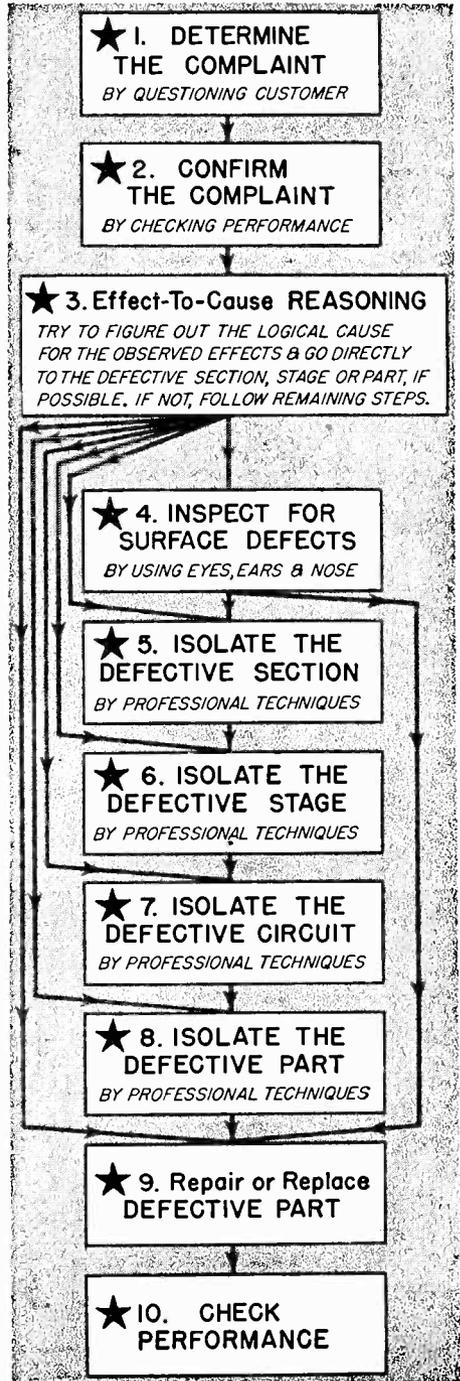


FIG. 1. The complete professional servicing procedure for radio receivers.

a.c.-d.c. receivers and d.c. receivers, the power cord plug must be inserted with the correct polarity to secure satisfactory d.c. operation; try reversing the plug if you encounter this situation.

With battery receivers, check the condition of all batteries and check all battery connections, because failure of any battery or any battery connection can definitely cause a dead receiver.

**2. Noisy Reception.** First determine whether the noise is originating inside the receiver chassis, outside the chassis but in some part of the receiver installation, or entirely apart from the receiver installation.

Disconnect both the antenna and ground leads, and short together the antenna and ground terminals of the receiver; if the noise now stops or is greatly reduced, there is an external noise source or the antenna-ground system is at fault. If the noise does not change appreciably, the trouble is in the receiver itself or the noise is coming in through the power line.

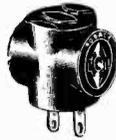
In the case of a set having a built-in loop antenna, rotate the loop or rotate the entire cabinet to secure the same effect; if this causes a change in the amount of noise, the trouble is external.

Noise may originate in the power line system and enter the receiver through the power cord. If a line noise filter is available, it can be inserted in the wall outlet, and the receiver plugged into the noise filter to block any noise signals coming in over this path.

If the noise starts or stops when a house light or other appliance in the house is turned on or off, the noise is definitely localized to a defective lamp, switch or appliance. If the noise occurs when someone walks across a room in the house, pipes in the plumbing system may be rubbing

together, or the receiver power plug may be making a poor contact in the wall outlet. Check by wiggling the power cord and plug. If the noise increases, examine the plug for broken wires, bend the prongs to make better contact, or replace the plug. Be particularly suspicious of any cube tap (three-outlet plug-in device). Try plugging the radio into another outlet, using an extension cord if necessary.

If the noise source is outside the set, check the antenna and ground leads by reconnecting them to the set.



This cube tap is a common source of noise. Poor contacts develop in these devices.

With the set turned on, shake these leads vigorously. If this increases the noise, look for a poor connection or break.

If noise is localized to the receiver, tap each tube while the receiver is in operation. Again, an increase in the noise level indicates that the trouble has been located. Be on the lookout for top-cap connections which are loose or touching the tube shields. Don't be misled by this test, however. Sometimes tapping a tube may jar the actual defective part or terminal and thus lead you to believe the tube is defective. If a new tube acts the same way, you have an under-the-chassis defect.

Variable condenser plates just barely shorting together may cause noise. This condenser trouble can quickly be identified, as the noise will be intensified when you move the rotor plates as you tune the receiver.

A loose cone or loose turns on the voice coil of the loudspeaker may produce a combination of noise and distortion, which can be recognized after

being heard a few times. Remember to notice such combination symptoms, as they frequently lead you right to the trouble.

**3. Annoying Hum.** There are quite a few causes of hum which are external to the chassis. Thus, a poor ground connection may permit the receiver chassis to pick up stray ripple fields. The resulting hum current in the chassis can enter a grid circuit and cause hum. Check the ground lead and the ground connection for this trouble. Try reversing the power cord plug in the wall outlet, because many sets are grounded through the power line.

Tubes in the a.f. stages with cathode-to-heater leakage can cause hum when the cathode is not directly connected to the chassis. Check the tubes for leakage in a tube tester.

Rectifier tubes are sometimes causes of hum. If the rectifier tube is of a vacuum type and has a purplish-blue glow between the elements, gas exists in the tube and is preventing normal rectification. When a gassy rectifier tube is found, remember that the gas condition may be the result of excessive current flow through the tube. The filter condensers should be checked, as excessive leakage will result in high rectifier current.

You will sometimes notice a white salt-like deposit on the cap of electrolytic condensers. This indicates the escape of some of the electrolyte. When the complaint is hum, these condensers should be checked.

Watch out for a line cord tucked into the set next to a grid circuit lead; pull out the cord to see if this reduces the hum.

**4. Howling.** In general, squealing (or howling) is due to electrical or mechanical coupling between circuits. A missing shield can cause this condition. Excessively long antenna and ground leads are sometimes

jammed into the cabinet and there cause coupling between stages, resulting in squealing. Shortening the leads is the remedy.

Howling can occur when some part vibrates under the influence of sound waves from the loudspeaker. Tubes made about ten years ago lacked internal bracing of electrodes, and hence were particularly subject to microphonic trouble of this nature. Touching a microphonic tube will usually identify it as the offender by stopping the acoustic feedback. Replacing the microphonic tube with a more modern tube of the same type, having a more rigid internal construction, will usually clear up the trouble.

Microphonic trouble may occur in the input audio stage of any high-gain audio amplifier, even with modern tubes. Try several tubes to find one which is less troublesome. Save the other tubes for use in a less critical stage.

If the rubber supports under the chassis of a receiver become hard, mechanical feedback from the speaker, through the cabinet will cause howling. Likewise, if the shipping bolts were not removed or loosened when the receiver was installed in the home, there will be mechanical coupling from the chassis to the cabinet through these bolts.

Front-panel control shafts should not touch the sides of the cabinet holes, and control knobs should be a short distance away from the panel. Felt washers are often used between the knobs and the panel to minimize chances for mechanical feedback which can cause howling.

Mechanical vibration of the rotor plates on gang tuning condensers will cause a howl, as this changes the capacity at an audio rate. Rubber mountings are usually used under the condensers to prevent this; be sure they have not hardened.

**5. Distortion.** As a general rule, it is a good idea to check all tubes in a tube tester when the complaint is distortion. A low-emission a.f. tube is a possible cause of distortion, because it changes the shape of the  $E_g-I_p$  characteristic curve. Strong signals then swing over a non-linear portion of the curve.

The use of an excessively long antenna on certain midget receivers may give an input signal so high that the receiver circuits are overloaded and distortion results. Hence, if you find that a midget receiver distorts on strong local stations but gives normal reception for weak locals and distant stations, try shortening the antenna.

A boomy, distorted tone can occur when a set is placed right up against a wall. Moving the set about two inches away from the wall will eliminate this trouble.

Defective loudspeakers are another common cause of distortion. Once you've learned to recognize the effects of loose turns on the voice coil, off-center voice coils, excessively hard or excessively soft cones, broken spiders, etc., you will be able to recognize these troubles when checking performance.

**6. Low Volume.** As you know, an audio system defect is indicated when both distant and local stations are heard but volume is low. Check the a.f. and detector tubes in your tube tester before going on with the service procedure, as a low-emission tube may be the trouble.

Be on the lookout for a phono-radio switch set in the phono position, particularly when the switch is on the rear of the set or on the phonograph mounting board. The set owner will usually turn the knobs on the front of a set but will frequently overlook others.

**7. Poor Sensitivity.** Unsatisfactory reception of distant stations can

be due to a defective pickup system, hence the antenna and ground should be checked thoroughly if this is the complaint. Tune in a local station, then disconnect the antenna and ground. If this doesn't cause much decrease in volume, the antenna system is defective.

An r.f. or i.f. tube with low emission can cause poor sensitivity, so again it is wise to check tubes first.

There are certain conditions about which you can do nothing, such as seasonal changes in reception, differences between day and night-time reception, or inability of the particular receiver to deliver the expected results, because of its basic design. If you don't recognize these, you may let yourself in for an impossible service job.

**8. Poor Selectivity.** Here again you must be on your toes, for many complaints of poor selectivity are really cases of station interference or inadequate receiver design for your locality. A set with sufficient selectivity for one locality may be inadequate in another, where there are more local stations. Experience with various types of receivers will soon teach you what to expect.

**9. Station Interference.** Local conditions determine the amount of station interference. A nearby transmitter will usually cause trouble of this kind, although there are several types of station interference.

Try a shorter antenna and try a wave trap before removing the chassis for further tests or adjustments. Other remedies will be discussed elsewhere in your Course.

**10. Intermittent Reception.** Any of the external defects which cause noise may also cause *intermittent reception*. You can often isolate a noisy or intermittent contact by pulling on leads, wiggling or pushing on parts,

and tapping on tubes. When you come to the defective part, such movements will cause a definite change in the receiver operation.

When the customer complains of changes in volume when an electric light or other appliance is turned on somewhere in the house, look for a poor ground or no ground at all. It may be necessary to run the ground wire to some other point in the home. (In many universal a.c.-d.c. receivers, a ground cannot be used.) Be

sure that a good antenna is used and try reversing the line plug.

In some localities, the power line voltage drops considerably during certain intervals of the day. In industrial localities this occurs during working hours, and in residential districts, it occurs in the early evening. When you find that the customer's complaint is due to this cause, all you can do is suggest that the customer bring the matter to the attention of the local power company.

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## Effect-to-Cause Reasoning

**Practical Importance.** You can now see that examining a receiver and its installation as just described will often reveal an external defect and save the time you would otherwise spend in removing the chassis from the cabinet. If the preliminary examination indicates internal trouble, however, we must proceed to localize the trouble.

Effect-to-cause reasoning is a process of reasoning from the observed effect back to its possible cause. If we omit this step, we could now go on to the isolation procedures and would eventually find the trouble. However, we must remember that we are attempting to find the trouble in the shortest possible time, so we should take advantage of effect-to-cause reasoning. A minute or two spent in diagnosing the trouble will frequently provide short-cuts around one or more of the servicing steps shown in Fig. 1. Further, a little reasoning will help you choose the proper technique or test methods for each complaint.

Effect-to-cause reasoning is built up from your knowledge of the purpose of each and every part in a ra-

dio and from your knowledge of how a part can fail. By tying together these facts and the observed symptoms, you will soon be able to make use of this step. Experience will thus serve to tie your fundamental-course learning to your practical work.

Effect-to-cause reasoning can successfully be used only if there are enough clues to lead you to the trouble. As you know, there are usually many possible causes for a particular effect or symptom of trouble. While confirming the complaint, you should look for additional clues or information which will narrow down the possible number of causes to a few probabilities or will lead directly to the source of trouble.

In many instances, there will not be enough clues at the beginning of a service job to allow effective use of this step. However, by carrying out another step or two of the trouble-localizing procedure, we will narrow down the possibilities considerably and will eventually reach a point where reasoning can be applied. In other words, although we have shown effect-to-cause reasoning as the third

step in Fig. 1, this is just the first time you can use it. If you cannot go directly to the trouble, isolate the defective section or stage, then try reasoning again.

When you start your servicing career, you will probably find that effect-to-cause reasoning does not work for you every time. Keep trying to use this step, however, as the ability to use it will come to you with experience. Also, experience will add to your store of clues, as you will soon learn that there are different kinds of hum, noise, etc., each indicating a particular source of trouble.

Now, let's take up a few typical examples to show how reasoning can be applied as a step saver.

### COMMON CAUSES OF VARIOUS RECEIVER COMPLAINTS

**1. Dead Receiver.** There are so many defects which can make a receiver dead that here it is usually wiser to start right away on an isolation technique. Nevertheless, there are certain symptoms associated with a dead receiver which are worth watching for because reasoning can often lead directly from them to the cause of the trouble.

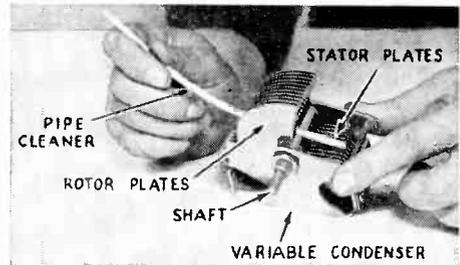
As one example, in an a.c. receiver you may find all tubes except *one* heat up normally, yet the one tube may test good. Since other tubes heat, there must be a break in the wiring to this one socket or a defective socket. A break in any circuit will remove an operating voltage, but a tube will become slightly warm if only the filament voltage is applied, so a completely cold tube indicates a filament circuit break.

As another example, you may find all tubes cold in a dead universal receiver. Remembering that tube filaments are in series in universal receivers, you logically check filament

circuit continuity (assuming you have tested the tubes and found them to be good).

**2. Noisy Reception.** The possible causes for noise are so numerous that here again it is best to start immediately with isolation procedures.

Of course, you should watch for obvious indications such as variations in a tuning indicator or visible sparking which may lead toward the trouble. Likewise, if the noise becomes



When the noise is loudest as the tuning knob is rotated, effect-to-cause reasoning indicates dust or metal particles between the tuning condenser plates. A pipe cleaner (obtained from a tobacco store) is ideal for cleaning, as shown above. This cleaning is done without removing the condenser from the chassis.

more prominent when a volume control, tone control, switch or tuning knob is adjusted, you would reason that the part itself is defective and would check it immediately.

**3. Annoying Hum.** Filter condensers are the first parts to suspect when hum is the complaint, because their chief job is suppression of hum. An electrolytic condenser which is open, has lost capacity or has a high internal resistance due to drying out will be less effective in reducing the power pack ripple voltage. Shunting each suspect with a good condenser is a quick check for any condenser troubles except leakage or short circuits.

The output filter condenser has more to do with hum than the input one, so it should be suspected first. As additional clues, an open output

filter condenser may result in motor-boating (oscillation or squealing), and there may be a loss in the low-frequency response (a form of distortion).

A defective electrolytic should be completely disconnected when discovered. Never connect a good condenser permanently across a bad one, even though this clears up the trouble at the time, because the bad unit may eventually break down and short the power supply.

You have learned that the input filter condenser builds up the d.c. voltage to a value near the peak value, in addition to reducing the a.c. ripple. Therefore, an open or low-capacity input filter condenser would result in low d.c. voltage and excessive hum. Also, a *leaky* output filter condenser will draw excessive current through the choke coil, which results in excessive hum and a drop in d.c. voltage. Hence, hum coupled with low d.c. voltage is a double clue to filter condenser trouble.

Notice how the basic theory you studied in the Fundamental Course is being put to use in finding the cause of trouble. Just reasoning from your knowledge of circuit and part actions this way will frequently lead right to the source of trouble.

Continuing with our reasoning, an open bypass condenser in an R-C filter for a tube may allow power pack ripple currents to enter the tube stage and produce hum. If a control grid circuit is open, the high input resistance allows small hum currents to set up large hum voltages which the tube will amplify. An isolation procedure will save time in locating these troubles.

If you can hear the hum only when a station is tuned in, the hum must be originating in the r.f. section of the set or at the broadcast station. Obviously, if the hum originates in the

audio amplifier it will be heard all the time, whether or not a station is tuned in, as it will pass directly through this amplifier. Hum originating in the r.f. amplifier is called modulation hum, as it travels through the set by modulating an incoming carrier. Tune in several stations to see if the hum comes in with every carrier. If the hum is heard on only one station, the trouble could be in the station itself. You can try picking up the signal from this station on another receiver, but be sure this set is capable of reproducing the low hum frequencies, as not all sets can do this.

If the trouble is modulation hum, we have automatically localized the trouble to the r.f. section, so we can concentrate on locating the defective stage, circuit or part. Notice that the opposite is also true; if hum is heard all the time, it must originate in the audio section or power supply. Again, reasoning has led us directly toward the trouble and has helped prevent blind, unnecessary checking of parts.

The ability to use effect-to-cause reasoning will grow as you gain practical experience. Further, you will begin to notice how different sources of hum and distortion produce characteristic sounds. On your first repair jobs, concentrate on remembering what you hear and find to be the cause of the trouble. Then, in the future, a particular sound or action will bring to mind a particular defect.

**4. Squealing or Howling.** Since oscillation can occur in any section of a receiver, the first step should be one of making a section isolating test before applying effect-to-cause reasoning.

A simple test will determine whether the trouble is in the a.f. or r.f. section. If tuning of the receiver has no effect on the squeal, and if pulling out the last i.f. tube does not

stop the squeal, you can definitely say that it is originating in the second detector or a.f. amplifier.

With the trouble isolated to the a.f. amplifier, a put-put sound (motorboating) indicates a low-frequency oscillation and leads you to suspect an open filter condenser, either in the power pack or an a.f. stage.

In the case of a high-pitched squeal originating in the a.f. amplifier, suspect an open plate bypass condenser for the output stage. This condenser is put in the circuit to make the plate load capacitive, thus preventing oscillations.

You learned that a squeal in the r.f. or i.f. sections can be produced only by two signals which differ in value by an audio frequency. Thus, a single stage may go into self-oscillation, producing a signal *which combines with the incoming signal* to produce the audio frequency squeals or a motorboating sound. Also, two stages may be oscillating at the same time, at frequencies which differ by an audio value. This produces a squeal which will be heard even when no stations are tuned in.

Therefore, when a squeal is isolated to the r.f. system, reasoning would lead you to try isolating the defective stage by bringing your hand near each r.f. tube in turn. When the pitch of the squeal changes or the frequency of the "put-put" sound varies as you approach a particular tube, you have very likely localized the trouble to that stage. Remember, however, that bringing your hand near the oscillator stage in a superheterodyne will shift its frequency, thus detuning the set and causing a change in the pitch of the squeal. Disregard this indication.

The first thing you would question in an oscillating r.f. stage is an open screen grid bypass condenser or an open screen grid-to-cathode bleeder

resistor. An open bypass condenser destroys the shielding action of the screen grid, while an open bleeder resistor raises the screen grid voltage to the point where the stage becomes unstable.

Reasoning also indicates that an open output filter condenser in the power pack can cause either degeneration or regeneration, depending upon the number of stages in the receiver and the method of coupling the stages. This condenser may therefore be questioned, but beyond this point it is best to begin a more positive isolation procedure.

**5. Distortion.** You have learned that distortion is most commonly caused by a defect in the a.f. system, so you would first check all a.f. tubes.

Distortion in the loudspeaker is usually easy to recognize. Thus, when the voice coil rubs against the pole pieces, only the higher audio frequencies are properly reproduced. Low notes, which require large movements of the voice coil, sound fuzzy or otherwise distorted. Other loudspeaker defects and their symptoms are considered elsewhere in the Course.

Gassy a.f. amplifier tubes (especially output tubes) and leaky a.f. coupling condensers are so often encountered when distortion is the complaint that they should come to mind immediately. To check these possibilities quickly, measure the d.c. voltage drop across the grid resistor. If the grid end of this resistor is positive with respect to the chassis end, pull out the following a.f. tube, leaving the receiver turned on. If the voltage still exists, you know the condenser is leaky; if the voltage drops to zero when the tube is pulled out, the tube is gassy and was drawing grid current through the grid resistor to produce the voltage drop across it.

Of course, in universal a.c.-d.c. re-

ceivers the tube cannot be pulled out, because this would open the filament circuits for all tubes. Instead, when you encounter a positive voltage across the grid resistor, unsolder one lead of the coupling condenser. If the



In a case of distortion, effect-to-cause reasoning has led this Radiotrician to check the voltage across a grid resistor. By locating the coupling condenser, this measurement can be made between the chassis and the end of the condenser which connects to the grid of the next stage, as shown here.

voltage is still there, you know the tube is gassy; if the grid resistor voltage drops to zero when the coupling condenser is disconnected, the condenser is leaky.

If these preliminary tests fail to reveal the defect, it is time to begin the defective stage isolation procedure for distortion.

**6. Low Volume.** Suppose you find that volume is low both for local and distant stations. The presence of distant stations shows normal sensitivity. Therefore, you would normally suspect a defect somewhere in the audio system. If the receiver has a tuning indicator which shows the expected variations when a station is tuned in, you have another indication of normal sensitivity. As low-emission tubes are a likely source of trouble, you would check them, if you

have not done so in a surface analysis.

Combination defects give you clues that guide you more definitely. For example, if low volume is associated with a high-pitched tone (little or no bass), you can reason that an open in an a.f. coupling condenser is lowering the coupling capacity so much that only the higher audio frequency can get through. You would then check the coupling condensers before starting isolation procedures. As a rule, however, immediate adoption of a suitable stage isolating technique is best for low volume complaints.

**7. Poor Sensitivity.** When distant stations are not picked up, the symptom is poor sensitivity. Even local stations may not come in with normal volume if the sensitivity has dropped greatly. If there is a tuning indicator on the set, it can be quite helpful. In case of low sensitivity there will be little or no indication from a tuning indicator when stations are tuned in.

Poor sensitivity is always an r.f. trouble, so you can concentrate on the r.f. amplifier and its tubes. The tubes should be tested first of all.

Poor sensitivity, accompanied by considerable hissing noise, indicates that r.f. gain exists but that little signal is reaching the input of the set. This observation would at once suggest that you check the primary of the antenna transformer with an ohmmeter to see if the primary is open, because this would prevent the signal from entering the receiver. Lightning discharges quite often open the primary and cause this trouble.

Improper alignment will cause poor sensitivity, as all stages will not be properly adjusted to give maximum response to the incoming signal. If stations come in at the proper dial settings, you know that oscillator trimmers are all right. Effect-to-cause reasoning here would lead you

to check only preselector and i.f. trimmer settings. If you find the trimmers badly out of adjustment or that stations do not come in at the correct dial settings, complete realignment of the entire receiver is required.

When poor sensitivity is accompanied by hum, check the filter condensers for an open by shunting with a good condenser. An open input filter condenser lowers operating voltages, while an open output condenser can cause degeneration, both causing poor sensitivity.

Open bypass condensers are possible causes of poor sensitivity, but there are so many of these condensers that it is usually best to start an isolation procedure. Then, when you have found the defective stage, you can check the relatively few condensers in it.

**8. Poor Selectivity.** First, determine whether the trouble is poor selectivity or station interference. If the interfering station is on the *same* frequency as the desired station, you have station interference. Also, if the interfering station is more than 40 kc. away from the desired station, the trouble is station interference, as the tuned circuits would remove any such interfering signals under ordinary conditions. It is only when the interference is from a station or stations *on adjacent* channels that the selectivity of the set can be questioned.

Next, some judgment is necessary in deciding if the complaint is actually due to a receiver defect or if it is caused by local conditions. The use of an excessively long antenna, or local stations assigned frequencies too close together for the particular receiver design, may cause the selectivity to appear worse than it actually is.

In actual cases of poor selectivity, something is wrong with the tuned

circuits. Here the first thing to do is to check the alignment. This will either correct the trouble or will suggest further tests.

If one particular trimmer will not peak or peaks very broadly, you know that trouble exists in that tuned circuit. If the trimmer peaks broadly, you know that the Q of the circuit is low and you should look for high-resistance leakage paths across the tuned circuit. Also, look for resistance in the circuit itself, caused by poorly soldered connections. If the trimmer is in a preselector circuit, the resistance may be due to defective or dirty wiping contacts on the condenser gang rotor sections.

If the dial pointer slips, you will not be able to adjust the trimmers to resonance at the new dial setting. In such a case, reset the pointer before realigning. Also, look for incorrect spacing between the rotor and stator plates of the gang section. If the spacing is not correct, the capacity will be abnormally high, and it will be impossible to reduce the capacity of the trimmer enough to give resonance at the proper dial setting. In addition, one or more turns on a tuning coil may short-circuit, reducing the inductance and making necessary a greater increase in capacity than the trimmer can supply.

In the alignment procedure, the trimmer condensers are adjusted one at a time and the observed reaction to each adjustment serves to isolate the defective stage. Of course, if the trimmer adjustment clears up the trouble, the circuit was out of alignment and no other defect exists.

**9. Station Interference.** Here, effect-to-cause reasoning plays a very important part in determining the correct cure. If a code station is heard all over the dial, you are safe in assuming that it is operating at the same frequency as the i.f. frequency

of the set, or is at one-half this frequency so the second harmonic of the code station produced by the frequency converter will travel through the i.f. amplifier. Shifting the frequency of the i.f. amplifier 5 or 10 kc. will usually clear this up. In addition, the use of a wave trap tuned to the frequency of the interfering station could be used.

In some cases a squeal will be heard at the dial setting for a favorite station, but is not heard on other local stations. This is usually due to a harmonic of the i.f. getting back into the mixer input and beating with the desired station to produce the squeal. You can't change the station frequency but you can shift the i.f. value of the receiver a few kc. to make the squeal fall to one side or the other of the desired station. In some localities, such troubles are never encountered, while in others the station frequency allocations result in numerous service jobs of this type.

If the interference is from another station on the same frequency, nothing can be done except experiment with different antennas to try to cut down the interference or raise the desired signal strength so it will drown it out. No guarantee can be given beforehand that this procedure will work.

Inexpensive midjets frequently do not have this trouble; their sensitivity

is too low to pick up the distant interfering station. If the customer should compare the operation of a midjet and another set, you may have to do some careful explaining to satisfy the customer that his large set is working normally.

**10. Intermittent Reception** If a radio receiver becomes weak or simply goes dead at intervals, effect-to-cause reasoning tells us that practically any part may have failed. An isolation procedure is first employed to localize the trouble as much as possible.

In some instances, the intermittent condition does not take the form of weak reception or a dead set; instead there may be hum, distortion or perhaps squealing at irregular intervals. Here, effect-to-cause reasoning is a powerful tool, as you may be able to determine what part could cause the particular symptom to occur. The same method of reasoning already described for these symptoms is then applied, bearing in mind, of course, that the defect must be such that it can occur intermittently. Because of its construction, an electrolytic condenser would not be likely to open and close intermittently and thus cause intermittent hum. However, cathode-to-heater leakage in a tube can be intermittent, as the heater can often move toward the cathode as it heats.

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## Defective Section and Stage Isolation

There will be many jobs where it is necessary to make more tests in order to localize the trouble. Hence, the professional technique of isolating the defect to a section, stage, circuit and part must be thoroughly understood.

A simple and quick isolation test often goes hand in hand with effect-to-cause reasoning and eliminates the necessity for going through complete isolation procedures. In other words, one or two carefully planned measurements can eliminate a large number of

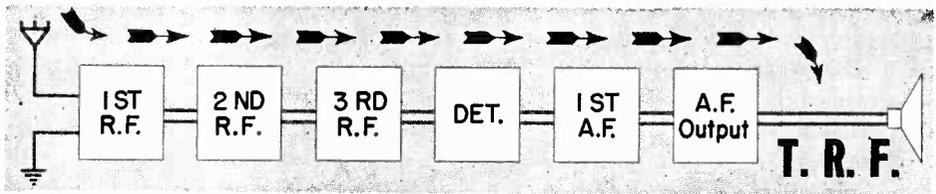


FIG. 2. Block diagram of a t.r.f. receiver.

suspects, thus making isolation easy.

With experience you'll soon develop your own quick isolation tests for different complaints. These tests will essentially be short-cut versions of the complete procedures now to be given. Mastery of these complete isolation procedures is highly essential before you can develop these short-cuts.

The isolation method to use will depend greatly on the conditions encountered and on the test equipment available. The tests you make in the home may be quite different from those made in your shop, yet either type of test will serve to locate the trouble. For instance, you may have a signal tracer or vacuum tube voltmeter in your shop, but you cannot carry this equipment around with you, as it is too delicate or too bulky. Therefore, the tests you make in the customer's home will have to be simple ones—tests that you can make with your multimeter or signal generator, or perhaps with no equipment at all.

Of course, we do not mean that simple tests cannot be used in the shop or that shop tests cannot be used in the home. The best method to use will depend on the complaint, the type of receiver and the available equipment. It is very important to be familiar with all basic methods, so you can choose the proper one for each case. Try all the methods your equipment permits, until you find the best procedure for each complaint.

## BASIS FOR ISOLATION TECHNIQUE

All isolation techniques depend upon the simple fact that signals normally travel through vacuum tubes only in one direction. Thus, in tubes having control grids, a signal in the grid circuit can move to the plate circuit, but cannot move in the reverse direction. This is even more true with screen grid tubes, because here the screen grid acts as a shield for any signal which might attempt to go in the reverse direction.

**T.R.F. Receiver.** The one-way-travel characteristic of signals in a radio receiver can be illustrated by means of the t.r.f. receiver block diagram shown in Fig. 2. Here a signal picked up by the antenna-ground system travels through each stage in turn, finally arriving at the loudspeaker. Under normal conditions the signal never travels in the reverse direction.

The amplitude or strength of the signal will, as a rule, be increased by each stage and will be greater in the plate circuit of a stage than in the grid circuit. These facts are of prime importance in trouble-isolating techniques.

Although we think of noises or howling as audio signals, it is possible for these interfering signals to originate in and pass through r.f. stages. This is possible because noise signals can set resonant circuits into natural oscillation, and the r.f. signal created in this manner will be modulated by the noise. The important point to

remember is that the noise signal will not pass from its point of origin back toward the antenna, and that the stages in this direction will be free from noise.

Hum signals get through r.f. stages in a different manner. If the r.f. stage is operating over a non-linear portion of its dynamic characteristic and a hum voltage is present, the hum voltage will be added to the incoming modulated carrier signal. Of course, the hum modulated signal moves along toward the loudspeaker, not in the reverse direction toward the antenna. This means that stages ahead of the point where the hum modulation took place will have the original hum-free signal.

**Superheterodyne Receiver.** When the additional signal transformations taking place in a superheterodyne circuit are clearly understood, isolation techniques can be applied just as easily as in a t.r.f. circuit. In a super, we still have our signals traveling in a forward direction toward the loudspeaker at all times, as shown in Fig. 3. Neither the desired input signal nor any undesired signals originating in the r.f. amplifier (pre-selector) can be forwarded through the i.f. amplifier with full strength if the local oscillator is dead or badly

misaligned. Some signals near the i.f. value may force themselves through the i.f. amplifier due to their initial strength, but these will be quite weak at the output because the i.f. amplifier is adjusted to pass with maximum gain only signals at the i.f. value.

In servicing a superheterodyne receiver, your test equipment must naturally be able to produce or detect signals at the i.f. value as well as at the other frequencies which exist in the receiver.

► We will now cover the basic methods of isolating the defective section and the defective stage, leaving circuit and part isolation for another lesson. Further, more complete details on using these methods will be given as the various complaints are covered elsewhere.

#### DEFECTIVE SECTION ISOLATION

Isolation of the defective section breaks the receiver into at least three parts, the r.f. section, the a.f. section, and the power supply.\* This procedure reduces the number of stages which must be checked, and narrows down the possible sources of trouble

\* The r.f. section of a superheterodyne includes everything before the second detector, such as the i.f. stages, first detector, oscillator and preselector stages.

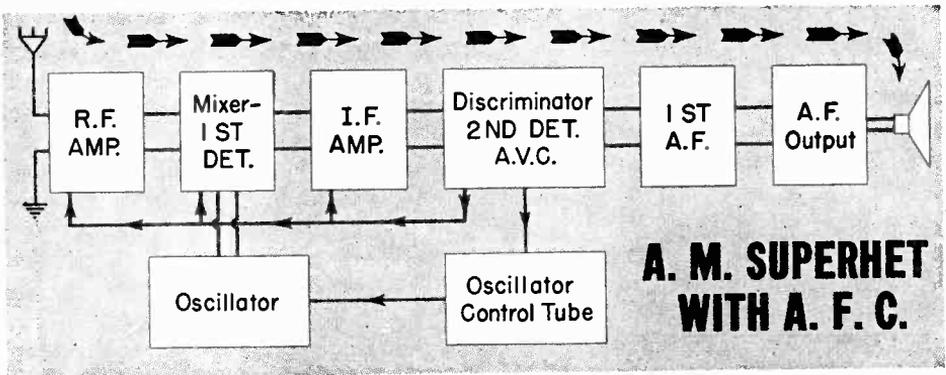


FIG. 3. Block diagram of a superheterodyne receiver having a.v.c. and a.f.c.

The symptom may indicate the section directly, or using effect-to-cause reasoning may lead directly to the defective section. For example, distortion is normally an audio trouble, while modulation hum is an r.f. problem. If the receiver works on one or more bands but not on others, the trouble must be in the preselector or oscillator circuits of the inoperative band, etc. However, if the trouble does not suggest which section is defective, isolation frequently can be accomplished by such simple acts as rotating the tuning or volume controls, pulling out a tube, or touching a grid cap.

If noise or oscillation is the complaint, pulling out the last i.f. tube or the detector tube will determine whether the trouble is in the r.f. or a.f. amplifier; if the noise or squeal disappears, it originates in the r.f. section. Tuning the receiver will determine if modulation hum is present; hum heard only when a station is tuned in originates in the r.f. section.

The volume control will either vary the sensitivity of the r.f. section or will be at the input of the a.f. amplifier in a modern set. Hence, if the control reduces hum, noise or squeals when rotated toward minimum volume, an r.f. trouble is usually indicated. Otherwise, the trouble is in the audio section. Remember, however, that the control itself may develop a poor contact or an open and thus may be the source of noise or cause hum.

When a receiver has a tuning indicator, its action will frequently enable you to identify the defective section. If the set is dead and the tuning eye does not have a green glow, there usually is a defect in the power supply. If the tuning indicator works normally when you tune in a signal,

the r.f. section, second detector, a.v.c., and the power supply are in good condition. This definitely places the defect in the audio section. If the indicator does not have normal action, you know that the r.f. section or the detector is not handling the signal properly. If the complaint is intermittent reception, watch for variations in the tuning indicator when the fading occurs; any variations indicate an r.f. trouble. Sometimes noise or oscillations can be localized by watching for variations. However, remember that power supply variations may change the tuning meter indication.

If the set does not have a tuning indicator, it is sometimes desirable to connect across the diode load a high-resistance d.c. voltmeter (5000 ohms per volt or higher, using the highest range which gives a noticeable deflection). The voltmeter will act like a tuning indicator when the receiver is tuned to a local station, if the sections ahead are working right. In the case of a triode or pentode detector having a plate load resistor, a measurement of the d.c. voltage between plate and cathode will give this same information, because this voltage will change as a station is tuned in if the detector and all preceding stages are intact.

#### **DEFECTIVE STAGE ISOLATION**

Signals travel from the input toward the output until the path is interrupted by some defect. Also, if noise, hum, distortion or oscillation occurs, the symptom will start at some point and will become a part of the incoming signal or will send through a signal voltage of its own. Therefore, if we have some instrument available with which to follow the signal along, stage by stage, we will come to the point where the break exists, where the noise, hum or oscil-

lation first appears, or where the distortion takes place. We will thus have located the defective stage.

We can go in the reverse direction also, by introducing a signal of the proper kind, such as from a signal generator or by a circuit disturbance. As the signal source is moved along from the output toward the antenna, the signal will come through normally as long as the stages between the loudspeaker and point of signal injection are working properly. As we move along, the disappearance of the signal or the appearance of the hum, squeal or distortion along with the signal will point to the defective stage.

If the trouble is hum, a squeal or noise which comes through the set even with no signal tuned in, we can use a stage interruption or blocking procedure to localize the trouble. The important facts about these tests will be explained now; later we will learn how to apply them for various receiver complaints.

#### **Circuit Disturbance Method.**

For a dead receiver, the simplest stage isolation method is the circuit disturbance test. This test depends upon the fact that any abrupt change in plate current, whether it be a rise or fall, creates a so-called surge signal which is relayed through following stages to the loudspeaker, and there produces an audible click or other indication.

The surges which produce indications in this test can be created by *disturbing* the circuit in various ways, hence this test is called the *circuit disturbance* test.

A surge introduced into the r.f. or i.f. section of a superheterodyne receiver will cause the resonant circuits to be shocked into momentary oscillation at the resonant frequency, producing an r.f. signal which carries the surge signal through the circuit. A disturbance introduced into an

audio amplifier is a complex audio signal or noise impulse which travels by itself through the following audio stages.

The necessary disturbance can be introduced in a number of ways. For example, pulling a tube out of its socket cuts off plate current suddenly, creating the required surge or disturbance. Reinserting a tube in its socket causes a sharp rise in plate current, and again a surge is created.

Tubes cannot be pulled out for this purpose in universal receivers, however, because all tube filaments are in series. Also, in some battery sets, removal of one tube may make filament voltages of other tubes too high and possibly cause burn-outs. The following alternative methods of introducing disturbances can be used in these cases as well as in standard a.c. receivers.

Opening a grid circuit by removing the top-cap connection usually makes the grid take on a high negative charge which sharply lowers plate current and creates a surge or disturbance. Putting the top-cap connection back on a tube brings the plate current sharply to normal, again creating a surge.

If the tube does not have a top cap, the control grid can be momentarily shorted to the tube cathode with a test lead. This removes the bias, causing a sharp change in the plate current. Remember, touch these terminals just long enough to get an indication.

Touching the grid lead at the socket, shorting the rotor and stator plates of a variable condenser, or touching a tube top cap is sometimes sufficient to create a disturbance.

#### **Stage Interruption Method.**

For locating hum, noise or oscillation, the stage interruption method will prove very helpful. The procedure is basically similar to the circuit dis-

turbance test; however, we are now interested in blocking the circuit, not introducing a surge. Therefore, we can move in either direction, although it is customary to start at the output and work back toward the antenna. As we pull out tubes, the hum or noise will stop as long as it originates further back toward the input. When we pull out a tube and find the noise or hum remains, the trouble is in the last stage interrupted, that is, the next stage toward the loudspeaker.

In universal and battery receivers, it may be impractical to pull out tubes. As an alternative procedure, circuits can be blocked by short-circuiting the control-grid input circuit. In other words, short-circuit the resistor or transformer secondary across which the signal normally appears. If one end of this part connects to chassis or ground, a test lead can be held between the control grid terminal and the chassis. However, if the return end of this part connects to some bias source, it will be necessary to find this terminal, so the test lead can be held right across the part. Otherwise, the test lead may provide a short-circuit path for the bias voltage.

To avoid having to worry about this, a .5-mfd. condenser can be used in place of the test lead. The condenser will bypass signals and thus block the stage without providing a short-circuit path for the bias.

Moving the test lead or condenser along from grid to grid toward the antenna, we will eventually reach a point where the hum, noise or squeal is still heard coming through to the loudspeaker, even while the condenser or test lead is across a grid circuit. The defect is then between this grid and the next grid circuit toward the loudspeaker.

Of course, if the symptom remains when the grid of the output stage is

blocked, the trouble is originating in the power supply or output stage. Further, if the intensity of the symptom just decreases instead of disappearing when a stage is blocked, the symptom is either being fed into all stages by way of the power supply, indicating a power supply defect, or it is originating in a prior stage and is being passed around the blocked stage through common power supply coupling.

**Signal Tracer Method.** Neither the circuit disturbance or circuit blocking methods require that a signal be tuned in. These simple procedures will work satisfactorily only when the defect does not depend on the presence of a signal. Hence, a trouble like distortion or intermittent reception definitely requires that a signal be tuned in before the trouble can be localized. Then, some means of tracing this signal from stage to stage is required. Therefore, with signal tracing equipment, we not only can determine whether signals are coming through; we also can determine the condition of the signal.

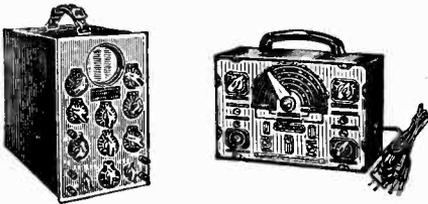
Instruments capable of following the signal through the entire radio include r.f. vacuum tube voltmeters, cathode-ray oscilloscopes, and special signal tracing instruments such as the Chanalyst, etc. Although only special instruments such as the Chanalyst are commonly known as "signal tracers," the other devices are used in a similar manner.

To use any kind of signal tracer, we first must have a signal voltage, which may be from a broadcast station or a signal generator. Starting at the antenna and working toward the loudspeaker, we would follow the signal along until it is interrupted or distorted in some manner. This localizes the defect to that particular stage. Similarly, a signal

tracer can be used to run down noise or hum which are also signals.

A defect causing low sensitivity can be found by noting the relative strength of the signal as you progress toward the output. If the signal level does not increase normally in some stage, you have located the defective stage.

**Signal Generator Method.** The signal generator is a source of voltage, used primarily to align receiver circuits. However, it can be



A typical cathode ray oscilloscope and a signal generator.

used for defective stage isolation where the complaint is a dead receiver. The signal generator can also be used to locate certain types of noise and hum which only exist when a signal is present. Further, when used with a cathode-ray oscilloscope, the source of distortion can be discovered.

With the signal generator tone-modulated and adjusted to the proper frequency, we can find the dead stage by starting at the input of the second detector and working back toward the antenna.

If no loudspeaker signal is heard with the s.g. at the second detector input, the trouble is in the audio section. If a signal is heard, work back toward the antenna until the signal stops, showing that you have passed through the defective stage.

**Summary.** The foregoing description is only intended to make you acquainted with the basic methods

used to localize sources of trouble. When each complaint is taken up later, you will get a more detailed description of these methods. You will learn that some methods work better than others for particular complaints, so you will be shown how to choose the best method for the complaint and your equipment.

**Special Receiver Circuits.** Any trouble in special circuits like a.v.c. and a.f.c. circuits can be isolated even though these circuits do not amplify or otherwise act directly on incoming signals. A knowledge of how the circuits work and what they produce is essential for any isolation procedure, however.

An a.v.c. circuit produces a d.c. bias voltage which increases in value as the receiver is tuned to resonance with an incoming signal frequency. In some sets, the amount of the a.v.c. voltage is indicated by an electric eye tuning indicator or a meter-type indicator. In sets without visual indicators, you can measure the a.v.c. voltage across the diode load or across individual a.v.c. filter condensers. Use a d.c. voltmeter having a sensitivity of 5000 ohms per volt or higher, since a low-resistance meter will change the circuit resistance and affect the production of a.v.c. voltage. A high-resistance meter will show the changes in a.v.c. voltage which occur as the receiver is tuned to various stations and will determine whether a.v.c. voltage is available for every controlled stage.

Turning now to a.f.c. circuits, we know that when they are working normally, the discriminator output voltage will be alternately positive and negative as the receiver is tuned through a station. At the same time, the volume will remain essentially constant for several kilocycles in either direction from the point at

which the local station should come in.

If the correct variation in voltage is obtained at the discriminator but there is no station-holding action as you tune to either side of the correct setting, a defect in the oscillator control circuit is indicated.

**F.M. Receivers.** Trouble in f.m. receivers is traced the same way as in a.m. receivers. The incoming signal passes through all of the stages in exactly the same way as in an a.m. superheterodyne receiver. The only major differences are the use of limiter and discriminator stages as last i.f. and detector respectively, and the fact that the a.v.c. voltage is taken from the limiter stage, because this stage has a d.c. voltage which varies with signal strength.

By feeding into an f.m. receiver a

constant-frequency r.f. signal produced by an unmodulated signal generator, you can trace continuity for signals from the antenna through the discriminator stage. By connecting a milliammeter in the plate circuit of the limiter, you can check the limiter saturation level and its ability to maintain almost constant output for all values of r.f. input. By varying the frequency of the r.f. input a certain amount in either direction while a d.c. voltmeter is connected across the discriminator output, you can measure the deviation voltage for a given frequency deviation and thereby determine if the discriminator is working properly. All other circuits in an f.m. receiver can likewise be checked exactly as in a.m. superheterodyne and t.r.f. receivers.

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## Test Equipment for Signal Tracing

Before going on with the use of isolation procedures for a particular complaint, we should learn something about the special equipment used for this purpose.

The basic test instruments used in radio servicing are the multimeter, the tube tester and the signal generator. In addition to these, we have special instruments, each of which has certain advantages for specific types of trouble-isolating jobs. These include vacuum tube voltmeters, signal tracing instruments and cathode-ray oscilloscopes.

Circuit designs vary considerably in instruments made by different manufacturers, because designers may use different methods for securing the same result. From your standpoint as a Radiotrician, however, you are

more interested in the *purpose* of each type of service instrument, the *basic design facts* which affect the use of the instrument, and the *limitations* of various instruments. Only signal tracing equipment will be considered here, as the basic multimeter and tube tester are covered in other lessons.

**Signal Generator.** This is a dual purpose instrument, necessary for receiver alignment and usable in a number of signal tracing procedures.

A frequently used circuit arrangement is shown in Fig. 4. Tube  $VT_1$  is the r.f. oscillator tube connected to the tank circuit  $L_1-C_1$ ; coil  $L_2$  provides plate to grid feedback. A band switch  $SW$  is provided to change coils  $L_1$  and  $L_2$  when different frequency ranges are required for alignment or signal tracing purposes. **Condenser**



probe for ordinary service work. However, when an extended frequency range is desired, an acorn-type tube is commonly used. This permits very short leads, making input resonance occur at a very high frequency and reducing the input capacity to a low value, 6 to 10 mmfd. or so. As a result, r.f. voltages up to at least 150 megacycles can be measured with a properly designed r.f. voltmeter.

In Fig. 5, the 955 triode tube is connected as a diode in a shunt-fed

to the input signal flows through the voltage divider resistors. This produces a d.c. voltage drop, so a d.c. vacuum tube voltmeter may now be used to indicate the signal level. The d.c. voltmeter circuit shown in Fig. 5 is of the balanced bridge type, having cathode degeneration to provide a linear scale. The combination  $R_6-C_2$  is an r.f. filter used to keep the r.f. off of the grid of the 6SN7 tube.

**Signal Tracing Instruments.** An r.f. vacuum tube voltmeter measures every signal (desired or otherwise)

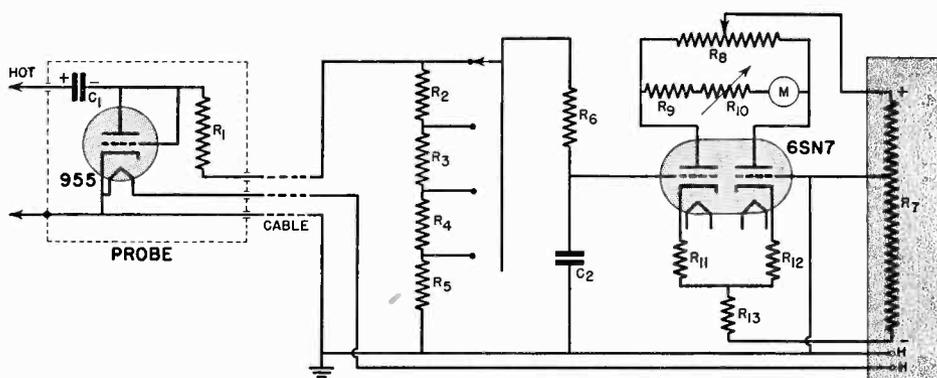


FIG. 5. An r.f. vacuum tube voltmeter capable of covering a wide frequency band without appreciable loading of the voltage source.

rectifier circuit. When an r.f. voltage alternation (half cycle) makes the *HOT* probe positive, electrons flow from the 955 cathode to the plate, charging condenser  $C_1$  to the polarity shown. For the other a.c. alternation, the tube is blocked, and condenser  $C_1$  discharges through resistor  $R_1$  and the voltage divider  $R_2, R_3, R_4$  and  $R_5$ . The time constant of  $C_1$  and the resistors is chosen so  $C_1$  discharges only a slight amount during a half cycle. Hence, the diode passes current to recharge  $C_1$  only on the extreme positive peaks of the a.c. cycle, thus negligibly loading the source of voltage.

A d.c. current which is proportional

at the point of connection. A signal tracing instrument eliminates this difficulty by placing a tuned r.f. amplifier ahead of the indicating circuit. Unfortunately, r.f. amplifier gain varies for different frequency settings, so that an indicator in the output could not be accurately calibrated in terms of input voltage, especially if a wide range of frequencies is involved. The chief purpose of a signal tracer of this type is to measure relative changes in signal strength and to indicate the presence of signals of particular frequencies.

The basic circuit for a signal tracing instrument is shown in Fig. 6. We

have a two-stage tuned r.f. amplifier using type 6AC7 or similar tubes, with provisions for band switching. The usual over-all frequency range is from about 90 kc. to about 16 megacycles. Some instruments will have only one r.f. stage, while others may have two or three.

The r.f. amplifier feeds a linearly-operated diode detector whose output may be used for visual or audible signal indication. R.F. and a.f. components in the output of the diode are filtered out, leaving only a d.c. volt-

AMP." in Fig 6 can be connected to a cathode-ray oscilloscope for analysis or to an a.f. amplifier and loud-speaker. Some instruments will have a built-in audio channel, with provisions for connecting a headphone or loudspeaker, while others may have a built-in loudspeaker. With an audio channel, a serviceman can listen for the signal and also hear distortion, hum or noise.

The tuned r.f. amplifier in a signal tracing instrument is designed to give high gain. Each frequency range is

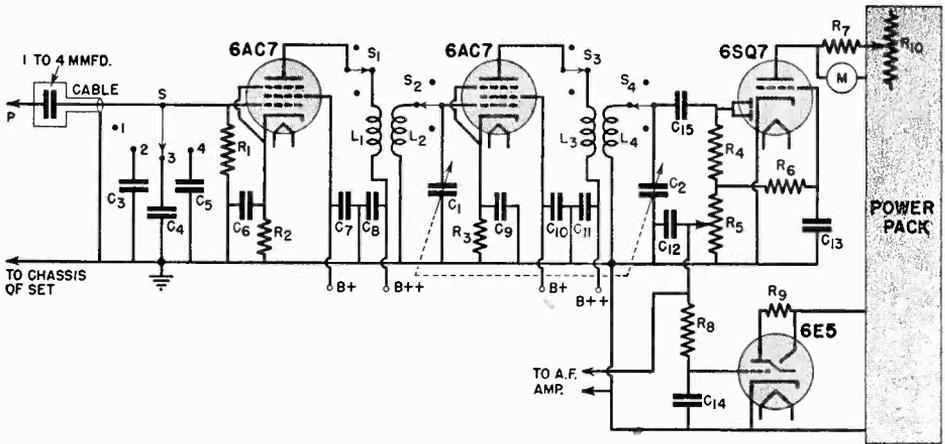


FIG. 6. A fundamental diagram of a multi-band signal-tracing instrument. Coils are shown for only one band.

age proportional to signal strength. This voltage can be used to actuate an electric eye tuning indicator, such as the type 6E5 tube shown.

Another indicating method involves applying the d.c. voltage to a d.c. amplifier (the triode section of the 6SQ7 tube), and using the resulting current to actuate an ordinary meter (*M*). The normal d.c. plate current of the triode through meter *M* is canceled out by adjusting potentiometer *R*<sub>10</sub> in the power pack. When the adjustment is properly made, the meter deflection is proportional to the carrier input signal strength.

The two leads labeled "TO A.F.

made as uniform as possible, and all bands are designed to have approximately the same gain. Only tuned r.f. amplifiers can be used here, because superheterodyne circuits which might give more uniform gain would introduce so many interfering signals that confusion would arise.

The shielded cable used as the test lead has a small series condenser built in the test probe. This condenser reduces the effect of the cable capacity by acting with it as a capacity voltage divider. This makes the input voltage division constant for frequencies up to about 5 mc. Above this frequency, the shielded cable will begin to upset

the voltage division by starting to act as a resonant line. As most radio service measurements are made below 5 mc., this does not greatly matter.

A capacity voltage divider is used at the input to extend the voltage range. The capacity required is added by switch  $S$ .

When an electric eye tuning indicator is employed, tuning condenser  $C_1$  and  $C_2$  are simultaneously adjusted for maximum deflection, and switch  $S$  is then set to a position which permits adjusting  $R_5$  so the eye will just barely close. The reading of  $R_5$  is multiplied by the multiplying factor of  $S$  to give an indication of the input voltage.

By shifting the probe to another r.f. terminal, tuning  $C_1$  and  $C_2$  to resonance and then adjusting  $S$  and  $R_5$  for closure of the electric eye indicator, another value is obtained. Dividing one value by the other gives a ratio corresponding to the voltage gain or loss between the two points. Thus, as you probe from the antenna step by step toward the loudspeaker, greater values should be obtained in a normal receiver.

Regardless of the indicating means, this signal tracer can be used in all the r.f. and i.f. amplifier stages. A separate audio amplifier channel is necessary to work from the second detector to the loudspeaker. This audio channel is usually built into a complete signal tracer, and usually uses the same kind of meter or tuning eye indicator as the r.f. channel. For audio work, the capacitance of the input cable is normally unimportant.

**Cathode-Ray Oscilloscope.** This instrument makes it possible to see the wave form of the signal at any point in a circuit. In the r.f. and i.f. sections of a receiver, the modulation envelope is studied. In the a.f. sys-

tem, any distortion, ripple current, regeneration, noise and motorboating are clearly evidenced when a pure sine wave audio signal is fed into the receiver circuit.

The functional diagram of a cathode-ray oscilloscope (c.r.o.) is shown in Fig 7. Probe  $P$  may be connected directly to a terminal which is hot with respect to the chassis. Any a.c. voltage on this probe causes vertical movement of the dot on the c.r.o. screen. Potentiometer  $R_1$  limits the vertical movement to any desired amount. At the same time, the horizontal sweep circuit moves the spot horizontally; if synchronized with the input signal, it will give a stationary curve.

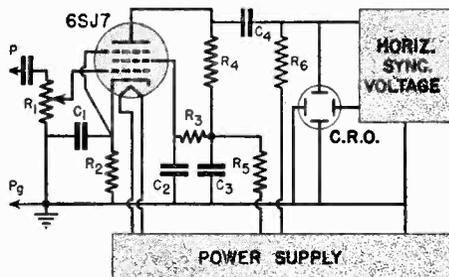


FIG. 7. Basic cathode-ray oscilloscope circuit, used for wave-shape analysis.

Although the oscilloscope usually has sufficient amplification to be used by itself, it can also be connected to the audio output terminals of a signal tracing instrument. Then any r.f. or i.f. stage in a superheterodyne receiver can be analyzed for defects which affect the modulation envelope, provided the input to the receiver is a sine-wave modulated signal from a signal generator. If the modulation frequency is low, about 400 cycles, sharpness of resonance in the test circuit can be ignored, because a highly selective circuit only cuts off (attenuates) the higher audio frequencies.

# Isolation Techniques for a Dead Receiver

We will now take up the complaint of a dead receiver and show just how the different isolation techniques can be applied to this complaint. Methods applicable to the various other complaints will be taken up in other lessons.

Whether you should make a section isolation or should go on to a stage isolation procedure will depend on your preferences and experience. Usually, a professional serviceman will go at once to the stage isolation tests when the complaint is "dead set," as the necessary tests can be performed rapidly. We will now take up these methods, starting with the circuit disturbance test.

## CIRCUIT DISTURBANCE TEST

To isolate the dead stage by means of the circuit disturbance test, start at the output stage and work backward\* toward the antenna terminal, creating in each stage in turn the required surge. If you hear the characteristic click or thud in the loudspeaker as a result of this surge, you can be reasonably sure that the plate circuit of the tube you are disturbing and all following stages are in good enough condition to permit passage of signals.

If no click, thud or howl is heard when a stage is disturbed, you have isolated the defect to that tube, the electrode circuits of that stage, the coupling to the next stage toward the loudspeaker, or the grid circuit of the next stage.

Quite often, touching a grid will set up circuit oscillations which cause a howl instead of a click or thud. Re-

gardless of the method used to create a disturbance, the indicated click or howl will not be heard in the loudspeaker if a stage between the point of disturbance and the loudspeaker is dead.

The only stages which may not give an indication of some sort when disturbed are a diode detector, a separate a.v.c. tube, an oscillator control tube in an a.f.c. circuit, a tuning indicator tube, and other tubes which are not in signal circuit paths or in the power supply system. The diode detector has no d.c. plate current to be disturbed. However, disturbing the preceding i.f. tube does cause a click to pass through the detector, as the surge is then acting as a signal. No click indicates either a defective last i.f. or defective detector stage, if the audio stages responded properly.

**Locating the Defect.** Having localized the defect to this extent, you would first test the tube in the section concerned, if you had not already done so. Next would come either a test of electrode voltages or circuit continuity, until you localized the defect to one of the suspected circuits and eventually to a particular part. This will be taken up when we discuss defective circuit and part isolation.

**Possible Complications.** The circuit disturbance test is admittedly not entirely fool-proof. Conditions can exist which produce confusing indications.

For example, when you have progressed well into the i.f. or r.f. section with a circuit disturbance test, the click or other indication in the loudspeaker may be quite weak if any of the stages between the disturbance

\* Backward is used here to mean in a direction opposite to the normal direction of signal travel.

point and the loudspeaker are badly out of resonance (out of alignment). Sometimes it is possible to hear an indication of some sort, even with an actual defect in the chain of stages between the disturbance point and the loudspeaker, because a circuit disturbance can be relayed through the power supply system to the loudspeaker. This case can be recognized with a little experience, as the indication will be unusually weak.

If no click or other indication is heard for the output stage, or if the indications are weak for all stages being disturbed, a loudspeaker or power supply defect is indicated. Measure the output voltage of the power pack or measure the plate or screen grid voltages of the output stage.

In the case of a push-push output stage employing zero bias, no indication will ordinarily be obtained when the control grid is shorted to the cathode. If a disturbance introduced in the preceding a.f. stage is heard, however, you know that the push-push output stage is normal.

Most servicemen start by pulling out the first audio tube instead of the power output tube, particularly when there is only one output tube. This saves a little time, as a satisfactory indication acts as a section isolating test and eliminates the need for an output stage disturbance. Furthermore, there is a possibility that the large surge produced by an output stage disturbance may puncture the insulation in the output transformer or in a filter condenser. This rarely occurs, and then only where the part is already weakened, so perhaps it is just as well that the part be replaced before it fails of its own accord.

### **SIGNAL GENERATOR TEST FOR A DEAD STAGE**

If you apply a test signal between

any grid or plate and *the chassis* in a dead receiver, and the loudspeaker reproduces the modulation tone of that signal, you know that the signal path is intact from the signal source to the loudspeaker. This is the basis for an isolation procedure involving a signal generator.

To isolate the dead stage with the aid of a signal generator which has independent a.f. output terminals, you can start at the input of the final a.f. stage and work backward toward the antenna, connecting to the input of each stage in turn and adjusting the signal generator to provide the proper type of signal for each circuit. When you first fail to hear the signal in the loudspeaker, you know that you have just passed through the defective stage.

If the signal generator does not have provisions for separate a.f. output, you can start at the plate of the last i.f. tube, with the signal generator set to the i.f. value. If the signal comes through, the second detector, a.f. stages and loudspeaker are all right; if not, some other isolation procedure must be used on these stages (you can listen to the output of each a.f. stage in turn with headphones which are used in series with a condenser to prevent shorting of the plate supply, or you can use a circuit disturbance test).

A condenser should be in series with the hot signal generator lead, in order to prevent shorting of d.c. electrode voltages by the signal generator when the hot lead is applied to a grid or plate terminal. Most signal generators have a built-in series condenser, but if none is present you can use any condenser value from .001 mfd. to .1 mfd., rated at 600 volts d.c. or higher.

In working from the input of the second detector up to the input of the mixer-first detector, the signal gen-

erator should be adjusted to the i.f. value of the receiver.

The i.f. value will usually be given on the schematic diagram of the receiver; if you do not have a diagram or if the value is not given, connect the s.g. to the grid of the last i.f. tube and sweep its frequency slowly down from 500 kc. to 100 kc. until you get maximum response. The s.g. setting will then be approximately the i.f. value. The correct value will probably be the nearest standard value, such as 175, 265, 370, 455, 465 or 480 kc. If the trimmer condenser adjustments have been tampered with, you may have to realign the i.f. stages as you move along toward the first detector.

After arriving at the input of the first detector and checking passage of the signal at the i.f. value, change the signal generator to a frequency corresponding to the receiver tuning dial setting. A response from the loudspeaker now proves that the local oscillator is working.

The signal generator can now be moved right up to the antenna and ground terminals, for a check of the preselector.

If no signal is heard when you are at the input of the first detector, with the signal generator set at the frequency corresponding to the receiver tuning dial setting, the oscillator is dead or out of alignment. Check the oscillator as follows: Tune the receiver to the frequency of a powerful local station, set the signal generator to the frequency of the local oscillator (to the receiver tuning dial setting plus the i.f. value), remove the signal generator audio modulation if a switch is provided to give unmodulated output, then vary the output frequency of the signal generator a few kilocycles in either direction. If the station is now heard, the defect is in the oscillator stage.

## VACUUM TUBE VOLTMETER TEST FOR A DEAD STAGE

With an r.f. vacuum tube voltmeter, you can check r.f., i.f. and a.f. voltages in both the plate and grid circuits of a dead receiver, and isolate the defective stage by verifying the presence or absence of the signal from a signal generator connected to the receiver input.

With the signal generator set to produce full output, you should be able to start at or near the antenna with the vacuum tube voltmeter, and proceed toward the loudspeaker, stage by stage, until you pass through the stage at which the signal vanishes. The signal will become stronger and stronger as you move toward the loudspeaker, making it necessary to change to less sensitive ranges on the instrument.

With a vacuum tube voltmeter, you must be on guard for conflicting readings due to hum and other undesired signals, which may give meter deflections even when the signal you are tracing does not exist. By tuning the receiver or signal generator slightly away from resonance, or by turning the signal generator off and on each time a test is made, you can determine whether the deflection is due to the desired signal or is due to hum, noise or other signals. The indication will disappear or drop considerably during this test if it is due to the proper signal.

It is necessary to retune any resonant circuit when the v.t.v.m. is connected across it, as the capacity of the device may detune the stage completely so that you get an indication of a dead stage, or may tune the circuit to some other signal. Remember to restore the original trimmer setting when you move on to another circuit.

With a vacuum tube voltmeter, you



Courtesy RCA Mfg. Co.

The RCA-Rider Chanalyst uses tuning eyes to indicate the strength of the input to the r.f., a.f., and oscillator channels. Another eye is used as a wattage indicator. The meter is part of a built-in d.c. vacuum tube voltmeter.

can measure the output of the local oscillator directly across its tank circuit. If the instrument is also designed for d.c. measurements, you can connect it across the diode detector load for use as a tuning indicator where the set does not have an indicator. If its reading changes as a station is tuned in, you know that the r.f. system is active, and can concentrate on the a.f. stages.

It should be emphasized that many complications can occur when using a vacuum tube voltmeter, because it does not have a preselector and hence responds to all signal frequencies. This led to the development of signal tracing instruments which do have tuned stages.

#### SIGNAL TRACING INSTRUMENT TESTS FOR A DEAD STAGE

A signal tracing instrument indicates the presence of signals at a particular frequency. It can be accurately tuned to any frequency

which normally exists in a receiver, and can thus be used to determine exactly what frequencies are present in each stage, and their relative strength.

Since the usual signal tracing instrument has a high input impedance, it may be connected between any electrode and the chassis without serious detuning of tuned circuits. The trouble can thus be isolated to a *particular circuit* in the dead stage.

In using a signal tracing instrument to isolate a dead stage, the receiver is connected to an antenna and tuned to a local station, or is connected to a signal generator and both receiver and signal generator are tuned to the same frequency. The ground probe of the signal tracing instrument is then connected to the receiver chassis, and the hot probe is connected in turn to the antenna, the grid of the first r.f. tube, the plate of the first r.f. tube, the grid of the mixer tube, the plate of the mixer tube, the

grid of the first i.f. tube, the plate of the first i.f. tube, etc., in sequence, working toward the loudspeaker.

In each stage, the signal tracing instrument should be tuned to the resonant frequency of the circuit to which it is connected. For instance, if the receiver is tuned to 1080 kc. and the i.f. value of the receiver is 455 kc., the signal tracing instrument would be tuned to 1080 kc. up to the input of the mixer-first detector. From there up to the input of the second detector, the signal tracer is tuned to 455 kc., corresponding to the i.f. value.

Always tune the signal tracing instrument for maximum output indication. This will either be maximum meter deflection, maximum closure of the electric eye indicator or maximum loudness of the audible signal, depending on the design of the instrument.

In this example, if you do not get a 455-kc. signal at the output of the frequency converter, you can tune the signal tracing instrument to 1080 kc. to see if the input signal is coming through the converter stage. Next, tune the signal tracing instrument to the oscillator frequency of 1080 plus 455, or 1535 kc., to see if the oscillator is working. Both signals will be present in the plate circuit of the first detector but will be weak, as the i.f. tuned circuit rejects these signals. However, they can be ob-

served if everything is normal.

Failure to get a signal of the correct frequency at a particular point in the circuit isolates the defect to that circuit. Then, a circuit continuity test, followed by parts testing, will usually reveal the defect in short order. However, with a signal tracer it is often possible to check and locate the defective part without using other instruments.

An audio channel is necessary to trace from the output of the second detector to the input of the loudspeaker. Usually such a channel is provided on the signal tracer.

As you progress toward the loudspeaker with a signal tracing instrument, the signal strength should increase, making it necessary to reduce the gain of the instrument. It should be noted that a great decrease will be observed when going from the primary to the secondary of an output audio transformer, as this is a voltage step-down device. An increase in signal strength will be noted, however, when going from primary to secondary in a step-up audio coupling transformer or in a transformer-coupled r.f. or i.f. stage in which only the secondary is tuned. In passing from a plate circuit to a following grid circuit where R-C coupling or double-tuned transformers are used, a slight decrease in signal strength is normally to be expected due to losses in coupling.

# Lesson Questions

Be sure to number your Answer Sheet 37RH-2.

Place your Student Number on every Answer Sheet.

Send in your set of answers for this lesson immediately after you finish them, as instructed in the Study Schedule. This will give you the greatest possible benefit from our speedy personal grading service.

1. If the pilot lamp does not glow and none of the tubes become warm when an a.c. receiver (not universal) is turned on, what things would you check before removing the chassis? *power socket, coupling cond. trans. No. 1 rectifier.*
2. If noise disappears when you short together the antenna and ground terminals of a noisy receiver, is the cause of noise likely to be inside the chassis? *No.*
3. Why are felt washers often used between control knobs and the front panel of a receiver? *To reduce mech. feedback*
4. When distortion is the complaint in an a.c. receiver, effect-to-cause reasoning indicates that the voltage across grid resistors in resistor-condenser-coupled a.f. stages should be measured. If one grid is positive with respect to the chassis, and this positive voltage still exists when the a.f. tube of that stage is removed without turning off the receiver, which part is defective? *Coupling Condenser*
5. When using effect-to-cause reasoning, what symptoms would you expect from an open output filter condenser? *Hum & lowered sensitivity*
6. State the basic fact upon which all stage-isolating techniques are based. *signal travels in one direction through circuit*
7. Suppose that you are carrying out a circuit disturbance test for isolating the defective stage in a 7-tube a.c. t.r.f. receiver. You pull out and replace tubes one by one, working from the loudspeaker toward the antenna, and get clicks in the loudspeaker for the a.f. output tube, the first a.f. tube, the detector tube, and the third r.f. tube, but no click for the second r.f. tube and the first r.f. tube. Which stage contains the defect? *second r.f.*
8. In making gain measurements with a signal tracing instrument, would you expect an increase in a.f. voltage when going from the primary to the secondary of the output transformer?
9. Suppose that the i.f. value of a receiver is not given and that you are reasonably sure the i.f. trimmers have not been tampered with. State the procedure that you would follow, using a signal generator, to find the i.f. value.
10. What is the advantage of a signal tracer over an r.f. vacuum tube voltmeter?



## DIG A LITTLE DEEPER

A poor South-African farmer struggled for years to gain a living out of his rocky soil, then finally gave up in despair and sought his fortune elsewhere. Years later, more wealth was being dug out of his rocky old farm every day than he had ever dreamed existed. His farm had become a diamond mine!

Many of us struggle along just like that poor Boer farmer, never dreaming that success could be ours if we dug a little deeper right where we are. Millions of men are just barely getting along today, when they have the ability to do much better things, simply because they lack confidence in themselves. They are victims of mental defeat; they don't believe they can do anything better.

If *you* lack self-confidence—if you lack a sense of mastership, a consciousness of power, and a victorious mental attitude, begin now to cultivate self-confidence. How? Make your decisions with confidence and speed, and stick to them. Dig into your work a little harder, keep going a little longer, and soon you'll have the self-confidence which carries you speedily to success.

*J. E. Smith*

**HOW TO ISOLATE  
THE DEFECTIVE CIRCUIT  
AND PART**

38RH-2



**NATIONAL RADIO INSTITUTE**

**WASHINGTON, D. C.**

**ESTABLISHED 1914**

# STUDY SCHEDULE NO. 38

For each study step, read the assigned pages first at your usual speed. Reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind. Study each other step in this same way.

1. Analysis of the Defective Circuit . . . . . Pages 1-8

Once you get the trouble pinned down to a particular stage in a receiver, a few minutes study of the circuits for that stage will often suggest one or two simple tests which will lead you directly to the defective part. A number of actual examples in this section show you exactly how to use effect-to-cause reasoning for this purpose in various types of circuits.

2. Isolating the Defective Circuit and Part . . . . . Pages 9-14

There will be plenty of jobs in which effect-to-cause reasoning fails to point out the defective part: Isolating procedures then come next, and you can choose between the methods for which general rules are given in this section. The electrode voltage test uses an ordinary d.c. voltmeter to check the d.c. voltage at each tube electrode, and the electrode continuity test uses an ohmmeter to check the continuity between tube electrodes and reference terminals in the power pack.

3. Examples of Part-Isolating Tests for Power Line Receivers . . . . . Pages 14-22

Basic circuits used in standard a.c., transformerless a.c., and universal a.c.-d.c. receivers are used as examples to show how electrode voltage tests and electrode continuity tests are used to isolate the defective part in a stage. Study the examples carefully, being sure you understand *why* each conclusion holds true, so you can interpret your own meter readings in a similar manner. A meter is of no value whatsoever unless you understand the story each meter reading has to tell.

4. Examples of Part-Isolating Tests for Battery and Vibrator Receivers . . . . . Pages 23-28

The large number of battery-operated portables, farm receivers and auto sets make this an important section. Many important rules are brought out which can be applied to all types of receivers. You will learn that the basic testing procedure is practically the same as is used for power line sets, except for variations introduced by the power supply.

5. Answer the Lesson Questions and Mail Your Answers to N.R.I.

6. Start Studying the Next Lesson.

# HOW TO ISOLATE THE DEFECTIVE CIRCUIT AND PART

## Analysis of the Defective Circuit

**T**HE professional servicing technique covered so far is planned to isolate the defect, first to one section (r.f., i.f., a.f., etc.), and then to one stage in that section. We thus narrow the trouble down to the few closely related parts that contribute to the operation of a single stage in a radio receiver.

In this lesson you are going to learn how to isolate, within the stage, the part that is actually defective and preventing normal operation. This is the objective of all radio servicing since as soon as the defective part is identified, it can be replaced and normal operation restored. Successful servicing means nothing more or less than finding that defective part with professional sureness and swiftness.

The methods you will learn in this lesson for finding that defective part are first, going directly to it by what we have called "effect-to-cause" reasoning, and second, running the part down by specific tests that lead from one symptom to another until the fault is isolated. The actual, practical examples of parts isolation tests to be given later in the lesson will enable the serviceman to handle a large majority of the defective receivers that make up the day-to-day work-load on any service bench.

### ISOLATING THE DEFECTIVE PART BY ANALYSIS

One may ask, why try to find the defective part by analysis? The answer

will be attested to by every successful serviceman—analysis is far quicker and in radio servicing, time is all-important. Instruments are always at hand to uncover additional symptoms if analysis fails. But if you learn to go directly to the trouble from the symptoms, it means that a major part of your time and labor will be saved.

Analysis is particularly useful when the trouble has been isolated to a single stage. It is much simpler to visualize clearly what is going on in a single stage than in all the sections of a complex receiver. It should be clear by now



One of the first steps in a service job is looking for surface defects. Tubes are usually checked at this time, as they make up a large percentage of the service complaints and the customer expects this service.

why you should frequently go back and review the lessons on basic theory whenever you get the chance. The more you get the real "feel" of what goes on in the various circuits of a radio receiver, the quicker and more accurate your analysis of defects is going to be.

The experience of all who study radio is that reviewing theory, the "how-it-works" parts of the course, from time to time after you have practical experience, is a sure way to make the theory vivid and real to the mind. Of course, you must get the fundamentals in the very beginning, but going back over them again will deepen your understanding of the "why" of each circuit and part in a radio. The real emotional satisfaction in radio comes from being thoroughly at home among the complex techniques of this fascinating, ever-expanding science. And this knowledge is one of your most important tools as a serviceman!

As a memory refresher, here is a quick check-list of common defects that occur over and over in receivers brought in for servicing:

1. *Tubes* — burned-out filaments; low emission; shorted electrodes; leakage between electrodes; intermittently open electrodes; gas.

2. *Resistors* — open; shorted; changed in ohmic value; intermittent internal connection; poor contacts in adjustable resistors.

3. *Coils* — open; shorted; some of the turns shorted together; poor connections at terminals, lugs or leads; leakage or ground to the core in iron-core units; intermittent or complete open due to electrolysis.\*

4. *Condensers* — open; shorted; leaky; loss in capacity; high-power factor; intermittent open.

5. *Connections* — open; shorted

leads; leakage between leads; intermittent or noisy connections (due to corrosion, thermal action, internal arcing or chassis vibration).

6. *Alignment*—this will be taken up elsewhere in the Course.

You will note three principal types of defects in the above list—shorts, opens, and changes in value. Poor connections and leaks are special cases of opens and shorts. In nearly every case, these defects will change the normal voltage, resistance and current relationships in the stage. That change is the symptom we look for, by means of a voltmeter, ohmmeter, or other instrument. After the discussion of analysis methods, we will learn the standard testing procedures for finding the defective circuit and part in any type of receiver by means of such tests. For our general discussion of "effect-to-cause" reasoning, applied to isolating the defective part within a stage, let us now consider the pentode detector circuit shown in Fig. 1. For purposes of analysis, we can "lift out" the i.f., a.f., and supply circuits as shown in Figs. 2, 3 and 4 respectively. By studying the normal paths, we can see just how a defect in any part will affect the circuit in which it operates. Some parts will affect more than one circuit, so they will be covered under the circuit where their defects are most noticeable. You should refer back to Fig. 1 from time to time, to get an overall picture of the stage operation, as the manufacturer's diagrams do not break the circuit down this way. You will have to do your own analyzing when servicing.

\*Electrolysis is the action which occurs whenever currents cause corrosion by flowing in and out of the surface of a conductor due to resistance developing in a joint or to conductivity in adjoining insulation. The corrosion causes green spots on the copper wire and will eventually break the wire at this point.

## I.F. SIGNAL CIRCUITS

Starting with the i.f. signal circuits of Fig. 2, we have a signal voltage coming in from the preceding i.f. stage. The i.f. plate current of the last i.f. amplifier tube flows through primary  $L_1$  of the i.f. transformer, inducing in the secondary  $L_2$  a corresponding i.f. voltage. This voltage undergoes resonance step-up. Then the voltage across  $C$  is applied across the grid and cathode of the detector tube, reaching the cath-

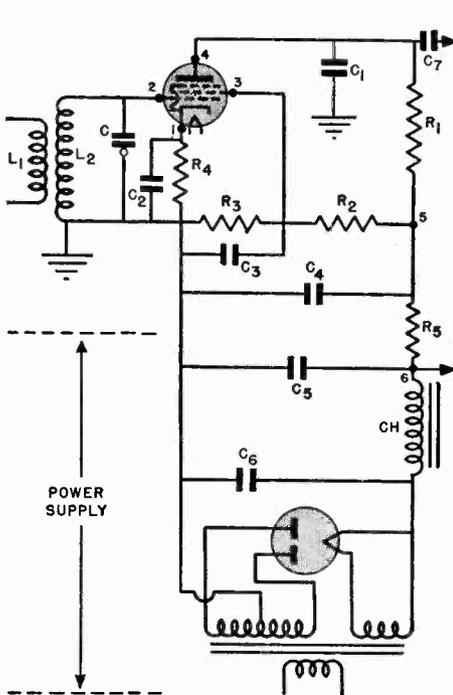


FIG. 1. This typical pentode C-bias detector stage is a good example of combined r.f., a.f. and power circuits.

ode through condenser  $C_2$ , which is a by-pass condenser for i.f. currents. The i.f. signal is carrying with it an a.f. component which represents the voice or music we want to hear. It is the job of the tube to separate the two. It accomplishes this "rectification" because cathode resistor  $R_4$  provides a  $C$  bias voltage which makes the tube operate

at the plate current cut-off point. Negative half cycles of the incoming i.f. signal have no effect on plate current because the grid is already so far negative due to the bias that the tube is in "no operation" condition. Positive half cycles counteract the bias voltage and allow pulses of plate current to flow. These pulses have both the i.f. and a.f. components in such form that the a.f. can be separated and sent on to

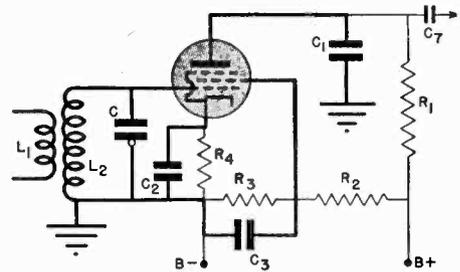


FIG. 2. The r.f. circuits of Fig. 1 are shown here by heavy lines. Refer back to Fig. 1 to see their relationship to other circuits.

the audio amplifier. Condenser  $C_1$  assists this separation, by-passing the i.f. currents to ground, where they travel back through  $C_2$  to the cathode.

Actually, the plate i.f. circuit is a series circuit, consisting of condensers  $C_1$  and  $C_2$ , and the internal tube plate-cathode impedance. Because  $C_1$  and  $C_2$  both have extremely low reactances at radio frequencies, the internal tube impedance is by far the highest, and hence most of the plate i.f. voltage is dropped in the tube.

Screen grid by-pass condenser  $C_3$  keeps the screen at i.f. ground potential so there is practically no r.f. voltage between screen and ground while this condenser is good. The screen then acts as an electrostatic shield between the plate and control grid, thus preventing i.f. voltage on the plate from feeding back to the control grid. Con-

denser  $C_2$  serves as a return path to the cathode for screen grid i.f. current by-passed by  $C_3$ , as well as plate i.f. current by-passed by  $C_1$ .

This analysis indicates that normal flow of i.f. signal current can be interrupted by a defect in any one of these parts, and we can proceed to study what happens to the operation of the stage with a defect in each one. This will enable us to reason back to the defect when we encounter the symptoms that are associated with it. However, certain defects in i.f. signal circuit parts may be easier to find from the effect on the supply or a.f. circuits. Hence, troubles will be covered here under the conditions where they are easiest to find.

**$L_2$  Open.** With a break in the windings of  $L_2$ , the i.f. voltage induced could not send a signal current around the resonant circuit  $L_2$ - $C$ . Hence, there would be no signal voltage developed across the circuit to pass on to the tube and the stage would ordinarily be dead. Whatever technique of isolating the dead stage you used would bring you to the grid circuit of the detector as the point where normal operation ceased. A signal tracer would show a signal across  $L_1$  but none across  $L_2$  or  $C$ ; this would indicate a break in  $L_2$ . A signal across  $L_2$  but none across  $C$  would indicate a broken lead between the two.

**$C_1$  Open.** Condenser  $C_1$  is used to by-pass the i.f. components in the plate circuit of the tube to ground, as explained above. An open condenser is the same as if the condenser was not there, so an open  $C_1$  would permit the i.f. components to enter and probably overload the following a.f. amplifier, causing distortion. Also, since the condenser will normally by-pass a portion of the very high audio frequencies, the tone quality will be more "tinny" than normal. You would have to be familiar

with the normal tone quality of the set to notice this, of course.

The possibility of this condenser being defective probably would be overlooked until the trouble has been localized to this stage. Even then, leakage in coupling condenser  $C_7$  would be suspected first, particularly since the distortion may appear in the first audio stage more definitely than here.

A signal tracer is the quickest way of confirming your analysis of this defect, since if the i.f. signal appeared between the detector plate and ground where it is definitely supposed to be absent, an open in  $C_1$  is indicated. Shunting the suspected condenser with a good one is another quick way of checking your analysis. If the distortion and tinniness clear up when the leads of the test condenser are touched across the old one, the fault is located.

## A.F. SIGNAL CIRCUITS

Let us see what constitutes the a.f. signal path, what parts are in the path, and how defects in those parts affect operation. The audio circuits are shown by heavy lines in Fig. 3. The plate-cathode path inside the tube is the source of a.f. voltage, and the a.f. current flows from plate to cathode through  $R_1$ ,  $C_4$  and  $C_2$ . Very small amounts of a.f. current will go through  $C_1$  to  $C_2$ , also from  $R_1$  through  $R_2$ , and then either  $C_3$  or  $R_3$  to  $C_2$ —or even through the power pack. However, these are not the intended paths for a.f. and more than a very small amount through any of them will interfere with normal operation. Following are the defects that can interrupt this normal flow.

**$C_4$  Open.** A.F. currents normally reach the chassis through  $C_4$ . If  $C_4$  is open, a.f. signals will take the path through  $R_5$  and  $C_5$ . Parts  $R_5$  and  $C_4$  together act as a hum filter to keep



voltage divider which supplies the correct voltage to the screen grid electrode.  $R_2$  alone could do this if of the correct ohmic value, but the addition of  $R_3$  completes a divider circuit which maintains a much more constant voltage under different currents than a single series resistor. Since  $R_3$  is much smaller in value than the screen grid-cathode path of the tube plus  $R_4$ , which are in shunt with  $R_3$ , variations in tube resistance make little difference in the total resistance of the circuit.

With this general picture of the plate and screen grid supply circuits, we can consider a few defects which affect the distribution of voltages in the circuit.

**$R_3$  Open.** Both  $R_3$  and the screen grid-cathode path of the tube in Fig. 4 normally draw current through  $R_2$ . If  $R_3$  opens, only screen grid current will flow through  $R_2$ . As a result, the voltage drop across  $R_2$  will be less and the screen grid will get a higher voltage than before. A high screen grid voltage makes the tube act more like an amplifier than a detector, resulting in weak and possibly distorted reception.

Suppose you were servicing a receiver when the complaint was weak reception. You tested the tubes, analyzed the chassis for surface defects, checked those parts suggested by an over-all effect-to-cause reasoning and failed to isolate the defect. A stage isolating procedure leads you to this detector stage. Several defects here could cause weak reception; an open in  $C_2$ , as previously considered; an open  $R_3$ , and other defects which we can temporarily ignore. You could check both  $C_2$  and  $R_3$ , but analysis may lead you to the defect directly. If distortion is present with the low volume we can forget  $C_2$  and concentrate on  $R_3$ , for an open here may produce distortion while the degeneration caused by an open  $C_2$  actually *improves* the tone quality. ex-

cept where stray capacity results in a predominance of higher audio frequencies. This technique of considering *all* symptoms helps you to go directly to the defect.

If  $R_3$  is suspected, a voltmeter test of the screen grid voltage or an ohmmeter check from screen grid to chassis would locate the defect definitely.

**$R_1$ ,  $R_2$  or  $R_4$  Open.** If any one of these resistors in the circuit of Fig. 4 opens, the receiver will be dead because one of the electrode supply circuits will be open. If you were using a signal tracer to isolate the defective stage, you would find r.f. signal voltage across  $C$  but very little (if any) a.f. voltage in the plate circuit. A complete absence of plate or screen voltage should immediately suggest itself.

If  $R_1$  is open, d.c. voltage measurements would indicate that there is no plate-to-chassis voltage but approximately normal voltage between point 5 and the chassis. With  $R_2$  open, there would be no screen grid voltage.

The voltmeter range used in checking from cathode to chassis for an open  $R_4$  will determine the actual reading. Should  $R_4$  be open, then the meter resistance will replace that of  $R_4$ . If a high-voltage range is used first (as it should for meter safety), then it is possible for the meter resistance to be high compared to the sum of  $R_1$  and the tube resistance which are in series with it. Hence, the voltage may be much higher than normal. However, if the meter has low sensitivity, or a low-voltage range is used the voltage may be about normal. The clue here will be the fact that the receiver comes to life as evidenced by noises or signals when the meter replaces the resistor.

**$C_1$ ,  $C_2$ ,  $C_3$  or  $C_4$  Shorted.** A short in  $C_1$ ,  $C_3$ , or  $C_4$  will remove d.c. voltage and kill the stage just the same as an open in  $R_1$ ,  $R_2$ , or  $R_4$ . Notice that

these condensers would furnish unwanted d.c. paths if shorted. Effect-to-cause reasoning cannot tell us whether the trouble is a shorted condenser, an open resistor, or both. A visual inspection might show a charred and blistered resistor, and this would be a strong indication that the condenser at the low-voltage end of the resistor was shorted, allowing excess current to flow through the resistor. Ordinarily, however, with a dead stage, there will be no visible indication, so the serviceman would make a few simple tests with a voltmeter to find whether the d.c. voltages were missing, and if so, which resistor or condenser was the cause.

A glance at the diagram shows that a break-down in  $C_1$  removes plate volt-

age by grounding the plate, while a short in  $C_3$  will remove voltage from the screen. A shorted  $C_4$  will remove voltage from both plate and screen. Hence, a measurement for these two voltages will determine which condenser or resistor should be suspected.

Should  $C_2$  become shorted, no voltage drop will occur across  $R_4$  and there will be no  $C$  bias. The removal of bias by a shorted  $C_2$  would not kill signal passage through the stage, but detection would be imperfect and the reproduction would be weak and distorted. Effect-to-cause reasoning would lead you to check all the operating voltages in the stage, if weak volume and distortion were the complaint. If normal  $C$  bias voltage did not show up, you would check for continuity from grid to ground, and check the resistance from cathode to ground. A short in  $C_2$  would show up on this test.

## CONTROL CIRCUITS

**A.V.C. Circuit.** Let us consider another kind of circuit, the a.v.c. circuit, to see what kind of help in running down defects we can get from a study of a circuit action.

In Fig. 5, the i.f. signal voltage developed across final i.f. trimmer  $C_4$  is applied between the plate and cathode of the diode  $VT_2$ , reaching the cathode through  $C_1$ . Rectified current flows from the cathode to the plate of the diode, then through  $L_4$ ,  $R_1$  and  $R_2$  back to the cathode. Combination  $C_1$ - $R_1$ - $C_2$  acts as a high-frequency filter which allows only a.f. signals and direct current to flow through volume control  $R_2$ . The a.f. signal is then forwarded to the input of the first a.f. amplifier stage through d.c. blocking condenser  $C_3$ .

The voltage across  $R_2$  will have both d.c. and a.f. components, hence both these voltages will be applied to any

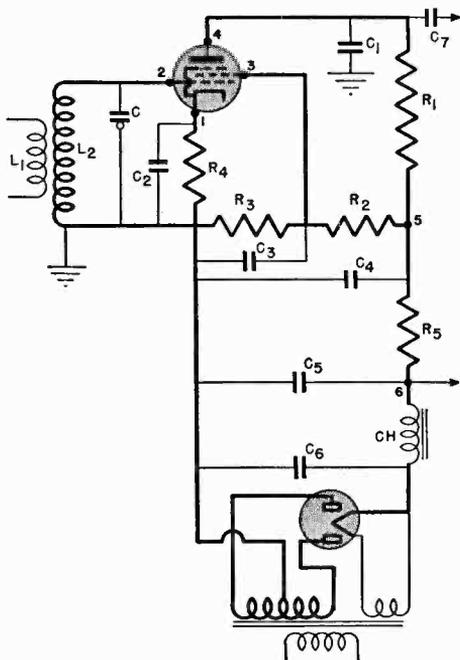


FIG. 4. The power circuits are d.c. paths for supply voltages. An open or short circuit will affect these paths and thus alter one or more of the operating voltages. This circuit is also taken from Fig. 1.

circuits which are connected between point 1 on  $R_2$  and ground.

One a.v.c. circuit which is connected between these points traces from point 1 through  $R_3$ ,  $R_4$  and  $L_2$  to the grid of i.f. amplifier tube  $VT_1$  and from this tube cathode through  $R_5$  to ground and thus back to  $R_2$ . Therefore, the d.c. voltage across  $R_2$  is applied between the grid and cathode of this tube. Condenser  $C_5$  and resistor  $R_3$  act as an a.f. filter, so only d.c. reaches the i.f. tube grid. Any a.f. voltage which is developed across  $C_5$  is further reduced by a similar a.v.c. filter made up of  $R_4$  and  $C_6$ .

In addition, we have a delaying action in these filters whenever any sudden changes in d.c. voltage occur. All this is basic a.v.c. action which you studied earlier in your Course.

Now let us see how certain actions of a receiver suggest an a.v.c. defect, and also see how the effects suggest what to suspect in the a.v.c. circuit.

**$C_5$ ,  $C_6$  or  $C_7$  Shorted.** The symptoms which indicate the *absence* of a.v.c. are a great difference in volume when tuning from local to distant stations, with overloading on strong signals. Should condenser  $C_6$  or  $C_7$  short, the a.v.c. voltage would be removed from one of the i.f. amplifier tubes and would be reduced on the other, causing similar symptoms. Such marked unevenness in volume between

stations should lead you immediately to check the a.v.c. filter condensers for shorts.

**$C_6$  Open.** If condenser  $C_6$  opens, weak reception will be the symptom because the high resistance of  $R_4$  has, in effect, been added to the path from the tuned circuit to the cathode. A good part of the i.f. signal voltage will be lost across  $R_4$  instead of being applied to the tube. The resistance will also isolate the grid-cathode capacity from across  $C_3$  which will detune the resonant circuit  $L_2$ - $C_3$ , further reducing sensitivity.

This is not a common trouble. However, if the low sensitivity has been isolated to an a.v.c.-controlled stage, consider cathode and a.v.c. condensers after eliminating tubes and alignment. Shunting a good condenser across the suspected one will show whether the original is open.

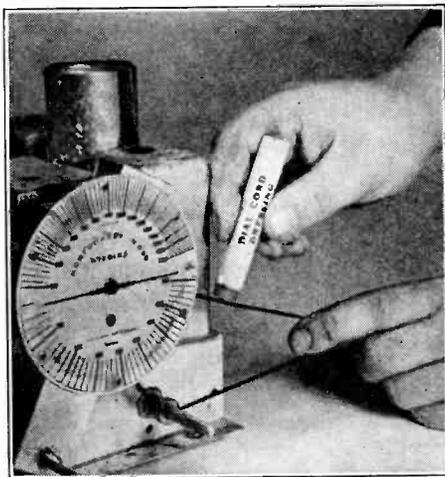
**$C_5$  Open.** If  $C_5$  is open, the sensitivity of the receiver will not be affected but the time constant of the a.v.c. system will be greatly shortened, that is, the a.v.c. voltage will respond to very quick changes in signal volume such as bursts of static or noise. Such sudden noises may make the receiver go dead for an instant or two. These effects might not be noticed either by the customer or the serviceman, so failure of this part might not be suspected.



circuit, but only a very weak audio signal comes out in the plate circuit.

This could be caused by low d.c. plate voltage, but other troubles must be considered. An open in condenser  $C_3$  or in condenser  $C_2$  could cause the same trouble. Checks across them with a signal tracer would isolate the defective one, as you would find a signal voltage across the defective part.

Signal tracing requires that you have some idea of how strong the signal



*Courtesy General Cement Mfg. Co.*

Sometimes a mechanical trouble, like a slipping dial cord occurs. This condition may cause the dial pointer to indicate incorrectly or may even prevent tuning, depending on the mechanism. Rosin can be used, or a handy stick-type dial dressing can be used as shown here. The material rubs off on the cord, forming a surface having more friction so the cord does not slip as readily.

should be at each point in the receiver, under normal conditions. Naturally, if you follow the signal from the input through the various amplifier stages, there should be a gain of voltage in each stage. However, converter tubes such as the first or second detector may not show an actual increase in voltage over the preceding stage. Many set manufacturers now furnish stage gain data, and experience will of course also

help you to know when signal voltage readings are normal.

Since defects in power supply circuits usually result in removing the d.c. voltages on the tube electrodes, the stage is likely to be dead, and signal tracing will not be found quite as useful in the case of a dead stage as with weak or distorted reception. Where electrode d.c. voltages are involved, the d.c. voltmeter and ohmmeter are most useful in running down the defective part in the stage. The general methods of using these instruments for maximum results will now be described.

### ELECTRODE VOLTAGE TESTS

Voltmeter tests in a defective stage naturally begin with a check of plate voltage, followed by measurements of the other electrode voltages in turn. The receiver is on but not usually tuned to a station. Abnormal voltage or lack of voltage will lead you to suspect some part or series of parts as being defective, and further tests will then isolate that part. The parts to be tested will of course be those in the supply circuit where voltages are missing or abnormal.

As an example, suppose we find no plate voltage on the tube of Fig. 4. Assuming we have isolated the trouble to this stage, we know that the power supply is working as other stages are operating, so the defect must be somewhere in the plate circuit of this stage only. Inspection of the diagram shows that the plate circuit consists of  $R_5$ ,  $R_1$ ,  $C_4$ , and  $C_1$ , and that defects in these parts could interrupt the plate voltage.

We could begin to run the part down with the voltmeter by checking the voltage at point 5. If voltage existed here, the most likely candidates are an open in  $R_1$  or a short in  $C_1$ . A short in

$C_1$  will make  $R_1$  take the full voltage from point 5 to chassis. The meter will show whether or not the voltage across  $R_1$  is the same as from point 5 to chassis. If it is, count on a shorted  $C_1$ .

If the voltage across  $R_1$  is less than from point 5 to the chassis, however, and the receiver shows signs of coming to life when the voltmeter probes are across the resistor, it means that  $R_1$  is open. You can easily see that in making the measurement you substituted the meter for the resistor  $R_1$ , so that plate voltage reached the tube through the meter resistance.

Making voltage measurements has the advantage of checking the receiver while all parts are under normal voltage stress. This is valuable because many troubles, particularly shorts in condensers, may appear only when full voltage is applied.

The serviceman has a wide choice of meters for use in voltage tests, but the meter should have a sensitivity of at least 1000 ohms-per-volt, and should have several ranges. The d.c. vacuum tube voltmeter can be used to measure all voltages with respect to chassis and does not usually load the circuit as does the ordinary voltmeter. However, where both ends of the part are above chassis potential, the vacuum tube voltmeter must usually be applied by measuring the voltage to chassis at each end of the part, and then taking the difference between the two. For example, the voltage between point 5 of Fig. 4 and chassis may be 200 volts, while that between the plate and chassis may be 75 volts. The difference, 200 — 75, or 125 volts, is the drop across  $R_1$ .

Other precautions well worth observing are to always start on the highest range of a multi-range meter in testing an unknown voltage, and to disconnect the meter from the voltage

under test when changing ranges. Failure to follow these rules is likely to result in a burned-out meter or in arcing at the contacts of the range-changing switch.

The question of what should be considered normal readings at the tube electrodes will depend on the type of meter being used. The values shown in service manuals are generally based on measurements with a 1000 ohms-per-volt meter. If you use a more sensitive meter or a vacuum tube voltmeter, you would expect to get higher values in any high-resistance circuits. Remember, too, that voltages in radio circuits are in most cases dependent on resistor values, and that these can vary as much as 20% from the stated value.

If these limitations are kept in mind, and as you gain experience in making voltmeter readings, you will find that the d.c. voltmeter can be a fundamental part of your servicing technique. Later in this lesson, you will learn exactly how the voltmeter is applied to uncover the defects you will actually encounter in servicing each of the main types of receivers now on the market.

## ELECTRODE CONTINUITY TESTS

After signal tracing and electrode voltage tests, continuity tests with an ohmmeter must be added to make your general servicing technique complete. Ohmmeter measurements must always be made with the receiver turned off and preferably disconnected entirely from any source of power. If you fail to turn off the set, voltages across the parts being tested will not only give erroneous readings but will likely damage the meter. And after turning the receiver off, wait for a minute or so for the filter condensers to discharge or else short-circuit them with a screwdriver or test lead to be sure they are

not storing voltage.

The real value of continuity tests in a servicing procedure is that by testing across the two ends of a given circuit, you get a quick check of every part included in that circuit. This applies, of course, to every circuit having d.c. continuity. By testing for continuity from each positive electrode of the tube to the highest positive point in the power pack, and from each negative electrode to the highest negative point in the power pack, you can quickly discover whether any of the d.c. supply lines is out of order. Checking all of these supply paths will cover most of the parts of a stage.

The location of positive and negative reference points will vary in different types of receivers, and will be covered in detail in the practical examples given later on in the lesson.

Of course, the symptoms of the receiver may give you a clue as to which of the paths to check first. For example, suppose we find the symptoms of no a.v.c. action in the circuit shown in Fig. 5. This could be caused by a shorted a.v.c. filter condenser, and an ohmmeter check from the grid of the i.f. tube to ground will quickly reveal this condition. This path should normally show a d.c. resistance equal to the values of  $R_2$ ,  $R_3$ , and  $R_4$ , and a shorted condenser will read very much less than this, in the case of  $C_6$ . A short in  $C_5$  would give a reading equal to  $R_4$ . A short in  $C_7$ , however, would not stand out clearly because  $R_4$  plus  $R_6$  is comparable in value to  $R_4$  plus  $R_3$  plus  $R_2$ . The measurement for  $C_7$  would have to be made from the grid of the first i.f. tube to chassis.

Ohmmeter tests will not reveal open condensers. Suspicion will fall on such condensers from the symptoms they set up in receiver operation. Many of the symptoms of open condensers have al-

ready been described, and other cases will be taken up later in the lesson. In each case the quickest and most practical test, unless signal-tracing equipment is at hand, is shunt a suspected condenser with a good one.

Many times you will find it most efficient to combine ohmmeter and voltmeter tests. For instance, take the



If the preliminary surface defect check and effect-to-cause reasoning do not lead right to the trouble, some means of section and stage isolation is used. If available, a signal tracer can be used as shown here. This instrument is particularly helpful in some of the more difficult jobs, such as weak or intermittent reception complaints.

case in the section on voltage tests where no voltage was found on the plate of the tube in Fig. 4. Two parts are suspected,  $C_1$  and  $R_1$ , and an ohmmeter test of each of these parts will show whether  $C_1$  is shorted or  $R_1$  is open.

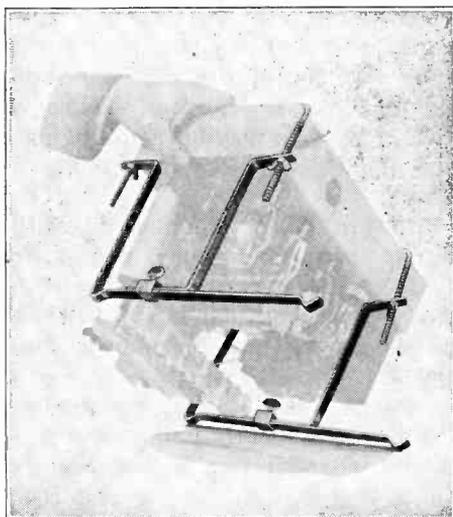
Some servicemen eliminate the voltmeter altogether and depend on the ohmmeter alone, using the d.c. supply

continuities from the tube to the power pack to cover opens, and checking across from positive to negative to find shorts. These tests will be covered in detail later on, but you can easily see that the serviceman has somewhat of a choice in his use of instruments. The voltmeter may be used to find trouble and the ohmmeter to check parts, or instead of using both instruments, either can be used alone. As you become highly experienced, you may find that one or the other works quicker for you. But in the meantime, it is necessary to understand thoroughly the use of both, since the serviceman who can use either instrument as the need arises will never be at a loss for means of finding the defective part in a receiver.

**Practical Pointers.** D.C. resistance values of coils are usually given on circuit diagrams, to help in continuity measurements. A few shorted turns in a coil have little effect on the total resistance of the coil, even though the short may cause the receiver to operate improperly. In power transformer windings, a few shorted turns may cause considerable heating and eventually ruin the transformer. In r.f. tuning coils, a few shorted turns can greatly reduce the inductance and the  $Q$  factor.

The fact that a measured value of coil resistance differs from that given on the diagram does not necessarily mean that the coil is defective. Normal variations in coil resistance of normal inaccuracies in the ohmmeter itself may result in different readings. Only when the ohmmeter reading is considerably different from the specified value can it be considered as pointing to a defective coil.

When measuring center-tapped coils of iron-core a.f. or power transformers, do not expect the resistance to be the same for both halves of the winding.



*Courtesy General Cement Mfg. Co.*

Radio receivers must be turned on edge or upside down to be worked on. They may be so made that they do not stand on edge readily, and the presence of an elaborate dial mechanism and the tubes may prevent turning the chassis upside down. Servicemen then must block up the chassis on old parts or use a manufactured support. The brackets shown here are quickly attachable and serve to protect the delicate parts on top the set as well as hold the set in an easy-to-work-on position.

An electrical center tap means equal turns on each half, but in the usual layer-wound construction of these windings the outer half of the winding will have a larger diameter and hence, more wire and more resistance.

When we test condensers with an ohmmeter, we must remember that it will show continuity only if the resistance being measured is within the range of the particular ohmmeter being used. A good paper or mica condenser should have a leakage resistance above 50 megohms-per-microfarad. Hence, practically any ordinary paper condenser found in a radio should have a leakage resistance above this value which is far beyond reach of the average ohmmeter. Therefore, when a steady reading is obtained

across a paper or mica condenser, we can suspect it of being leaky. However, it is advisable to disconnect the condenser to make a check of it alone, as there may be a possible shunting path through which the reading is being obtained.

Be sure to keep your hands off the ohmmeter probes when testing for leakage, because if you touch the probes with both hands you will read your body resistance in shunt with the part.

When a condenser larger than about .01 mfd. is tested, the pointer will jump when the connection is made, returning slowly to a high-resistance reading. Be sure to wait for the pointer to stop moving. This will take several seconds with large condensers, or with any circuit in which high-capacity condensers are connected.

Electrolytic filter condensers have much lower leakage resistance than other types. Two values will be found, depending on the polarity of the test

probes. The higher value is the correct one.

Whenever there is the least possibility of doubt regarding your interpretation of ohmmeter tests for condensers, always disconnect one condenser lead and try a new condenser of approximately the same capacity in the position of the old one. The working voltage of the test condenser must be as high or higher than the old one. This will give you a positive test and will often save considerable time in trying to interpret confusing ohmmeter readings.

If the receiver has just been turned off, wait for the tubes to cool before making ohmmeter tests. Otherwise, you may get readings through the tube itself if the positive ohmmeter probe is connected to some element while the negative probe is connected to the cathode. Emission continues until the cathode cools, so the ohmmeter battery causes a current flow through the tube.

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## Examples of Parts Isolating Tests For Power Line Receivers

Now that we understand the basic procedures for using the voltmeter, ohmmeter, and signal tracer, we can learn the actual tests that form a major part of the day-in, day-out work on any efficient service bench. The examples to be given will cover all of the usually found defects that occur in each of the standard types of power-line receivers:

1. A.C. receivers using power transformers (also vibrator-rectifier systems).

2. A.C. receivers without transformers.

3. Universal a.c.-d.c. receivers.

We repeat, the tests to be described, if thoroughly studied and understood, will give you the ability to solve smoothly and swiftly the major part of the receiver troubles that you will meet in professional servicing. Remember that we have narrowed the trouble to one stage, and that our problem in each case is to find the defective circuit and part in that stage

that is actually the seat of the trouble. The replacement of the defective part is the end toward which all testing methods should be directed.

Further, remember the general rules we will now develop by examples can be applied to all types of receivers—even those using other power supplies as long as the circuit is basically similar.

### A.C. RECEIVER WITH POWER TRANSFORMER

The circuit diagram in Fig. 6 shows the first a.f. amplifier stage and the power pack of a typical a.c. receiver employing a power transformer. The other stages in this receiver have been omitted from the diagram for the essential facts regarding the isolation of the defective part can be more easily realized if we do not complicate our circuit.

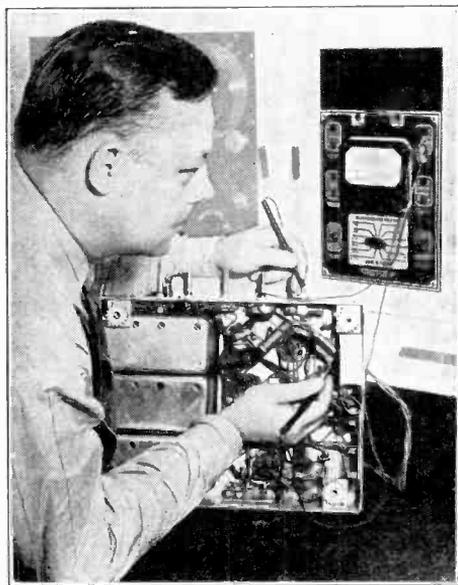
**Electrode Continuity Tests.** In all a.c. receivers having power transformers and all receivers using both a vibrator and a rectifier tube, the following basic continuity rules apply:

1. All positive electrodes, such as plates and screen grids, will have a conductive path to the *cathode of the rectifier tube*, which is the most positive reference terminal.

2. All negative electrodes, such as control grids, suppressor grids and cathodes, will have a conductive path to the *center tap of the high-voltage secondary*, which is the most negative point in the receiver. Since the plates of the rectifier tube connect to this center tap through the high-voltage secondary, and since these plates are easier to locate than this transformer terminal, it is common practice to use one of the plates of the rectifier tube as a reference point when checking the continuity of the negative electrodes. Furthermore, if the trouble has been

localized to a stage and we can be sure that nothing is the matter with the power pack through previous tests, the set chassis can be used as the negative reference point.

To make continuity tests in the first a.f. amplifier stage shown in Fig. 6, you could place one ohmmeter probe on a filament terminal of the type 80 rectifier tube (the filament here serves also as the cathode), and place the other probe on the plate of the type 75 tube.



The final step is one of localizing the defective circuit and part. The service multimeter is now used. Voltage and resistance readings, properly interpreted, will lead right to the trouble whenever the defect is of such a nature that any supply circuit is affected.

The ohmmeter should read the total resistance of this circuit which is the sum of 75,000 ohms ( $R_{22}$ ) plus 50,000 ohms ( $R_{23}$ ) plus field coil  $CH$ , another 2000 ohms, or a total of about 127,000 ohms. Any value from about 100,000 ohms to 150,000 ohms should be considered correct.

If the value is radically different, there is a defect of some kind in the

circuit. For instance, if there is no reading at all, (infinitely high resistance) there must be an open in the circuit. To localize this open, you would leave one ohmmeter probe on the rectifier filament, and move the other one toward it along the circuit, from point 1 to point 3, to point 4 and point 5. When the meter first shows continuity, you know that you have passed through the break. The defective part must then be the one next to the test probe. If the meter reads at point 3 but not at 1, for example, the open must be in  $R_{22}$  or its connecting leads.

If the continuity from plate to power pack checks correctly, you can proceed with the grid and cathode continuity tests. These are the negative electrodes and the check should be made to the highest negative point in the power pack. Let us use either plate of the rectifier tube. From the grid, this circuit traces through  $R_{21}$  to the chassis to  $R_{14}$  to the secondary of the power transformer to the rectifier plate and should read about 1 megohm. An open in the circuit can be found by moving the test probe at the grid back along the circuit toward the other probe until continuity is discovered.

The third electrode continuity, from the cathode of the 75 tube to the plate of the rectifier tube, includes  $R_{20}$ ,  $R_{14}$  and the transformer secondary. This should obviously read 2300 ohms plus the transformer resistance, which may have any value between 100 and 600 ohms or so. Any radical difference from this value indicates a defect which can be localized by again moving one of the probes along the circuit toward the other until normal conditions are found.

These three tests, from plate, cathode, and grid, to the reference points in the power pack, will quickly uncover abnormal d.c. resistance in the

three d.c. supply circuits to the tube. As you can see, this includes  $R_{14}$ ,  $R_{21}$ ,  $R_{20}$ ,  $R_{22}$ ,  $R_{23}$ ,  $CH$ , and the transformer secondary for opens. Also, should the reading between the 75 tube cathode and the rectifier plate be lower than normal,  $C_{25}$  or  $C_{30}$  may be shorted. The tests have real significance as to the operating condition of the stage because they show the condition of the supply circuit right up to the tube electrodes.

Other shorts may be somewhat harder to find with continuity tests. A study of the diagram is necessary to visualize all the possible shunt paths which may apply to any test made for a short circuit. Whenever in doubt, unsolder a connection in a shunt path temporarily. This labor saves time in the end by making the circuit under test stand out alone for a conclusive test.

For example, suppose we are trying to find a short to chassis in the plate circuit of Fig. 6. A test from point 1 to point 2 (the rectifier filament) would read correctly whether the short to chassis exists or not. The way to find it is by a test from point 1 to chassis. A very low reading indicates a short in condenser  $C_{27}$ . A reading of 75,000 ohms points to  $C_{41}$ . Higher readings may be the result of  $C_{41}$  having appreciable leakage resistance, or may be due to the path through  $R_{22}$ ,  $R_{23}$  and the electrolytic condensers. Paper condensers used across plate and screen grid supply voltages usually break down completely rather than just become leaky, so we would normally expect  $C_{41}$  to be completely shorted. Remember, however, that many receivers use electrolytic condensers in this position and that there are always exceptional cases.

If the leakage or short is not between point 1 and the chassis, the ohmmeter



from the socket, with the power on, and measure the drop across  $R_{22}$  and  $R_{23}$ . Naturally, unless the condensers are leaky, there is no path for current to be drawn through the resistors, and there should be no voltage drop across them, with the tube out of the socket. A drop through both resistors points to leakage in  $C_{27}$  or  $C_{32}$ , while a voltage drop across  $R_{23}$  alone points to  $C_{41}$ .

A d.c. vacuum tube voltmeter can be used for this test very effectively by making the measurements to chassis. With the tube out and no leakage, points 1, 3, and 4 should all have the same potential to chassis. If it drops at 3 but no further drop at 1, then  $C_{41}$  is leaky; if it drops from 4 to 3 to 1, then leakage in  $C_{27}$  or  $C_{32}$  is indicated. Only the vacuum tube voltmeter can be used to read the voltages to chassis on this test, since any other meter draws current through the resistors so that drops will occur anyway, making it hard to tell if leakage exists.

The second important electrode voltage in this stage, the grid bias voltage, can be tested with a voltmeter if the resistance of the meter is carefully accounted for. The bias voltage is developed between cathode and ground by the passage of plate current through resistor  $R_{20}$ . If you put a 1000 ohms-per-volt meter, using the 3-volt range; across cathode and ground, you are actually connecting 3000 ohms in shunt with  $R_{20}$ . This reduces the bias resistance to 1200 ohms, and the voltage will go down accordingly. Plate current will increase, but not enough to bring the bias up to normal. If you remember that the voltage you get in making this measurement is lower than the voltage when the meter is removed, the measurement is a valuable indication that some bias voltage exists. A high-sensitivity meter or a vacuum tube voltmeter will of course read close

to the actual bias voltage.

The bias voltage should naturally exist between grid and cathode, but direct measurement across these elements with a low-resistance meter is even more misleading than across the bias resistor. The meter would join with  $R_{21}$  as a voltage divider across  $R_{20}$ , and almost the entire drop would be across  $R_{21}$ , which is several hundred times the value of the meter resistance. This measurement can be made only with a very high-resistance meter.

If you have found approximately normal bias voltage across  $R_{20}$ , you can make sure that it reaches the grid by removing the tube (to prevent false readings through it) and measuring the voltage from plate to grid. If you get a reading here, it proves that the grid circuit has continuity to chassis as the voltmeter is actually measuring the plate-chassis voltage through the grid resistance. Should the circuit be open, there would be no complete path for the meter current so it would not read. The value of the voltage on this reading is unimportant and highly variable with meter sensitivity and the plate and grid resistors. The test is for the existence of any voltage at all, to prove that grid and chassis are connected.

A last important voltage measurement in the stage of Fig. 6 would be across resistor  $R_{21}$ . Normally, there is no d.c. voltage drop across this resistor because in a receiver voltage amplifier, under normal conditions, there is no d.c. current flow through the resistor. A voltage drop across this resistor, indicating current flow, means that  $C_{26}$  is leaky or the tube is gassy. To eliminate the tube from the test, remove the top cap from the grid. If this stops the current flow, the tube is guilty, but if not, it is  $C_{26}$ .

The foregoing examples show clearly

that it is a combination of logical reasoning with simple measurements that forms the real backbone of professional servicing. If you study your theory to get the real "feel" of circuit action and learn the simple, straightforward testing procedures described in these lessons, these two will go to work together for you and you will honestly go places in radio.

### A.C. RECEIVER WITHOUT POWER TRANSFORMER

Another of the major types of receiver you will encounter as a serviceman uses the voltage-doubler circuit of Fig. 7, which includes one i.f. stage with the power pack. The electrode continuity and electrode voltage tests that serve to run down a defective part in this stage will now be described.

**Electrode Continuity Tests.** Considering the power pack first, note that  $C_{20}$ -CH- $C_{21}$  make up the power pack filter system. Point 3, the cathode of rectifier tube B, is the highest positive terminal. Point 2, which is conductive to the plate of rectifier tube A, is the most negative terminal. We thus return to the same basic reference terminals in the power pack as in an a.c. receiver having a power transformer.

To locate the correct cathode terminal for use as the highest positive reference point, trace visually a lead from the choke coil to the rectifier cathode. (This lead may go to an electrolytic condenser terminal first.) Presuming the power supply is normal, as proved by the fact that a part of the receiver is functioning, we do not have to go to this cathode. We can use the electrolytic condenser positive terminal 3 or 4 if more convenient.

The negative reference point is B — and there are a number of points connecting to it which are easy to locate by visual inspection. Among them are

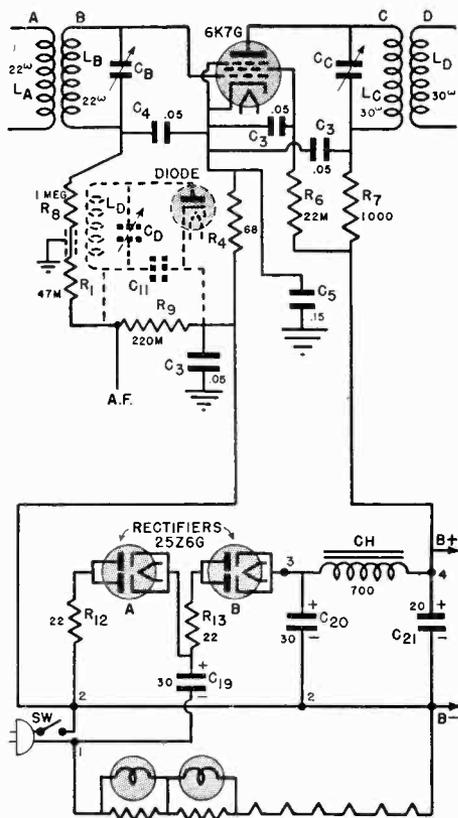


FIG. 7. Power supply and i.f. stage used in the Zenith 5719 receiver. The diode detector circuit has been redrawn in dotted lines to show its relationship to the grid return circuit of this i.f. stage. Coil  $L_D$  feeds this detector. Notice the manufacturer identifies parts having the same electrical size with the same code number. Thus, there are three condensers marked  $C_3$ , all having .05 mfd. capacity.

the set side of the ON-OFF switch SW and the negative leads of  $C_{20}$  or  $C_{21}$ . The switch terminals must be identified unless you turn on the switch, in which case either terminal can be used. The cathode of the diode detector tube (shown by dotted lines) connects to B —. Also, the 6K7G cathode can be used, if you are sure that the 68-ohm resistor  $R_4$  is in good condition.

Remember, the set *MUST* be disconnected from the power line when

making continuity tests, to avoid shocks and possible damage to your ohmmeter. There is no necessity of using the plate of rectifier tube *A* for this reference point.

Continuity tests for screen grid and plate supply circuits are made by using the positive reference point (3 or 4) in the power pack as the common reference terminal for one probe of the ohmmeter and connecting the other ohmmeter probe to the plate and screen grid terminals in turn.

Similarly, point 2 or some part connected to it directly is used as a reference point for continuity tests of the cathode and control grid supply circuits. Notice that the *chassis is not a part of this circuit*. Don't use the chassis as a reference point in a transformerless power pack set (or an a.c.-d.c. set) unless the diagram shows that it is part of the circuit.

Careful analysis of continuity test readings is required to locate shorts in this stage. For example, in the plate supply circuit we have  $L_C$ ,  $R_7$  and  $CH$  in series. A direct short across either  $CH$  or  $R_7$  will be indicated as lower resistance than normal for this circuit (considerably less than 1700 ohms), but a short in  $L_C$  would not be revealed by the initial continuity test because the resistance of this coil is quite low in comparison to the rest of the circuits. In other words, you cannot clear coil  $L_C$  of suspicion of being shorted by making a resistance measurement between the plate and terminal 3; you must measure directly across the coil, on a low ohmmeter range, to make 30 ohms stand out. Should you locate a short across  $L_C$ , you would be required to unsolder one lead of  $C_C$  to prove that the trimmer condenser is not defective. Also remember that you must allow some tolerance from the values shown in the manual, so that the re-

sistance change of a few shorted turns could not be detected by an ohmmeter test.

Shorted turns would change the inductance of the coil greatly, however, and make it difficult or impossible to "peak" the transformer during alignment as you will learn later. Observation of the action of the trimmer condenser or the use of signal-tracing equipment will localize this defect.

Shorts in screen grid and plate bypass condensers are revealed by ohmmeter tests. However, notice that the short circuit will be to the 6K7G cathode, *not* to the chassis. You could use either the cathode of the i.f. amplifier tube or point 2 as a reference point and move the other ohmmeter probe to the 6K7G plate to check for a short in the plate by-pass  $C_3$ , then to the screen grid for a short in the screen grid by-pass. (This is also marked  $C_3$  on the diagram. Many manufacturers number capacities having the same value with the same number on diagrams. You must study the circuit to see which is meant.)

Sometimes a serviceman will check for continuity and will find an open, such as an open  $R_7$ . Replacing this resistor, he finds the set still does not function and the replacement smokes and overheats. Plate bypass  $C_3$  is probably shorted, which caused the original burnout of  $R_7$ . It is well to check for such possible shorts whenever an open circuit is found, before replacing the open part. Combination troubles like this are quite common.

**Voltage Measurements.** Electrode voltage measurements on this receiver are made in the same way as on an a.c. receiver with power transformer. The only difference is that the measurements are not made from positive points to chassis, *because the chassis in this set is not in the d.c. supply cir-*



through the high resistance  $R_{11}$ , so the chassis cannot be used for voltage measurements.  $R_{11}$  is provided to discharge  $C_{26}$  when the set is turned off, to reduce the danger of shock.

The 30 volts shown on the diagram as the plate voltage of the triode in Fig. 9 is based on measurement with a 1000 ohms-per-volt meter. It is good to remember that the actual operating voltage is higher and in this case, the difference between measured and actual voltages will be considerable because of the very high resistance of  $R_7$ . If you use a 1000 ohms-per-volt meter, you should of course get a reading not far from 30 volts, but a high ohms-per-volt meter in this plate circuit will give a higher reading.

Grid convection current flowing through grid resistor  $R_6$  of the triode section will produce a self-bias voltage of about 1 volt for the grid. If you measure the voltage between grid and cathode with a 1000 ohms-per-volt voltmeter during electrode voltage tests, however, the reading will be practically zero, since the meter shunts resistor  $R_6$  down to a low value and thus makes the voltage drop across it minutely small. A vacuum tube voltmeter would measure this voltage across  $R_6$  or the continuity of the circuit can be checked with a high-range ohmmeter.

If  $R_6$  opens, you would probably expect a dead receiver on the theory that the grid would then acquire a high enough negative charge to block plate current in the triode section of the tube. Actually, however, it is likely that there would be enough leakage between the grid and cathode terminals of the tube socket (due to dust and moisture) to provide almost normal grid bias.

Hence, instead of a dead set, a more

likely possibility is that with high leakage resistance here, the stage might block at regular intervals with accompanying distortion and hum. An ohm-meter test of  $R_6$  would be the proper procedure here if an ohmmeter with a 15-megohm range is at hand; otherwise, try shunting  $R_6$  with a good resistor having a value around 15 megohms.

When testing the grid circuit for possible defects, condenser  $C_{16}$  should concern us. If  $C_{16}$  becomes leaky, a portion of the d.c. voltage developed across  $R_5$  would be applied across  $R_6$ , producing distortion which becomes more noticeable as volume is increased. As resistor  $R_6$  has a high ohmic value, even a small amount of leakage in  $C_{16}$  will develop an appreciable voltage across  $R_6$ . To check for leakage in  $C_{16}$ , you can unsolder one of its leads and measure its resistance with a high-range ohmmeter, or try another condenser.

Here is a case where a voltage measurement across  $R_6$  would indicate the leakage through  $C_{16}$ . Disconnect the lead to the grid of the tube to eliminate the self-bias normally developed in this resistor. Tune in a station and set volume control at maximum so the maximum a.v.c. voltage across  $R_5$  is applied to  $C_{16}$ , then check for a voltage across  $R_6$ . A high-sensitivity d.c. meter must be used to check this voltage across  $R_6$ .

This is a case where a leaky coupling condenser produces a negative voltage (with respect to B—) across the grid resistor, due to the polarity of the a.v.c. voltage which is acting as the d.c. source. In cases where the coupling condenser connects from the preceding tube plate, the grid end of the following resistor will be made positive.

# Examples of Part-Isolating Tests for Battery and Vibrator Receivers

The basic procedures for these receivers are very similar to those you have just finished studying. Therefore, we can start right in on battery-powered receivers.

## BATTERY RECEIVERS

There are two kinds of battery receivers, those operating entirely from batteries and the combination a.c.-d.c.-battery types. The combination types must be studied both as battery receivers and as a.c.-d.c. receivers. The main difference between combination types and universals is that in the combination types the filaments must get d.c. power, usually from the power supply or the cathode circuit of the output tube, and hence cannot be connected directly to the power line. Here we will limit ourselves to the type operating from batteries only.

What are the differences in the usual battery receiver from those we have already studied? Remember that the methods of using continuity tests and voltage tests which we have learned in this lesson are basic to all types of receivers. Once the fundamental tests are learned, it is only necessary to consider the differences in the different types of receivers in order to figure out the applications of the tests to any type of receiver encountered.

First of all, in battery receivers the filaments are used directly as cathodes. The filaments may be in series or in parallel. In cases where the filaments are in parallel, the chassis will usually be part of the d.c. circuit so it can be used as the negative reference point.

A quick indication of whether the filaments are series or parallel can be had from the relation of the "A" voltage to the rating of the tubes. Natural-

ly, the tube filament voltage ratings must add up to the "A" voltage for series connection, but the "A" voltage is approximately equal to that of a single tube for the parallel connection.

A caution—it is usually not wise to pull out tubes in battery sets with the voltage on. In parallel-connected sets, there may be a series resistor to drop the "A" voltage to the proper value and if a tube is pulled out, the voltage on the remaining tubes may rise enough to damage them.

The leads connecting to the *B* battery are very convenient continuity reference points in battery sets, and continuity tests will be described below. Voltage measurements can be made between the *B*—battery lead and any positive point in the set or from chassis if it is in the circuit, and are exactly like those already described earlier in this lesson.

**Electrode Continuity Tests.** Before an ohmmeter is applied to a battery receiver, it is necessary to disconnect all batteries to avoid damage to the meter.

For practical examples of ohmmeter tests, consider the typical battery receiver of Fig. 10. This is the way the diagram appears in service manuals with parts identified only with numbers.

Note that the cathode of the *1C5G1* is connected directly to chassis, as is one terminal of the "A" battery. Since the other side of the filament goes directly to the battery, it is evident that the filaments are in parallel and that each one in the set will be grounded on one side.

The double-pole, single-throw switch *SW* is typical of battery receivers, as

it will interrupt both "A" and "B" supplies when turned off. This is a necessary arrangement to prevent leakage through electrolytic condenser 13 from ruining the "B" battery if only the filament was turned off. This switch must be closed before making continuity tests from the battery leads after disconnecting the batteries.

In case the battery leads are not marked to show polarity, you must write down the color code before disconnecting the leads from the battery. In the case of battery leads connecting to a plug which fits into a socket on the batteries, you will have to trace the leads into the set or use a voltmeter to find positive and negative terminals of the battery socket, so the corresponding plug pins can be identified, after which the leads can be labeled. A voltmeter will be essential in the case of a receiver having "A," "B" and possibly "C" batteries all in one block and with a plug connector.

Once the polarity of the battery leads is established, continuity checks to the tube electrodes are made in the same way as for the other receivers already discussed.

The isolation of defective parts in the simple circuits of the typical battery receiver is relatively easy. For instance, suppose that part 15 in Fig. 10, an 800-ohm resistor, is open. This is the bias resistor for the output stage but note that the plate and screen currents of all stages go through it. An open in this resistor will remove plate and screen voltages from the whole set, killing all operation. The continuity check from B+ to plate and screen would read correctly, but the test from grid to chassis would indicate an open which would have to be in part 10 or part 15. A check across each with the ohmmeter will quickly find the open.

**Voltage Tests.** This same open re-

sistor could be found with voltage measurements. By any method of stage isolation we would have found the receiver dead and would suspect a lack of voltage on the plate. A measurement from plate to B— would show voltage, but a measurement from plate to chassis would show no voltage. This definitely puts the open be-

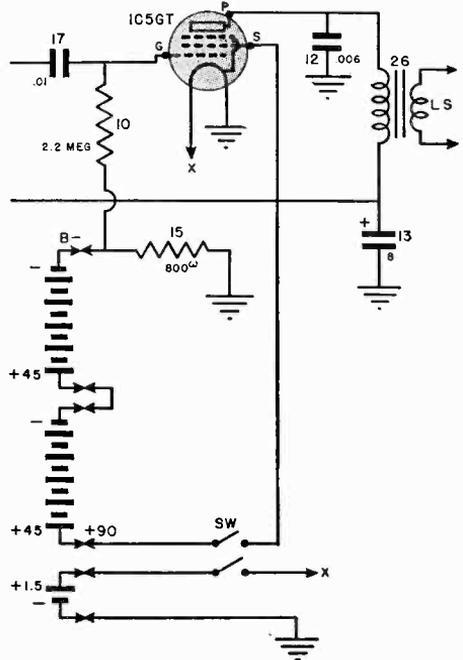


FIG. 10. Output stage of the Stewart-Warner 02-4A battery powered receiver.

tween chassis and B —, and part 15 is the only such connection.

This example shows the importance of using proper reference points, and also shows that the use of more than one reference point is valuable if the circuit diagram is carefully studied.

**Other Defects.** A complaint often found in battery receivers arises from an open in the 8-mfd. electrolytic condenser, part 13. At first thought, it would seem unnecessary to use a large

filter condenser in a battery receiver, with no a.c. hum voltages to filter out. The condenser is not for hum filtering, however, but to provide a path around the "B" battery for r.f., i.f., and a.f. signal currents. With a new battery, this would not be necessary because the resistance of the battery is low but as the battery ages, its resistance will often rise markedly. The signal currents through the battery then develop signal voltages across it which are fed from one stage to another since the "B" battery is common to all stages. This "feedback" is a potent cause of oscillation. With condenser 13 across the battery, such oscillation is eliminated.

Thus, oscillation in a battery receiver should lead you to shunt condenser 13 with a new one. If the trouble stops, replace the condenser.

Opens in condensers 17 and 12 of Fig. 10 would not be revealed by an ohmmeter or voltmeter test but must be found, as previously described, from the operational symptoms of the receiver or with signal-tracing equipment.

### RECEIVER USING SYNCHRONOUS VIBRATOR

Receivers having *vibrators followed by full-wave rectifier tubes* are handled in exactly the same way as ordinary a.c. receivers with power transformers, once the vibrator circuit is checked and cleared of suspicion.

However, in *full-wave synchronous vibrator* systems, there is no rectifier tube with cathode and plate terminals to serve as reference points for continuity tests. Furthermore, even after we locate reference terminals in receivers of this type, certain special precautions are necessary to avoid confusing readings. Therefore, let us go through the general procedure involved here for isolating the defective part,

using the typical circuit in Fig. 11 as an example.

**Electrode Continuity Tests.** The diagram shows that the cathode and control grid circuits connect to the chassis. This is typical of synchronous vibrator receivers. Hence, the chassis is the reference terminal for all continuity tests to negative electrodes.

If we trace through the entire plate supply circuit of the 6S7G tube, starting from the plate, we see that there is continuity through parts 47, 46, 2 and 3 to point *c*, which is the center tap of the secondary of part 66, the power transformer. The center tap *c* of the high-voltage secondary winding is, therefore, the *highest positive terminal in a power pack of this type*.

It may not be easy to locate the secondary center tap *c*, as the transformer may be mounted in a shielded container. This container may even include choke coil 3, condenser 16 and, in some cases, choke coil 2.

Therefore, we may have to find some other point as near the power supply as possible for a positive reference point. The positive terminal of a filter condenser is usually easily found, or you might use the plate or screen grid terminal of the output power tube. Using these output tube terminals as positive reference points—if the output stage is found to have normal voltages—is a very convenient method and can be used in any type of receiver.

When the receiver of Fig. 11 is turned off to make a continuity test, the vibrator arm may be touching contact "B." This grounds one end of the high-voltage secondary, making it impossible to check the positive circuits for shorts to the chassis. This difficulty can be avoided by pulling the vibrator out of the socket. If the vibrator can not be removed but the moving parts are accessible, paper can be put

between the contacts. If the vibrator is sealed up in a metal housing, it is best to break the circuit at point *c* and use the lead so disconnected as the reference point.

Suppose that condenser 27 in Fig. 11 becomes leaky, drawing a high current through resistor 46 and burning it out. Ordinarily, a continuity test from the plate of the tube to the highest positive point would immediately show an open, but if vibrator contact *A* or *B* is closed, there will be a d.c. circuit through 47 and through shorted condenser 27 to the chassis, through contact *B* and the secondary winding to point *c*. This looks as though the plate supply circuit is unbroken. If the vibrator is removed, the open will be immediately apparent.

A continuity check to the plate of the 6S7G would normally read about 400,000 ohms, the total ohmic value of parts 47 and 46. If this value is found, then we can check for shorts to the chassis. These might be in parts 12, 27, 11, or 16. With the chassis as reference point, start with the ohmmeter probe on the plate of the 6S7G tube. As in previous similar tests, a very low reading here means a short in condenser 12. The test could continue as in previous examples by moving the test probe along the circuit to find very low readings or points of lowest resistance to chassis. For instance, if the resistance to chassis at point *d* is lower than at points *p* or *e*, we know that the leakage must be in condenser 27. If point *e* gives a lower reading than *d* or *p*, the leakage must be in the power pack. This principle of searching for the point of lowest resistance to chassis is extremely useful for finding parts that are not completely shorted, when the leakage may be obscured by the resistance in the circuit.

The leakage through the electrolytic

condensers in the power pack should be always kept in mind when checking for shorts from the positive circuit to chassis.

A further useful procedure for finding leakage in a circuit where many parts might be involved, especially if a diagram is not available, is to disconnect parts so that individual circuits or parts can be tested alone. For instance, in Fig. 11, if there is a very low reading from point *f* to chassis, you could begin by unsoldering the choke 2 terminal from *f*. If this does not remove the short between *f* and the chassis, disconnect the lead to the other stages at *g*. If the low reading at *f* disappears, the short is beyond *g* in the other stages and can be run down. This method can be continued by disconnecting junctions right up to the plate of the tube until the short is found.

It is not advisable to make a continuity check of the grid circuit, due to the presence of the bias cell. Even the ohmmeter current flow through this kind of cell can ruin it. Never try to measure the voltage of a bias cell either, except with a vacuum tube voltmeter. If you notice the presence of a bias cell on the diagram or in the receiver, you will avoid using either an ohmmeter or voltmeter in any tests that put the bias cell in the test circuit.

Continuity can be checked through grid circuit parts 40 and 45 by placing one ohmmeter probe on the chassis and the other at point *h*, the junction of part 45 and the bias cell. A normal reading of about .5 megohm will be obtained with the volume control set at maximum, and about 50,000 ohms with the volume control set for minimum.

To test the voltage of a bias cell, if a vacuum tube voltmeter is not available measure the plate-cathode voltage with a d.c. voltmeter, first with the

cell in the circuit and then with the cell removed and the bias-cell socket terminals shorted or connected together. If there is no difference between the two readings the cell must be defective since the difference in plate current caused by a removal of bias should normally cause a sharp drop in the voltage at the plate. If the cell is defective, putting in a new one will cause the plate voltage to rise considerably and will clear up the distortion that is almost certain to be present with a defective cell.

If a new cell is not available, the cell can be replaced in most circuits by a high resistance, on the order of 10 megohms, and the bias is then obtained by convection current as in the circuit of Fig. 5. However, in Fig. 11, the high resistance cannot be placed

in series with the circuit. The problem is solved by placing a condenser of about .01 mfd. across the bias-cell socket, and connecting a resistor of about 10 to 15 megohms between grid and cathode as shown in Fig. 12. The bias cell is removed entirely. With a gassy tube, this system can never be used since the grid will go positive and the tube will soon be ruined.

**Electrode Voltage Tests.** Voltage tests in the stage of Fig. 11 can be made to chassis in the same way as in other receivers having the chassis in the d.c. supply circuit. The first check, as before, is usually from plate to chassis. Even though no voltage values are given on the diagram, comparative readings taken at points *f*, *d*, and the plate, will serve to isolate a defective part. For instance, if the plate reads

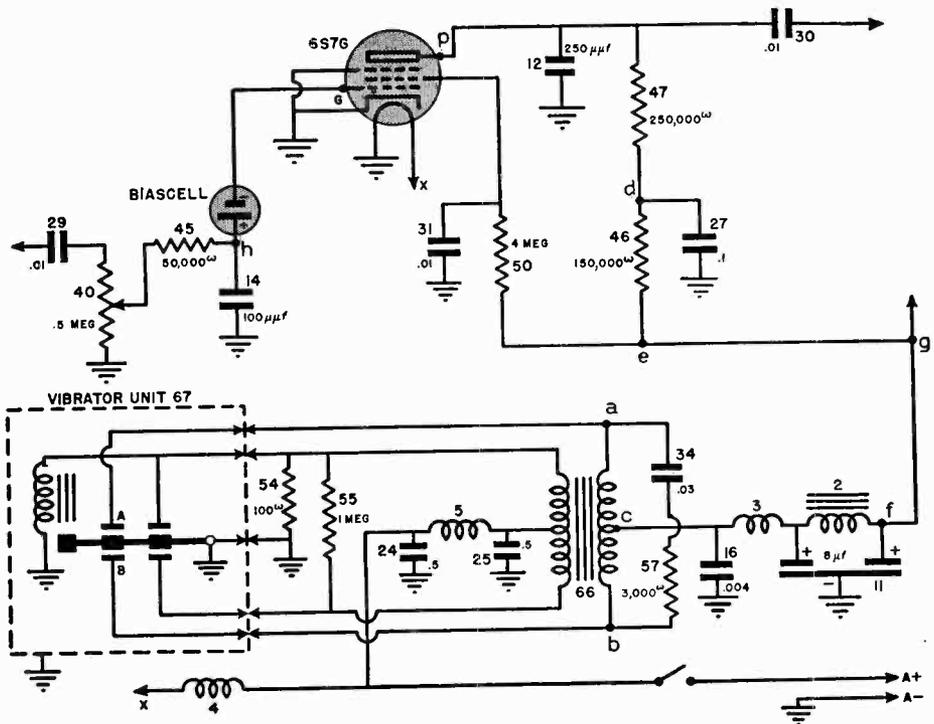


FIG. 11. Power supply and first a.f. stage of the Stewart Warner R-192-D vibrator-powered receiver

no voltage and you find voltage at *d* and higher voltage at *e*, it is evident that condenser 12 is shorted. Or if the plate again shows no voltage and you get about the same reading at *d* and *e*, it is obvious that resistor 47 is open, as no current is flowing through 46. This is in line with the standard voltage procedure you have been learning all through this lesson and with your analysis of circuit action in the plate circuit.

If you found about the same voltage-to-chassis reading right up to the plate of the tube, for instance, it means that the circuit is unbroken and unshorted up to the tube terminal but that no tube plate current is flowing. This throws suspicion on the tube itself. To take the opposite case in which we suspect that leakage from the plate circuit to chassis is causing higher than normal current through 46 or 47, we could not tell accurately from voltage readings that leakage current existed because no voltage values are given on the diagram. However, by removing the tube from the socket, as previously explained, the normal plate current is interrupted and any remaining current through the plate circuit must be leakage current.

Remember from our previous example of this kind that measurements for voltage drop across resistors 47 and 46 should not be made by measuring to chassis with a low-resistance voltmeter, since the current through the meter will confuse the readings. The measurement can be made directly across the resistors themselves.

### LOOKING AHEAD

We have now completed the study of the basic testing methods and the

effect-to-cause reasoning which are needed to isolate the defective part and restore the receiver to operation that you, as a professional serviceman, will face in nine out of ten of the servicing jobs. These methods, if thoroughly understood and learned not by heart but so that you really know why they are used, will give you the ability you need for success. And never miss an opportunity to reread the sections that give you the theory of action in every kind of radio circuit, because in this way you will get the finest satisfaction that comes from radio, that of having the basic workings of radio at

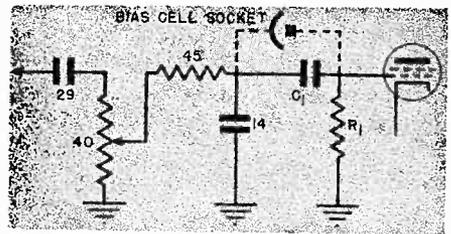


FIG. 12. Parts 29, 40, 45 and 14 are the same as in Fig. 11. Parts C<sub>1</sub> and R<sub>1</sub> are the extra parts used to obtain convection bias.

your fingertips so that you can enjoy being "in" on the thousands of new developments that come in this expanding science every year.

One more most important servicing procedure, the proper adjustment of trimmers—known as alignment—will be taken up in the next lesson. And then we will go on to those highly unusual troubles that require special testing methods and more detailed knowledge. We intend that the serviceman who makes an honest study of this Course shall never be "stymied" because he came up against a case that is far off the beaten run of the average serviceman's experience!

# Lesson Questions

Be sure to number your Answer Sheet 38RH-2.

Place your Student Number on every Answer Sheet.

Send in your set of answers for this lesson immediately after you finish them, as instructed in the Study Schedule. This will give you the greatest possible benefit from our speedy personal grading service.

1. A signal tracer measures a large i.f. signal between the screen grid and ground (chassis) of Fig. 1. Which part is defective and what is the matter with it? *R3 open or shorted*
2. If a receiver, using the a.v.c. circuit of Fig. 5, goes dead momentarily during strong bursts of static, which part is defective and what is the matter with it? *C5 open*
3. In Fig. 1, would you expect to get the same ohmmeter reading between each rectifier tube plate and chassis? *yes*
4. Name three easily identified points in Fig. 7 which could be used as the common negative reference point for ohmmeter measurements.  
*cathode 6A7G, rectifier tube, megator, C20-C21*
5. If an ohmmeter check between the plate and cathode of the 6K7G tube in Fig. 7 shows a resistance of about 35 ohms, what part is defective?  
*C3 plate to case*
6. If resistor  $R_7$  of Fig. 7 is replaced, and the replacement smokes and overheats, what part is probably defective?  
*C3 - PC*
7. When checking across  $R_6$  of Fig. 9 with a voltmeter to determine if  $C_{16}$  is leaky, should the positive or the negative voltmeter terminal go to B —?  
*Positive*
8. In Fig. 10, normal voltage is found between plate and B — but no voltage is between plate and chassis. What part is defective?
9. In the circuit shown in Fig. 11, what must be done before continuity measurements can be made, in addition to turning OFF the set?
10. With the tube in Fig. 11 removed, suppose your voltage readings between  $d$  and chassis are considerably lower than that between  $e$  and chassis. What other voltmeter reading will tell you whether the leakage is in condenser 12 or is in condenser 27?

Be sure to fill out a Lesson Label and send it along with your answers.



## MAKE BELIEVE IT'S TRUE

A young man once asked a successful friend to state just one rule for success. "*Look as though you have already succeeded,*" the friend advised. Following this rule eventually made this man president of a great bank in New York City.

As Shakespeare expresses it: "*Assume a virtue if you have it not.*" Dress like a successful man, act like a successful man. Be successful in all your thoughts, and you will be successful in the world as well.

David V. Bush states this same thought in terms of radio: "*If we think poverty thoughts, we become the sending and receiving station for poverty thoughts. We send out a 'poverty' mental wireless, and it reaches the consciousness of some poverty-stricken 'receiver.' We get what we think. If we are going to be timid, selfish, penurious and picayunish in our thinking, these thought waves will go forth until they come to a mental receiving station of the same caliber. 'Birds of a feather flock together,' and minds of like thinking are attracted one to the other.*"

Once you learn that you have a legitimate control over your own affairs, that you have the right to win, *you will win.*

*J. E. Smith*

# **TUNING CIRCUIT TROUBLES**

## **ALIGNMENT**

39RH-1



**NATIONAL RADIO INSTITUTE**

**WASHINGTON, D. C.**

**ESTABLISHED 1914**

# STUDY SCHEDULE NO. 39

For each study step, read the assigned pages first at your usual speed, then reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind. Study each other step in this same way.

1. Tuning Circuit Troubles.....Pages 1-9

Usually, alignment is all that is needed when tuning circuits cause trouble. However, sometimes other troubles may occur, such as shorted coil turns, excessive leakage or increased series resistance. These cause troubles which can be localized by the usual effect-to-cause reasoning or stage elimination procedures. You can then find whether the tuning circuit is at fault by noting whether the trimmer tunes broadly, whether a readjustment produces better results or whether it is impossible to reach a peak. All these clues point to definite tuning circuit conditions.

2. Preliminary Considerations for Superheterodyne Alignment....Pages 9-16

Before starting to align a receiver, you must locate the trimmers, have the necessary equipment and properly set the receiver dial and controls. This section explains these preliminary steps in detail.

3. Methods of Alignment.....Pages 16-23

The method to follow depends on whether the receiver is just being "trimmed" for better results, or whether it is badly out of alignment, due to tampering. Also, different procedures are followed for the different types of all-wave receivers. Learn the basic alignment procedures "backwards and forwards"—you can then go on to all-wave sets and unusual cases with confidence. Remember, some practice will be necessary before these operations become "second nature" to you.

4. Special Alignment Cases.....Pages 23-28

Here are details on special jobs, such as F. M. receivers, receivers using regeneration, T. R. F. sets and variable selectivity receivers. You can always refer to these instructions when in doubt as to the proper method to use. The rules are simple, though, so you will find practice soon fixes them in your mind.

5. Answer Lesson Questions and Mail Your Answers to N. R. I.

6. Start Studying the Next Lesson.

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# TUNING CIRCUIT TROUBLES

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## ALIGNMENT

### Tuning Circuit Troubles

**I**N THIS lesson you will learn how the professional serviceman gives quick, accurate service for another class of receiver troubles, those connected with defects or misalignment of the tuning circuits. You have already learned the professional methods of finding a defective part in the supply circuits of a receiver. These methods might be called the "ground-work" of radio servicing, which gives you a basis for your career as a serviceman. Now we are going to take you a big step further, with the ability to handle tuning circuit troubles and alignment. The man who can work with the same speed and accuracy on any receiver brought in for service and is not limited just to simple types of receiver breakdowns is in a special class, far above the average "repairman" in profits and in professional standing.

▶ With tuning circuit troubles, effect-to-cause reasoning is of the greatest importance and will save the serviceman large amounts of his most precious commodity—time. The actual defect may be an open circuit, short circuit, increased series resistance, change in coil Q, etc., but these troubles are unique in that the manner in which the receiver reacts, when tuned or when alignment adjustments are made, often serves to localize the trouble.

We will show first how to apply this principle to complaints associated

with the tuning circuits—weak reception, poor selectivity, dead spots and station interference. Then we will cover alignment procedures fully, and you will learn just how to restore maximum sensitivity and selectivity to a receiver.

#### WEAK RECEPTION

When the customer says "weak reception," he means either low volume or poor sensitivity. If both local and distant stations are weak, the audio stages are likely at fault. If distant stations only are missing, or if they come in at the wrong points on the dial, the r.f. section contains the defect, so we will deal only with poor sensitivity here.

▶ Question the customer carefully. If the trouble came suddenly, it is probably a supply circuit failure, whereas a gradual loss of pep is more often in the tuning circuits or in alignment. The stage and parts isolation techniques you have already learned should generally be applied first, to make sure the trouble is not caused by a defect in a tube or supply circuit.

Alignment may be off in one stage or in several. With all stages slightly out of line, the over-all sensitivity will be low, but individual stages may be close enough to normal to give misleading results when using a signal tracer. Such slight misalignment may occur from aging of the trimmers,

destroying their springiness so their plates change position.

A slight shift of all trimmers in the same direction would merely cause a change in the dial reading of station reception; a 1400-ke. station would come in at 1420 kc., say. However, it is more likely that one stage will be off in one direction and another stage in the other direction. Then, when the set is tuned, each stage reduces the signal. Tuning for better response in one transformer reduces it in the other, so that the total effect is one of weak response with broader tuning than normal.

► How can we make a quick test to determine whether or not weak reception is due to misalignment? Move a suspected trimmer adjuster slightly in one direction and then in the other; any signal *increase* when a trimmer is moved slightly indicates misalignment. If only decreases are found, alignment is probably not at fault and the trimmer should be restored to its original setting.

*Don't blindly "play" with the trimmers on a set.* Complete alignment may be necessary in any case if you move the trimmers any great amount in making tests.

► Weak reception naturally results when the oscillator frequency shifts, because it is necessary to tune the receiver dial to a different point than that where the preselector was aligned, so the preselector cuts down the signals by being tuned off-resonance. Another clue to this is the fact that stations come in at the wrong dial points. Also, there would be an unusual amount of noise, because the preselector fails to pass enough signal to over-ride the converter noise.

► Weak reception can be caused in the tuning circuits not only by misalignment, but by defects which lower the  $Q$ . Corroded connections, intro-

ducing resistance in the tuned circuit, leakage paths across the coil or condenser, or moisture absorbed by the coil will all lower the  $Q$  and reduce the step-up of the tuned circuit. A careful check of the trimmer adjustments will help in finding a low  $Q$  circuit. A tuned circuit with proper  $Q$  will come to a sharp peak as the trimmer is adjusted. If the trimmer has very little effect on the signal, with broad adjustment—no sharp resonant point—look for the defects which lower  $Q$ .

Considerable experience with radio receivers will be necessary before you can judge the normal sharpness of adjustment of a tuned circuit. A preselector circuit, for example, is rather broad anyway. In other cases, the design of the receiver may be such that it is intended to have much sharper tuning than you at first expect, so you might overlook the fact that the set tunes more broadly than it should. Be sure to note the results of your first alignment jobs carefully, so you can develop the ability to judge whether or not you are getting a normal trimmer adjustment.

► When a trimmer adjusts properly, it should be possible to pass through a point of maximum response. In other words, as you adjust the trimmer you reach a maximum, and further turning in the same direction causes response to fall off again.

Should you notice that as you turn the trimmer in or out, the volume steadily rises but you never reach or pass through a maximum peak, there is something wrong with the condenser or coil in the tuned circuit or the wrong frequency is being used.

► From the foregoing, you can see that weak reception, if combined with loss of the distant stations or misplacement of stations on the dial or a lack of proper peaking of the

trimmers, points to a tuning circuit trouble. Usually, if the alignment seems to be fairly good and the trimmers show a reasonably good peak, the trouble is not in the tuning circuit.

## DEAD SPOTS IN A RECEPTION BAND

There are three cases where dead spots may occur. In all instances of dead spots, you have an r.f. trouble. However, sometimes the tuning circuits are not at fault.

**Case 1.** In this case there is no signal received at all, except for one low-frequency station. Usually this one station comes in over the entire low-frequency end of the band.

This invariably means that the oscillator in a superheterodyne receiver has stopped operating. Without oscillator voltage, the first detector may become an amplifier, and with the help of the preselector may force a low-frequency station through the i.f. amplifier. Such single-station reception means we must restore operation to the oscillator stage.

**Case 2.** In the next case, signals are picked up either at the low-frequency or the high-frequency end of the dial, but not at both ends.

When signals are received only at the low-frequency end of the dial and at approximately their normal position, this usually means that the oscillator stops at the high-frequency end. However, it is possible that the oscillator may be misaligned so that it tracks with the preselector only at the low-frequency end of a band.

If signals are received only at the high-frequency end of the dial, the tuning condenser gang may be short-circuited at the low-frequency end by a bent plate or dust and metal particles between the plates. If you do not find any rotor plates which appear to be touching a stator plate, particu-

larly if no unusual noises are heard as the gang condenser is rotated, the oscillator may be stopping or again we may have a case of misalignment. Measure for a voltage across the oscillator grid resistor, if one is used. Lack of voltage over the portion of the dial where no signals are received indicates that the oscillator stops on that section of the dial.

*Oscillator Troubles.* The cure for various oscillator troubles depends on the type of circuit in use. Fig. 1 shows a typical circuit using a pentode tube as a combination mixer-oscillator. When the tube ages it may fail to oscillate at the low-frequency

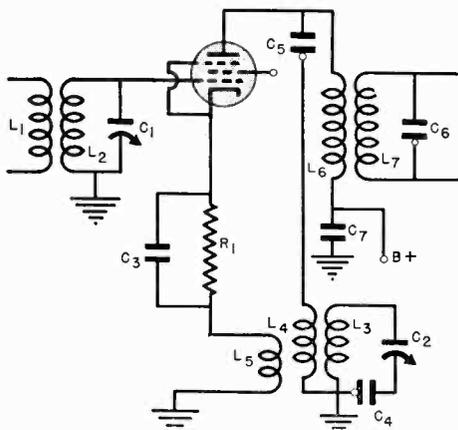


FIG. 1. A typical pentode type mixer-oscillator stage.

larly if no unusual noises are heard as the gang condenser is rotated, the oscillator may be stopping or again we may have a case of misalignment. Measure for a voltage across the oscillator grid resistor, if one is used. Lack of voltage over the portion of the dial where no signals are received indicates that the oscillator stops on that section of the dial.

Replacing the tube may be only a temporary cure, as these circuits are sometimes designed for selected tubes. In such cases, reduce the value of the oscillator bias resistor  $R_1$  by about one-third of its original value. Thus, if the resistor was originally 9000 ohms, replace it with a 6000-ohm resistor.

By-pass condenser  $C_4$  may cause trouble and should be replaced if a smaller resistor does not help. High-resistance connections in the tank circuit  $L_3-C_2-C_4$  could also cause this trouble so go over each joint in the circuit with a hot soldering iron. Electrical connections are usually made to the stator of the tuning condenser through the bolts holding the

stator in place. Corrosion between the threads of the bolts and the stator assembly may form a high-resistance joint. Usually, simply loosening and tightening these bolts one at a time is enough to remove the corrosion and the resistance.

If oscillation stops at the high-frequency end of the dial, the trouble is generally increased r.f. resistance in the tank circuit, lowering the coil Q. Moisture absorption by the oscillator coil form is to be suspected. Either replace the oscillator coil or remove it and dry it out in an oven. After baking, the coil should be dipped in melted paraffin to prevent further trouble of this kind. Cracked mica in i.f. trimmer  $C_5$  or low-frequency padder  $C_4$  will also cause this oscillator trouble.

▶ A typical pentagrid-converter tube circuit is shown in Fig. 2. If this circuit refuses to oscillate at the low-frequency end of the dial, the trouble may be due to high-

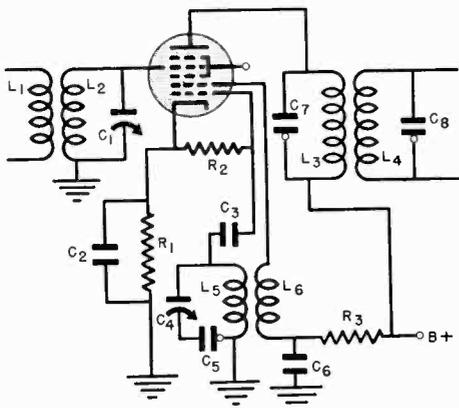


FIG. 2. The standard pentagrid-converter, functioning as a mixer-oscillator.

resistance joints in the tuned circuit. If an extra peppy tube is required to maintain oscillation, the ability of the oscillator to perform may be improved by increasing the value of the oscillator grid resistor  $R_2$ . Generally, a value between 50,000 and 75,000 ohms should be used with 6.3-volt tubes; too large a resistor will cause blocking of the oscillator.

Where battery pentagrid-converter tubes are used, oscillator failure can often be traced to lowered filament voltage. The circuit-disturbance test will indicate that

the mixer section is alive but the oscillator simply will not function. The remedy is to replace the A battery or to use a more peppy tube. In the case of the 1.4-volt types particularly, the value of the grid resistor  $R_2$  may have to be increased up to as high as 275,000 ohms.

**Case 3.** In this case signals are received at both ends of the dial, but not satisfactorily in the middle. This is a rather interesting case of misalignment.

Curve 1 in Fig. 3 represents the *pre-selector alignment* of a typical receiver. As the dial of the receiver is changed from 0 to 100, the band from 1500 to 500 kilocycles is covered.

If the i.f. is 260 kilocycles, the proper oscillator tracking is shown by curve 2. The oscillator range must be from 1760 to 760 kc., in order to receive all frequencies in the band. Curves 1 and 2 are exactly parallel, having the same spacing (260 kc.) between them at all points.

In many receivers, the adjustable range of the high-frequency trimmer is so great that it is possible to switch the oscillator frequency *below* the incoming frequency instead of *above* it, particularly in the older receivers having low intermediate frequencies such as 175 kc. or 260 kc. (In receivers having higher i.f. values, this trouble will normally be encountered only on the short-wave bands.)

Naturally the i.f. value will be produced whether the oscillator is above or below the incoming frequency, so long as the difference is equal to the i.f. Therefore, if we adjust the oscillator to produce 1240 kc., a 1500-kc. signal can be received, as a 260-kc. i.f. is produced. If the oscillator now ranged from 1240 kc. to 240 kc., the difference between the oscillator frequency and the incoming signal would again produce the i.f. signal at all points on the dial. Again

we would have a curve parallel to curve 1, but underneath it.

However, the usual oscillator circuit will not work this way, as the tuning circuit values force it to be *above* the incoming frequency at the low-frequency end of the band, even though it is *below* the signal at the high-frequency end. (The only time this will not be true is when the oscillator is specially designed, as in certain ultra - high - frequency wave bands.)

Thus, if you have an improperly adjusted trimmer, putting the oscilla-

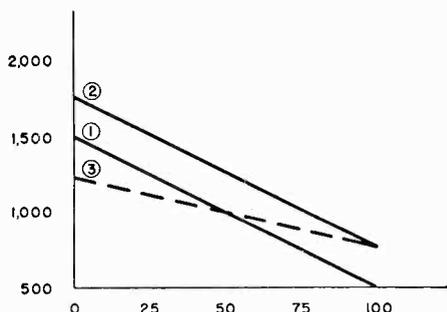


FIG. 3. Curves showing how cross-over produces improper tracking.

tor *below* the signal at the high end of the dial, the frequency of the oscillator will usually range as shown in curve 3 of Fig. 3—it will “cross over” to be *above* the signal at the other end of the dial, so this condition is called “cross-over.”

With this condition, stations at the high-frequency end of the dial will be received, and so will stations near the low-frequency end of the dial, because in both instances the oscillator frequency is such that the desired i.f. can be produced. However, at the center of the band, the oscillator frequency approaches and may even be the same value as the preselector alignment. As a result, we would have to tune away from the correct dial position, to produce the i.f. This detunes the preselector so stations in the

middle of the band will be attenuated to such an extent that they will not be received.

► The way to avoid cross-over, when aligning the standard receiver, is to be sure that the oscillator frequency is *above* the incoming frequency. When aligning the oscillator, rotate the high-frequency oscillator trimmer throughout its range. If you find that there are two oscillator trimmer positions which give a maximum output, choose the trimmer position which has lower capacity (plates more open) and the oscillator will be *above* the incoming signal.

### POOR SELECTIVITY

In general, *poor selectivity* means that when attempting to pick up one station, another station 10 or 20 kc. away is heard at the same time.

► Poor selectivity is common in t.r.f. receivers. Here, a check of the alignment is desirable. In addition, check for missing shields since this will frequently permit some circuit nearer the detector to pick up an undesired signal in such a way that it does not have to pass through the tuning circuits.

► With superheterodyne type receivers, broad tuning is not a common complaint. However, certain design conditions may produce the symptoms of poor selectivity.

The automatic volume control circuit will affect the broadness of response of a receiver. When you tune in a local station exactly, the a.v.c. voltage reduces the sensitivity of the i.f. amplifier a great amount. As you tune away from the proper dial position, however, the reduction in signal strength also reduces the a.v.c. voltage and increases sensitivity so that the receiver tends to “hold” the strong signal past its true dial position. In fact, the local station may

be held to such a point that the signal sounds distorted due to side-band cutting in the tuned circuits. Any other stations which are tuned within the "holding" range will suffer interference. However, the local station will not be interfered with itself, because the a.v.c. reduction in sensitivity will discriminate against weaker stations on the same frequency as the local.

► As another instance of design affecting selectivity, consider the high-fidelity receiver. In this type, the i.f. amplifier is band-passed to give a broad response. The selectivity curve is not sharp but instead has a broad top with fairly steep sides, which is very necessary for high fidelity. However, you will frequently encounter owners of receivers of this type who complain of the lack of selectivity.

In these cases selectivity can be improved only at the expense of fidelity, unless it is possible to make use of wave traps or other similar devices, or to change the antenna. Surprisingly, you will find that sometimes a smaller antenna, and other times a larger antenna may be necessary. With the larger antenna, usually the stronger stations will develop more a.v.c. voltage, which will tend to lower the sensitivity of the receiver to a point where perhaps the interfering stations will not be picked up. On the other hand, sometimes a smaller antenna will discriminate against the interfering stations, particularly when it is erected in such a direction that the desired stations are favored.

► Should the i.f. amplifier of a receiver be out of alignment, the selectivity of the receiver may be broadened. When this occurs, we have a real case of poor selectivity which alignment can correct. Unless you are familiar with the characteristics of the particular receiver, you may find it impossible to determine whether the

condition is due to the design of the set or to misalignment. In such a case, try realigning the receiver to note whether this makes any improvement. Improper alignment will cause some loss of sensitivity, so the characteristics of a set with poor selectivity because of misalignment would be: *Strong local stations spread over the dial without a great amount of distortion; local stations blank out distant stations; distant stations are weak.*

► Lack of selectivity can be produced by lowered Q also, produced by such circuit defects as moisture and dirt in coils or in condenser wiping contacts, all of which will lower the circuit Q and the selectivity. If a stage is overloaded, the positive signal peaks cause a grid current flow and make a conductive path between the grid and cathode, which places a fairly low resistance across the grid resonant circuit, thus lowering Q.

► As you can see, poor selectivity may be due to circuit defects, to misalignment or to the design of the receiver. Localizing the trouble to one stage by using a signal tracer or signal generator, then testing selectivity and gain by shifting trimmer settings will prove whether alignment or a circuit defect is the trouble.

## INTERFERENCE

**Station Interference.** There are two kinds of station interference. In one case the interfering station is on the *same* frequency as the desired station, and in the other case the interfering station is on a frequency more than 40 or 50 kc. away from the desired frequency.

► When the interfering station is on exactly the same frequency as the desired one, the receiver's tuning circuits cannot possibly separate the two and there is nothing to do but try

different antenna systems. We must either increase the strength of the desired signal so that the a.v.c. can lower the sensitivity of the set enough to cause the interfering station to drop out, or we must erect an antenna system which will discriminate against the unwanted station.

► In the other case, where a strong local station causes interference over 40 or 50 kc. on the dial, the local signal is usually getting into the receiver without passing through the tuned circuits. Absence of shielding may permit this signal to get directly into the second detector, or it may come in over the power line and be radiated to some signal circuit.

In these cases, by-passing the power line, installing proper shielding, and sometimes using a wave trap in the antenna circuit to reduce the amount of energy from the interfering station will usually eliminate the trouble. Under ordinary circumstances, this is not a condition which alignment will help at all. Remember, the tuned circuits provide selectivity only because they are resonant to a particular frequency to which they happen to be tuned at the moment. If a station quite far removed from that frequency is so powerful that it can force its way through, or can get into the circuit at some point other than through the normal paths, the tuned circuits cannot be considered at fault.

**I.F. Interference.** The usual superheterodyne receiver will have an i.f. somewhere in the 175 to 465-kc. range. Any local commercial or government station that happens to have a frequency the same as the i.f. value may force its way through the preselector and then be amplified by the i.f. stages to give interference. You can recognize this condition usually by the fact that this interfering station is heard over a wide range of the station

dial, particularly at the low-frequency end, since at these positions the preselector is less effective in its rejection. More important, the signal will be code or a type of message quite distinct from a broadcast signal.

The offending station need not be at the i.f. value—it may be at some subharmonic and still interfere. If the i.f. is 460, a 230-kc. station can produce in the r.f. stage or first detector a second harmonic of 460 kc. which causes interference.

Since the interfering code station is on the i.f. or is some sub value of this frequency, we might at once believe that the cure is to shift the i.f. value of the set somewhat. Where it is possible to do this, all we need to do is tune the i.f. 5 or 10 kc. away from the interference and it will be entirely cleared up.

If the receiver design does not permit an i.f. shift without upsetting the dial tracking, it may be necessary to add shielding to any exposed coils or tubes to eliminate any direct pickup from the interfering station. Sometimes, r.f. filters in the power line will prove helpful.

► However, before doing this, examine the set or diagram carefully. Many receivers have built-in i.f. wave traps in the antenna circuit which are tuned to the i.f. and will usually block any signal of that frequency out of the receiver.

If the set has no i.f. trap, one can frequently be added. The proper type will depend on the type of receiver. A parallel-resonant circuit may not be effective in the antenna lead if the receiver input circuit is of high impedance, while a series-resonant circuit shunted across the receiver antenna and ground posts will not be effective if the receiver has a low input impedance. Try both types of traps to see which works better.

**Image Interference.** A signal frequency *above* the oscillator frequency by the i.f. value will cause interference, if it can get through the preselector. We assume the station having a frequency *below* the oscillator frequency by the i.f. value is the desired station and is tuned in by the preselector. Since the undesired signal is of the proper frequency to produce an i.f. signal, it is called the "image signal."

The existence of image interference depends on the selectivity of the preselector and on the i.f. value. The lower the i.f. value, the greater should the preselector selectivity be, because a low i.f. puts the image signal closer in frequency to the desired signal. High i.f. values of about 460 kc. permit less-selective preselectors, for now when the receiver is set to, let us say 600 kc., the image signal for a 460-kc. i.f. system will be  $600 + 460 + 460 = 1520$  kc.; even a single-tuned-circuit preselector can separate frequencies so far apart.

Usually image interference occurs when a weak station is tuned in and a strong local is at the image frequency. Under these conditions, the a.v.c. voltage is low and the stages are running almost "wide open" (high gain).

You can recognize the trouble as image interference if you can recognize the program on the interfering station as coming from a station which is twice the i.f. *above* a desired station frequency.

► If a single station comes in at two points on the dial, one correct and the other an image point, without causing interference, the action is called double-spot tuning. Often no action is taken to correct this, as long as no interference results.

► There are two usual remedies for images which do cause interference. Where one local is giving trouble, a

wave trap will usually reduce its penetration sufficiently to eliminate image interference. You may also shift the i.f. value so the image signal will come in at a dial position where it will not be objectionable. Complete alignment of the i.f. and preselector stages is important, and any defect that reduces stage selectivity should be eliminated.

**Oscillator Harmonic Interference.** We have so far assumed that the oscillator generates only one signal at a time. You realize, of course, that the average receiver oscillator will produce a series of harmonics which can combine with some local high-frequency station to give interference.

For example, the receiver is tuned to 1200 kc.; the i.f. is 460 kc. The fundamental oscillator frequency is  $460 + 1200 = 1660$  kc. Its second harmonic is 3320 kc. A station 460 kc. above or below 3320 kc. can produce the intermediate frequency. A government station in your locality may operate on one of these frequencies with sufficient strength to penetrate the preselector, giving squeals and station interference. Also, this explains why police calls are sometimes received on the broadcast band. Should these signals interfere with desired stations, a shift of the i.f., or the use of a wave trap tuned to the offender are the usual remedies.

**Intermodulation Interference.** This interference usually occurs in localities where many high-powered broadcasters exist. Let us say we have this combination of conditions—the i.f. is 260, two local stations exist, one of 1000 kc., the other 1260 kc. Both signals are strong enough to penetrate the preselector. Once they get to the first detector, they will combine to produce a signal at 260 kc. which rides through the i.f. stages. You will

hear garbled programs over a wide tuning range, being strongest in the 1000 to 1260-ke. tuning range.

If you suspect this trouble, eliminate temporarily the oscillator action (short oscillator tuning condenser, or pull out tube if separate). If the interference continues, you have intermodulation interference. The logical solution is to change the i.f. value.

**Cross-talk.** When a strong local station rides in with one or more of the other desired stations—sometimes with all of them—and is not at a frequency which can produce intermodulation or image interference, the condition is called “cross-talk” or “cross-modulation.”

Cross-talk occurs because something in or near the radio is acting as a rectifier. If the receiver is close to a powerful local station, it is possible for this cross-modulation to occur in the receiver itself, by overloading one of the stages. A wave trap will usually eliminate this kind of cross-talk.

► In other instances, this cross-modulation is occurring outside the receiver, for instance in a poor or corroded contact somewhere in the antenna system. Such a contact may conduct current better in one direc-

tion than in the other, forming a rectifier.

To check for this, disconnect the antenna entirely and move its lead-in well away from the receiver. Try a short indoor antenna to see if you can pick up the desired station without cross-modulation. If so, go over the entire antenna installation carefully. Be particularly suspicious of window lead-in strips which use bronze or steel clips for connecting to the copper lead-in wire. This junction of dissimilar metals is a frequent cause of trouble. Eliminate the strip entirely as a test for this condition, by connecting the two ends of the lead-in wire together directly.

You may find that the plumbing or heating system of the house will pick up radio energy and will rectify this energy due to poor joints in the system. This energy will then be radiated from the pipes and will be picked up by the receiver. One clue to this condition will be the fact that moving the radio to your shop will apparently clear up the trouble, while it recurs as soon as the receiver is brought back to its original location. Try moving the receiver to another part of the room, as this is often an effective way of clearing up the trouble.

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## Preliminary Considerations for Superheterodyne Alignment

A receiver may be out of alignment because of aging of parts, from the effect of heating and cooling, which causes expansions and contractions, or you may have upset the alignment in your search for a tuning circuit defect. Regardless of the reason, alignment must be properly performed to restore a radio to the maximum pep

and selectivity of which it is capable.

We have given you so far in this lesson the methods the advanced serviceman uses to recognize and locate tuning circuit defects, and to uncover a case of misalignment. You are now going to study alignment itself—how to prepare the receiver for it, and then exactly how to do it.

## OSCILLATOR TYPES

All sections of a receiver determine to some extent the location of a station frequency on the station dial, but the oscillator has far more influence than any other circuit. Each receiver is designed so that when the oscillator is properly adjusted, the stations are received at the proper points on the dial scale, and all the tuning circuits work effectively together.

Dial tracking depends on the design of the oscillator circuit, as you learned in other lessons. Here is a brief review of the basic circuits.

**Type 1.** The most used tracking method uses a padding condenser in series with the oscillator tuning condenser. The padding condenser reduces the tuning capacity in the oscillator circuit so that it will tune to higher frequencies than will the preselector.

This type of oscillator circuit requires two adjustments, one at each end of the band, to get proper tracking over the entire band. A trimmer condenser, in parallel with the main tuning condenser, is adjusted at the high-frequency end of the band. Then the padding condenser is adjusted at the low-frequency end of the band.

**Type 2.** This circuit makes use of a tuning condenser having specially cut rotor plates for the oscillator. These plates are cut so that they are smaller physically than are the other tuning condenser sections, so there is less capacity in the oscillator tuning condenser. It will therefore be automatically tuned to higher frequencies than the other sections.

A trimmer condenser is used for high-frequency adjustment. However, tracking over the entire frequency range is now dependent upon the proper design of the tuning condenser, the oscillator and preselector coils, and the choice of the proper inter-

mediate frequency. There is no padding condenser with this type of oscillator.

**Type 3.** The third type is used in short-wave bands in many all-wave receivers. A carefully selected fixed condenser is used as the padder for these particular short-wave bands. The only adjustment which is made is at the high-frequency end of the band. The tracking at the low end of such a band is then dependent upon the accuracy with which the padding condenser was chosen, its ability to maintain its originally chosen value, and the use of the proper i.f.

**Type 4.** Sometimes the oscillator coil will have a movable powdered-iron core so that its inductance can be changed. If variable tuning condensers are used, the coil inductance is adjusted for proper tracking at the low-frequency end of the dial. This type may be found with specially cut oscillator tuning condensers or with fixed padders. In either case, this moving core is the low-frequency adjustment. (This is one case where there will be a low-frequency adjustment even with specially cut oscillator plates.)

**Type 5.** Many modern receivers do not use tuning condensers. Instead, fixed condensers are used, while the preselector and oscillator coil inductances are varied by the tuning mechanism. In these cases, the coil inductance is adjusted at the high-frequency end of the band. Adjustments are made by moving the coil with respect to the core, or the cores can be individually adjusted with respect to the tuning mechanism. In a few cases, high-frequency trimmers may be found. If so, these are usually adjusted at the extreme high-frequency end of the band, then the coils are adjusted at a somewhat lower frequency (still near the high-frequency end,

however). No low-frequency adjustment is usually provided on these sets.

## IDENTIFYING TRIMMERS

If at all possible, obtain and follow the instructions of the manufacturer of the receiver you are to align. This is particularly true when you are just beginning your service career, or the receiver is a complicated type.

After you have gained experience, you will know the basic alignment procedure to follow, but you will still find the manufacturer's instructions very helpful. In particular, you should try to get information giving the position of the trimmers for the various adjustments and wave bands. In addition, the manufacturer's instructions usually give the intermediate frequency and the order in which the adjustments should be carried out.

If the manufacturer's instructions are not available, it will be necessary to identify the trimmers on the receiver so the alignment can be properly made.

**Single-Band Receivers.** In a single-band superheterodyne of the standard broadcast type, the i.f. trimmers will be inside the i.f. shield cans in most instances, or will be mounted very close to the i.f. coils. If the cores of the i.f. coils are varied, the adjustments will be right on the shield can. If you don't see the adjustments on the top or side of the i.f. shields, they may be underneath or at the rear of the chassis.

Standard practice puts the high-frequency trimmers right on the tuning condenser gang in a single-band receiver. It is rather rare to find them elsewhere.

An examination of the tuning condenser gang will show whether you should expect a padding condenser. If all of the rotor sections are the same size, expect a padding condenser. This

may be most anywhere on the receiver chassis. It may even be necessary to trace to the padding condenser from the oscillator.

On the other hand, if the oscillator section of the tuning condenser gang has special cut plates, you have no reason to expect the presence of a padder, except where variable coil cores are used.

If you see no tuning condensers, expect variable coil cores, plus other trimmers in some instances.

► In checking the trimmer condensers, you are very likely to find that



Alignment requires a signal generator and an output meter. Separate instruments, as shown here, or a combination multimeter-signal generator can be used. The signal generator furnishes a signal of known frequency and controllable output. The adjustments are made for maximum indication on the output meter. A simple procedure, yet it is vital to the radio performance that the proper steps be followed.

you have an extra trimmer or so. Consider such extras carefully. They may be wave trap adjustments, regeneration controls or, on older receivers, an i.f. adjustment for the individual transformer feeding a separate a.v.c. tube. The manufacturer's instructions should be followed in such cases, if possible.

**All-Wave Receivers.** In all-wave

receivers, the i.f. amplifiers are just like those used in a single-band receiver. When we come to the problem of identifying the preselector and oscillator trimmers, however, we find an increased number of trimmers. In some instances, there may be separate high-frequency trimmers and separate padders for every wave band. In others, there may be a complete set of trimmers for certain bands and only some of the trimmers on other bands. More information will be given on this when you come to the instructions for aligning all-wave receivers.

## EQUIPMENT

In order to align a receiver properly, a well calibrated signal generator is a definite requirement. It is possible to align by using broadcast stations and a signal tracer, but the use of a signal generator is much the preferred method.

You will require, in addition, some kind of output meter. Alignment tools will also aid in making the alignment.

**Signal Generator.** The s.g. (signal generator) used for alignment purposes should preferably be an all-wave type, with accurate calibration, and an attenuator or volume control to vary the output from maximum all the way to practically zero output.

*Signal Generator Connections.* The signal generator should have a condenser in series with the connecting leads, to prevent the d.c. path through the attenuator from short-circuiting plate supply voltages or grid bias voltages. If this condenser is not already included in the s.g., one should be added. The condenser can be most any size when aligning i.f. stages, as it just acts as a blocking condenser.

When a signal generator is connected to the input of a receiver, it replaces the regular antenna, which

removes the effect of the regular antenna inductance and capacity and affects the alignment of the preselector circuit. The effect of the antenna can be restored by using a "dummy antenna" in series with the signal generator hot lead. Recommended values are a 200-mmf. condenser, a 20-microhenry coil and a 25-ohm resistor in series for broadcast band alignment. However, a condenser of about 200 mmf. can be used alone and the alignment of the receiver will be reasonably correct for the average antenna. For the short-wave bands, a 400-ohm resistor is usually considered a good dummy antenna and is placed in series with the hot signal generator lead.

► Receiver manufacturers will frequently advise the use of rather elaborate coupling devices and dummy antennas. In most instances these are not an absolute requirement. When the alignment of a receiver has been properly carried out, the preselector stage is the only stage which will be at all out of alignment if you don't use these devices. Most servicemen make a practice of touching up the preselector trimmers when the receiver is installed to compensate for antenna effects, in cases where distance reception is an important requirement. This has the advantage of correcting for actual conditions.

► Another decoupling device which is occasionally used is the one shown in Fig. 4. In many instances, a signal generator is connected between the grid circuit of a tube and ground. Unless a condenser is used in series with the signal generator, the d.c. path through the attenuator will short-circuit the a.v.c. voltage supply. Also, the resonant circuit may be tuned to some frequency which will cause it to act as a partial short circuit on the output of a signal generator, thus

reducing the amount of energy which can be fed into the circuit. In such cases, the signal generator can be coupled to the grid top cap through a blocking condenser, and a resistor can be placed in series with the grid circuit. The resistor has a value between 10,000 and 50,000 ohms and is in series with the tuned circuit so far as the signal generator is concerned, so no matter how the tuned circuit may be set, the signal generator is feeding into an impedance at least equal to the value of the resistor.

In the case of single-ended tubes, which have their grid terminals on the bottom, it is not easy to add the series resistance. In such cases, the signal generator is connected to the grid terminal through a condenser, and the tuned circuit is set near the low-fre-

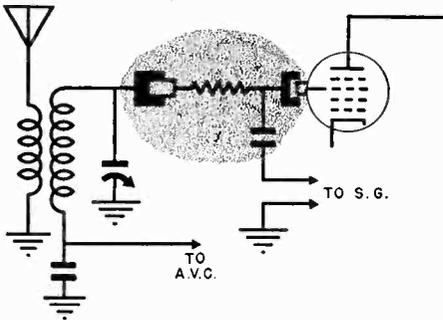


FIG. 4. An R-C coupler for the s.g.

quency end of the band, where its effect is minimized.

**Output Meter.** The most common output meter is an ordinary a.c. voltmeter using a copper-oxide rectifier, such as would be found in most any service multimeter. Using a voltage range of from 15 to 50 volts, this meter is usually connected between the plate of the output tube, point *e* in Fig. 5, and chassis. (To prevent short-circuiting the plate supply, a 600-volt condenser rated at about .5 mfd. must be used in series with this voltmeter, if one is not already built into the analyzer.)

The reason for using this position is that the a.c. voltage is reasonably high, and this is a point which is easy to find, as any tube chart will give the location of this socket terminal, which can also be found by tracing back from the output transformer.

Of course, if one has a lower-range a.c. voltmeter, the connection can be between *d* and ground or even between *b* and ground. In each case it is necessary to have a series blocking condenser to prevent d.c. current from flowing through the a.c. rectifier.

Many servicemen make use of an a.c. voltage range of from 1.5 to 3 volts, by connecting across the voice coil of the loudspeaker. This is between terminal *f* and ground of Fig. 5.

All of these points will give equally good indications, using the output of a modulated signal generator. Usually, standard signal generators use a modulating frequency of about 400 cycles, which is indicated adequately with any standard a.c. voltmeter. It is important that you realize in all cases you are looking for an alignment adjustment resulting in *maximum* response rather than any particular *amount* of response. You don't care whether the output is 10 volts or 12 volts or 25 volts, just so you have obtained the most there is by adjusting any particular trimmer.

Should the receiver have a tuning indicator either of the meter or tuning-eye type, you can use it to indicate maximum response, because it works from the automatic volume control circuit of the receiver. This is not quite as accurate as using an output meter, however, as the a.v.c. circuit may level off.

A high-sensitivity d.c. voltmeter or a d.c. vacuum tube voltmeter can be used by measuring the a.v.c. voltage. If the instrument does not indicate on a.c., it may be connected between

point *b* and ground of Fig. 5. If audio signals might interfere, the signal generator can be used without modulation, or the connection can be between point *a* in the a.v.c. circuit and ground.

Point *c* is an undesirable position due to the presence of some r.f. voltage. It is also a more difficult point to locate, as  $R_2$  may actually be inside the i.f. transformer shield, while point *b* is a terminal on the volume control and is easy to locate.

**Alignment Tools.** The usual trimmer condenser adjustment is an ordinary screw or, less often, a hexagonal nut. In a few instances, two trimmers will be combined, so there is a screw in the center of a nut, both being adjustable. Primary and secondary trimmers of a single i.f. transformer might be thus combined.

► Although we normally think of an adjustment as varying a trimmer, in many instances the inductance of the coil will be changed instead. This doesn't matter, as the same basic alignment procedure is still followed.

The same screwdriver or socket wrench adjustment is made on a variable core, except on the type where the core is fastened on a plunger which is held by a friction locking nut. Here, the locking nut is released and the plunger is then pushed in or pulled out. A special hook-shaped tool is recommended for this adjustment.

► When the adjusting device is grounded, any ordinary screwdriver or socket wrench of a suitable size can be used for alignment. However, if the adjustment is not grounded, an ordinary metal screwdriver or socket wrench and the operator's hand will add capacity to the circuit. When the tool is removed, the circuit is no longer in alignment. You can over-adjust in an attempt to have the maximum occur when the tool is removed,

but this is a guess-and-try method and is not recommended. Instead, servicemen use a tool made of a rod of bakelite or other insulating material. At the end there is a very small metal screwdriver blade or a suitable socket for adjusting purposes. As the body of the rod is an insulating material and there is only a very small amount of metal in the device, body capacity effects are practically eliminated.

## CHECKING AND CORRECTING DIAL CALIBRATION

Since proper dial tracking is one of the important indications of the correct alignment procedure, it is necessary that the dial itself be in the proper relationship with respect to the tuning condenser gang.

There are many kinds of receiver dials, ranging all the way from those used on inexpensive midgets, where calibration marks may be only every 50 or 100 kc., to those on expensive communication receivers where the calibrations may be closer than 10 kc. The closer the calibration marks, the closer the alignment should bring the stations to the proper points.

However, before we can align a receiver properly, we must be sure that the dial itself has not slipped from its proper position. An attempt at alignment with respect to an erroneous dial position will result in misalignment of the receiver.

In some dial mechanisms, the dial pointer is fastened mechanically right to the shaft of the tuning condenser gang, so that there is usually not much trouble. In most cases, however, the dial pointer is operated by gears or by a dial cord employing friction at some point in the mechanism, so there is considerable chance of slippage.

► If the manufacturer's instructions are available, you will usually find

information on checking the dial position. In most instances, the pointer should sweep over the entire calibration range of the dial face, with a certain amount of overlap at each end. There is often a calibration mark either at the low-frequency or high-frequency end of the band. If there is such a mark, you are supposed to set the dial pointer at that point, with the tuning condenser gang either completely open or completely closed, as the case may be.

In the case of dials having pointers which sweep over a range of one-half circle or  $180^\circ$ , the pointer is usually

else misaligning the receiver, or if the receiver is out of alignment anyway, this may not be true.

### SETTING RECEIVER CONTROLS

In setting the controls before alignment, follow the manufacturer's instructions or the simple rules that follow.

While the i.f. section is being aligned, it is very desirable to set the receiver tuning dial at some point where no station is tuned in. Usually, the manufacturer's instructions will have you set the dial at the low-frequency end of the band, as there are

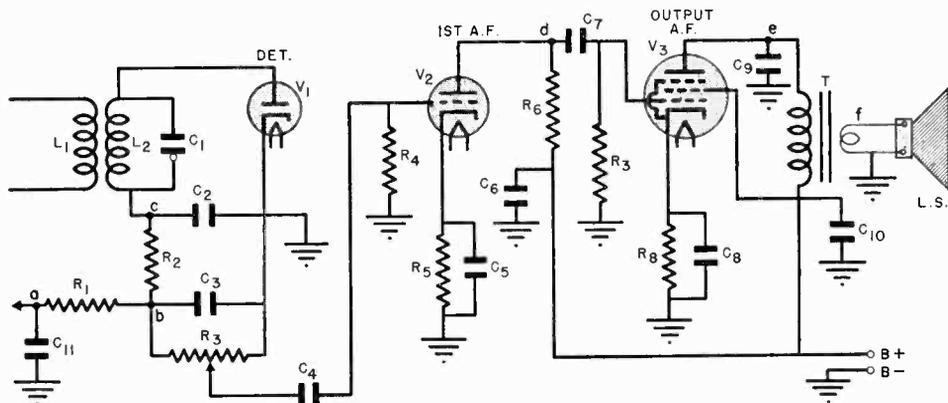


FIG. 5. Typical output meter connecting points are shown at a, b, c, d, e and f.

intended to be horizontal at each end of the tuning range.

► Should no instructions be available, check the dial sweep. For example, if you find the dial pointer right on the end of the markings at one end but with an over-sweep of  $\frac{3}{8}$  inch or more at the other end of the dial, the pointer should be moved so that the amount of overlap at both ends is about the same.

If the dial slippage is not complicated by misalignment, usually all the stations will be off in the same direction, being 10 or 20 kc. high or low, as the case may be. However, if the dial slippage has resulted in someone

fewer stations picked up at this end under normal conditions. Another reason for this will be given when you take up alignment procedures.

The high-frequency adjustment of a receiver is usually given in terms of some arbitrary setting such as 1400 kc., 1500 kc., or some other value. The manufacturer is merely indicating a value which *can* be used in a case of this kind, and if you have a local or interfering station on the recommended frequency, you will naturally shift a little to avoid such interference. If 1400 kc. is recommended but has an interfering station on it, you

will go, for instance, to 1450 kc. or some nearby value for alignment.

The same thing is true of the low-frequency end. Usually, 600 kc. is recommended. Again, it may be that you will find some local station interferes. If so, use 580 kc. or 620 kc., or any nearby value which is free from interference.

Naturally, during the alignment procedure an external antenna is not normally connected to the receiver. However, it may still pick up strong local stations, particularly if the set uses a built-in loop antenna which cannot be disconnected.

► The volume control on the receiver may be set at any point which will permit the output indicator to function in a normal manner. It may be desirable to turn down the volume control to prevent the 400-cycle note from being too loud, if the a.v.c. voltage is being used to indicate signal strength.

But if the output meter is an a.c. voltmeter in the audio amplifier, the

volume control setting must be high enough to give a useful reading. Common practice is to set the volume control on the receiver at maximum and control the signal with the signal generator attenuator.

► The setting of the tone control does not matter, as long as it has no effect on the selectivity of the set. In some instances the tone control will change the selectivity by providing band-spread in the i.f. amplifier. In such cases, it should be set as described when we discuss this type of alignment.

► Of course, the wave-band switch must be set for the band you are aligning. When aligning the intermediate frequency amplifier, the wave-band switch is normally set at the broadcast band position.

► Should the set have automatic frequency control, this control should be turned off during the regular alignment procedure. The manufacturer's instructions should be consulted if other special controls are found.

---

## Methods of Alignment

We are now all ready to align, with a signal generator and output indicator at hand and with the receiver thoroughly warmed by being turned on for a half hour. This warming up period is necessary because heat may change the values of trimmers, etc., slightly, and the set should be lined up in its operating condition. We can now go through the actual alignment for several different conditions and types of receivers.

### BASIC ALIGNMENT

By far the most common service condition will be the one where alignment is only slightly off. The set

sensitivity and selectivity may be below normal, and stations may have shifted 10 or 20 kc. on the dial. However, the set is not completely out of alignment, nor has it been tampered with.

Presuming that you have carried out all of the preliminary steps, you can proceed as follows:

**Step 1.** Suppose the receiver is similar to the one shown in Fig. 6. Here we have a single-band, conventional first-detector-oscillator tube stage, followed by an intermediate frequency stage, which in turn feeds into the second detector.

The signal generator is connected

between the control grid of the converter tube (point *b* in Fig. 6) and chassis to align the i.f. section. The receiver is normally tuned to the low-frequency end of the broadcast band, at some point where no local station interference will exist. Furthermore, by tuning the receiver to the low-frequency end of the dial, the tuned circuit  $L_1-C_1$  will not act quite so much as a short circuit to the output from the signal generator.

As mentioned previously, the signal generator should have a condenser in series with it to avoid short-circuiting the a.v.c.

Should there be any trouble about making the connection to the control grid of the converter tube, many servicemen connect the oscillator to the antenna and ground terminals of the receiver (point *a* and chassis). If the signal generator output is high enough, it will force its way through the preselector tuned circuit with sufficient energy to permit proper alignment of the i.f. amplifier.

► With the signal generator connected, it should then be set at the i.f. value for the receiver. This can be determined from the manufacturer's instructions. If these instructions are not available, it is usually rather easy to determine the proper i.f. value. Start with the signal generator near 500 kc., and reduce its frequency slowly but steadily. The *first* signal which comes through the set when proceeding from 500 kc. downward, is at or near the correct i.f. value. In the past, standard intermediate frequencies of 175, 260, 370, 455, 460, 465 and 480 kc. have been used. If the value indicated by your signal generator is near one of these frequencies, you can shift to this standard frequency and proceed with the alignment. Should there be any doubt about the signal, tune the receiver

over a 20 or 30-kc. range. If the signal varies or disappears, a signal generator harmonic is coming through the regular preselector channel. If the signal remains about the same strength, you have the signal coming through at the i.f. value properly.

After the s.g. is set at the right frequency and the signal is coming through to the indicator, adjust condensers  $C_6$ ,  $C_7$ ,  $C_8$  and  $C_9$  to give maximum output. It is usually desirable to go back over the trimmers, because there is a certain amount of interaction between primary and secondary trimmers. You can start with any of the four condensers and then go to any of the others. Usually one transformer pair is aligned before going on to another transformer, however.

► In the i.f. amplifier stage shown in Fig. 6, tuned primary and secondary (double-resonant) i.f. transformers are used. These are extensively used in supers, yet other types will be encountered. Transformers having only the primary or only the secondary tuned are often found in the less expensive four-, five- and six-tube supers. No difficulty should be experienced when you run across these units, as the single adjustment per transformer is made for maximum output, as with other i.f. transformers.

If variable iron cores are used instead of trimmers, the adjustment is also for maximum output.

Remember, we are watching the tuning indicator or the output meter at all times to determine just when the maximum is obtained. If you turn a trimmer beyond the point of maximum response, the output will drop again, so you must turn back to the maximum point.

**Step 2.** Having adjusted the i.f. amplifier to the proper frequency, we now move the signal generator to the

antenna and ground terminals of the receiver (if not already there).

Should the receiver input be a loop antenna, either use a special loop coupler or make one by using a short piece of wire and making a two- or three-turn loop with it. This wire loop is then connected to the hot signal generator lead and is brought near the receiver loop antenna.

The receiver dial is now set at the highest frequency marked on the dial, if it extends to 1600 or 1700 kc., and the generator is set to the same frequency. The oscillator high-frequency trimmer ( $C_4$  in Fig. 6) is then adjusted for maximum response.

The dial and the generator are now set at some frequency between 1400 and 1500 kc. and the preselector trimmers are adjusted for maximum output, thus insuring better preselector tracking at medium frequencies. Should the maximum tuning range extend only to 1500 kc., the oscillator and preselector can be adjusted at the same frequency setting.

**Step 3.** The next adjustment is made at the low-frequency end of the dial, when the receiver has a padder. The dial is then set somewhere near 600 kc. and the signal generator is set to the same frequency. Then, padding condenser  $C_5$  is adjusted. If the dial calibration at the low-frequency end of the band is not far off, this adjustment is simply made for maximum output.

*Rocking.* Should the dial calibration be off somewhat at the low-frequency end of the band, it may then be necessary to "rock" the low-frequency adjustment. To do this, set the *signal generator* at 600 kc., or some similar frequency where no local stations come in. Then tune the receiver for maximum response, disregarding the tuning dial setting altogether. You may find that the 600-kc.

signal from your signal generator comes in at 610 kc.

Now turn the tuning dial slightly in either direction from the maximum response position and readjust the padding condenser  $C_5$ . Notice whether these two adjustments give a higher output than the first position. If the output decreases, turn the receiver dial in the other direction. Pay no attention to whether or not you are approaching the correct point on the dial during this first adjustment. All you are trying to do is move the dial and the padding condenser to find the point where the actual maximum in response is obtained.

For example, you may find that at 610 kc. the output is 20 volts. By changing to 605 kc. and readjusting the padder, the output may become 22 volts. As you go closer to 600 kc., the output may drop again. Go back to the point which gave you maximum response.

This procedure is known as "rocking," because you rock the dial back and forth until you find the actual resonant point of the preselector, while you are adjusting the padder to get the maximum output indication.

**Step 4.** Now return to 1400 kc. and readjust  $C_4$ , as there is an interlocking action between the low-frequency and high-frequency adjustments of the oscillator.

**Step 5.** Step 4 causes the padder adjustment to be off somewhat, so repeat Steps 3 and 4 until the signal generator frequencies are picked up at the correct points at both ends of the band. The final adjustment is made at the high-frequency end.

## PROCEDURE FOR SETS FAR OUT OF ALIGNMENT

The alignment procedure described next is like the basic alignment procedure, except that methods are added

to handle a set that has been tampered with, so that it is far out of line.

**I.F. Alignment.** We must first decide what the correct i.f. value is for the particular receiver. If the manufacturer's instructions are not available, look for a service manual listing intermediate frequencies. Failing this, consider the type of receiver and its approximate age. Most of the recent receivers have intermediate frequencies between 455 and 480 kc. As a starter, therefore, you can use one of these frequencies on any standard receiver made in the last five years or so. On older sets, as your first trial, find and use *some frequency which allows adjustment of the i.f. trimmers near the center of their ranges*. Trimmer condensers should not be wide open, nor should they be completely tight. One-quarter to one turn back from the tight position is usually the correct range.

When the i.f. amplifier is far out of alignment, it will usually be necessary to feed the signal generator output through fewer tuned circuits.

Therefore, you will have to move the signal generator along as you align, starting near the second detector and moving back toward the input. This is particularly necessary when two i.f. stages are used.

In a circuit such as shown in Fig. 6, we should start the alignment with the signal generator connected to point *d*.

If the adjustment is too far away from the correct i.f. value, no signal may go through transformer  $T_3$  until you find for what frequency it happens to be adjusted. You should vary your signal generator over a wide range, until a signal comes through the transformer, permitting the adjustment of trimmer condensers  $C_8$  and  $C_9$  for maximum response. If you are far away from the correct i.f. value, you can now move the signal generator toward the proper value until you can just barely hear the signal. Then readjust  $C_8$  and  $C_9$  for maximum output at the new setting. Repeat this procedure until you have brought this transformer up to the right i.f. value.

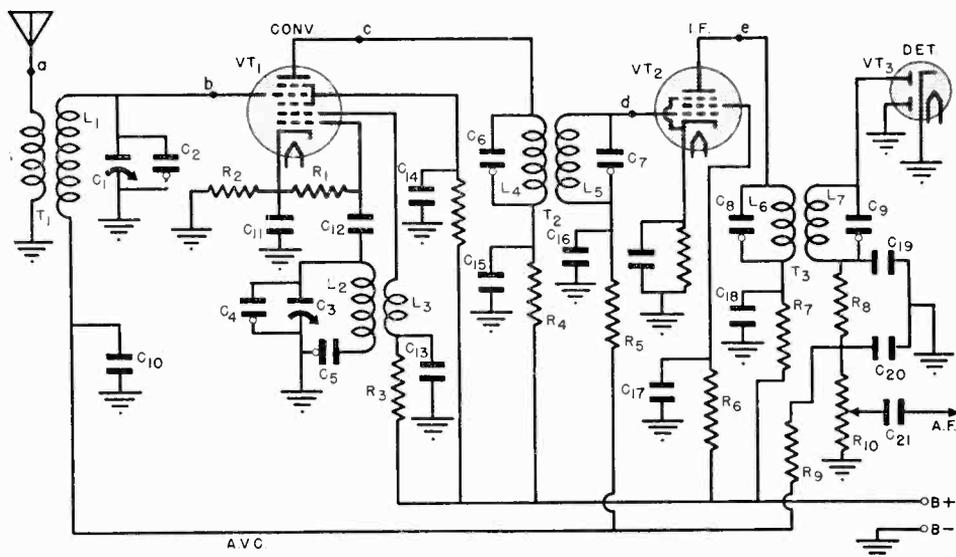


FIG. 6. This single-band superheterodyne diagram shows the connecting points for the s.g.

► After getting  $C_8$  and  $C_9$  to the correct i.f. value, move the generator back to point  $b$  to adjust  $C_6$  and  $C_7$ . With the i.f. tube amplifying normally and with transformer  $T_3$  tuned to the right frequency, it is usually possible to force the correct frequency through transformer  $T_2$ . Condensers  $C_6$  and  $C_7$  are then adjusted for maximum output.

► For the final adjustment, leave the signal generator at point  $b$  and

that stage in order to force a signal through the converter tuned circuit first, before moving back to the antenna for final preselector alignment.

With the preselector in reasonable alignment, you should examine carefully the calibration of the dial scale. If the receiver makes use of a padding condenser, the foregoing adjustments will probably give a reasonable tracking. On the other hand, should the oscillator be one of the cut-plate type,

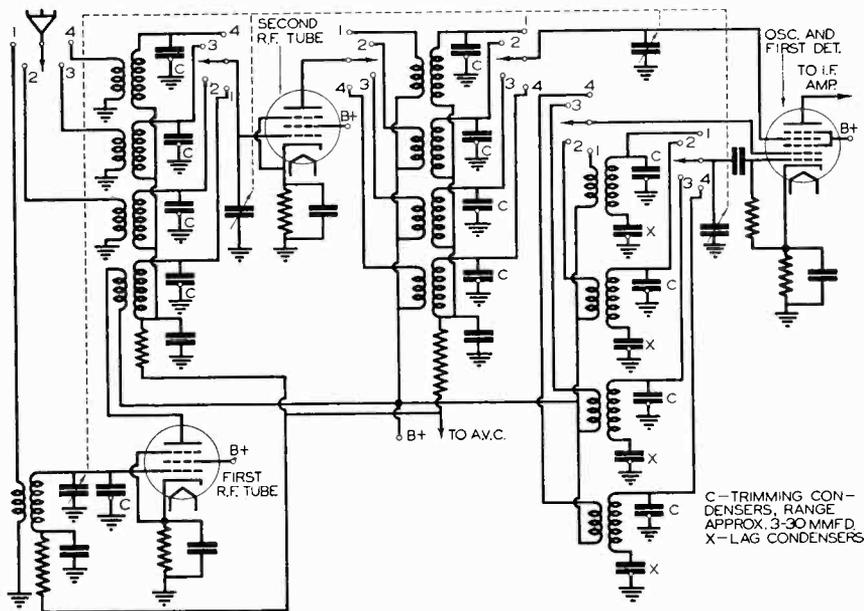


FIG. 7. An all-wave receiver with independent bands.

readjust all four trimmers for maximum output, following the simple basic procedure.

**Preselector and Oscillator.** If there is no r.f. stage in the preselector, we can now move the signal generator to the antenna and ground terminals and proceed with the oscillator and preselector alignment in the manner described before. It will definitely be necessary to rock the dial at 600 kc. if a padding condenser is provided.

If there is a radio frequency stage, we may have to go to the plate of

the stations at the low-frequency end of the band may be considerably off from the proper dial position, indicating an incorrect i.f. value. If the station comes in at a *higher* frequency setting, use a *lower* i.f. value, and vice versa. Shift the i.f. 10 kc. or so as trials, and then realign the receiver to see how this affects the dial calibration.

## FURTHER ADJUSTMENTS

Although we have adjusted the high- and low-frequency ends of the

band, tracking in between will depend on the tolerance of the tuning condensers and coils. In most cases, the alignment will be entirely satisfactory. However, to insure exact tracking at more dial frequencies, many receivers have tuning condensers with segments or slots cut in the end rotor plates of each section. By bending these segments properly, the capacity can be changed to correct the alignment over the entire tuning range, where the maximum is to be obtained. You would set the station dial near the high-frequency end, so the first segment meshes with the stator—note dial frequency value—set the signal generator to this frequency, and bend the segment on each section either out or in to give maximum output. Use an insulated aligning tool or probe and, before making a permanent bend, move the segment first one way and then the other to determine which way gives greater output. Now move the rotor so the second rotor segment meshes with the stator (note the frequency value on the station dial), set the signal generator to this value, and bend the segment for maximum output. Repeat this operation until all segments are adjusted.

### ALL-WAVE RECEIVER ALIGNMENT

With an all-wave superheterodyne, the i.f. stages are always aligned first in the usual manner, then each r.f. range is considered independently. Often the band-switching circuits in an all-wave receiver will make it necessary to align the bands in a definite order, as set forth in connection with typical circuits discussed now.

**Independent Bands.** Fig. 7 shows a conventional, highly flexible preselector system for all-wave reception. One stage of r.f. amplification is used in all bands except the highest one, where an additional r.f. stage is em-

ployed. Each tuned circuit in the preselector has a high-frequency trimmer  $C$ . When a different range is switched in, another set of high-frequency trimmers is used. Hence, the adjustment of  $C$  for one band in no way affects another band.

The oscillator circuit for each range has its own high-frequency trimmer  $C$  and its low-frequency padder  $X$ . The tracking alignment may be carried out in any order desired, as the bands are completely independent. Follow the usual procedure for each band, adjusting at the high and low ends. If you decide to bend rotor segments, remember you may do this for only one range and that range must be aligned first.

You may find a receiver with this preselector system which has trimmer condensers on the gang of variable condensers. Check to see which range does not have the high-frequency trimmers on the coils and *align this band first*, using the trimmers on the ganged condensers. As a rule, they will be used for the highest-frequency range. If you use them to align the wrong band, you may add so much capacity into the circuit that the trimmers in some other band cannot be reduced sufficiently to make an alignment.

**Cascade Alignment.** A number of manufacturers wind all preselector coils on one winding form and all oscillator coils on another coil form. This reduces costs but introduces a problem in alignment that you should recognize. Such a construction is used in the circuit shown in Fig. 8. Oscillator coils  $L_4$ ,  $L_5$  and  $L_6$  are all wound on one coil form; preselector coils,  $L_1$ ,  $L_2$ ,  $L_3$  and broadcast primary coil  $L$  are also wound on a single coil form.

Switches  $SW_1$ ,  $SW_2$  and  $SW_3$  together make up the wave-band switch. In position 1, the receiver is set to the highest-frequency band. Observe that

coils  $L_3$  and  $L_2$  are shorted by switch  $SW_2$ , leaving coil  $L_1$  to be tuned by condenser  $C_P$ . The high-frequency preselector trimmer is  $C_1$ . In the oscillator section, switch  $SW_3$  (when placed at position 1) shorts coils  $L_6$  and  $L_5$ , leaving coil  $L_4$  to be tuned by condenser  $C_0$ . The high-frequency trimmer is  $C_4$ .

When all switches are set to position 2, only coils  $L_3$  and  $L_6$  are shorted. In this position, coil  $L_5$  is the major oscillator inductance, but coil  $L_4$  is still in the resonant circuit and is also

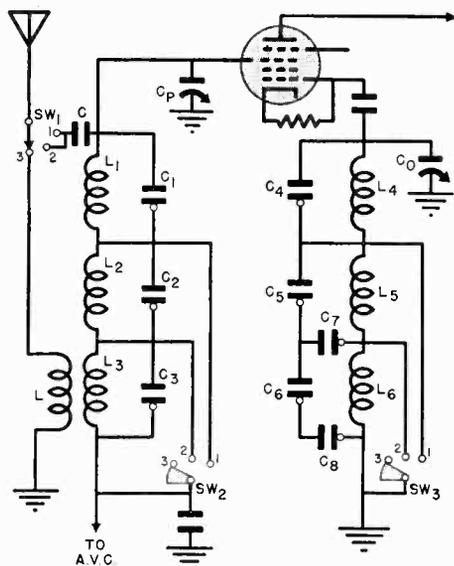


FIG. 8. The preselector-oscillator circuits of a receiver requiring cascade alignment.

included in any tuning performed by  $C_0$ . In this case, condenser  $C_4$  (the trimmer) has the effect of reducing the effective inductance of  $L_4$  for this band. Similarly, preselector coil  $L_2$  is the major inductance but  $L_1$ - $C_1$  are in the tuning circuit. So, if you adjust band 2 first, then band 1, adjusting  $C_4$  in the latter case will upset the previous alignment. We must realize, therefore, that if we align the high-frequency band first, all other trimmers are shorted and of no importance.

By adjusting bands 1, 2 and 3 in order, each adjustment made is included in the next one. That is why we call it cascade alignment.

To align this all-wave preselector, we set the receiver at a frequency near the high-frequency end of the highest-frequency band, set the signal generator to the same frequency, and adjust  $C_4$  and  $C_1$  for maximum output. There is no low-frequency oscillator padding condenser on this band.

Next, you align the second frequency band, in this case the  $L_2$ - $C_2$  and  $L_5$ - $C_5$  circuits. Under no condition should you touch trimmers previously adjusted. Set the receiver and signal generator to a high frequency on this band, then adjust  $C_5$  and  $C_2$  for maximum output. Set both the receiver and signal generator to a low frequency and make a rocking adjustment with  $C_7$  for maximum output. Check alignment at the high-frequency end, adjusting  $C_5$  if necessary.

The third or broadcast band is handled the same way, adjusting  $C_6$  and  $C_3$  at the high frequency and  $C_8$  at the low-frequency points.

Reasonable dial tracking is expected on band 1, without a padding condenser, due to careful manufacture and the proper choice of the oscillator coil value. The low-frequency padding  $C_7$  in the second highest band may be omitted in some receivers or may be a fixed condenser. If trimmers are mounted on the variable condenser, use them for the highest-frequency band.

In a cascaded preselector circuit, slotted rotor plates are not usually provided, as they can be adjusted only in the highest-frequency band. If you try to bend them in the broadcast band, you upset all previous adjustments, and realignment offsets the gains obtained in the broadcast band.

**Special Data.** Identifying trim-

mers on all-wave receivers is quite a problem. They may be mounted on strips in some definite order or they may be scattered throughout the set. If at all possible, obtain the manufacturer's instructions for such jobs. Otherwise, you will have to spend some time finding the proper adjustments. This problem is complicated by the fact that padding condensers, or even high-frequency trimmers, may not be used on all bands.

Identifying trimmers by tracing from the coils may be possible, if the coils are not in shields and can be traced to the band-switch or otherwise distinguished.

If this fails, turn the band-switch to the highest-frequency band, tune in a signal and then touch the trimmers one after the other with a metal screwdriver. You will hear noises or detune any signals being received when you touch the trimmers in that band. Repeat this procedure for the other bands to pick out the remaining trimmers.

Padding condensers will usually have more plates than the trimmers. Also, the trimmers themselves may be larger or have more plates for the lower-frequency bands.

The manufacturer's instructions should give you any unusual data, such as operating the oscillator *below* instead of above the incoming frequency, as is done in a few cases on the highest-frequency short-wave band. If you have no instructions, you should go ahead with the standard procedure and note results.

Be careful in adjusting the oscillator in the high-frequency (short-wave) bands, to avoid "cross-over." Because the i.f. value is so low, compared to the frequency of alignment (usually 18 mc. or so), it runs the usual two points of maximum response close together. Select the setting with the *least* trimmer capacity (plates opened widest) when aligning, to keep the oscillator *above* the incoming frequency, then check response at the low-frequency end of the band.

► You may hear a third signal between the two strong ones. This results where the incoming and local oscillator signals differ by one-half the i.f. value, producing a signal which the frequency converter may double. If you keep the level of the signal generator down, this signal can be readily detected by its low output.

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## Special Alignment Cases

### IRON-CORE COIL EXAMPLE

A number of receivers, such as the battery receiver of Fig. 9, use powdered iron adjustable cores in the coils to give alignment.

The i.f. is aligned in the usual way by connecting the signal generator between grid No. 3 of tube  $VT_1$  and chassis. With the signal generator at the i.f. of 455 kc., adjust cores  $A_6$ ,  $A_5$ ,  $A_4$ , and  $A_3$  to give peak output.

To make the preselector and local

oscillator track, connect the signal generator to the antenna and ground leads and set both at about 1400 kc., adjusting  $C_6$  first and then  $C_1$  for peak output. Note the two oscillator trimmers  $C_6$  and  $C_6$  which are on the variable condenser. Two were used by the manufacturer because the capacity of one was too small. This shows that you must expect design features that seem illogical. (The manufacturer's service manual tells us to set one trimmer one full turn off

the tight or maximum capacity position.)

Now shift the signal generator and receiver dial to about 600 kc., and adjust core  $A_2$  first and then core  $A_1$  for peak output. Repeat the high- and low-frequency adjustments for possible interaction. In a receiver that is considerably out of line, the core adjustment at the low-frequency end of the dial is recommended before the high-frequency adjustment. This is natural, as the variation in inductance of  $L_2$  and  $L_4$  has the greatest effect on dial tracking, particularly at the low frequencies.

Since the *preselector* has a low-frequency adjustment as well as the oscillator, the usual rocking adjustment is unnecessary. If this preselector or low-frequency adjustment is omitted, a rocking adjustment may be required.

### VARIABLE-SELECTIVITY I. F. TRANSFORMER ALIGNMENT

In a high-fidelity receiver, the i.f. stages are designed for band-pass action, usually by providing critical coupling between primary and sec-

ondary coils. Fixed band-pass action is not desirable in a receiver intended also to receive distant stations through powerful local or nearby broadcasters. Therefore, a control of selectivity from peak to near flat response is incorporated in such receivers. Several such schemes are shown in Fig. 10.

In another lesson, you will learn how band-pass alignment can be obtained by using visual methods—a cathode-ray oscilloscope technique—which allows you to see the resonant response curve of the r.f. system. Here, we will consider only those systems where a peak adjustment is made in the peak position of the system and band-passing is automatically obtained when set in the high-fidelity position. Therefore, when a variable coupling i.f. transformer is encountered, *set it for selective reception and align the section for peak maximum output.*

The theory of these transformers has been covered elsewhere in the Course, but let us briefly review their action:

Fig. 10A shows a double-tuned resonant transformer with variable cou-

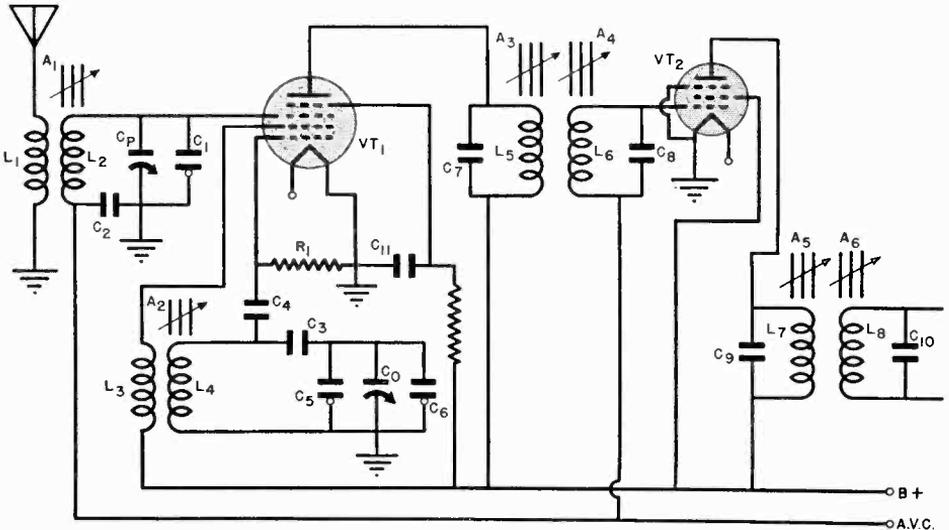


FIG. 9. Movable powdered-iron cores are used as the alignment adjusters here.

pling between primary coil  $L_1$  and secondary coil  $L_2$ . To make a peak alignment, set the control for minimum mutual inductance and adjust the trimmers for maximum output. Band-pass action is produced automatically as the control is adjusted for increased  $M$ .

When a critically-coupled double-resonant circuit is peaked, the response may be made broad by loading the circuits with resistance. A third winding may be used to introduce loss (equivalent to loading with resistor),

three resonant circuits,  $L_1-C_1$ ,  $L_2-C_2$ , and  $L_3-C_3$ . When  $C_3$  is properly adjusted, the third winding introduces reactance into the other two circuits, so as to provide double peak response. Increasing the resistance of  $R$  thereafter reduces this apparent detuning of input and output resonant circuits, and returns the system to peak response. Note that this i.f. transformer will have three trimmers mounted on it.

To adjust this circuit,  $R$  is set at maximum resistance. Trimmer condensers  $C_1$  and  $C_2$  are now adjusted

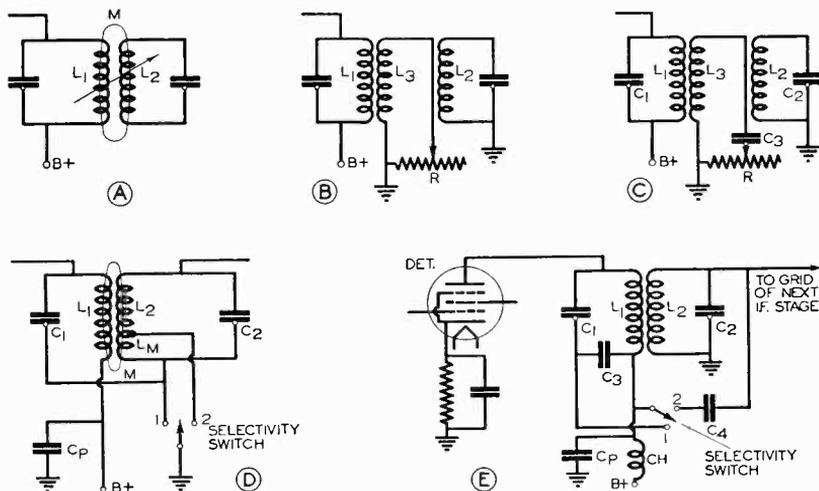


FIG. 10. Several examples of methods used to obtain variable selectivity.

as shown in Fig 10B. When  $R$  is zero, coil  $L_3$  is short-circuited and, through mutual coupling, acts as a load on both tuned circuits, giving broad response. When  $R$  is at its maximum resistance value, negligible loading exists and the circuits can operate with peak response. To align an i.f. stage with this i.f. transformer, set  $R$  for maximum value and adjust both trimmers for maximum output. Decreasing  $R$  thereafter produces band-pass action.

The circuit shown in Fig. 10C is a similar band-pass circuit. There are

for maximum output. Resetting  $R$  for minimum or zero resistance, adjust  $C_3$  for *minimum* output.

Some receivers employ a two-position fidelity control—a selective and a high-fidelity position. Two such circuits are shown in Figs. 10D and 10E. In both circuits, the selective reception is obtained in position 1, so use this position for peak alignment.

With the switch shown in Fig. 10D in position 2, coil turns  $L_M$  are added in the  $L_1-C_1$  circuit, lowering the resonant frequency of the primary circuit. The primary current flowing in

$L_M$  and closely coupled to  $L_2$  is out of phase with the secondary current and thus reduces the flux in  $L_2$ , reduces its inductance, and increases the resonant frequency of the secondary circuit. This produces double peak response automatically by resonating one circuit above and the other circuit below the i.f. peak value.

► In the circuit shown in Fig. 10E,  $C_3$  is shorted in position 1, while condenser  $C_4$  is inactive; the circuit is then peaked. With the switch set to position 2,  $C_3$  is in series with  $C_1$ , reducing the circuit capacity and increasing the resonant frequency. At the same time,  $C_4$  shunts  $C_2$  and thus reduces the resonant frequency of the secondary. By proper choice of  $C_3$  and  $C_4$ , this provides double peak response.

### ALIGNING F. M. RECEIVERS

It is standard practice to align the i.f. amplifier of an f.m. receiver before aligning the preselector and oscillator, just as in a.m. receivers. However, the i.f. alignment is somewhat unique in that the i.f. section up to the input of the limiter tube is aligned first, then the output of the limiter and the input of the discriminator are aligned as a separate step.

In practically all cases, f.m. receivers use tuned circuits that are loaded by resistors. Hence, they have a broad peaked response. For this reason, it is possible to align an f.m. receiver by using a standard signal generator as long as it covers the required frequency band—the same kind you would use on an a.m. receiver.

Since the adjustment of the trimmers is relatively broad, we must have some means of output indication. The limiter grid current is used for two reasons. First, the limiter grid current varies directly with the

signal strength, so it is an accurate output indicator. (The limiter holds the discriminator input constant, and there is no output from a properly aligned discriminator when an a.m. signed generator is used. Hence, an ordinary output meter is useless.) Secondly, since the limiter draws a grid current of 50 microamperes or more when stations are tuned in, it loads the preceding resonant circuits. We must measure the limiter grid current to be sure it is the normal amount so that proper alignment of its input circuit can be insured.

A recommended output indicator is a microammeter having a range of about 200 microamperes. If one is used, it should be inserted in the grid return circuit at the point marked  $X$  in Fig. 11. It should be by-passed by a .05 mfd. condenser.

A vacuum tube voltmeter or 20,000-ohm-per-volt multimeter can also be used by reading the voltage drop across the resistor  $R$  caused by the grid current. The voltage should be at least .00005 (50 microamperes) times the resistance of  $R$ . The positive meter lead is connected to the chassis and the negative lead to the ungrounded end of resistor  $R$ . For alignment, the signal generator is connected between the control grid of the first detector tube and chassis. Here it may be necessary to use a condenser-resistor decoupler to prevent the first detector tuned circuit from reducing the signal generator output too much.

The signal generator is tuned to the i.f. resting frequency of the set; this is generally from 10.1 to 10.7 megacycles. (Older sets used 2.1 and 4.3 mc.) Adjust the i.f. trimmers for maximum meter reading. Reduce the signal generator output if the grid current exceeds the range of your output indicator. When you reach the

limiter input be sure that at least 50 microamperes are flowing before you adjust the trimmers corresponding to  $C_1$  and  $C_2$  of Fig. 11.

► Should the i.f. amplifier be so badly out of alignment that an output indication cannot be obtained, you can start with the signal generator hot probe connected to the control grid of the last i.f. tube (the one just ahead of the limiter), and can then work back a stage at a time, aligning each stage as you go. Finally, you can make an over-all i.f. alignment from the first detector to the input of the limiter.

ground, in Fig. 11, with the negative probe at point  $A$ . Then adjust condenser  $C_3$  for maximum meter reading. When this adjustment has been made, remove the meter probe from point  $A$  and connect it to point  $B$ , so the meter is between  $B$  and ground. Now, adjust condenser  $C_4$  for *minimum* reading on the output meter. Theoretically, the reading should decrease to zero, but if it does not, use the adjustment giving the lowest meter reading. It is advisable to reverse the meter connections as a check, because the voltage may have passed through zero and reversed polarity.

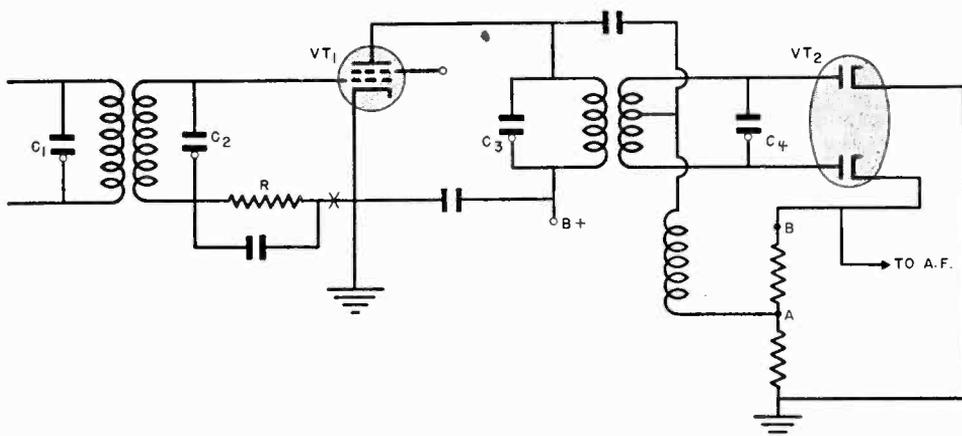


FIG. 11. The limiter and discriminator stages of a typical f.m. receiver.

When the i.f. amplifier has been aligned satisfactorily, the output indicator is removed from the limiter stage. If a microammeter was used in the grid circuit, close this circuit when the meter is removed.

► You are now ready to adjust the output of the limiter and the input of the discriminator. Leave the signal generator connections and its frequency setting just as they were for the limiter and the i.f. adjustments. You will need either a vacuum tube voltmeter or a high resistance multimeter as an output meter. Connect your meter between point  $A$  and

Since many vacuum tube voltmeters will not read reverse polarities, you may have passed the zero point without knowing it. A minimum at point  $B$ , regardless of meter polarity, completes the i.f., limiter and discriminator adjustments, and none of these trimmers should be touched again.

The preselector and oscillator adjustments are made in the same manner as those for the high-frequency bands of an all-wave receiver. The output meter is connected again as for limiter and i.f. adjustments, and all trimmers should be set for maximum readings.

## ALIGNING T. R. F. RECEIVERS

Tuned radio frequency circuits, used extensively years ago, are today found mainly in a few inexpensive receivers.

There are normally as many sections to the variable condenser gang as there are resonant circuits, and each section is provided with a trimmer condenser, usually mounted on the variable condenser gang. Their greatest influence is at the high-frequency end of the range and, assuming that the variable condensers have not warped or been forced out of alignment by a mechanical shock, a single-point alignment usually suffices. An a.c. voltmeter is usually used as an output meter, connected in the a.f. amplifier as for a superheterodyne.

Employ a signal of about 1400 kc., either from a signal generator or a local station. Set the receiver dial exactly to the frequency of the test signal. Then adjust all trimmers for maximum output. Go through the alignment at least twice. Check sensitivity at a low frequency to verify normal alignment at this end of the range. If sensitivity is low, you may then consider bending rotor segments for better alignment through the entire range.

## ALIGNING SUPERS HAVING REGENERATIVE SECOND DETECTORS

In some inexpensive superheterodynes the second detector is regenerative. The amplification and selectivity so gained makes it possible to omit the i.f. amplifier tube and still obtain fair results. In such a set, the output

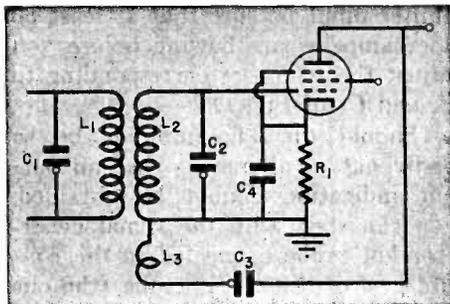


FIG. 12. Regeneration is used to increase the i.f. gain.

of the first detector is fed through the i.f. transformer into the second detector input. The second detector circuit is shown in Fig. 12, the remainder of the receiver being conventional in design. When aligning a set of this type, the regeneration condenser is adjusted for minimum capacity (least regeneration), and the i.f. trimmers for maximum signal output. The regenerative condenser is then adjusted until a squeal or howl is heard, after which the control is backed off just enough to stop the howl without reducing the output too much. Regeneration condensers are generally controlled by a fiber nut which may or may not be colored red.

Whenever the second detector tube is changed, readjustment of the regeneration control is necessary. Do not go through the entire alignment procedure; just throw the set into oscillation, back off the control until oscillation ceases and give the control an extra 1/4 turn in the minimum capacity direction.

# Lesson Questions

Be sure to number your Answer Sheet 39RH-1.

Place your Student Number on every Answer Sheet.

Send in your set of answers for this lesson immediately after you finish them, as instructed in the Study Schedule. This will give you the greatest possible benefit from our speedy personal grading service.

1. In a pentagrid-converter, is the oscillator grid resistor value *increased* or *decreased* to restore operation at the low-frequency end of the band?  
*decreased*
2. What produces the interference called "i.f. interference"?  
*station - beating with desired to cause a difference in f. =*
3. Name three defects which could cause reception of stations only at the high-frequency end of a tuning band.  
*low cut section, misalignment, oscillator detuning*
4. When the oscillator tuning condenser has specially-cut plates and the oscillator coil is an air-core type, would you expect to find a low-frequency adjuster in the oscillator circuit? *no*
5. Can an a.c. meter be used as an output meter between point *a* and chassis of Fig. 5? *no*
6. Suppose your a.c. meter is not accurately calibrated at 400 cycles. Could you use it for an output indicator?
7. Briefly describe the procedure known as "rocking."
8. How can you avoid getting the oscillator frequency below the incoming signal frequency when making a high-frequency adjustment?
9. Why is rocking unnecessary for the circuit shown in Fig. 9?
10. Briefly give TWO reasons why the limiter grid current should be used for the output indication in an f.m. receiver.

Be sure to fill out a Lesson Label and send it along with your answers.



## STICK—AND YOU'LL WIN!

“How can I be a success?” The simplest answer to that question is “finish whatever you start.”

Until success is attained, the individual tasks confronting you are relatively unimportant in themselves. The really important thing is their effect upon you, what you *learn* from them, the *practice in succeeding* that you acquire in accomplishing these tasks.

We can always find plenty of reasons for quitting, but what do we accomplish, what do we learn by quitting? We only learn how to quit—how to fail.

If success in any undertaking is dependent upon effort (and we all agree that it is), then the more effort we make, the greater our success. You may not be able to trace the success present in every effort, but it is there just the same. You don't attain success in one jump. You succeed by degrees, and the degree of success is in exact proportion to the degree of effort.

*J.E. Smith*

**USING THE CATHODE RAY  
OSCILLOSCOPE IN RADIO  
SERVICING**

40RH-2



**NATIONAL RADIO INSTITUTE**

**WASHINGTON, D. C.**

**ESTABLISHED 1914**

# STUDY SCHEDULE NO. 40

For each study step, read the assigned pages first at your usual speed, then reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind. Study each other step in this same way.

1. How the Oscilloscope Works . . . . . Pages 1-6

A review of the fundamentals of the c.r.o. is followed with more details on the construction and operation of this valuable tube. How the electron beam is obtained and a deflection obtained must be known, so the patterns formed can be clearly understood.

2. A Typical Commercial C.R.O. . . . . Pages 6-12

A standard c.r.o., designed for service work, is described in a step-by-step manner, so you will know just what the controls are for, and can operate any similar type you may encounter. The special amplifiers and their switching arrangements are covered. Here you find practical information and warnings, so you will not damage the tube. Peculiarities produced in the image by improper control settings and by stray voltages are portrayed so they can be minimized or avoided.

3. Putting the C.R.O. to Work . . . . . Pages 12-22

How to use the c.r.o. for determining frequency; signal tracing for hum, oscillations and distortion; types of distortion and the meaning of the patterns seen; determining phase. All these uses are covered in this section, which is packed with practical information. The every-day applications of the c.r.o. are set forth, so you can properly use this service tool, should you obtain one.

4. High Fidelity Alignment . . . . . Pages 23-36

The c.r.o. is an absolute requirement for speedy alignment of a high-fidelity receiver. By using a frequency-modulated signal generator, the c.r.o. will show the response curve of the r.f. and i.f. stages, so that the desired band-pass characteristic can be obtained. Further, the wave form shows exactly when resonance is reached, and the exact width of the pass band, permitting accurate adjustment.

5. Answer Lesson Questions and Mail Your Answers to N.R.I.

6. Start Studying the Next Lesson.

# USING THE CATHODE RAY OSCILLOSCOPE IN RADIO SERVICING

## How the Oscilloscope Works

THE cathode ray oscilloscope has become one of the most useful measuring instruments known to modern science. It not only can measure electricity, it actually makes the whole *character* of the wave visible at one time, so that every aspect of it can be studied. Even more—the time relationships between waves are instantly visible. It thus gives an actual picture of what is going on, not only in radio circuits but in every kind of moving or vibrating phenomenon that can be converted into equivalent electrical wave forms. Any variation, such as sound through a microphone; vibration through a pressure-actuated pick-up; light variations through a photoelectric cell; or heat fluctuations through a thermocouple, all become visible and give up the secrets of their nature on the screen of the oscilloscope.

► To the radio serviceman, the oscilloscope is not a basic instrument like the multimeter or signal generator, but is one of the “extras” that can be acquired, as finances permit, to make speedier, better service work possible. It can be used to compare voltages, to measure frequency, to show response curves of an amplifier, to show the presence of, and quickly find the source of distortion, hum or oscillation. It can be used as a signal tracer and is essential for proper alignment of high-fidelity equipment.

► This lesson will describe how to use the c.r.o. for these purposes, showing how the instrument can make service work more rapid, accurate and efficient.

### THE C.R.O. TUBE

First, let us review briefly the fundamentals of the c.r.o. itself, which have been covered elsewhere in your Course.

**The Electron Gun.** A cathode ray tube, like any other radio tube, has an electron-emitting cathode, control elements and an anode. However, the c.r.o. shoots its electrons in a thin stream at a fluorescent screen to produce light. Besides the cathode, which emits the electrons, there are three elements in the electron gun which are used to make the electrons move toward the fluorescent screen at high velocity and in a very narrow stream. These elements are shown in Fig. 1A and are the grid  $G$ , the first anode  $P_1$  and the second anode  $P_2$ .

The grid, as shown in Fig. 1B, is like a tin can open at one end, with a hole in the closed end. Electrons are pulled through the small hole in its closed end.

The first anode,  $P_1$ , is a cylinder with a “baffle” in the middle, having another small hole. It is made positive to attract the electrons and get them moving so rapidly they tend to move in a

straight line, and nearly all go through the small hole in the baffle.

The second anode,  $P_2$ , is more positive than the first anode. It thus gives

the element  $G$  controls the quantity of electrons in the stream, thus controlling the brilliancy of the spot of light on the screen.

**Obtaining a Deflection.** Now, with a stream of electrons striking the screen and producing a spot of light, we must deflect the stream up and down and to the right and left, to sweep the spot of light across the screen and get varied line patterns.

Two pairs of deflecting plates are used, the beam passing between the plates of each pair. (See Fig. 1B.) If one plate of a pair is made positive with respect to the other plate, the electron beam will be attracted toward this positive potential, since the beam is made of negatively charged particles.

► One plate of each pair, as shown in Fig. 3, is connected to anode  $P_2$  so that the plates will normally be at the same potential as  $P_2$  and thus not defocus the beam. Notice that it is the voltage difference *between* the plates which produces the deflection, not the fact that a connection is made to  $P_2$ .

even more speed to the electrons which are emerging from the hole in the baffle. In addition, its electrostatic field, in combination with that of  $P_1$ , produces curved lines of force which actually focus the moving electrons into a point at the screen, much in the way that a lens focuses light rays.

► Thus, as shown in Fig. 2, the electrons are emitted by the cathode, are controlled as to quantity by the adjustable negative potential on the grid, are accelerated and made to emerge from a small hole by the first anode, are further accelerated by the second anode, and are "focused" so that, after traveling the length of the tube, they come together in a small spot on the screen at the end of the tube. The potential on  $P_1$  is variable so that an exact focusing of the beam at the screen can be obtained.

► The screen is coated with fluorescent chemicals which will produce light when struck by electrons. The greater the number of electrons striking a given point, the brighter the light produced there. Varying the voltage on

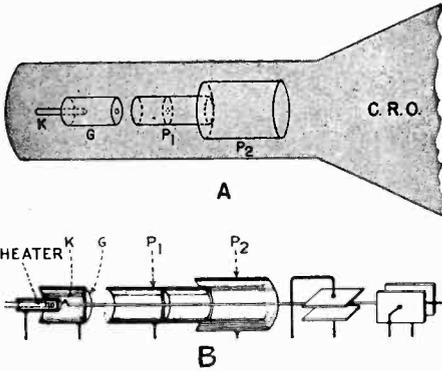


FIG. 1. The parts of the electron gun, arranged in the neck of the tube.

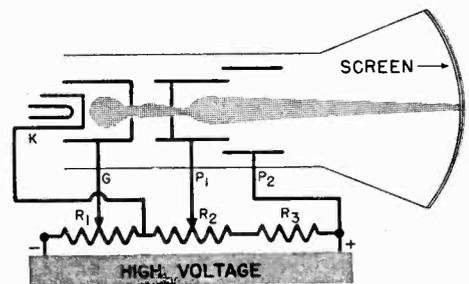


FIG. 2. How the electron beam is obtained.

Since this deflecting voltage will usually be found between chassis and some other point, it is convenient to ground the two plates connected to anode  $P_2$ . This means that the high positive potential of  $P_2$  is connected

to the chassis, which is different from the usual receiver practice, but does not affect the power supply operation.

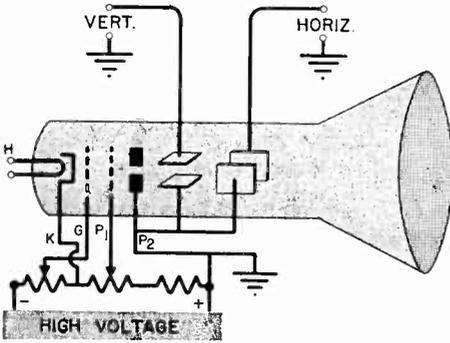


FIG. 3. The symbols used here are those commonly found on schematic diagrams representing the c.r.o. tube. Notice the connections to the deflecting plates.

► Fig. 4 shows how the position of the spot of light is affected for different deflecting voltages applied to the deflecting plates. At *A* you see a view of the tube as you would see it when looking right at the screen. (The deflecting plates are indicated on the drawing, although they cannot actually be seen from the front of the tube.) The spot here is in the middle between the vertical deflecting plates 1 and 2 which make up and down motions of the spot, and the horizontal deflecting plates 3 and 4 which cause side-to-side motion. (Notice the plates are named for the direction of deflection, not their physical position.)

A voltage on *V*, with none on *H*, making plate 1 positive, moves the spot up as shown in Fig. 4*B*. Reversed polarity sends the spot down as in Fig. 4*C*. An a.c. voltage applied to *V* will obviously keep the spot moving up and down in a line, as in Fig. 4*D*. If the spot moves faster than 16 times a second, the persistence of vision in the human eye makes us see a solid line instead of a moving spot.

Figs. 4*E*, 4*F* and 4*G* show what happens to the spot if the voltage is applied to the horizontal plates instead of the vertical plates. The solid line formed by an a.c. voltage is now from side to side rather than up and down. Since the length of this line is proportional to the peak of the applied voltage, the voltage can be measured by comparison with a known voltage, or by actual calibration in terms of the length of the line.

► The great value of the c.r.o. does not lie in measuring voltages, however, but in making visible the actual form of an a.c. wave. You might reason that we could apply the a.c. voltage to both sets of plates at the same time. This makes one plate of each pair positive at the same instant, as shown in Fig. 5*A*. The electron beam is pulled to the left and upward at the same time, so it moves along the diagonal to the

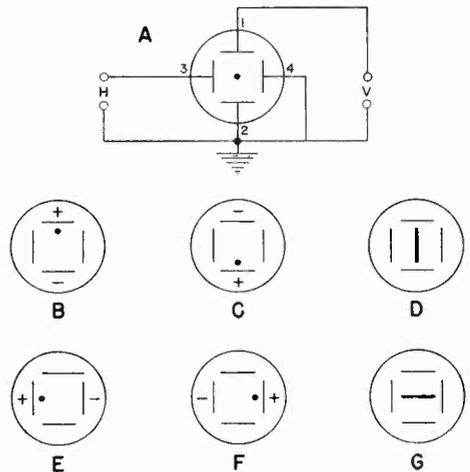


FIG. 4. The result of applying a voltage to just one set of deflecting plates.

upper left. When the polarities reverse, we have the condition shown in Fig. 5*B*. Thus, we have formed a diagonal line as shown in Fig. 5*C* instead

of getting a picture of the wave. This same line will be formed regardless of the wave shape, as long as the *same* voltage is applied to both sets of plates directly.

► Let us see what can be done to make the wave shape visible. Since we are trying to view the wave form, this gives a clue as to what is necessary.

When we draw a curve showing the form of a wave, we are actually showing how the variations (voltage, in this case) are increasing and decreasing *with respect to time*. That is, the line of our curve moves steadily from left to right, to indicate the passage of time, and *at the same time* moves up and down, to show how the voltage rises and falls *during that time*. The resultant of these two motions is the curve which traces the form of the wave.

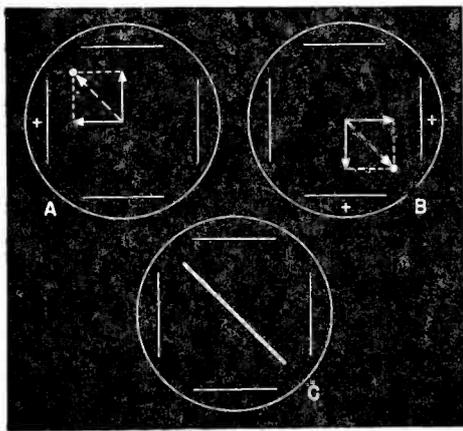


FIG. 5. Applying the same identical voltage to both sets of deflecting plates produces a tilted line.

Applying this to the c.r.o., we see that for the spot to trace the actual form of the wave, it must be moved steadily from left to right to indicate time, and at the same time must be moved up and down in response to the

voltage we want to “see.” Therefore, it is customary to apply a “timing” voltage to the horizontal deflecting plates, and to apply the voltage that is to be analyzed to the vertical deflecting plates.

The resultant of these two movements will sweep the spot across the screen in the form of the a.c. voltage applied to the vertical plates.

► But one cycle of that wave would be swept across the screen and gone too quickly for the eye to see. Even the lowest audio wave, 20 cycles per second, would be gone before we had a chance to glimpse its shape. The c.r.o. makes the wave stand still by making it repeat itself over and over on the screen, for as long as we want to see it.

This repetition of the wave is accomplished by applying a “sweep” voltage to the horizontal plates. The sweep voltage carries the spot steadily across the screen from one side to the other, as the a.c. wave on the vertical plates is going through its cycle, and then the sweep voltage snaps the spot back to the beginning just in time to start the process over again, at the same point in the a.c. wave. Such a voltage is known as a “saw-tooth” voltage and is produced by a circuit like that in Fig. 6A.

Essentially, condenser  $C_1$  is charged through resistor  $R_2$  to the point where the voltage across  $C_1$  equals the breakdown voltage of the special gas tube  $VT$ . When this critical voltage is reached, the tube begins to pass current, which ionizes the gas and reduces the tube plate-cathode resistance to a very low value, so the condenser discharges through the tube. The very low plate-cathode resistance causes most of the plate supply voltage to be dropped in  $R_2$ , leaving too little for the tube to maintain ionization, so the tube ceases to conduct. At this point,

condenser  $C_1$  is again charged from the B supply through resistor  $R_2$  and the cycle is repeated. The frequency of the sweep voltage can be adjusted by varying condenser  $C_1$  and resistor  $R_2$ , as the frequency is dependent on the time constant of the  $R_2$ - $C_1$  combination. The voltage developed across the condenser has the saw-tooth wave shape shown in Fig. 6B, and is taken from the output terminals through condenser  $C_2$ . This condenser blocks the d.c. so that the desired saw-tooth a.c. voltage can be applied to the horizontal plates.

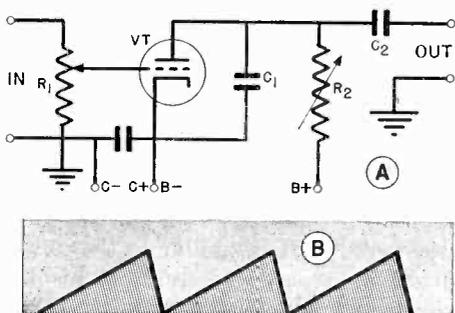


FIG. 6. A typical sweep generator and the wave form produced.

► The gas discharge tube  $VT$ , although it has a grid, does not work exactly like the ordinary vacuum tube. The amount of grid bias will set the voltage at which the tube will start conducting, after which the grid has no further control over plate current, until the plate voltage is stopped by the plate voltage dropping. This allows the grid again to assume control, repeating the cycle.

The grid is normally used as a means of synchronizing the saw-tooth frequency accurately with the voltage under study. By setting the sweep controls so that the saw-tooth frequency is just under that of the applied signal, and then applying a small voltage from this signal to the grid of the

sweep tube, the saw-tooth voltage is made to "lock in" with the signal under study. The tube is unblocked at exactly the right intervals by the control pulses making the grid less negative, forcing the  $R$ - $C$  circuit to "jump" into step every cycle. This overcomes any frequency drift in the sweep circuit because of changes in plate voltage, etc., and forces the sweep and the applied signal to stay together, or to synchronize.

## PRODUCING THE IMAGE

With this sweep voltage applied to the horizontal plates and an a.c. sine wave applied to the vertical plates, the resulting image can be plotted as shown in Fig. 7, to give a clear picture of how the two combine in producing the motion of the spot.

The sweep frequency  $B$  must exactly equal the sine wave frequency  $A$  to produce the image shown at  $C$ . Sup-

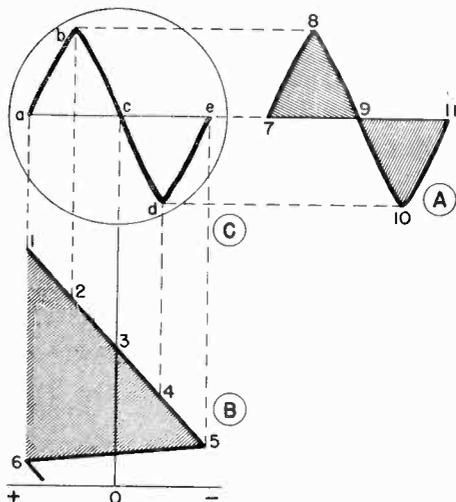


FIG. 7. Applying a sweep voltage to the horizontal deflecting plates and a sine wave to the vertical deflecting plates produces a sine wave pattern on the screen.

pose that point 1 on the saw-tooth wave represents a positive voltage on

the plate to the left, when looking at the face of the tube. Then the electron spot will be moved to position *a*. At this moment the sine wave is just starting from position 7.

As a saw-tooth wave changes from position 1 to position 2, the sine wave during the same time has moved from 7 to 8 so the spot will have moved to the position where the dotted lines intersect. This will be position *b*. Similarly, the spot is moved to positions *c*, *d*, *e* and then back to *a*.

The return of the spot from *e* to *a* is produced by the saw-tooth wave between points 5 and 6. The spot is being moved very rapidly across the face of the tube and, as a result, few electrons have time to hit any one point on the screen. This means that this backtrace is very faint; in fact, you may have to turn up the intensity control somewhat even to see it.

When the spot returns to *a*, the pattern is retraced and this action is repeated over and over again.

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## A Typical Commercial C.R.O.

We have now discussed the fundamental action of the c.r.o., and shown in detail the formation of an image on the screen by the combination of a sweep voltage and an a.c. signal. Before going on to explain the methods of using the c.r.o. in practical servicing, let us look at an actual commercial c.r.o., to see what controls are usually provided and how they affect operation. An understanding of this instrument will allow us to use most commercial instruments intelligently, since they are usually operated in a similar manner, even though the controls may have different names and positions.

### THE C.R.O. CIRCUITS

Before studying the setting of the controls, we should have an understanding of the circuits in the c.r.o., so we can know just what the controls do.

Fig. 8 is the wiring diagram and Fig. 9 shows the controls and general appearance of the RCA 155, chosen as an example of a typical service instrument. The instrument has in it a cathode ray tube, a high-voltage power supply, amplifiers for the horizontal

and vertical voltages, and a sweep generator.

**Power Supply.** There are two complete power supplies in Fig. 8. The section using rectifier  $VT_2$  is the low voltage supply. It is similar to the power pack in a large receiver, furnishing about 400 volts for the operation of the amplifiers and sweep circuit. As this is a rather high d.c. voltage, the filter condensers are in series, to increase the voltage rating. Thus, condensers  $C_{10}$  and  $C_{20}$  together give half the capacity and twice the voltage rating of a single condenser.

The c.r.o. tube requires about 1000 volts for operation. Such a high voltage from a single power supply would mean a special rectifier tube and expensive power transformer. Instead, here the high voltage pack, using tube  $VT_1$ , furnishes 600 volts, and its output is connected in series with the 400-volt supply, thus furnishing the needed 1000 volts.

A single high-voltage power transformer winding is used, tapped for the low voltage supply. The rectifier  $VT_1$  is connected in the *negative* side of its circuit, so the output voltages can be

added and this single transformer winding used.

As the d.c. current required by the c.r.o. tube is extremely small, resistor  $R_{13}$  can be used in the high voltage filter without appreciable d.c. drop. The filter condensers  $C_{12}$  and  $C_{13}$  are paper condensers, oil impregnated, capable of withstanding 1000 volts. Small capacities can be used here as the low drain does not appreciably discharge the input condenser and the  $R_{13}$ - $C_{13}$  combination gives sufficient filtering.

► High voltages are dangerous—NEVER take an oscilloscope out of its cabinet for repairs until it is disconnected from the power line. DON'T WORK ON ONE WHILE IT IS CONNECTED TO THE POWER LINE. You can use an ohmmeter, with the oscilloscope OFF and disconnected from the power line, to find any ordinary trouble.

**The C.R.O. Tube.** The type 906 c.r.o. tube is shown as  $VT_6$ , just to the right of the power pack. The elements, reading down, are: the filament, the cathode, the grid element, anode 1, anode 2, the vertical deflecting plates, and the horizontal deflecting plates. The d.c. potentials for the elements are obtained by tapping at different points on the voltage divider  $R_{14}$  to  $R_{21}$  inclusive, which is across the full output of the two power supplies in series. The c.r.o. grid is negative with respect to the cathode, and this voltage can be varied by adjusting  $R_{14}$ , the intensity control.

Anode 1 is positive and  $R_{16}$  is adjustable for focusing. Anode 2 is near the positive end of the divider, and thus has nearly the full high voltage between it and the cathode.

The deflecting plates have one of each pair connected to anode 2, keeping them at the same potential as the

anode to prevent defocusing the beam.

**Centering Controls.** It is impossible to locate the tube elements in every tube so as to get the spot in the exact center of the screen when no deflecting voltages are applied. Controls  $R_{22}$  and  $R_{23}$  are included to permit accurate centering of the spot. Since the controls are connected across  $R_{20}$  and  $R_{21}$ , which are a part of the voltage divider, the controls have the same voltage across them. Anode 2 and the two deflecting plates connected to it are connected to the center of the divider section  $R_{20}$ - $R_{21}$ . Thus, by moving the arms of  $R_{22}$  and  $R_{23}$ , which are connected to the other two deflecting plates, the potential on each of the "free" deflecting plates can be made either above or below that on the opposite one of the pair. This d.c. difference between the pair will move the spot toward the more positive plate; thus it is possible to make it move to the center of the screen.

After the beam has been centered and focused, and the intensity adjusted, we can apply voltages to the deflecting plates through  $C_2$  and  $C_5$ , thus developing a.c. voltages across resistors  $R_{24}$  and  $R_{25}$ .

**Sweep Generator.** The sweep generator is marked "timing" in Fig. 8 and uses the gas discharge tube  $VT_3$  and a selector switch  $S_4$ , which connects the proper condenser into the circuit for the frequency range desired. The condensers  $C_7$  to  $C_{10}$  cover the range 15 to 22,000 cycles in four overlapping ranges. The chosen condenser is charged through  $R_{11}$  and  $R_{12}$ , the latter being adjustable to give a "fine" frequency control.

► The synchronizing transformer feeds the control signal into the timing tube grid through control  $R_8$ . This control is set so locking is obtained but is not over-set—since too much con-

trol signal will distort the shape of the sweep voltage.

**Vertical Amplifier.** The signal to be viewed is usually applied to the vertical deflecting plates. The terminals are shown at the upper left in the diagram. When switch  $S_1$  is set to the *OFF* position, the input signal is applied through  $C_2$  directly to the vertical deflecting plates of the c.r.o. When  $S_1$  is set to the *ON* position, the signal is fed through  $C_1$  to the grid of the amplifying tube  $VT_4$ . To obtain a flat response characteristic over a wide frequency band (excellent between 20 cycles and 35 kc., and usable to 100 kc.), the plate load is a combination of resistances  $R_3$ ,  $R_4$  and inductance  $L_2$ . The coil  $L_2$  compensates for the normal high-frequency output drop (caused by plate-to-ground capacity) by acting as an increasing load with frequency, thus raising stage gain. The output is then fed through  $C_2$  to the vertical deflecting plates.

**Horizontal Amplifier.** This stage is similar to the vertical amplifier. However, switches  $S_2$  and  $S_3$  need some explanation.

► In the position shown,  $S_2$  and  $S_3$  are set for an external synchronizing voltage. This voltage is applied from some standard frequency source to the *SYNC* terminals at the right of the diagram, going to the grid of the sweep tube through  $T_2$ . The sweep output, thus controlled, is fed through  $R_{26}$  to the horizontal amplifier tube  $V_5$ , and then through  $C_5$  to the horizontal deflecting plates, the amplitude being controlled by  $R_5$ .

► When switches  $S_2$  and  $S_3$  are set to the next position, 60-cycle a.c. is fed from the filament supply of  $VT_5$  through  $S_3$  to the input of the sweep, so the sweep frequency can be locked to this frequency or some multiple of it.

► In the next position of  $S_2$ - $S_3$ , a part of the voltage under study on the vertical amplifier is fed to the sweep input and "locks" the sweep with it, as described in the basic discussion. This is the most used position of these switches.

► In the next position of  $S_2$ - $S_3$ , the sweep is not in the circuit. Instead, an external signal can be fed into the *HORIZONTAL INPUT* terminals at the right, and through tube  $VT_5$  to the horizontal plates.

► In the final position of  $S_2$ - $S_3$ , the horizontal input terminals connect directly to the deflecting plates through  $C_5$ , without the horizontal amplifier or sweep generator.

### SETTING THE CONTROLS

The controls on different oscilloscopes may have different positions and names, so the manufacturers' instructions should be followed in all cases. However, suppose we wish to see the wave form produced by a 60-cycle power line, and intend to use the RCA 155 for this purpose.

Turning to Fig. 9, you will see that the controls on the front of the instrument are divided into four groups. Right underneath the tube, there are two controls labeled *INTENSITY* and *FOCUS*. Then there is a vertical group at the left, consisting of three controls and two binding posts. The head of this column is labeled *VERTICAL*.

Next to this is a group of three controls and two binding posts which are labeled *TIMING*. Then we have three controls and two binding posts which are labeled *HORIZONTAL*.

In order to put the instrument into operation, we should proceed to set these controls as follows:

**Step 1.** The *ON-OFF* switch is a part of the *INTENSITY* control. Turn the *INTENSITY* control until the



switch clicks *ON*, and wait for about a minute to permit the circuits to warm up. Then advance the *INTENSITY* control until a spot of light is seen on the screen. Don't turn the *INTENSITY* control too high, just turn it up so that the spot of light is barely visible. THIS IS IMPORTANT—IF A BRIGHT SPOT OF LIGHT IS PERMITTED TO REMAIN AT ONE POINT ON THE SCREEN FOR AN APPRECIABLE PERIOD OF TIME, THE SCREEN CHEMICALS CAN BE BURNED AWAY, WHICH DESTROYS THE ABILITY OF THAT PARTICULAR SPOT ON THE SCREEN TO REPRODUCE LIGHT.

**Step 2.** The spot focus should be adjusted, with both *GAIN* controls turned to zero to keep voltage off of the deflecting plates and thus eliminate any deflection of the spot into a line.

Rotating the *FOCUS* control will make the spot of light increase in size and spread into an irregular shape. The control should be adjusted so that the spot is as small as possible and perfectly round. If the spot is now too bright, turn back the *INTENSITY* control until the spot is just easily visible, being sure to eliminate any halo of light around the spot.

**Step 3.** The spot must now be centered, still with no voltages on the deflecting plates. The *VERTICAL* centering control (underneath the *INTENSITY* control) will move the spot up or down, and the *HORIZONTAL* centering control (underneath the *FOCUS* control) will move it to left or right. The spot should be moved by means of these controls to the exact center of the screen.

**Step 4.** As we will use the internal sweep, we must set the *HORIZONTAL AMPLIFIER* switch,  $S_2$ - $S_3$  on the schematic (at the lower right of the

front panel of the photograph) to *INT*, the position for control of the sweep by the voltage under study. This position is shown on the photograph. This connects the timing circuit input to the vertical amplifier, and its output to the horizontal amplifier.

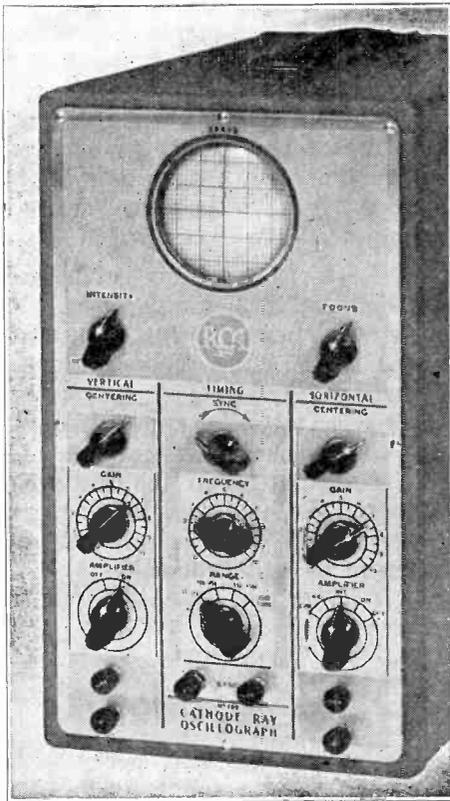
Turn up the *HORIZONTAL GAIN* control ( $R_5$  in Fig. 8) and the spot will turn into a faint horizontal line. Increase the intensity setting and the horizontal line will become brighter. When the spot begins to move, fewer electrons strike any given point on the screen in a given time, and brightness drops. It is safe to increase the number of electrons in the beam by adjusting the intensity control, since the spot does not stay in one position long enough to damage the screen. Adjust the *HORIZONTAL GAIN* control so that the line extends about two-thirds the distance across the face of the tube.

**Step 5.** To analyze a 60-cycle wave from the power line, the *RANGE* switch in the *TIMING* group must be set to the first range, which covers frequencies between 15 and 120 cycles. Remember, we want the sweep voltage frequency to be the same as our 60-cycle frequency.

**Step 6.** Now connect the voltage to be analyzed to the *VERTICAL* terminals, at the bottom left of the panel. You can get a voltage from the power line by using a step-down transformer, connecting the low-voltage winding to the c.r.o. terminals.

**Step 7.** Next, turn the *VERTICAL AMPLIFIER* switch to the *ON* position. Then advance the *VERTICAL GAIN* control until the line of light on the screen changes into some kind of pattern, most likely a series of rapidly moving lines or figures.

**Step 8.** Now rotate the *TIMING SYNC* control ( $R_8$  in Fig. 8) to about mid position. We are now ready to ad-



Courtesy RCA

FIG. 9. The controls on the RCA 155 oscilloscope—a typical service type instrument.

just the *FREQUENCY* control ( $R_{12}$ ) in the *TIMING* group, so that the sweep frequency will be the same as that of the wave on the vertical plates. This is accomplished by turning the control until the pattern on the screen stands still, instead of moving about in a scramble. Several “stationary” points will be found within the range of the control, and the one should be chosen which gives on the screen a single complete cycle of the sine wave. The sweep circuit is now set to 60 cycles, and “locked” with the incoming wave.

The *VERTICAL GAIN* control will now vary the amplitude (height) of the sine wave, while the *HORIZONTAL GAIN* control will vary the

width of the image. These two controls should be adjusted so that the image is well within the dimensions of the screen of the cathode ray tube.

It now may be necessary to readjust the *SYNC* control until the image stands still. (It may still get out of step momentarily, causing the image to “jump” occasionally.) Use the least amount of *SYNC* voltage (knob turned farthest toward the left) which will give a stationary image, as too high a setting can distort the image by causing a non-linear sweep.

### DISTORTION PRODUCED BY THE CONTROLS

Image distortion may occur if the signal amplitude is so increased that the image extends out near the edge of the screen. Since only a portion of the screen can be made useful without distortion, a large size screen is desirable, to permit a large image which does not reach the edge of the screen. Service oscilloscopes usually have screens of 3 to 5 inches in diameter. The 3-inch tube limits the useful image to about 2 inches in each direction.

FIG. 10 shows how the proportions between the horizontal sweep and ver-

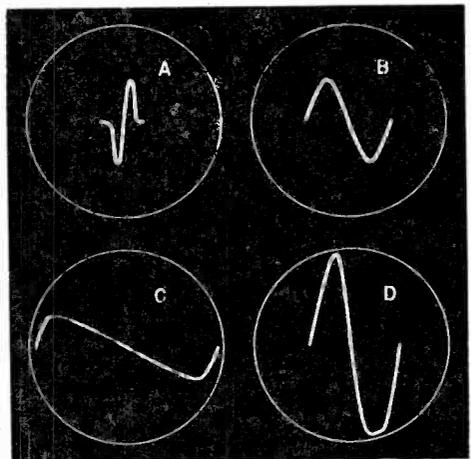


FIG. 10. Improper adjustment of the controls can distort the produced image as shown here.

tical amplitude affect the usefulness of the image. In *10A* the sweep is turned too low; the horizontal gain control should be advanced to "pull out" the image lengthwise, to avoid the too-sharp peaks. In *10C*, the image has been stretched too far, and in *10D* it has been increased in height too much, producing a square shape at the bottom (distortion). Proper adjustment of horizontal and vertical gain controls produces the well proportioned image of *10B*.

### STRAY VOLTAGES

Stray voltages, in addition to those being studied, may be picked up by the c.r.o. and appear on the screen. Strays are most apt to be picked up in the neighborhood of industrial machinery, and when test leads are connected

to the vertical input terminals but are not connected to any source of voltage. Under these conditions, a straight vertical line will appear on the screen with no horizontal voltage applied. Should the sweep be on, you can even synchronize with a distorted pattern, particularly when the stray voltage is from a near-by power line.

If the test leads are touched together or connected to a source of voltage, the stray voltage will usually be minimized. If, however, the stray is very strong, it may come through even with the leads connected to a voltage source, and will appear as a series of humps that "crawl" around the stationary pattern. The only cure for this is to shield the test leads or move the instrument to another location.

## Putting the C.R.O. to Work

So far, we have learned how the c.r.o. operates, and how to make it reproduce an image of the applied a.c. voltage. In order to show how this instrument is used on the service bench, we are now going to take up the actual methods of applying it to service problems, such as determining frequency, signal tracing, finding hum, oscillation or distortion, and determining phase relationships. Then we will show how high-fidelity alignment depends on this instrument. The c.r.o. is literally an instrument of a thousand uses, but the ones to be described here are those which the advanced serviceman is most likely to find of the greatest value in speeding up his work and making it accurate, efficient and professional.

### FREQUENCY DETERMINATION

In the c.r.o. operations already described, we have obtained a stationary image of a single cycle on the screen by

adjusting the sweep frequency to be exactly equal to the incoming frequency. Actually, a stationary pattern will be obtained if the sweep is a sub-

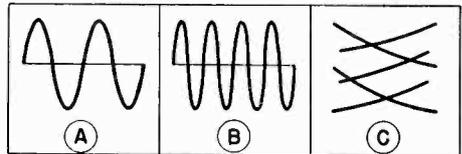


FIG. 11. Frequency can be determined by the number of cycles or lines in the pattern.

multiple of the incoming frequency, and the number of complete cycles in such a pattern immediately gives the relationship of the two. For example, if the incoming wave is 1000 cycles and the sweep is set to 500, we will get a 2-cycle pattern, as shown in Fig. 11A. Two cycles of the incoming wave occur during each sweep cycle. Similarly, a wave having a frequency four times

the sweep frequency makes four complete cycles, and so on up to 15 or 20, depending on the size of the screen.

► When the sweep is adjusted the other way, so it is *above* the incoming frequency, we no longer get an image of the incoming signal. Instead, there will be a series of more or less horizontal lines. When adjusted so that there are two lines stationary, the sweep frequency is twice the incoming signal. When there are three lines, the sweep is 3 times this frequency, etc. Fig. 11C shows five lines, so the sweep is five times the incoming signal. These patterns can be used only for frequency determination.

► In between these points where the sweep voltage and incoming frequency lock to produce a steady image, you will get various scrambled patterns. As these scrambled patterns are continually changing, they cannot be analyzed.

► The fact that we can get a number of patterns which do lock permits us to calibrate the sweep generator controls approximately, if an accurate signal can be fed to the vertical amplifier. Then, by making a chart of the setting of the sweep controls, we can use the settings as rough frequency determinants for unknown incoming signals.

For example, if we apply an accurate 1000-cycle signal and then adjust the sweep circuit to get a single sine wave, we have the sweep circuit set for 1000 cycles. When we adjust to where we get two sine waves, the sweep is set at 500 cycles, etc. Thus, by using one frequency we can find perhaps 10 or 15 points on the sweep controls accurately.

When an unknown signal is applied, we can estimate the frequency from the adjusting points on the frequency controls. Of course, this is not an extremely accurate method of determin-

ing frequencies, as the sweep frequency drifts, but it may prove entirely good enough for many purposes.

### DETERMINING FREQUENCY BY USING LISSAJOUS FIGURES

If we apply a.c. voltages to both sets of plates, instead of a.c. to the vertical and a saw-tooth to the horizontal, a type of pattern known as a Lissajous figure is obtained which can also be used for frequency determination.

If exactly the same a.c. voltage is applied to both pairs of deflecting plates *in phase*, a straight slanting line is obtained, as you will recall from Fig. 5. Should the voltages be of the

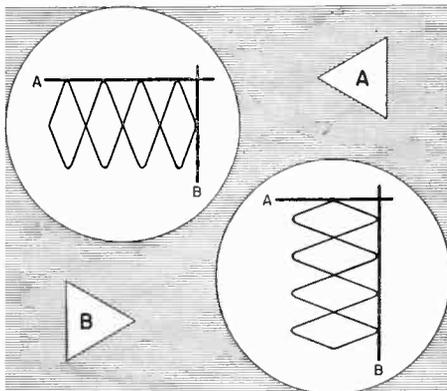


FIG. 12. Using imaginary lines A and B to determine frequency in a Lissajous figure.

same frequency, but out of phase, as may occur if they are from different sources, then a loop or circle will form. However, any pattern which is a single line, oval or circle shows that the two a.c. voltages have the same frequency. A series of closed loops means that the two frequencies differ, and the position and number of the loops gives the ratio of the two applied frequencies. Even fractional ratios can be determined in this way.

For example, suppose we find that we get the pattern shown in Fig. 12A. To determine the frequency ratio,

draw or imagine the lines *A* and *B*. Notice the number of times each line is touched by the image. The line *A* is touched four times, while the line *B* is touched once. Therefore, the frequency ratio is 4 to 1 and the frequency of the signal applied to the vertical plates is 4 times that of the signal on the horizontal plates in this case. Thus, if a 60-cycle voltage is applied to the horizontal plates, the vertical plate voltage has a frequency of 240 cycles.

It is possible for the image to turn around, as shown in Fig. 12*B*. Now the figure touches the line *A* once and the line *B* four times, so the ratio is now 1 to 4 and the horizontal signal frequency is 4 times the vertical signal

zontal voltage, a 5 to 3 ratio gives  $5/3 \times 60$ , or 100 cycles as the vertical frequency, etc.

By carefully studying the patterns, with one a.c. voltage from a source of known frequency, the frequency of the unknown a.c. wave can be determined.

### PATTERN PECULIARITIES

Certain peculiar patterns become intelligible when the controls on the c.r.o. are adjusted. A good example of this is a case where the rapidity of movement of the spot at certain portions in the pattern makes those parts invisible until the intensity is brought up. The disappearance of the back trace in Fig. 7 is an example of this. In Fig. 14*A*, a disconnected pattern is shown which becomes Fig. 14*B* when the pattern is made brighter. The motion of the spot from *a* to *b*, from *c* to *d*, from *e* to *f* and *f* to *a* is so rapid that no line may be visible at low intensities. Square waves in general will have very thin vertical lines, since the spot moves very rapidly in such patterns.

Other examples are shown in Figs. 14*C* and 14*D*, with the faint lines being shown here as dotted lines.

### SIGNAL TRACING

When signal tracing in dead receivers with a c.r.o., a signal generator is connected to the input of the receiver, and the c.r.o. vertical input is connected across the plate circuit of each stage in turn, working toward the output. The sweep can be set so that the image has some recognizable character, giving positive identification of the signal, and we can tell how far the signal gets through the receiver. Of course, the same can be done by using the signal generator and the receiver loudspeaker and working in the other direction—from output to input, so the c.r.o. is not a requirement in this case.

In the case of hum, noise, oscillation

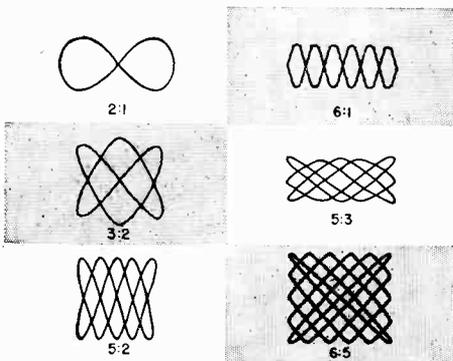


FIG. 13. Several typical Lissajous figures showing frequency ratios.

frequency. Thus, if the horizontal voltage is still a 60-cycle signal, the vertical voltage must be a 15-cycle signal.

A number of typical patterns are shown in Fig. 13. In all cases, counting the number of times the loops touch the imaginary lines *A* and *B* will give the frequency ratios.

Notice that Fig. 13 shows some of the fractional patterns for ratios such as 3 to 2, 5 to 3, etc. Should these patterns be turned around, the opposite ratios are indicated, such as 2 to 3, 3 to 5, etc. Using 60 cycles as the hori-

or distortion, the c.r.o. is more valuable, as you can not only find the source of trouble—the wave shape gives a clue as to the exact nature of the trouble.

**Hum.** By setting the sweep to 60 cycles and moving the vertical input connections along from the second detector to the output, the presence and *exact location* of hum voltage is determined. Hum picked up from the wiring or due to cathode-to-heater leakage is 60-cycle hum and shows a single-cycle pattern. However, hum caused by defective filter condensers in a full-wave power supply is 120 cycles and would give a 2-cycle pattern. Of course, you would normally check filter condensers and for cathode-to-heater leakage before using the c.r.o. Then the c.r.o. can be used for those unusual cases where the hum arises in an audio stage.

**Noise.** A signal generator connected to the input may or may not be necessary to find noise, depending on whether the noise source requires a shock to produce it. Most noises arise in circuits carrying d.c. and are caused by electrolysis producing corrosion, which in turn causes a poor contact. These noises can be heard and found without a signal. Any noise occurring in a signal circuit, or which is dependent on a signal, requires the use of a signal for quickest location.

When viewing a signal pattern, an added noise voltage causes irregular tearing and breaks in the pattern, with high, sharp peaks. Thus, by moving along from the radio input toward the output, when you reach the stage in which the noise is produced a jagged signal with irregular peaks will appear, either added to the steady signal or as a separate pattern, if a generator is not being used.

**Oscillations.** Oscillation is also found by moving the vertical input

leads from stage to stage in the receiver. Oscillations may occur in either the r.f. or the a.f. sections. No signal source is necessary to localize the trouble.

An exception is an audio amplifier which has a response peak, so it favors strongly some particular frequency. This amplifier may not oscillate until a signal at or near the resonant frequency is fed to it; the signal can then usually be turned off. Using a variable frequency audio oscillator, the signal frequency can be varied until oscillations start, then steps can be taken to damp this oscillation, once the offend-

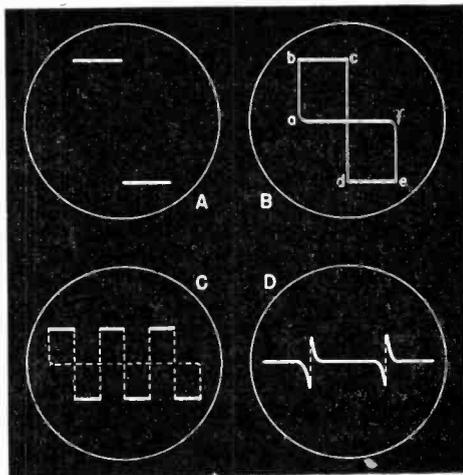


FIG. 14. Bring up the intensity enough to see the complete pattern so you won't be puzzled by disjointed images.

ing stage is discovered. When the signal generator is being used, the oscillation will appear as an interference to the generator pattern, or a sharp increase in image size.

**Distortion.** The c.r.o. is particularly useful for distortion complaints. Proper interpretation of the patterns seen will give a valuable clue as to just what is wrong with the amplifier.

Before beginning a search for distortion, it is necessary to understand that we are not going to try, in radio servie-

ing, to create perfect audio amplifiers. All amplifiers have a certain amount of distortion, and the human ear, in addition, is an imperfect instrument. A receiver which sounds good to one person may sound poor to another.

Thus, to the radio serviceman, distortion is any condition which makes a receiver sound worse than it normally does. It is only when something has happened to the receiver that a serviceman is likely to get a call.

Of the three types, amplitude, frequency and phase distortion, the radio serviceman is most interested in the first, since serious amplitude distortion is very offensive to the ear. Ordinary amounts of phase distortion cannot be heard at all and any defect producing a remarkable amount of frequency distortion will normally cause weak reception or hum also, which is easily located by other means. A television serviceman, on the other hand, is equally as interested in phase distortion because of its great effect on the picture. With these facts in mind, we can proceed to describe exactly how the c.r.o. is used to locate the different forms of distortion.

### AMPLITUDE DISTORTION

With amplitude distortion, the shape of the wave is altered by the addition, in the amplifier, of harmonics that were not in the original signal.

Therefore, we can determine the amount of amplifier distortion by comparing the shape of the wave that goes into the amplifier with that which comes out of it. We can also determine the location of the distortion by moving along stage by stage to determine just where the wave shape changes.

A signal must be sent through the audio amplifier by an audio oscillator or the audio modulating tone from a service generator. The c.r.o. vertical input is connected across the generator

and the sweep adjusted until a stationary pattern of 1 to 3 cycles is obtained. The input should be a constant sine wave frequency, for easy comparison.

By connecting the c.r.o. as shown in Fig. 15, we can quickly shift the c.r.o. signal from that at the input to that at the output of the amplifier. The wave shape at the output can then be compared with that at the input, and any change, indicating distortion, will be apparent.

Because it may be difficult to remember accurately the input wave shape when the output appears on the screen, and due to the increase in amplitude of the output, it is convenient

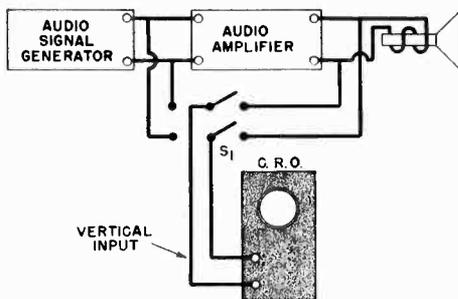


FIG. 15. How to connect the c.r.o. to an audio amplifier when checking for distortion.

to put a piece of tracing paper over the screen and trace the input wave form on it. This is easily compared with the output wave form, when the gain is adjusted so the tracing and output signal have the same amplitude (size).

Very small amounts of amplitude distortion cannot be determined by study of a sine wave. However, as pointed out, in radio servicing only rather large amounts of amplitude distortion will cause complaints, and such quantities are detectable on the c.r.o. screen.

**CAUTION.** When making the tracing, be careful not to punch or strike the end of the c.r.o. tube. These tubes are vacuum types and, due to their

size, are under terrific pressure. A scratch or puncture will cause an "implosion" (air rushing inward) which shatters the tube, blowing glass slivers about with great force.

Most modern c.r.o. units have a celluloid or plastic shield over the tube face for protection. This shield will be marked with convenient vertical and horizontal guide lines or scales, so the size of an image can be easily compared with another. There is no great danger when making a tracing over this shield.

► The oscilloscope has two important advantages in distortion complaints.

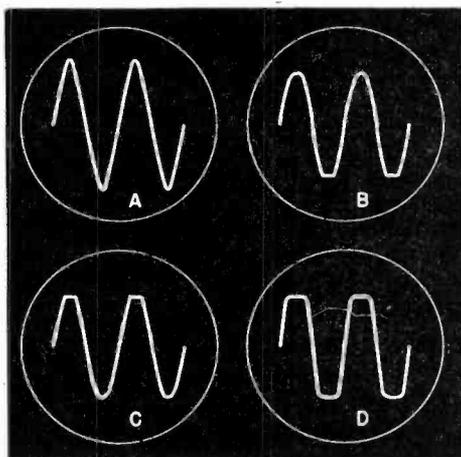


FIG. 16. Basic distortion patterns.

By moving the c.r.o. stage by stage through the receiver, the exact location of the distortion can be found. In addition, the shape of the distorted wave will often indicate the cause of the trouble.

For example, suppose we start with the sine wave shown in Fig. 16A. This should pass through the amplifier with no change other than an increase in amplitude. However, suppose that some stage produces a wave like that in Fig. 16B. As you will notice, the upper portion of the wave is still a sine

wave, but the lower section has been distorted. This immediately indicates that the tube in the stage in question is operating over a curved region of its characteristic. One-half the applied voltage is amplified normally but the other half is being cut off because of the operation over the curved tube characteristic. We cannot know whether this is the upper or lower tube characteristic, however, unless we know the direction of deflection of our c.r.o.

**Determining the Direction of Deflection.** The direction of deflection can be determined by using a battery. Turn off the horizontal amplifier, leaving a spot (or line if strays are picked up) of light on the screen,

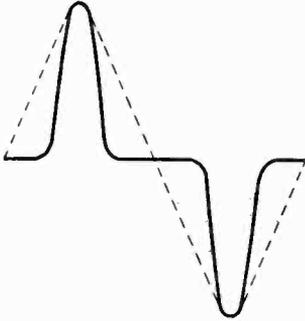
turn up the vertical amplifier, and touch the vertical test leads to a battery. The spot or line will jump up or down, returning slowly to its original position. The spot jumps because the battery causes a momentary charging current through the input blocking condenser, which is passed on like an a.c. signal.

After touching the battery, discharge the blocking condenser by touching the test leads together. Reverse the battery, and the spot should jump in the opposite direction.

The direction of movement of the spot depends on the number of stages in the c.r.o. amplifier and on the tube connections. Notice the battery polarity which causes the spot to jump up. Should the hot (or "high" lead on the RCA 155) lead go to the *negative* bat-

tery terminal, the c.r.o. has a negative input polarity; otherwise, a positive input polarity.

**Causes of Distortion.** If the signal shown in Fig. 16B is obtained right across the plate load in the plate circuit of the stage causing the distortion, it indicates that we have too much bias, too low a plate voltage or a weak tube, when the oscilloscope has a *negative input polarity*. Either condi-



The solid line shows the wave form obtained when a push-pull stage is over-biased. Compare with the dotted sine wave.

tion causes a tube to operate over the *lower* curved portion of its characteristic.

If the signal is inverted, as at C, so the top half is distorted while the lower half is normal, then under the same input negative polarity condition, this would indicate that we are operating near the upper bend in the tube characteristic. A logical cause for this would be a lack of sufficient bias. In either case, look for the usual causes of these operating voltage changes and you will go right to the source of trouble.

► Should the oscilloscope have a *positive* input polarity, the exact opposite conclusions are to be reached in these two cases.

► We can reach these particular conclusions about the causes of the distortion of this wave only if we are in the plate circuit of the offending stage. If we get the distorted wave shape at the

output of an amplifier, you must remember that each stage is inverting the signal 180°, so the wave is inverted, once for each stage it passes through. Hence, if we connect to the output of an amplifier and get this kind of wave, we know only that some tube is operating over a curved characteristic and we must localize the distortion further.

► The wave shown in Fig. 16D has both halves distorted or chopped off. This at once is a sign of an overloaded audio stage. Either the signal fed to the stage is too high, or something within the stage is causing the tube characteristic to curve excessively, so that very little of the straight portion

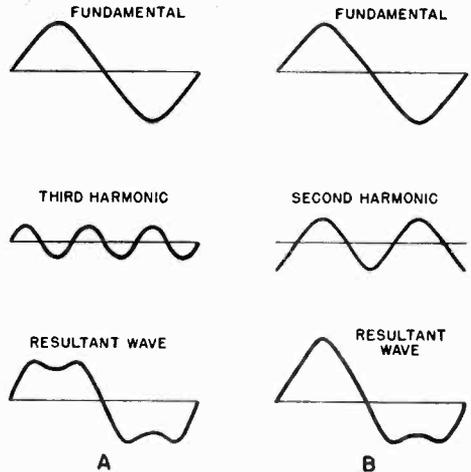


FIG. 17. Adding harmonics to a fundamental produces a wave shape change. Notice the difference between the resultant for odd harmonics and that for even harmonics.

is left. This condition is commonly the result of a poor tube, or lower than normal plate voltage, accompanied by a drop in bias voltage.

**Even and Odd Harmonics.** In studying such distorted sine waves to determine the source of the distortion, it is useful in addition to know whether the added harmonics are *even or odd*

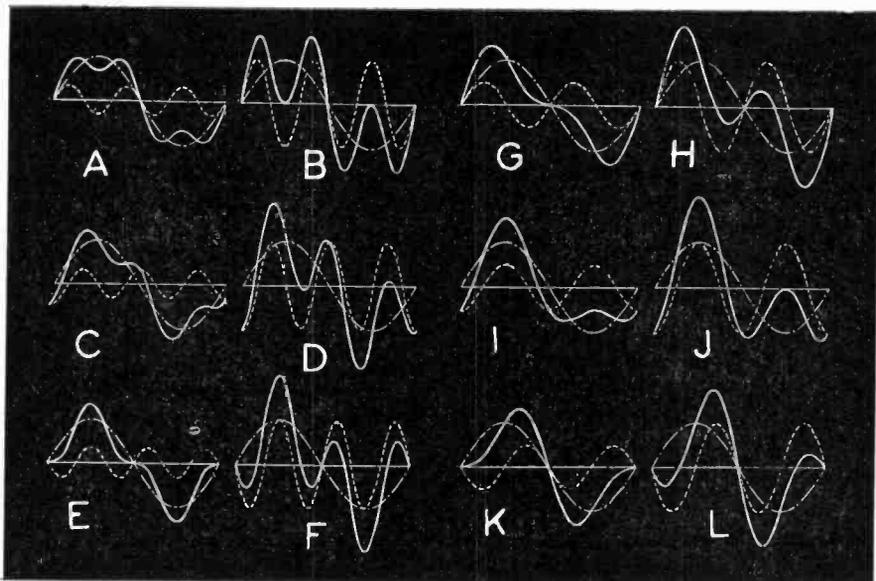


FIG. 18. Fundamental, harmonic and resultant waves superimposed on each other. The patterns A through F show odd harmonics, while G through L show even harmonics. The shape of the resultant depends on the amplitude and phase of the harmonics. Thus, A, B, G and H show harmonics in phase; C, D, I and J show a  $90^\circ$  phase shift between fundamental and harmonics; E, F, K and L show a  $180^\circ$  phase shift. Patterns A, C and E show the harmonic one-third the amplitude of the fundamental, while B, D and F show equal amplitudes. Similarly, G, I and K show harmonics with one-half the fundamental amplitude, and H, J and L show equal amplitudes. Regardless of the phase or amplitude, odd harmonics produce unsymmetrical resultants while even harmonics produce symmetrical waves.

harmonics, since this is often a further clue to the source of the trouble. Figs. 16B and 16C represent *even* harmonic distortion, because they result from the addition of the second, fourth, sixth or higher harmonics to the original sine wave. The wave shown in Fig. 16D, on the other hand, represents *odd harmonic* distortion, as it is primarily the result of combining a fundamental with the third, fifth and other odd harmonics.

**Harmonic Combinations.** Figs. 17 and 18 show graphically how various types of harmonics, of various amplitudes and phases, combine with a fundamental to make a complex wave. You will note that adding *odd* harmonics to a sine wave produces a *symmetrical* wave, that is, both halves of the cycle are exactly the same shape.

The lower half could be flipped up and would fit over the upper half.

On the other hand, when even harmonics are combined with the fundamental, the two halves of the waves are not symmetrical (not exactly alike). Of course, the actual shape of the wave will depend on the phase relationship and the relative amplitudes between the voltages, as shown in Fig. 18. Incidentally, when looking at some of the even harmonic waves of Fig. 18, you will find that the halves of the resultant waves are *mirror images*, that is, the lower half, if flipped up, would be a complete *reversal* of the upper half.

► Of course, it is perfectly possible to have a combination of even and odd harmonics at the same time. Then the resulting shape of the wave will de-

pend on the relative amplitudes and phases of the harmonics.

► As a bit of extra information, *even harmonic* distortion is not as noticeable to the human ear as is *odd harmonic* distortion. The even harmonic distortion may be as high as 10% before it is readily noticed, while odd harmonic distortion of 3% to 5% is usually noticeable.

## FREQUENCY DISTORTION

Frequency distortion, the inability of the amplifier to amplify equally all frequencies in its operating range, is usually detected by sending sine waves of different frequencies but constant amplitude through the amplifier. The output is measured at each frequency, and a comparison of the outputs at different frequencies to a reference frequency output, usually by means of a curve, will show whether or not the amplifier is "flat" enough for the purpose for which it was designed. Fig. 19 shows a typical curve. Notice that the output is measured, as no change in the wave shape occurs, other than its size.

► Developing a response curve by

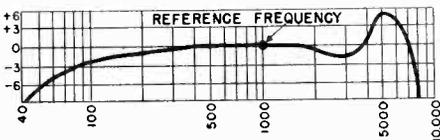


FIG. 19. A typical frequency response curve.

measuring output on a c.r.o. would be rather tedious, so it is usually not done in this way. A method of developing the response curve of the amplifier on the screen of the c.r.o. in a single pattern will be described later.

► Few defects in an amplifier change the frequency response without producing other changes in operation. An open coupling condenser would reduce the response to low frequencies, but would also result in an over-all lower-

ing of volume. A defective output filter condenser will affect the frequency response but will also introduce hum. A saturated transformer core reduces low-frequency response and will cause low volume as an additional symptom. A change in the characteristics of the loudspeaker is about the only defect which will produce frequency distortion without some accompanying symptoms leading you directly to the defect.

## PHASE DISTORTION

Phase distortion is a shifting of the phase relationships between various frequencies, in passing through an amplifier. Voltage peaks of different frequencies do not emerge with the same

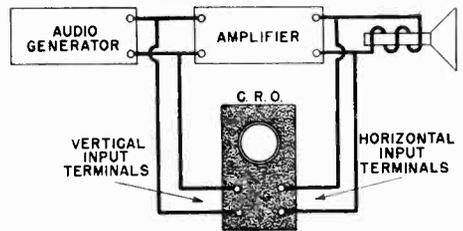


FIG. 20. The c.r.o. is shown connected here for determining phase shift.

time relationship they started with, so some peaks occur earlier or later than normal. This distortion is usually the result of some condenser-resistor combination, which delays or slows down the passage of some frequencies more than others. The extreme high and low frequencies are particularly affected in this way.

For ordinary music, phase distortion is of relatively little importance. The ear cannot detect phase shifts unless these shifts become large with respect to the time taken by a musical note. The amplifier will tend to cut off the signal altogether before any such large phase shift can occur.

However, in television, phase shifts

are extremely important. The c.r.o. provides a ready means of measuring phase shift in an amplifier, and the television serviceman must understand thoroughly how to perform this measurement.

► The connections are shown in Fig. 20. The vertical amplifier is connected to the signal coming from the audio signal generator. Then, the horizontal amplifier input is connected to the output of the amplifier, instead of to the timing circuit.

If there is no phase shift, we should have the same signal at the output as we had at the input, so this is like feeding the same a.c. signal to both sets of deflection plates. As you will recall, this will produce a slanting line. However, remember that the line is produced only if the voltages applied to the two sets of plates are completely

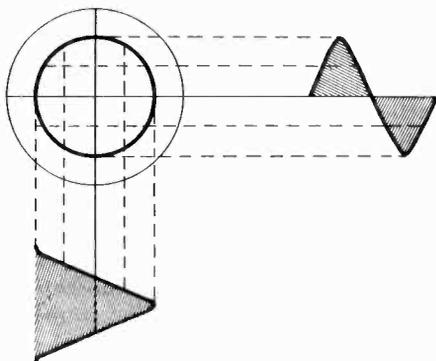


FIG. 21. Applying sine waves 90° out of phase to the c.r.o. produces a circular pattern.

in “step” or phase. Should there be a difference in phase, other patterns will be produced.

For example, if the voltages are equal, but are 90° out of phase, we will get a perfect circle as shown in Fig. 21. The two applied voltages are 90° out of phase as shown here, because one is at its maximum point while the other one is at zero.

Several typical patterns are shown

in Fig. 22 for different phase shift conditions. Notice that the pattern changes gradually from the slanting line through an ellipse to the circular pattern, then goes back to a slanting line pattern, tilted in the opposite direction.

► Of course, the exact shape of these patterns will depend upon the relative amplitude of the voltages. Usually, the amplitudes are adjusted to be ap-

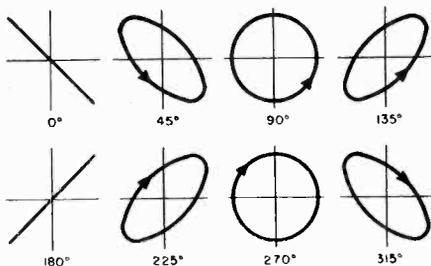


FIG. 22. The pattern changes from a slanting line to an oval, then to a circle, as the phase difference is changed.

proximately the same. To do this, turn off the vertical gain and adjust the horizontal gain to produce a line of some reasonable length, noting the horizontal gain control setting. This line is then measured. The horizontal gain is now turned off and the vertical gain turned up until a vertical line of the same length is formed. Then, the horizontal gain control is reset to the position which gave our first line.

When the figure is formed, its shape will indicate the phase difference. The pattern will form around the point where the horizontal and vertical scale lines on the screen of the tube cross, if the spot was originally centered at this point. Usually, we are interested only in finding whether a phase shift has occurred and are satisfied with a very rough approximation of the amount, as can be determined by comparison of the size and tilt of the oval patterns.

Should the exact value be necessary then we must use trigonometry. This method permits determining the phase angle, even when the amplitudes are not exactly equal. Measure the distance along the vertical line, represented by  $B$  of Fig. 23 and also the amplitude of the ellipse, which is  $A$ . Divide  $B$  by  $A$  and the result will be a number which is the sine of the phase angle. Then, by looking up this value in a trigonometric table, we can find the angle. For example, suppose we find that  $B$  is equal to  $\frac{1}{2}$  inch, while  $A$  is equal to 1 inch. Then, dividing  $\frac{1}{2}$  by 1, we get .5. A sine table shows this to be the sine of  $30^\circ$ ,  $120^\circ$ ,  $210^\circ$  and  $300^\circ$ . If the  $0^\circ$  line tilts as shown in Fig. 22, by looking at the direction of tilt of the oval, we can tell whether we have:  $a$ ,  $30^\circ$  or  $300^\circ$ , or  $b$ ,  $120^\circ$  or  $210^\circ$ . (The tilt will be similar to the  $45^\circ$

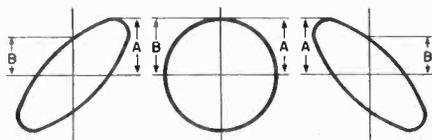
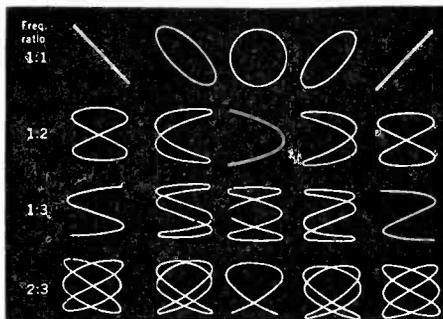


FIG. 23. Measure  $A$  and  $B$  to determine the phase shift.

and  $315^\circ$ , or  $135^\circ$  and  $225^\circ$  degree ovals shown in Fig. 22.) We cannot determine which one of group  $a$  or  $b$  it is, unless the direction of rotation of the spot can be determined, or we can increase the phase shift by a special circuit. Then, if the pattern becomes circular, we have either  $30^\circ$  or  $210^\circ$ , while if it changes to a line, we have either  $120^\circ$  or  $300^\circ$ . Thus a tilt to the left (as the  $45^\circ$  pattern in Fig. 22) which changes to a circle as the shift is increased indicates a  $30^\circ$  pattern.

The phase shift patterns shown in Fig. 22 depend on the actual connec-



The Lissajous figures we studied earlier are affected by phase shifts as shown here. The vertical columns from left to right show phase shifts of  $0^\circ$ ,  $45^\circ$ ,  $90^\circ$ ,  $135^\circ$  and  $180^\circ$ . The patterns will drift from one form to another if the frequencies do not have exactly the ratios indicated. Hence, you can let the pattern drift to a shape where the ratio is easily found rather than worry about some of the intermediate forms.

tion of the plates of the c.r.o. tube. It is possible for the tube to be rotated itself  $90^\circ$  or for the plates to be connected differently. If so, the patterns will be interchanged. The pattern we have shown for  $180^\circ$  will be the zero pattern while the zero pattern will be that for  $180^\circ$ . Similarly, the other patterns will interchange, the  $45^\circ$  pattern becoming then the pattern for the  $225^\circ$  and vice versa. The only way you can determine which is the true zero for your particular oscilloscope is to connect the same voltage, from the same source, to both the vertical and horizontal plates and notice in which direction the line slants.

# High-Fidelity Alignment

The c.r.o. finds one of its most important service applications in the alignment of high-fidelity receivers. The problem of alignment of such receivers is entirely different from that of the ordinary receiver, which requires only a standard signal generator and an output meter.

If the i.f. amplifier of a high-fidelity receiver is adjusted for maximum response at a single frequency, the band-pass characteristic will be destroyed and the receiver will lose the "fidelity" which it was designed to have.

The necessity for a band-pass i.f. in a high-fidelity receiver is plain if we remember that a modulated carrier consists of a *group* of frequencies, which extend *above and below* the carrier as far as the highest frequency to be transmitted. Thus, a 5000-cycle modulation range requires a band of frequencies 10-kc. wide, and this is the usual broadcast practice, entirely adequate for voice or dance music. Many stations now modulate to 7500 cycles, however, and some special ones go even higher. Such higher frequencies are necessary for true fidelity with fine music.

Receivers designed for distant reception cannot pass a band 15 kc. or 20 kc. wide which is necessary for high fidelity. Fig. 24A shows how such receivers are designed for a much sharper response, in order to eliminate interference between stations only 10 kc. apart in the broadcast band. Naturally, when frequencies only 5 kc. from the carrier are so sharply "cut" as shown in Fig. 24A, the high-frequency modulation will be practically eliminated and the receiver will heavily emphasize the bass frequencies. This may suit some "ears," but is not true fidelity.

In order to get the higher side band frequencies through the i.f. amplifier, high-fidelity receivers have the "flat-top" band-pass characteristic shown in Fig. 24B. This can be made wide enough to pass 5000 cycles without attenuation, as shown in Fig. 24B, or may be made even wider to obtain higher fidelity. Naturally, such receivers can be used on strong local signals only, since their wide response would permit serious interference from distant stations, if the sensitivity were also high.

► To align a band-pass circuit properly, we must adjust the response so that it is even over the whole width

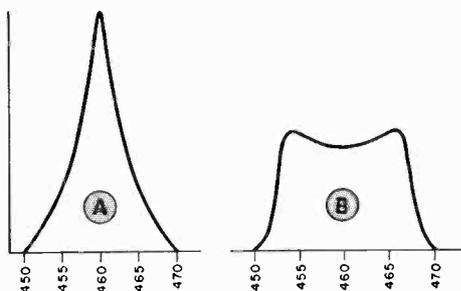


FIG. 24. Resonance curves. A shows a high-selectivity, high-sensitivity response, while B shows a broad pass-band for better fidelity.

of the pass band, and not "peaked" at one frequency, as in an ordinary i.f. circuit. This could be done by feeding different frequencies throughout the pass band into the amplifier, and plotting the various output levels as a response curve. This is a tedious process, however, and would have to be repeated each time the circuits were adjusted.

With the c.r.o., the response over the whole pass band can be made instantly visible on the screen, and the effect of adjustments will be apparent as they are made.

This is done by changing the frequency of the generator signal back and forth over the pass band at a regular rate, and at the same time applying a horizontal sweep on the c.r.o. which carries the spot across the screen and back at the same rate. If the input level is held constant, the *vertical* height of the spot above the center line will be proportional to the signal

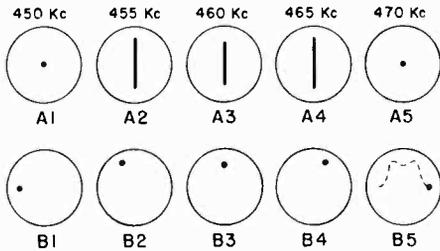


FIG. 25. The application of voltage to the vertical plates only will produce lines of varying length as in A1 to A5. By sweeping the spot as in B1 to B5, we get a trace of the wave as shown by the dotted line in B5.

output, and since the spot is also moving *horizontally* in step with the changing carrier frequency, it will plot a graph of the amplifier response over the band covered. Fig. 25B shows the spot at different positions in the sweep and in B5 indicates the curve that is formed.

► To use this system, we must vary the generator frequency over a wide band at a steady, rapid rate, and synchronize the c.r.o. horizontal sweep with these variations, to make the curve stand still. This is obviously impossible to do by hand, but there are two means of doing it, one mechanical and one electrical. Both systems, which use special signal generators known as “wobblers,” use a variation rate of 40 to 60 times a second, and sweep the signal frequency over a band 20 or 30 kc. wide. As this is frequency modulation, such generators are also called f.m. signal generators. Both systems will be described.

## MECHANICAL WOBBLATION

Mechanical wobulation is obtained by placing a small variable condenser in parallel with the regular tuning condenser of the oscillator tank circuit, and driving this small condenser by means of a motor. The condenser is made so it rotates through a complete circle, its capacity going from maximum to minimum and back, once for each rotation. This carries the signal frequency up and back again in step with the rotation.

For example, suppose the oscillator is set at 1400 kilocycles and the trimmer, such as the one shown in Fig. 26, is being rotated. The circuit is set for normal frequency when this condenser is half in mesh, as shown at C. Then the motor is started, which rotates the condenser through its range 30 or 40 times per second. As it changes from maximum to minimum capacity, this signal frequency goes

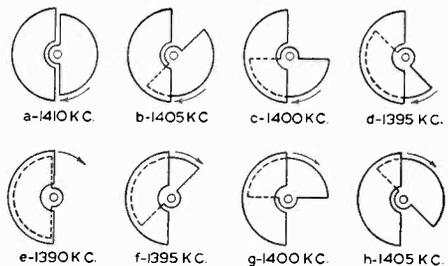


FIG. 26. Rotating the trimmer produces a frequency change with the change in capacity.

through a range of about 20 kilocycles, 10 kilocycles on each side of the normal or resting frequency.

As this condenser has a certain fixed capacity change, it will give a different band width as the main tuning condenser is adjusted for a different resting frequency, because this changes the ratio of the capacity *change* to the total capacity. As a fixed band width is desired, standard practice is to make use of a fixed oscillator. A frequency

of 700 or 800 kilocycles may be used by some manufacturers, while others may use 2000 kilocycles. In any event, the motor-driven condenser varies the frequency of the fixed-frequency oscillator, so the band width is fixed. Then, the output of this fixed-frequency oscillator is mixed with the output of a variable-frequency oscillator in a mixer-detector circuit. The resulting beat frequency is the output, so the resting frequency can be varied at will, but it is swept over a fixed band width. The dial on the signal generator is calibrated in terms of the frequency which actually comes out of it as a result of this mixing process.

**Obtaining Synchronization.** Various means of synchronizing the c.r.o. horizontal sweep with the rotations of the condenser can be used. A magnet on the condenser motor shaft,

switch on the condenser shaft, which allows the condenser voltage to build up and then discharges it every rotation.

## ELECTRONIC WOBBULATION

The oscillator control circuit, which is the heart of the automatic frequency control system, provides a means of getting electronic wobbulation. You will recall that by connecting a tube to

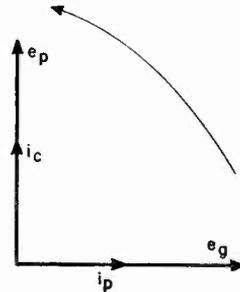


FIG. 28. The current and voltage relationships in the oscillator control circuit.

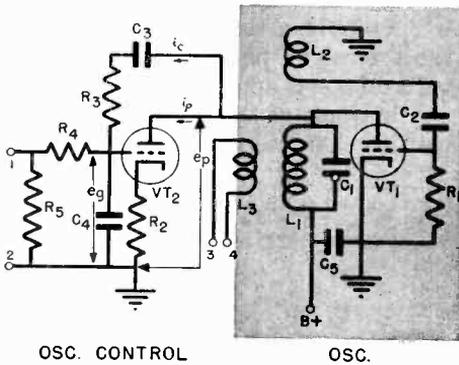


FIG. 27. The oscillator control circuit for electronic sweeping is just like the control circuit used in a.f.c. systems.

the oscillator circuit properly, we can make the tube act as an inductance coil. Then, by feeding a control signal to the tube, we can obtain an inductance which varies at a regular rate, thus varying the main oscillator frequency at the same rate.

**The Oscillator Control.** You will recall that one of the important characteristics of an inductance is the fact that the voltage leads the current by  $90^\circ$ . Therefore, any device which will exhibit this characteristic is acting just like a coil. As a review, refer to Fig. 27. Here we have a tuned-plate oscillator, connected so the plate current of the control tube must flow through tank coil  $L_1$ . This plate current is made  $90^\circ$  out of phase with the plate voltage.

The phase shift in the plate current is obtained by using the network  $C_3$ ,  $R_3$  and  $C_4$ , which is connected across the tank circuit. As the resistance of  $R_3$  is large compared to the reactances of  $C_3$  and  $C_4$ , the current  $i_c$  is in phase with  $e_p$ , as shown in Fig. 28.

As the current  $i_c$  flows through condenser  $C_4$ , the voltage drop across this condenser

turning near a fixed pickup coil, will give a regular pulse for each rotation. The magnet can be shifted on the condenser shaft to give the pulses at just the right time. Or, an R-C sweep circuit can be built right in the wobbulator and its output fed through the horizontal amplifier instead of using the c.r.o. built-in sweep. This R-C circuit can be connected to a shorting

will be  $90^\circ$  behind the current  $i_g$ . This is the normal action of a condenser. Therefore, the voltage  $e_g$  lags current  $i_g$  by  $90^\circ$ .

The plate current flowing through tube  $VT_2$  follows the grid voltage  $e_g$ . When  $e_g$  becomes positive, the plate current increases, and vice versa. Therefore, the plate current  $i_p$  is in phase with the grid voltage  $e_g$ . Since  $e_g$  is  $90^\circ$  out of phase with  $e_p$ , this means the plate current is likewise  $90^\circ$  out of phase with the plate voltage. Furthermore, it lags

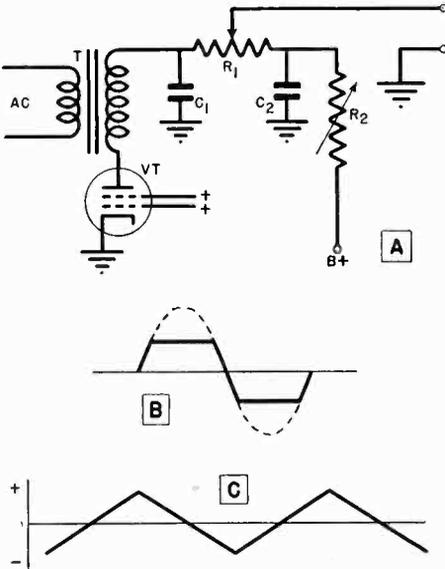


FIG. 29. A circuit for producing a triangular or pyramid wave shape.

behind the plate voltage, which is correct relationship for an inductance.

The value of resistor  $R_2$  is chosen so that the tube draws some particular amount of current, so it acts as an inductance of a certain value. This inductance is in parallel with coil  $L_1$ . This lowers the effective amount of inductance, but condenser  $C_1$  is adjusted to where the proper frequency is obtained.

**Control Pulses.** Now, when we apply control pulses to the input of the oscillator control tube, the plate current drawn by the tube will vary according to these control pulses. When the grid becomes more positive, the tube draws more current, thus acting as a smaller inductance. This reduces the effective inductance of the tank

circuit so that the circuit is tuned to a higher frequency. On the other hand, when the control pulses add to the bias, this reduces the current, making tube  $VT_2$  act as a larger inductance, and thus tunes the circuit to a lower frequency. By adjusting the value of the control pulse amplitude, we can get a fixed amount of swing about the normal or resting frequency.

For the purposes of drawing a characteristic curve, we want the frequency to be swept over this band of frequencies at an even and steady rate, first up, then back down over the same frequencies. This means the control pulse must swing gradually positive, then gradually negative. This form of change is produced by a voltage having a triangular or pyramid shape.

A circuit for producing this special wave is shown in Fig. 29A. Essentially, this is an amplifying tube, adjusted to operate on a characteristic having a short straight portion. Then a high value of a.c. is applied, so that the tops and bottoms of the swings are cut off by the characteristic curvature, pro-

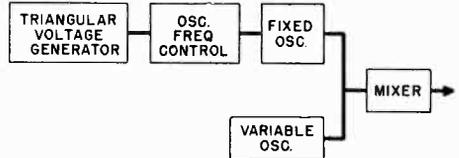


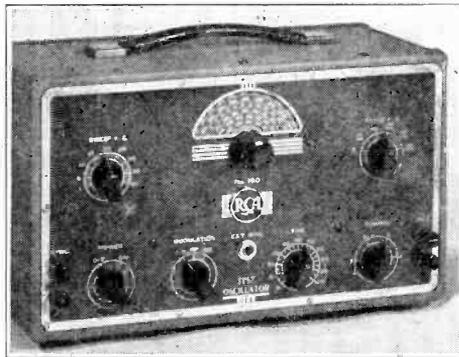
FIG. 30. This block diagram shows how the sections of an electronically wobulated oscillator are connected together.

ducing a square-wave shape as shown in Fig. 29B. These square-wave current pulses must pass through the filter  $C_1-R_1-C_2$ . As a result of the time constant of the filter, a wave shape which is like that shown in Fig. 29C will be obtained. This is the desired wave shape. The voltage can be varied by adjusting the slider on  $R_1$ , which makes it possible to vary the range over which the oscillator frequency is

changed. Thus, the sweep can be adjusted to be any desired width from perhaps 5 kc. to 40 kc.; this is not possible with a mechanical system.

**Complete Wobbulator.** The triangular voltage generator is connected to the oscillator frequency control, which in turn varies the frequency of the fixed oscillator over the desired band. The output of the fixed oscillator is fed to a mixer-detector circuit, along with the output from a standard variable frequency oscillator, as shown in Fig. 30. The "mixing" action is exactly similar to that in the mechanical wobbulator already described, producing a signal that varies over the pass band at a regular rate.

This particular oscillator can be used for ordinary alignment, as well as high-fidelity alignment. However, most servicemen purchase a standard oscillator, then obtain a separate wobbulated oscillator when their high-fidelity business warrants. The reason for this is that the wobbulated oscillator, because of the mixing action, has in its output several frequencies, just as there are several frequencies in the plate circuit of a converter tube. The extra frequencies would prove very confusing if a receiver is far out of alignment. For such receivers, a standard signal generator should first be used to get the receiver approximately lined up, and the wobbulator can then



*Courtesy RCA*

FIG. 31. A typical electronic wobbulator. It looks like an ordinary signal generator except for having an adjustable sweep control and an extra position on the modulation switch.

► Fig. 31 shows the picture of an electronic wobbulator of this kind. The controls are not greatly different from those of a standard signal generator. However, in this particular case, there are two adjustments for attenuating the output, and the modulation switch has an extra position for frequency modulation or wobble. The control at the upper left varies the amount of the pyramid voltage fed to the oscillator control, and thus varies the width of the sweep.

be used for final alignment, in high-fidelity cases.

## WOBBULATOR ALIGNMENT

Let's take a wobbulator and actually use it to line up a high-fidelity receiver. We would connect it to the first detector grid just as usual for i.f. alignment, through a blocking condenser. Then, assuming a diode second detector, the c.r.o. vertical input would be connected across the load resistor.

A connection must then be made from the wobulator to the sweep control circuit on the c.r.o. The receiver, oscillator and c.r.o. are turned on and allowed to warm up.

► Now, suppose the oscillator is set to the i.f., and the sweep frequency is ad-

460-kc. i.f. value (450 kc. to 470 kc.).

Each one of the humps on the screen will have a shape similar to the response characteristic of the amplifier. However, it is difficult to tell with this kind of curve just when the middle or resting frequency of the oscillator is at

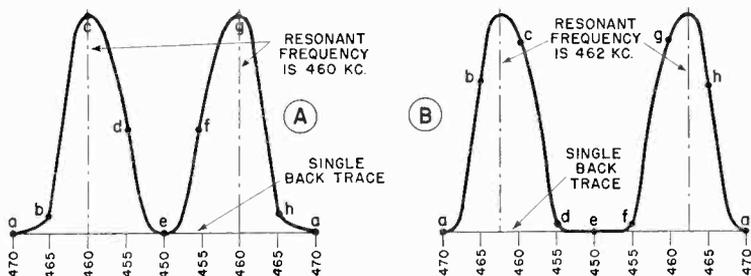


FIG. 32. The double hump resonance curve pattern is obtained when the c.r.o. sweep is set to the same rate as the wobulation. This pattern does not indicate clearly when the circuits are resonant to the resting frequency of the oscillator.

justed for a stationary pattern, *at the same rate as the wobulation*. The result will be a double-hump pattern as shown in Fig. 32A. You can see the reason for this if you will remember that the wobulator goes through the resting frequency twice for each cycle

the mid-frequency of the i.f. pass-band. For example, as shown in Fig. 32B, the oscillator has been moved to 462 kilocycles. Comparing A with B shows that there is very little difference in the appearance of these waves.

► The solution to our difficulty is to

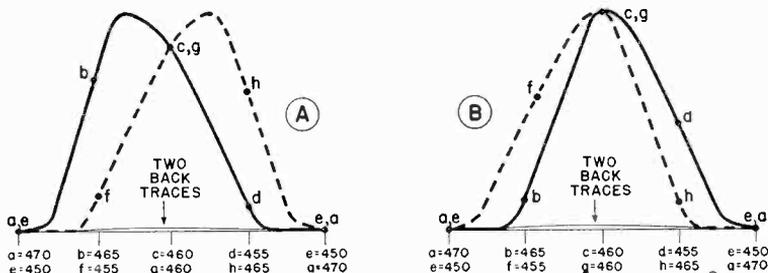


FIG. 33. This overlapped pattern is the one usually used for band-pass alignment. It is obtained by making the sweep frequency twice the wobulator rate.

of revolution of the revolving condenser, or twice for each cycle of the triangular sweep wave. In this particular instance, we have a wobulator sweeping over a band 20 kilocycles wide, 10 kilocycles on each side of the

adjust the sweep frequency of the c.r.o. to twice the wobulator rate. Then, we only have time for one of the variations for each cycle of the sweep, with the result that the two "humps" are placed one on top of the other, as

shown in Fig. 33. This overlapped pattern is the one normally used for high-fidelity alignment, so most mechanical and all electrical systems provide a controlling voltage twice the wobulator rate.

► Starting from point *a* and moving to point *b*, etc., you will find that the tracing is made through the resting frequency, then snapped back to the beginning and another trace is formed, apparently behind the first one. Notice the frequency goes up for one trace and down for the overlapped trace. This means the upper and lower halves of the band pass curve are overlapped, so we can see directly when both halves are alike.

► If we are not exactly in resonance, the tracings will separate to a position similar to that of Fig. 33A. As we tune to resonance, however, one tracing moves behind the other one. When we are exactly in resonance, the two peaks will exactly coincide, as shown in Fig. 33B.

Therefore, the receiver can be tuned to the proper resting frequency by making these peaks coincide. Now, the sides of the traces do not coincide until the proper alignment has been obtained. Therefore, the alignment of our band-pass, high-fidelity i.f. can now be made by causing the peaks to spread out to the proper flat-top shape, with the two humps exactly coinciding with each other. Therefore, we first must peak-align the i.f. amplifier, then must increase the capacity of one trimmer and decrease that of the other on each transformer, so the primaries and secondaries are tuned above and below the proper frequency, to get the band pass curve.

If you turn the adjustments too far, you will detune the circuit so the shape of the wave will change or the peaks may even separate. Typical distorted

waves are shown in Figs. 34A and 34B.

► When a band-pass circuit is perfectly adjusted, you will get a figure something like Fig. 34C. Incidentally, you can determine the amount of band-passing (an important adjustment), because the length of the back-trace line between points 1 and 3 of Fig. 34C corresponds to the sweep band width of the oscillator. The resting frequency is at point 2. If the sweep is adjusted for 30 kilocycles, the distance from point 1 to point 3 will be 30 kilocycles. Therefore, you will have to be sure that you have obtained the proper amount of band-passing by comparing the spacing of the sides of the image with the back-trace length. For ex-

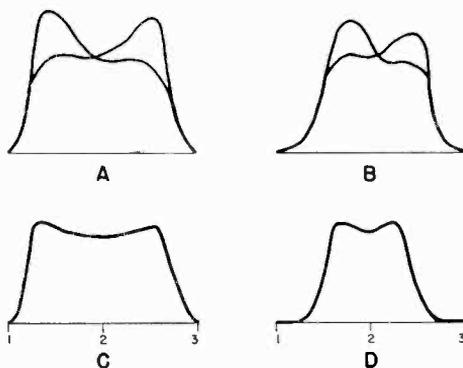


FIG. 34. Typical distorted patterns are shown in A and B. The band width of the pass-band can be determined by comparing the spacing of the sides of the pattern with the length of the back-trace line. Thus, C shows a circuit band-passed to the full limit of the wobulator range, while D shows a pass-band of about half as much.

ample, in 34D the circuit is band-passed, but only for a 10- or 15-kilocycle band width. This may be all the set is designed for. However, if the set should have a wider band width, the adjustments should be continued so the flat top is retained, but the sides are moved further apart.

**Connecting the C.R.O.** Practically all high-fidelity receivers will have a

diode second detector, using a resistive load, as in Fig. 35. The c.r.o. should be connected across resistor  $R_2$  when band-passing the receiver. When the receiver uses A.F.C., the connection is made across the audio load. This is  $R_{24}$  in Fig. 36.

Should the receiver happen to use transformer coupling, or should the connection be made to an audio stage

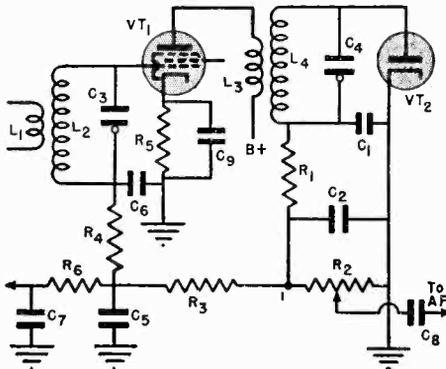


FIG. 35. Tube  $VT_2$  is a typical diode detector of the type usually found in high-fidelity receivers. The c.r.o. should be connected across  $R_2$ .

instead of the detector, phase distortion may occur. This causes part of the image to exist below the back-trace line as shown in Fig. 37. Phase shifts distort the wave to a useless form insofar as determining proper band-passing is concerned, so must be avoided. Where transformers are found, replace the primary temporarily with a 10,000- to 50,000-ohm resistor, or connect a 20,000- to 25,000-ohm resistor in parallel with the primary, with the c.r.o. connected across the resistor.

**Interference Patterns.**

Several kinds of interference may develop, depending on set design and local conditions. Typical patterns are shown in Fig. 38. If an interfering signal comes in, patterns like *A* and *B* may be found. Tune the preselector to some frequency

causing no interference when this occurs.

Noise produces similar breaks and jagged patterns, but is usually more irregular than the patterns shown.

► Modulation of the wobulator with an audio tone, due either to a fault in the instrument such as hum modulation, or to having the amplitude modulation turned on, will produce patterns like *C*. Most wobulators now have a switching arrangement which makes this double modulation impossible except when defects occur.

► Ghost images as in *D* occur when harmonics of the wobulator get into the circuit. When aligning the i.f. amplifier, a harmonic may feed into the

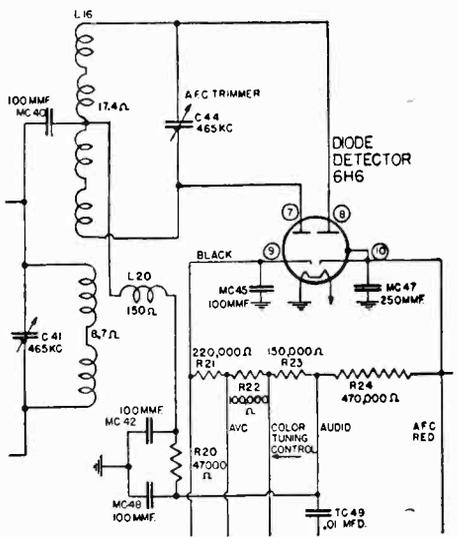


FIG. 36. The discriminator circuit of a General Electric receiver with a.f.c. The audio output is obtained across  $R_{24}$ , so the c.r.o. should also be connected across this resistor.

preselector and come through like an interfering signal. Tune to a different point on the dial or stop the receiver oscillator to clear this up.

► R.F. voltages in the detector load may produce a blurred image like *E*,

instead of the clear, sharp image shown at *F*. In a well-designed receiver, this occurs only when the detector r.f. filter is defective.

► Should regeneration occur in the i.f. amplifier, there may be a sharp peak at the resting frequency, with an asymmetrical side condition. It may be impossible to get true band-passing until this trouble is cleared up, as the peak will remain sharp instead of broadening properly. This same pat-

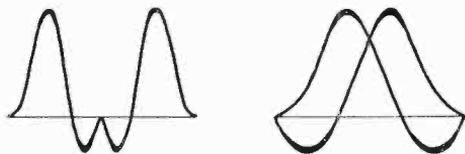


FIG. 37. Connecting the c.r.o. across a transformer or to an audio stage instead of the second detector produces a phase shift, pulling part of the pattern below the back-trace line and so distorting the pattern that it cannot be used.

tern would occur if the coupling between the i.f. transformers is changed in any way.

If oscillations develop, particularly at a frequency slightly different from the wobulator output, an interference pattern like that shown in Fig. 38C may be produced.

► From this section, you can see that the combination of wobulator and c.r.o. is necessary for the proper alignment of high-fidelity receivers, and will in addition indicate the presence and nature of a number of misadjustments and faults that could be difficult to find by other means.

### PRACTICAL HINTS

Let us now review the basic procedure for high fidelity alignment, to be sure the proper procedure is followed.

► First, the receiver must be peak-aligned by following the standard pro-

cedure, particularly if any tampering has occurred. Then, the wobulator is connected to the grid of the first detector and the c.r.o. is connected across the second detector load. The pre-selector is tuned away from any interfering station frequency—usually to the low frequency end of the band.

The wobulator is tuned to the i.f. frequency, then the c.r.o. sweep is adjusted to give the double or overlapped trace, with a fixed image. Assuming the wobulator to be producing the correct i.f., the adjusters are varied to get coinciding peaks, then adjusted for a flat-topped, square-shaped band-pass characteristic, with the sides of both traces coinciding. The width of the pass band should be adjusted as is proper for the set, comparing with the length of the back-trace line, which corresponds to the band width for which the wobulator is set.

► To complete the high fidelity alignment procedure, the wobulator should be moved to the input of the receiver and a check of the overall response made. The c.r.o. remains connected to the diode load of the second detector.

The wobulator is now of course

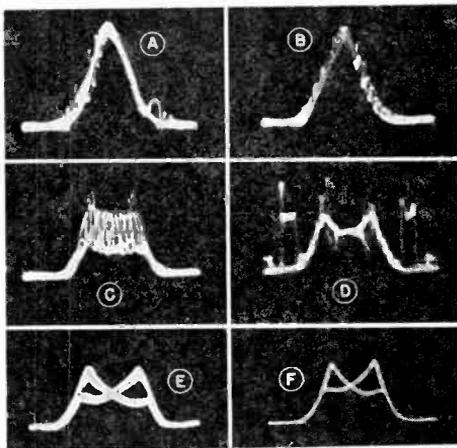


FIG. 38. Typical interference patterns.

tuned to some frequency within the broadcast band and the receiver is tuned to the same frequency. The output wave shape should be very similar to that attained through the i.f. amplifier alone. Normally, the preselector is not band passed, but is so broadly tuned that it does not appreciably affect the shape of the output wave. If any alignment adjustments are made on the preselector, it is important to see that the wave shape is not greatly changed.

► In some cases you may find that the band pass adjustments of the i.f. amplifier produce a wave having two rather pronounced peaks with a valley between. Check to be sure you are not trying to adjust for too wide a pass-band, as the receiver may be of limited fidelity. If the peaks are separated the proper distance for normal band passing, and the i.f. adjustments do not seem to make the top of the band pass wave more flat, then wait until the overall check has been made. You will usually find in such cases that the preselector selectivity is somewhat more sharp with the result that, when tuned exactly to resonance, it introduces a third peak between the two i.f. peaks. This peak "lifts" the valley, producing a more flat characteristic.

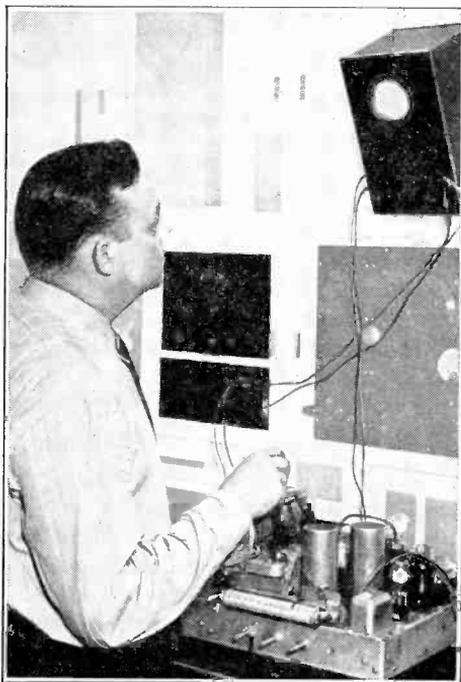
► Band pass adjustments normally will be made only on high fidelity receivers which are particularly designed for this purpose. However, you may get a request for improving the response characteristic of a receiver with variable selectivity, or some standard receiver. Let's see just what can be done in either case.

### Variable Selectivity Receivers.

These receivers have been covered elsewhere in the course. However, they are receivers where the selectivity and fidelity can be changed by introducing a variation of the i.f. amplifier band

width. The coupling between the transformers may be changed, loading may be introduced, or the primary and secondary may be detuned from the normal response frequency to give a flatter overall response. Two types of controls may be available, a switch type having two positions, or a continuously variable control.

► In any case, the normal adjustment is one of peaking the receiver with the control set in the maximum selectivity position. Ordinarily, the standard alignment procedure would be followed assuming the receiver is designed to



The wobulator at the left is connected to the receiver and a synchronizing connection is made to the horizontal input of the c.r.o. The vertical input is connected to the diode load of the second detector. As adjustments may be above the chassis, the connections to the set must be firmly made and of sufficient length to permit moving the set about. Hook-up wire is frequently more useful than test leads, as you can solder the wires to the desired points in the radio, thus being assured the connections won't jump off during alignment.

give reasonable fidelity in the broad selectivity-high fidelity position.

► Where there is any question about this, you can proceed with the use of a wobulator in the manner described in this lesson, with the control set in the high-fidelity position. In other words, first make the peak alignment, then move the control to its maximum fidelity position and proceed to band pass the amplifier.

Always remember that these two adjustments interlock. If you adjust to where you have the maximum in fidelity, you will probably find that when you return to the sharp tuning position you no longer have quite as much selectivity. It probably will be necessary to reach some compromise between selectivity and reasonable fidelity in these cases. Be sure the receiver owner understands this before you start this alignment procedure.

**Standard Receivers.** The ordinary receiver is usually designed to have high selectivity, so peak alignment is the only alignment which is practical in most instances. True band passing cannot be obtained unless the i.f. transformer windings are critically coupled. However, in some instances, the transformers have sufficient coupling to give somewhat broader response if the band pass alignment procedure is followed.

Obviously, it is a waste of time to band pass a receiver where the audio amplifier is incapable of better response. However, if the owner does desire better response, you can try band-passing.

In such cases, peak align the receiver as in the normal procedure. Then proceed to try to make a band pass adjustment (that is, decrease the capacity of one trimmer and increase the capacity of the other on the same i.f. transformer). In this case you are ac-

tually tuning one winding of each transformer above the proper frequency and the other winding of the same transformer below the proper frequency, so as to get a broader response. If the transformers are under-coupled, the sharp peak will drop, forming a broader, more rounded peak. Of course, this adjustment cannot be carried too far, for the loss in sensitivity and selectivity will be too great to permit anywhere near normal distant station response.

► Where the coupling is close to the critical value, more of a flat-topped, band-pass characteristic is obtained. Even so, it is rarely advisable to try to band pass such an ordinary receiver beyond the point where a pronounced valley begins to appear between the peaks. However, it is sometimes possible to load the i.f. transformer windings by placing resistors between 10,000 and 100,000 ohms across the secondary of each of the i.f. transformers. Use the *largest* resistor value which will give you the response you are trying to obtain. Smaller resistor values improve the response further, but at too great a sacrifice of selectivity and gain.

► Again, it is necessary to explain to the receiver owner the fact that more interference may exist, and fewer distant stations will be picked up when any kind of band pass adjustments are made on these sets.

## F. M. ALIGNMENT

Although a high sensitivity meter is usually used as an indicator for F.M. alignment, it is possible to make use of the c.r.o. and a wobulator, giving a visual alignment method which will show the exact i.f. and discriminator response characteristics.

Suppose we have to align an F.M.

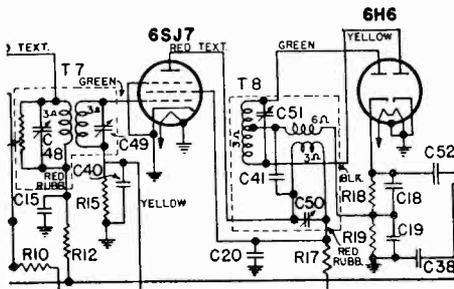


FIG. 39. The limiter-discriminator section of the General Electric Model GM-125 frequency modulation receiver.

receiver having a discriminator and limiter as shown in Fig. 39. Normally, the indicator is connected across the grid resistor of the limiter first. This is resistor  $R_{15}$  in the diagram. Then, after having aligned for a maximum indication here, the output indicator is moved to the output of the discriminator and the discriminator transformer adjusted. As you will recall from your study of frequency modulation, the

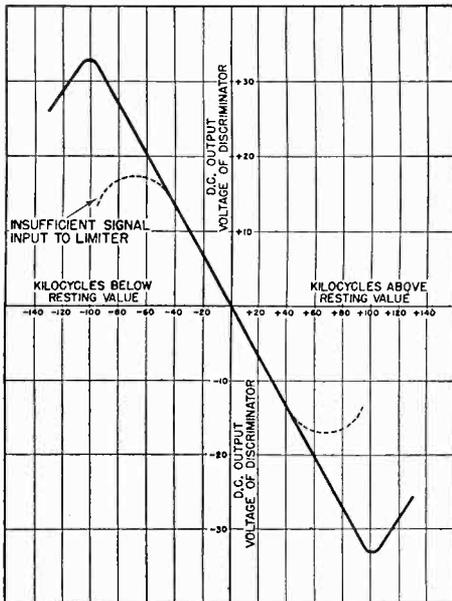


FIG. 40. The S curve shows how the output voltage of the f.m. discriminator varies with frequency deviation.

operating curve of this discriminator forms an S curve, when comparing frequency shift with the discriminator output. A typical S curve is shown in Fig. 40.

In order to use the c.r.o., the frequency modulated signal generator must be capable of covering the i.f. band width of 200 to 300 kc., which means that the sweep must be from 300 to 400 kilocycles wide. This requires a special frequency modulated oscillator, as the ordinary types previously discussed for A.M. radio receivers are incapable of going much beyond 40 kilocycles.

With this wobulator connected to the input of the first detector in the F.M. receiver, and set to the i.f. frequency of the receiver, the c.r.o. is then connected across resistor  $R_{15}$ . Usually a resistor of about 500,000 ohms is used in series with the c.r.o. lead to

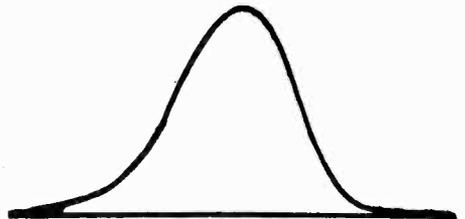


FIG. 41. The response curve of a typical f.m. receiver i.f. section. It does not look broad, but actually is, as the wobulator sweep is plus or minus 100 to 150 kc., so the back-trace line represents 200 kc. or more wide. (Plus or minus 100 kc. gives a band width of 200 kc.).

keep the c.r.o. from affecting the circuit action.

With the c.r.o. connected to the grid circuit of the limiter, the i.f. adjustments are made for a maximum peak response as shown in Fig. 41. This may seem surprising until you consider the fact that the back trace now represents perhaps 400 kilocycles, so the sides of the image in Fig. 41 are from 150 to 200 kilocycles apart. This definitely is

a very broad pass band even though the overall shape of the curve does not appear to be as flat-topped as one might expect. However, remember the band-pass is obtained by connecting resistor loads across the transformers and this does not give a flat-top band-pass characteristic.

After this adjustment has been made satisfactorily, the c.r.o. connections are then moved to the output of the discriminator. Connect across both resistors  $R_{18}$  and  $R_{19}$  of Fig. 39. One lead goes to the  $C_{52}$ - $R_{18}$  junction and the other to the chassis.

With the frequency modulated oscillator still connected to the first detector and tuned to the same frequency, you will now get a representation of the S curve. The c.r.o. sweep should be adjusted to where you get two of these curves, as shown in Fig. 42. Now, ad-

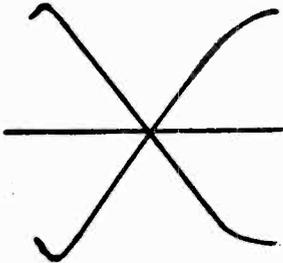


FIG. 42. This double S curve obtained at the output of the discriminator tells when the discriminator transformer is correctly adjusted.

just the trimmers on transformer  $T_8$ . As you adjust trimmer  $C_{51}$ , the curves will cross each other at different points. The correct adjustment of this trimmer is indicated when the curves cross about midway in a vertical plane. (This position is indicated by the horizontal line in Fig. 42.) Proper alignment of condenser  $C_{50}$  is indicated when the sides of the curves near the cross-over point are nearest to a straight line.

It is important that sufficient signal

be fed in to saturate the limiter. This level can be determined by increasing the output from the signal generator to the point where the amplitude of the cross-over curves no longer increase in height.

► Should the receiver be a combination F.M.-A.M. receiver, each section would be aligned separately in the standard manner. A typical output for such a receiver is shown in Fig. 43. Connect across  $R_4$  for aligning the amplitude modulation section and across the  $R_1$ - $R_2$  combination for the F.M. discriminator alignment.

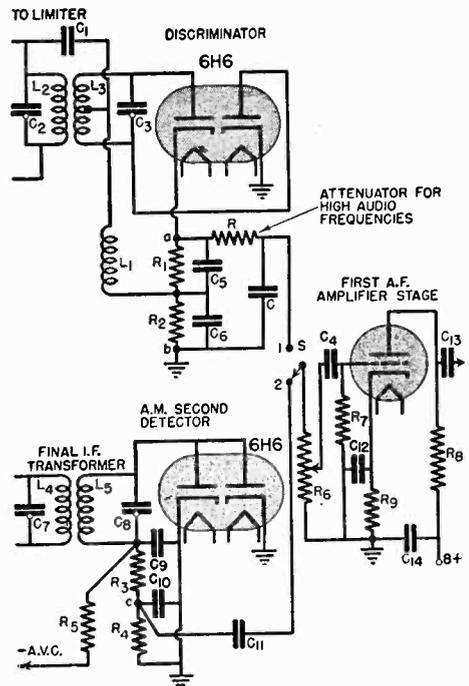


FIG. 43. The discriminator and second detector of a combination a.m.-f.m. receiver.

### AUDIO RESPONSE CURVES

You will recall that an audio response curve like Fig. 19 requires a number of measurements over the frequency band. This tedious procedure can be avoided by using an f.m. oscil-

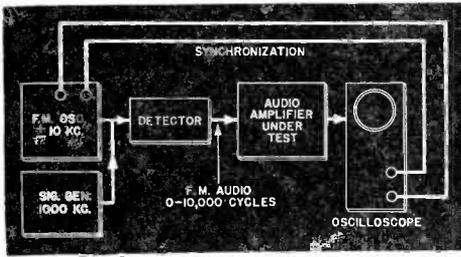


FIG. 44. How to connect a wobbled signal generator and a standard signal generator to a detector so as to obtain an audio sweep.

lator or wobblator to get an approximate curve.

The wobblator output is fed to the second detector, along with the output of another signal generator, as shown in Fig. 44. The two generators are adjusted for exactly the same frequency, so as to get a zero beat between them.

Now, as the wobblator output is varied over the sweep band, beat frequencies are produced by the detector between zero and half the sweep band width. If the sweep band is 20 kc., we will get audio frequencies from zero to 10,000 cycles.

As these frequencies are being swept over the audio band, synchronizing the c.r.o. will produce a tracing like Fig. 45. The wave runs from zero to 10,000 cycles, then repeats itself in reverse, so only half of the pattern is used. Also, the lower half (below line A-B) is ignored. The outline represents the audio response to a fair degree, provided the output of the wob-

blator is reasonably flat and provided the detector does not introduce frequency distortion.

The response curve for low frequencies below about 10 times the sweep rate is not accurate. Thus, for a sweep rate of 60 times per second, frequencies between zero and 600 cycles are attenuated, because the amplitude does not have time to build up before the beat frequency has moved on to the higher frequencies. However, over the rest of the band, the response curve is fairly accurate.

► You, as a serviceman, should now realize that the c.r.o. is an instrument that can help you in many ways. The thorough knowledge of the c.r.o. which you have gained from this lesson will not only enable you to make the specific service applications described, but will give you a valuable basic understanding of the many, many uses of this wonderful instrument throughout science.

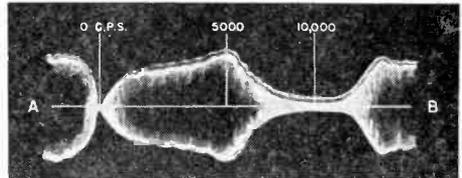


FIG. 45. By using the portion between zero and 10,000 cycles and ignoring the tracings below the A-B line, we have a pattern showing the frequency response of an audio amplifier. The pattern is not reliable below 600 cycles, but above this point is reasonably accurate.

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# Lesson Questions

Be sure to number your Answer Sheet 40RH-2.

Place your Student Number on *every* Answer Sheet.

*Send in your set of answers for this lesson immediately after you finish them, as instructed in the Study Schedule. This will give you the greatest possible benefit from our speedy personal grading service.*

1. What is the purpose of varying the voltage on anode  $P_1$  in Fig. 2?
2. Is the voltage to be analyzed normally connected to the *vertical deflecting plates*, or to the *horizontal deflecting plates*?
3. Why should the synchronizing voltage (SYNC control) be adjusted so the least amount giving a stationary image is used?
4. Suppose the sweep is set for 60 cycles, and as you trace for hum, *you get a two-cycle pattern*. Is the hum due to: *a*, cathode-to-heater leakage; *b*, pick-up from the filament wiring; or *c*, defective filter condensers?
5. Suppose only one-half a wave image is distorted. Does this show: *a*, even harmonic distortion; or *b*, odd harmonic distortion?
6. Which **two** of the following phase shifts will produce a perfectly circular pattern. (assuming equal amplitudes):  $45^\circ$ ;  $180^\circ$ ;  $90^\circ$ ;  $0^\circ$ ; or  $270^\circ$ ?
7. Is the c.r.o. sweep set to the *same* frequency as the wobulator rate, or is it set to *twice* this rate, to get an overlapped band-pass pattern?
8. If the peaks of the overlapped alignment pattern do not coincide, what condition is indicated?
9. When the beam is not being swept, it is necessary to prevent the spot from burning the screen of the c.r.o. tube. Name the control used to do this, and tell how it is adjusted.
10. In what directions are the i.f. trimmers varied to get band pass response, after the initial peak adjustment has been made?

Be sure to fill out a Lesson Label and send it along with your answers.



## KEEPING PROMISES

To have the reputation for always meeting obligations and paying debts is a most valuable asset in social as well as professional life. If you owe money, pay promptly when payment is due. If you owe allegiance to a person or organization, grant it. If you have given your promise to some one, keep it. To hedge on any type of obligation is to start a moral disintegration which may eventually ruin your chances for enjoying life and its successes. Failure to keep an obligation is a sin which has resulted in many a failure in life itself.

*J. C. Smith*

**HOW TO ELIMINATE HUM,  
SQUEALS AND MOTORBOATING**

41RH-2



**NATIONAL RADIO INSTITUTE**

**WASHINGTON, D. C.**

**ESTABLISHED 1914**

# STUDY SCHEDULE NO. 41

For each study step, read the assigned pages first at your usual speed, then reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind. Study each other step in this same way.

1. Hum as a Service Complaint. . . . . Pages 1-8

This section answers the important question: "What causes excessive hum?" A review of power pack troubles is followed by practical examples of troubles which may arise within the receiver itself. Be sure to notice how the serviceman can unwittingly introduce excessive hum, so you can avoid these headaches.

2. Localizing Hum. . . . . Pages 9-17

The type of hum automatically localizes the defective section, so we need only effect-to-cause reasoning, stage and part isolation. These procedures are given in detail for both steady hum and modulation hum.

3. Residual Hum Problems. . . . . Pages 18-21

Once in a while you may be called on to reduce the residual hum level in a receiver. Hum-bucking circuits or changes in design are required, most of which take considerable time. These procedures should be tried only after reaching an agreement with the owner, as results cannot be guaranteed.

4. Oscillations, Squeals and Motorboating. . . . . Pages 21-28

Oscillations occur only when there is a feedback path, a feedback with the proper phase and sufficient feedback energy. The cure is usually one of reducing the feedback energy or destroying the path. You will study the paths and then take up the methods of localizing the offending stage and curing the trouble.

5. Answer Lesson Questions and Mail Your Answers to N. R. I.

6. Start Studying the Next Lesson.

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# HOW TO ELIMINATE HUM, SQUEALS AND MOTORBOATING

## Hum as a Service Complaint

**H**UM is a very common service complaint. However, hum is not a trouble which is "there" or "not there". Only a battery operated set can be completely "humless"; there is always some residual hum in even the best receiver operating from a power line or power converter. Your object, therefore, in servicing a set for hum is not to remove the hum altogether, but to reduce it to an acceptable level. This lesson will teach you the many ways in which this can be done.

You can learn to recognize hum very easily. Just listen carefully to the output of any properly operating power line receiver when the surroundings are quiet and no program is coming through the loudspeaker. Turn the volume control to zero to eliminate stray noises and other signals. You will soon become conscious of a low-pitched humming sound if your ears can hear low frequency sounds. If the set is a standard a.c. receiver using full-wave rectification and operating on a 60-cycle power line, you will be hearing 120-cycle hum. (It is called 120-cycle hum because the fundamental frequency is 120 cycles; there are harmonics present also.)

Next, connect a headphone in series with a 10,000 to 50,000 ohm resistor, connect this combination across the filament terminals of any tube in an a.c. receiver except the rectifier tube, and slip on the headphones. You will

hear 60-cycle hum. You should learn to distinguish the difference between 60- and 120-cycle hum, because this will be important in recognizing the source of hum in service work

► Usually the hum level is considered satisfactory if the hum is not noticeable when the device is in normal operation, but this standard allows wide variations in the amount of hum present. For example, an outdoor public address system can have more hum than a radio receiver, because in its normal use surrounding noises will make hum less noticeable.

Incidentally, there may be a considerable difference between the amount of a.c. hum *voltage* in the output circuit and the amount of hum *sound* which actually comes from the radio loudspeaker. Midget or table type receivers give relatively little hum output, because the design of the speaker and output transformer and the limited amount of speaker baffle do not favor the passage of low frequencies. As a result, half-wave rectification and less efficient filter designs are permissible in these sets because even relatively large amounts of hum voltage are not appreciably reproduced as hum sounds. On the other hand, even an extremely small hum voltage at the voice coil of a loudspeaker in a high-fidelity receiver may produce a relatively large amount of hum sound, both because the system is capable of reproducing

low frequencies, and because the speaker baffle or speaker cone may have a response peak at low frequencies or have resonant points at or near the hum frequency.

Practical experience is the only guide to the amount of hum normally found in any particular kind of set. Make a point of noticing the hum level of radio receivers you service for other complaints, so you'll learn to know the amount of hum to expect in various models.

► Remember—the amount of hum permissible in a device depends a great deal upon the listener. Some ears do not hear low frequency sounds well. If you find you can't hear hum unless you bring your ear quite close to a loudspeaker, be careful about handling customer complaints—your customers may hear hum much more readily than you do.

Usually you will be asked to service a set for hum only if some defect or change in the circuit has raised the hum level to an abnormal degree. Surprisingly large amounts of hum may be tolerated by the receiver owner until his attention is called to the fact that the receiver is humming, or until some defect suddenly causes an even louder hum. But once he has become definitely conscious of the hum, he may then become so critical that he listens for the hum instead of the program.

► Incidentally, when you are working on a receiver with an elusive case of hum, you may yourself reach the point where even a normal hum level seems excessive. Getting away from the receiver for a while will frequently restore your sense of proportion. This will also often work for the customer—just keeping the radio in your shop for an extra day or two, so that he begins to forget the hum annoyance, may satisfy a customer if his radio is apparently normal and the

usual hum elimination procedures do not lower the hum level. Of course, if the customer insists, and is willing to pay for the large amount of time that would be necessary, then there are some elaborate procedures which can be followed. These will be described in this lesson.

Before we study service techniques for reducing hum, let's review quickly the causes of excessive hum in radio circuits. This will make it easier for you to see just what must be done to localize and eliminate hum troubles.

**Power Lines.** We will assume a standard 60-cycle power line is used in the examples in this lesson. Should some other frequency line be used, then the frequencies will be changed. For a 25-cycle line, the frequencies will be 25 and 50 cycles instead of 60 and 120 cycles in these examples.

## POWER PACK TROUBLES

Most a.c. receivers have a power supply like Fig. 1, in which an a.c. voltage from the power line is rectified. The output of the rectifier contains a large amount of a.c. ripple, so it is passed through a filter before being used to supply the radio. Of course, if there is any defect in the filter, the hum level in the receiver will be abnormal. Let's see what can go wrong with a filter—starting with the condensers, as they are the most likely sources of trouble.

**Filter Condensers.** It is possible for a filter condenser to open, to short circuit, to develop leakage, to lose capacity, or to develop a high power factor. (A good condenser will have a low power factor, or low series resistance. As this resistance increases the capacity becomes less effective.) When a condenser is *open* the effect is the same as if the condenser were not present at all, so the hum level will increase. A *short-circuited* con-

denser will kill the receiver altogether instead of causing hum.

*Leakage* in a condenser has the same effect as connecting a resistor in parallel with the condenser. As this does not affect the capacity, a leaky condenser by itself will not cause hum, but may cause another part to do so, as we will show later.

*Loss of capacity and high power factor* may be caused in wet electrolytic condensers by evaporation of the electrolyte and in dry electrolytic condensers by drying out of the electrolyte. Either defect produces more hum, particularly if  $C_2$  (Fig. 1) is the offender.

**Choke Coil.** Excessive d.c. current through the choke coil, such as may be produced by a leaky output filter condenser, will saturate the choke coil core and thus reduce the effective inductance of the coil. Reducing the choke inductance of course lowers the filter effectiveness, so the hum level will rise. In this case the condenser is the actual cause of trouble but it is the change in choke characteristics that causes the hum. Replacing the condenser will allow the choke coil to resume normal operation.

As you know, choke coils have an air gap to prevent core saturation. The gaps of well-made chokes are filled with some non-magnetic material such as paper, cardboard, copper or brass spacers, etc. However, the gap of an inexpensive choke often has nothing in it; once in a while the gap of such a choke will close up, if the clamps holding the core happen to loosen. This permits core saturation which lowers the choke coil inductance and raises the hum level.

If some of the coil turns short together, the inductance of a choke will also be reduced, increasing hum. This difficulty is not common with ordinary choke coils, but does happen occa-

sionally when a speaker field is used as a choke.

**Rectifier Tube.** A full-wave rectifier normally supplies current with a ripple frequency of 120 cycles. If the rectifier tube sections become unbalanced, so that one section passes more current than the other, the fundamental ripple frequency will change to 60 cycles because the output is more nearly like that of a half-wave rectifier. This often causes hum in the set, because a filter which is adequate for 120-cycle ripple may be unable to filter the 60-cycle ripple sufficiently. (Remember, a poor socket connection

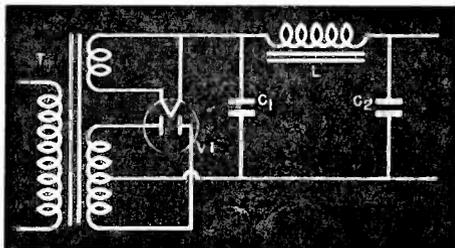


FIG. 1. Although this standard a. c. power pack is the type most commonly encountered, the same filter troubles described in the text will be found in a. c.-d. c. packs.

or a defective power transformer, in which only half the high voltage secondary supplies voltage, will also cause half-wave rectification and create hum. If the tube tests good, investigate these points.)

Sometimes a rectifier tube becomes gassy, due to a faulty tube or because of excess current flow. The high concentration of ions makes the tube continuously conductive, so it becomes an imperfect rectifier and passes some current on the reverse cycle. This again introduces 60-cycle a. c. in a circuit designed to eliminate only higher frequencies and may produce hum. Also, an r.f. oscillation may develop, producing hum modulation as we shall learn later. When you see the rectifier light up with a purple-

pink glow, and know it is not a mercury vapor tube, you're justified in assuming it is gassy. To cure the condition you must usually replace the tube. If there is an excess current flow, it must be reduced by replacing the defective part (usually leaky filter condensers).

**Tuned Filters.** Some of the older receivers have tuned filters. These are adequate hum eliminators as long as they stay tuned to the proper frequency, but are very difficult to retune if they drift off. If you find something wrong with a tuned filter nowadays, the easiest thing to do is to remove the tuning feature so the filter is no longer resonant, and replace the filter condensers with modern, high capacity electrolytic condensers. The hum reduction of the new

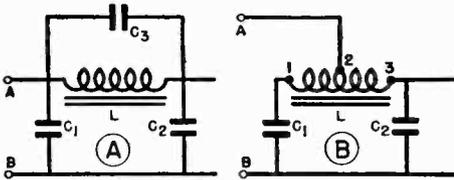


FIG. 2. Two common tuned filters.

filter will equal or better that of the tuned circuit.

To change over tuned filters of the type shown in Fig. 2A, remove  $C_3$  and increase the capacity of  $C_1$  and  $C_2$ ; to change over one like that in Fig. 2B, move the wire coming from terminal A from point 2 to point 1 and increase the capacity of  $C_1$  and  $C_2$  each to 8 mfd. or more.

## RECEIVER DEFECTS

Quite apart from the power pack, there are many possible sources of hum in the receiver itself. One of the most common is cathode-to-heater leakage in any tube in which the filament is operated from a.c. voltage directly. Normally, the cathode of such a tube is insulated from the filament.

However, the cathode may short to the filament or leakage may develop between them. If either happens in a set in which the cathode is connected to the chassis through a bias resistor, a hum voltage may exist between the cathode and the chassis. (Cathode-to-heater leakage is not troublesome if self-biasing is not used and the cathode is grounded directly, as there is then no way of developing a cathode-chassis voltage.)

Cathode-to-heater leakage is most likely to occur in a.c.-d.c. receivers, where the difference in potential between a tube cathode and its filament increases as you progress along the filament string (see Fig. 3). For this reason, the first audio tube (the tube most sensitive to hum) is always placed first in the filament string of such sets, where the potential difference is least, to minimize hum.

A tube tester may or may not show up cathode-to-heater leakage, depending on the sensitivity of its leakage indicator. Try another tube when there is any doubt.

**Decoupling Circuits.** Figure 4 shows typical decoupling circuits used

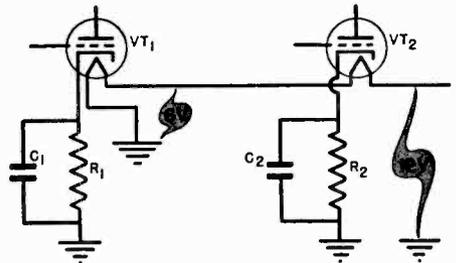


FIG. 3. The filament-to-ground voltage increases as you move along the filament string in a.c.-d.c. receivers.

in plate and grid circuits, particularly in audio stages. The filter  $R_4-C_4$  is used in the plate supply. The combination normally forces signal currents through  $C_4$ , keeping them out of the power supply, and also acts as an additional filtering section between

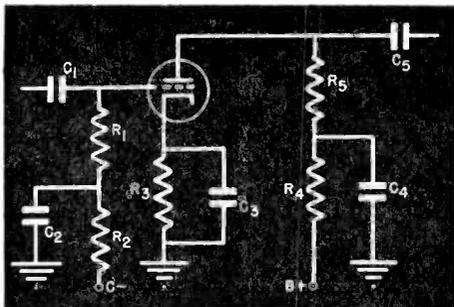


FIG. 4. Plate and grid decouplers.

the power supply and the tube. If  $C_4$  should open, a certain amount of hum may get into the circuit from the power supply and be fed through  $C_5$  to the following tube. It is also possible for signal currents to get out of the circuit and enter other circuits through the power leads; this may cause motorboating, or other difficulties.

► Similarly filter  $R_2$ - $C_2$  prevents signals from getting into the  $C$  supply and keeps hum voltages from getting into the grid circuit. If  $C_2$  opens, hum voltages may be applied to the control grid.

► We don't ordinarily think the bypass condenser  $C_s$  has much to do with

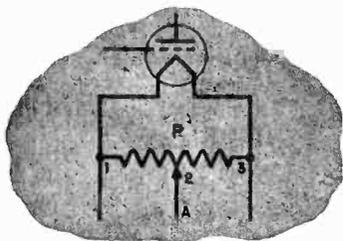


FIG. 5. How a hum balancer is connected to give the effect of a filament center tap.

the hum level. However, if there is even a very slight amount of cathode-to-heater leakage in the tube, there will be a small 60-cycle current flow from cathode to chassis. This will not cause much voltage drop across the low-reactance condenser  $C_s$  as long as

it is in good condition; but, if the condenser opens, the cathode-to-chassis impedance will rise to the value of  $R_s$ , and this increased impedance will cause a considerably higher hum voltage drop. Since this voltage will be between the grid and cathode, it will enter the stage and be amplified.

**Hum Balancers.** In a few radios, you may find a triode tube in the audio amplifier. The grid return to such a tube is made to some point representing the center tap of the actual tube filament. Sometimes a center-tapped resistor or potentiometer will be used as shown in Fig. 5. The potentiometer  $P$  is normally adjusted so that hum is at a minimum. However, if this potentiometer burns out or becomes disconnected at *either* terminal 1 or terminal 3, then it effectively just connects terminal A to one side of the filament. This will introduce hum, as will improper adjustment of the tap.

**Grid Circuits.** If the grid circuit of a tube (particularly a first audio

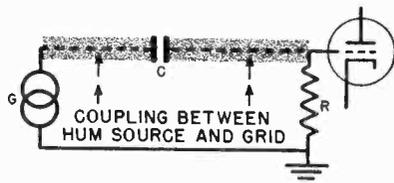


FIG. 6. Capacity coupling exists between grid wiring and the hum source.

tube) increases in impedance or becomes open, hum is practically certain to result. To see why, let's look at the circuit in Fig. 6. Here the stray hum field is represented by a generator  $G$ , and the capacity between the grid wiring and the hum source by a condenser  $C$ . Since  $C$  is of course very small, it has a high reactance.

Since the hum generator feeds into the voltage divider formed by  $C$  and  $R$ , the hum voltage divides between the reactance of  $C$  and resistance of  $R$ .

The larger  $R$  is, the more hum voltage appears across it between grid and ground, and the greater the hum output.

When the grid input circuit has low impedance ( $R$  is small), it takes a considerable amount of stray hum voltage to develop an appreciable hum signal. On the other hand, when the grid circuit has high impedance, very small amounts of stray hum voltage may cause trouble. Sometimes a set

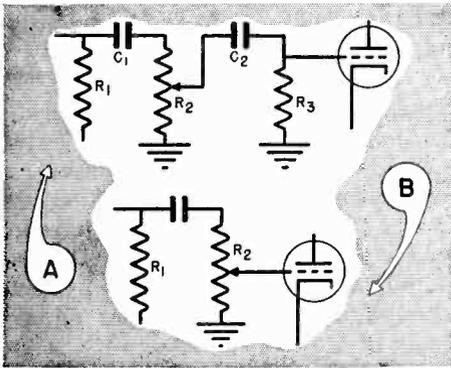


FIG. 7. Two methods of connecting volume controls. A defective control can cause hum in either instance.

develops hum because of an increase in the grid circuit impedance; the most common cause of such an increase is a defective volume control. As the control wears, a poor contact develops between the rotating arm and the strip; causing an increase in the resistance between the grid and ground. In many types, the strip itself may wear out; this, of course, increases the resistance a great amount.

This effect may even occur when the volume control is decoupled from the grid circuit, as in Fig. 7A. Here, the a.c. impedance between grid and ground is formed by  $R_3$  in parallel with  $C_2$ - $R_2$ , with the volume control in turn shunted by  $C_1$ - $R_1$ . Since the condenser reactances are negligible, we have  $R_3$ ,  $R_2$ , and  $R_1$  in parallel when  $R_2$  is set for maximum volume. If the

control opens circuits at the ground end, the a.c. impedance between the grid and ground will increase because it will no longer be shunted by  $R_2$ .

In a circuit like that shown in Fig. 7B, where the volume control is right in the grid circuit, a defective control can break the d.c. path. The a.c. impedance rises to a high value, causing a loud hum. This open also produces a "floating grid" (no bias), and will frequently cause distortion in the set output as well as hum. In either case, the volume control will not properly control volume, which is an additional clue leading you directly to the trouble.

## INDUCTIVE COUPLING

Iron core transformers and chokes have leakage magnetic fields which exist in the space about the devices and frequently travel through the chassis. Most of them use soft iron magnetic shields to limit this field, but even so audio transformers must usually be placed well away from power transformers and filter chokes to avoid hum pickup.

If you replace an iron core choke or transformer, you must take special precautions not to introduce hum in the receiver—unless you can get an exact duplicate, in which case there is no problem. If you must use one of the "universal" replacement types, or some other type having similar electrical characteristics, you may find after making the installation that the leakage magnetic field of this particular part is different, or the shielding poorer, so that some magnetic coupling exists to another circuit or part and hum develops. The amount of hum increase will usually be slight, but it may be noticeable to the owner of the receiver.

Frequently a simple change in the position of the part, such as rotating it or tilting it at an angle, will clear up

this trouble. Usually, however, there are so many wires connected to such parts, and the mounting space is so limited, that this is not practical; if so, try using a shield. Soft iron is the best shield at these low frequencies. Make a practice of saving the cases from defective transformers for possible future use as shields for others.

Since both inductive and capacitive coupling can exist between wires, any wires carrying large amounts of a.c. current must be kept away from grid and plate leads to prevent hum. Often the filament leads are twisted together so that induction from them to neighboring wires is minimized.

Because of inductive and capacitive coupling, you may accidentally raise the hum level by moving the wires about while hunting for trouble or while making a replacement. Be careful about this. Before moving wires or parts, make a note of their exact locations, and return them to these positions when repairs have been made.

Set owners often tuck excess lengths of line cord into their radios. Watch out for this—it may bring the a.c. cord close to a grid, producing enough coupling to cause hum.

### REVERSED HUM-BUCKING COIL

If you disconnect a speaker for any reason, be very careful when you connect it up again. It is very easy to make the mistake of connecting the hum-bucking coil up backward—even factory assembly workers have been known to do so.

This hum-bucking coil is a small coil wound next to the field coil. The field coil usually carries a certain amount of a.c. (particularly when used as a choke coil in the power supply) and induces some hum voltage in the voice coil, making the loudspeaker produce hum. The hum-bucking coil is arranged so that when it is properly

connected to the voice coil, the voltages induced in the hum-bucking coil by the field cancel those induced in the voice coil. This prevents speaker field induction from causing hum. However, if the hum-bucking coil terminals are reversed with respect to the voice coil terminals, then twice as much hum current flows through the voice coil and the hum level increases considerably. Figure 8 shows how the hum-bucking coil is usually represented on diagrams. Here,  $L_1$  is the speaker field,  $L_3$  the voice coil, and  $L_2$  the hum-bucking coil.

Even without a diagram, you can always tell when a hum-bucking coil is used by tracing the connections

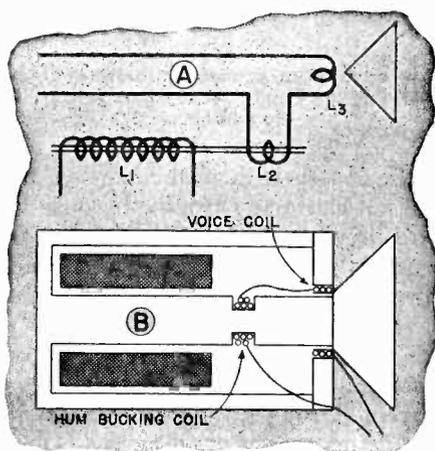


FIG. 8. The hum-bucking coil is placed between the voice coil and field, and is so connected that the hum voltage induced in it and the voice coil by the field are out of phase and cancel.

from the output transformer to the voice coil. If the two leads from the voice coil go directly to the output transformer, there is no hum-bucking coil. However, if one of the leads from the voice coil and one of the leads from the output transformer secondary go to a small coil in the field coil enclosure, a hum-bucking coil is used.

## POOR CONNECTIONS

It is quite possible for a poor connection to cause hum, particularly when the joint is common to more than one circuit, as in Fig. 9. Here the grid resistor  $R$  and one filament terminal must both be grounded. As it is difficult to solder to the chassis lug  $G$  with parts in the way,  $G$  is connected to terminal 7, and the resistor is then connected between terminal 7 and terminal 5. Now, as long as the ground

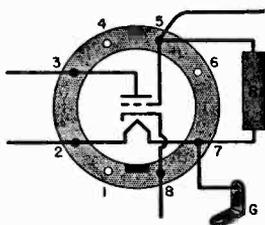


FIG. 9. A poor connection between 7 and grounding lug  $G$  introduces a hum voltage into the grid circuit.

at the chassis is in good condition, this will not cause any trouble because the wire between 7 and  $G$  is usually too short to have appreciable resistance. However, suppose a poor connection develops at  $G$ . This is the same as adding a small resistor in the circuit between 7 and  $G$ . The a.c. filament current flowing through the resistance of this poor connection causes a voltage drop; tracing from the grid to ground, you can readily see that this a.c. voltage drop will be introduced into the grid circuit. Feeding directly into the grid circuit like this, even a small hum voltage may produce a very loud hum. You can cure such a condition, after isolating its source, by resoldering the connections or by using a different ground for the grid return.

Hum may also be introduced into a circuit by a poor ground connection to a shield. A wire in a circuit critical to

hum pickup is often shielded—particularly a grid lead which must be run any great distance. The shield around such a wire must be grounded to be effective; if the ground connection is poor or non-existent, the effect of the shield is lost and hum can be introduced into the shielded wire.

Sometimes the shield must be grounded at both ends, as in Fig. 10, to reduce the hum level. A poor ground at either end of the shield permits the shield itself to pick up hum voltages and transfer them to the wire within.

## MECHANICAL HUM

All the examples so far discussed are classed as electrical hum, because they cause a hum voltage which is reproduced as sound by the loudspeaker. Hum may also be caused mechanically by a vibrating part which produces sound directly. Almost invariably, mechanical hum is

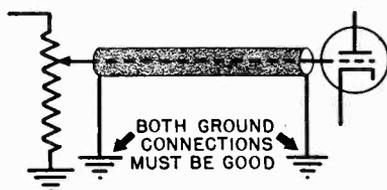


FIG. 10. Watch for poor grounds.

produced by an audio or a power transformer whose laminations are so loose that the transformer core can vibrate under the influence of the varying flux.

The source of mechanical hum can easily be discovered by careful listening. You can remedy the condition just as easily, either by tightening the clamping bolts or by driving a small wedge between the transformer laminations. Either method will usually secure the core so that it cannot vibrate.

# Localizing Hum

The same general procedures are used to locate the source of hum as are used to localize any other radio trouble. Your first step should be to confirm the complaint. Next, use effect-to-cause reasoning—which can eliminate many localizing steps. For example, the manner in which the hum is reproduced will let you determine at once the section of the receiver in which the hum originates.

If the hum is produced steadily, whether a program is tuned in or not, then the hum *must originate in the power supply or in the audio amplifier*, since an audio frequency voltage cannot travel through the r.f. amplifier by itself. Of course, the hum will be more evident when you are tuned off a station, because then there is no program to mask or drown out the hum.

On the other hand, if the hum is tunable (is heard only when a signal is tuned in) then we have *modulation hum, due to hum voltages entering a defective r.f. or i.f. stage, or due to modulation outside the receiver*.

Let's repeat these two important facts, to fix them firmly in your mind:

*Steady hum* originates in the power supply or in one of the a.f. stages.

*Modulation hum* originates outside the receiver or because of modulation in an r.f. stage.

Since steady hum is the complaint you'll meet most often, we'll study it first. Modulation hum will be taken up later in this lesson.

## STEADY HUM

Let's assume the power line frequency is 60 cycles. Suppose you are asked to service a standard a.c. receiver with full-wave rectification, and a case of steady hum. You know at once that the cause lies somewhere in the power supply or a.f. circuits.

Next, ask yourself some mental questions about the operation of the receiver.

Is the hum accompanied by any other complaint? How loud is the hum? Is it 60-cycle or 120-cycle hum?

The answers to these questions will help you greatly in localizing the source of hum. Let's see what some typical answers reveal.

▶ A steady 120-cycle hum of medium loudness, accompanied by weak reception and possibly distortion, may indicate the input filter condenser is open, has lost capacity, or has developed a high power factor. Any of these conditions will lower the operating voltages, causing weak reception. The fact that the hum is only of medium loudness shows the following choke-condenser sections are still effective.

The same combination of defects may also mean the output filter condenser is leaky. This causes an excessive voltage drop in the choke, and therefore lowers the operating voltages supplied to the set.

You can be reasonably certain a filter condenser is causing this sort of hum if you find a condenser with a swelled-up container, or one which grows hot after a few minutes' operation—or if you discover a deposit of white chemicals about the vent on a wet electrolytic. Replacing a suspected condenser with a good one and listening to see if the hum is reduced is often a simple way to confirm your diagnosis. If you believe the condenser is open, has lost capacity, or has increased in power factor, shunt it with a good one (watching polarity, of course, if you use an electrolytic). If you suspect leakage, however, you must disconnect the original conden-

ser before trying another one in its place.

The choke coil will probably overheat if you have a leaky output condenser. However, this is not a very helpful clue, since it's hard to tell just how much heat should be developed in a choke coil or speaker field. Speaker fields are frequently quite warm to the touch, even during normal operation. If you know the voltage drop which should exist across the field or choke, and measure a voltage which shows that the drop is considerably higher than normal, then you do have an indication of excess current through the field or choke coil.

▶ A very loud 60- or 120-cycle hum, accompanied by weak reception or a dead receiver, usually indicates an open grid circuit, particularly in the first audio stage. The hum frequency depends on the source of the stray field; the power transformer would induce a 60-cycle voltage while the filter choke would cause a 120-cycle hum. Watch for grid caps off tubes or on the wrong tubes in such cases. The hum will be especially loud if the open is near the grounded end of the input device, because the parts and leads still connected to the grid will help pick up stray fields.

A very loud 120-cycle hum with a loss of low frequency response, possibly accompanied by oscillation, motorboating, or distortion, usually means an output filter condenser is open or has lost capacity.

▶ A 60-cycle hum, in a standard receiver using full-wave rectification, usually indicates cathode-to-heater leakage in an audio tube. The loudness depends on which tube has the leakage, since tubes near the input of the amplifier cause more hum than those farther along in the amplifier.

Remember that a hum voltage is amplified just like a signal voltage, so even a small amount of hum origi-

nating in the first audio stage can cause more hum output than a large amount of hum in a tube farther along in the amplifier. From this you can see, except for the power supply, the first audio stage is the one most likely to be the source of abnormal hum.

▶ A push-pull output stage will tend to cancel any hum voltage coming from the power supply to that stage alone. (The two tubes draw equal plate currents in opposite directions through the output transformer, so any ripple introduced in this plate supply will cause opposing fluxes in the output transformer which tend to cancel the hum.) Many receiver manufacturers, taking advantage of this fact, obtain a higher plate voltage by connecting the plate supply for the push-pull stage at a point in the filter circuit where there is less filtering. This is safe enough as long as the push-pull tubes stay balanced, but the hum level will rise as soon as they become unbalanced.

Finding tubes which will balance is often something of a problem. Since the push-pull tubes are power tubes and so draw high currents, a tube tester will often not show up an unbalance unless the tubes are radically different. Sometimes the only way you can find two tubes which draw approximately equal plate currents, and so minimize hum, is to keep trying tubes in the set or to insert a plate current meter in each tube circuit.

**Defective Stage Isolation.** Almost always, proper effect-to-cause reasoning and a few tests will lead you immediately to the source of steady hum. This is particularly true of modern receivers, where there are rarely more than two audio stages and usually only two or three filter condensers. These few items can be checked quickly and the source of hum located in a few minutes. However, if there is some rather unusual steady hum

condition (particularly in an elaborate receiver), some of the following methods of further localization may have to be used. Before trying any of these methods, tune away from any signals or turn the volume control to zero volume, so you can concentrate on the hum.

► The stage interruption test is an easy way to localize the hum-producing stage. This method consists of blocking the circuits, one at a time, to determine which is causing the hum. To apply the method to an a.c. receiver, start at the input of the amplifier and pull out the tubes one at a time, moving toward the output each time. (Of course, you must replace each tube before pulling out the next.) If pulling out a particular tube makes the hum disappear, then the hum is originating in that stage or in the preceding coupling device.

If you notice that pulling out tubes one at a time makes the hum decrease each time, but does not stop it altogether, then probably the hum is getting into all the stages simultaneously. This immediately indicates that the trouble is in the power supply common to all the stages.

The tubes in universal receivers can't be pulled out to perform the stage interruption test. It is possible to short circuit the input of each stage, however, and achieve the same result. If one end of the input part (across which the signal appears) is connected to the set chassis, you can short it out by touching the control grid terminal and the chassis with the probes of a test lead. Move from stage to stage, toward the output. However, you should not do this if bias sources are placed in the grid circuit between grid and chassis, because your test lead will short out the bias. (An example is shown in Fig. 11, where  $VT_3$  gets bias from the tapped choke  $L_4$ .) In-

stead, hold the test lead right across the input device itself ( $R_6$  for example). This blocks hum signals from preceding stages but will not stop hum coming from the bias source, such as might be caused by an open  $C_7$ .

If the set chassis is not part of the circuit, then you must identify the B— return lead and short the grid terminal to it, or else short across the input part itself.

► A signal tracer is very useful in localizing hum sources. If the tracer has an audible output indicator, all you have to do is connect it to the output of each stage in succession and listen for the hum. Moving from input to the output, the first stage you encounter having hum is the source. Remember, this hum will get louder as you move toward the set output. If the signal tracer has only meters or electric eye indicators, with no provision for phones or loudspeaker, then you may find it more difficult to locate a relatively low-level hum source.

A pair of headphones in series with a blocking condenser can be used as a signal tracer, moving from input to output in the a.f. stages.

► There are a few precautions to observe in using either a signal tracer or a pair of headphones. Either device may be more sensitive to hum voltages than is the amplifier. Since the headphone is bound right to your ear, hum levels will always sound louder than those normally produced by the loudspeaker, so even the normal residual hum level may sound high when you are listening directly to the receiver stages. Be careful not to confuse this hum level with the abnormal hum for which you are looking.

► The c.r.o. is useful as a signal tracer if the hum level is high enough to give a noticeable deflection. By using a 60-cycle sweep, and noticing whether you have a one- or two-cycle

pattern, you can tell whether you have 60- or 120-cycle hum.

**Defective Part Isolation.** Once the defective stage has been isolated, hum localization is relatively easy. Naturally, the first step is to test the tube in the defective stage, or try a new one. If this gives no result, a careful check of the grid, plate and other electrode circuits will show the part which is causing the trouble.

► If there is any evidence that the wiring has been changed about during previous servicing, try shifting the wires in that stage with a non-metallic rod. If you find any wire causes variation in the hum level as it is moved, move it carefully to the position of least hum.

It is usually possible to break circuits in a manner which will help in localizing hum. For example, you may hear hum when a signal tracer is connected at terminal 5 of Fig. 11. You cannot tell whether this hum originates in the grid circuit of  $VT_3$  or in the plate circuit of  $VT_2$ . By disconnecting condenser  $C_6$ , however, and listening between terminal 5 and the chassis, you can definitely tell where the hum is getting in. If you hear it between 5 and ground with condenser  $C_6$  disconnected, then it must be getting into that grid circuit; the most logical cause of this would be an open condenser  $C_7$ . (Notice that  $C_7$  and resistor  $R_7$  are used as a decoupling filter in this grid circuit.) Of course, disconnecting  $C_6$  effectively raises the grid circuit impedance by removing the shunting effect of  $R_5$ , and so may make the residual hum level rise at the  $VT_3$  grid. Remember to make allowances for this.

Disconnecting  $C_6$  while listening to the output of the set will also help point out the hum source. If the hum decreases or vanishes, then it must originate in some previous circuit; if it remains or increases, it is being devel-

oped in the output circuit, the power supply, or the loudspeaker.

► You can connect a signal tracer across a possible defective part and listen for excessive hum. For example, if you are suspicious of condenser  $C_7$ , connect a signal tracer between terminal 8 and chassis. Any appreciable hum at this point indicates an open condenser, because a good condenser would practically short circuit hum voltage between these two points. Similar use of the tracer will show you whether condensers  $C_{10}$  and  $C_8$  are defective.

► In the same way, you can separate the circuits of  $VT_1$  and  $VT_2$  and discover which is causing hum by unsoldering condenser  $C_4$ . Again, remember to make allowances for a possible rise in hum caused by disconnecting the shunting resistor  $R_2$  from the grid circuit of  $VT_2$ . If the hum source is in the  $VT_2$  grid circuit, the most logical suspect would be the volume control.

► To separate circuits which are transformer coupled instead of resistance coupled, disconnect the primary winding and leave the transformer connected to the following grid circuit. Assuming the hum is present at the grid, and remains with the primary disconnected, the transformer is probably picking up the hum inductively. This means you will either have to move the transformer to a different location or shield it.

**Practical Hints.** When filter condensers are mounted in a common block, you will frequently find that only one or two of the condensers are defective. However, it is best to replace them all, because when one condenser in a block goes bad, the others will usually soon follow. (This is not so true of condensers that are separated from each other.) Many servicemen make a practice of replacing electrolytic condensers rather generously, particularly if the receiver has

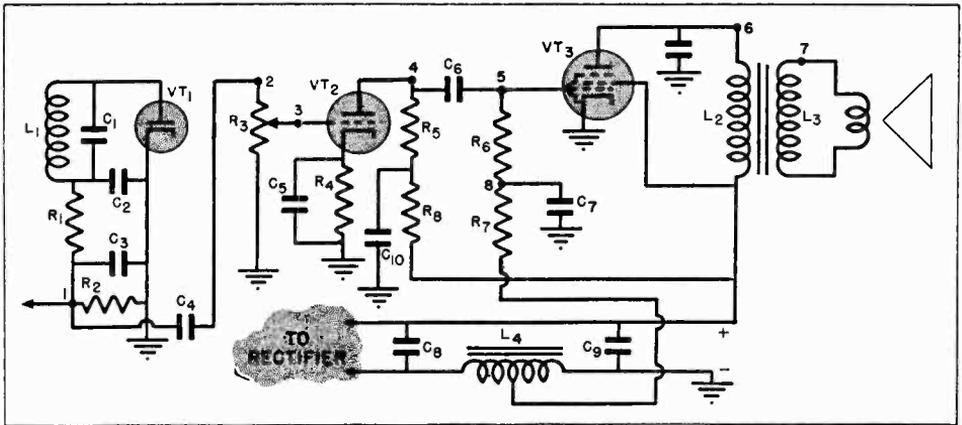


FIG. 11. Here are the testing points for localizing hum.

been operating for several years with no condenser replacement. It is gradually being recognized that condensers, like tubes, can wear out.

When replacing condensers, be very sure to make the proper connections. See that the replacement condenser has ratings like those of the original, and that the replacement is connected with the proper polarity and between the right points.

► An unusual source of hum is found in some receivers in which the filter condenser block is made up of dry electrolytic condensers contained in a waxed cardboard container, and is mounted on the set chassis by means of a metal strap around the block.

If the filter choke is in the negative side of the circuit, the negative lead of one of the condensers will not go directly to the chassis (see Fig. 12A). If leakage develops between the ungrounded negative terminal of the filter condenser and the grounded mounting strap, the choke coil will be shunted by the leakage resistance  $R$ . This will reduce the effectiveness of the choke coil and introduce hum.

If the set has a common bus or lead as the B— return so the set chassis is not a part of the circuit (see Fig.

12B), none of the negative leads of the filter condensers will go to the set chassis. Leakage between any condenser section or lead and the chassis through the strap will set up hum currents in the chassis between the leakage point and the  $C_3$  return, thus providing chassis voltage drops which may link with other circuits.

To stop hum in circuits like these, either replace the condenser block or

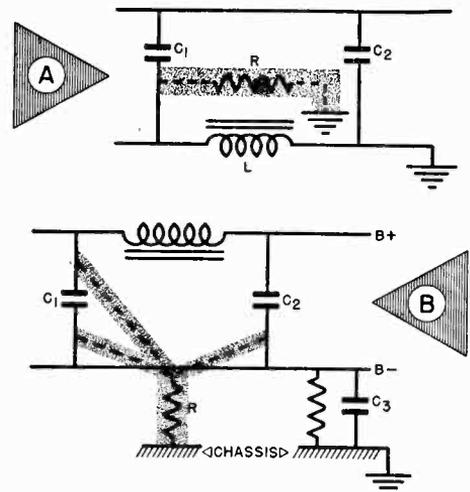


FIG. 12. Leakage to chassis through the cases of filter condensers is a source of hum in many a. c.-d. c. receivers.

cut the strap and make the condensers self-supporting.

## MODULATION HUM

Modulation hum develops in the r.f. amplifier or outside the receiver. Since an audio signal will not travel through the r.f. or i.f. stages by itself, it must be mixed with or modulate an incoming carrier to cause hum. Hum of this sort is more evident when a station is temporarily not modulating, as during the silent period between programs, *but you can hear it only when you are tuned to a station.*

The fact that mixing is necessary for modulation hum to exist means that there must be both a hum voltage and a curved tube characteristic in the stage where the hum is introduced. When an r.f. amplifier has a linear Eg-1p tube characteristic, the presence of both r.f. and hum voltages will not produce hum. If you were to take a cathode-ray oscillograph picture of the plate current when both the ripple and r.f. components exist in a stage operating as a linear amplifier, the pattern would be as shown in Fig. 13A. As you can see, the hum signal is varying the r.f. signal, but the amplitude (height) of the r.f. pulses remains exactly the same. (That is, if you measure the distance between points  $x$  and  $y$  and compare it with the distance  $m-n$ , or the distance  $s-t$ , you will find they are the same.) This means we have both an audio and r.f. voltage existing together, but they are not modulated. In the plate circuit of this stage, the tuned circuit will pick out the r.f. voltage, and the audio voltage will be ignored. Thus, a hum voltage introduced in a linear r.f. amplifier will not be passed on to the next stage.

This does not mean, however, that only a detector stage can give hum modulation. The Eg-1p characteristics of all amplifiers, including those in the

radio frequency system, are not perfectly linear, so in practice we may get modulation hum from any amplifying stage.

Operating an amplifier at a point on the curved region, as shown in Fig. 13B, permits normal variation in one direction but tends to cut off variations in the other direction. This does not greatly matter to the incoming signal, since the flywheel effect of the following resonant circuit will restore the original wave shape. However, if hum is introduced, the plate current will have the appearance shown. Comparing the  $x-y$  distance with the  $m-n$  or the  $s-t$  distance in this figure shows you that the amplitude of the r.f. signal has been changed by the hum voltage—in other words, a modulation has occurred. The following tuned circuit will now reconstruct a completely modulated wave like Fig. 13C, so the hum voltage will be carried along on the r.f. voltage just like any original modulating signal.

**Effect-to-Cause Reasoning.** When you are tracking down modulation hum, notice whether it is heard on all stations, only on certain stations, or just on one. These are important clues.

► Usually, if the source is within the set, you will notice that the modulation hum is heard on all reasonably strong local and distant stations. The most common sources of such hum are cathode-to-heater leakage in a tube or a defective filter system in the plate supply for the oscillator of a superheterodyne. (Very frequently a separate condenser-resistor filter combination is used in the oscillator plate supply, because any hum in this supply will modulate the oscillator voltage and so be mixed with the incoming signal in the first detector.)

If these points appear normal, then check operating voltages carefully,

noticing in particular the grid bias voltages developed for the r.f. amplifiers. In addition, a lower than normal screen grid voltage will cause operation on a curved characteristic just as excess bias will—so changes in the values of bleeder resistors, or leakage in a screen-grid bypass condenser, can cause modulation hum by producing a curved characteristic in an amplifier.

► If modulation hum appears to be stronger at one end of the station dial (usually the high frequency end), often some stage is in a regenerative state which allows it to be easily affected by stray hum fields or hum voltages. Eliminating the excessive feedback in such a case will frequently remove the modulation hum. This condition sometimes means that realignment or a check of the bypass condensers is needed.

► If the modulation hum appears only on the stronger local stations, it may be originating outside of the receiver. The power line picks up a considerable amount of r.f. energy; if the antenna or ground installation is in poor condition, this power line r.f. may feed through the chassis to the input of the set. As the power line may have non-linear characteristics, it is quite possible for these r.f. signals to arrive with a hum modulation. Also, if the antenna is near power lines, a poor joint may cause rectification and mixing right in the antenna system. The cure is to restore the antenna and ground to full effectiveness, and, if necessary, to bypass the power line connections at the set with a small condenser. One condenser from each side of the power line to chassis, which in turn is grounded, is effective. (A separate ground cannot be used on an a.c.-d.c. universal set unless a ground terminal is provided on the set by the manufacturer. Don't connect a ground to the chassis of such receivers.)

Sometimes you can make the modulation hum disappear just by reversing the power line plug in the wall socket. If this reversal also changes the apparent strength of the signals, then you have a definite indication of a poor antenna or ground installation.

► If modulation hum is heard on only one station, the signals from that station are strong enough to drive one of the tubes into the curved region of its characteristic, external hum modulation occurs, or else the hum is originating in the station itself. Station hum

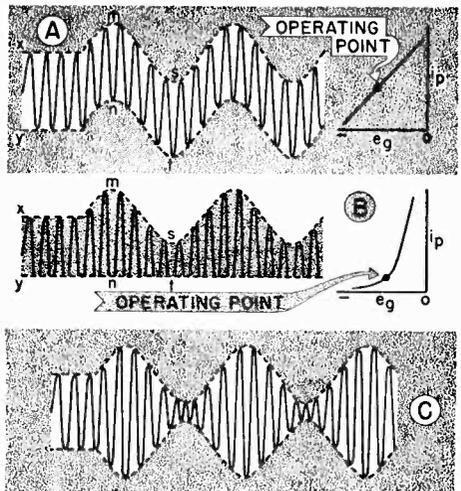


FIG. 13. How modulation hum occurs.

is much more noticeable on high-fidelity receivers where there is a greater response to low frequencies. To determine which of these possibilities exists, try another receiver at the location or move the defective receiver to your shop. If hum is heard regardless of location or changes in antenna, then the station is to be suspected. If the test receiver hums at the location of the defect, overloading or external hum modulation is occurring.

If some nearby station or some very powerful station actually drives a tube into detection, usually a wave

trap tuned to that particular frequency will reduce the incoming signal level enough to prevent hum modulation.

**Defective Stage Isolation.** When you are tracing a steady hum, all interfering signals must be tuned out. However, when you hunt for the source of modulation hum, you need an r.f. voltage source. To isolate a defective stage, you can use a broadcast signal and a signal tracer with an au-

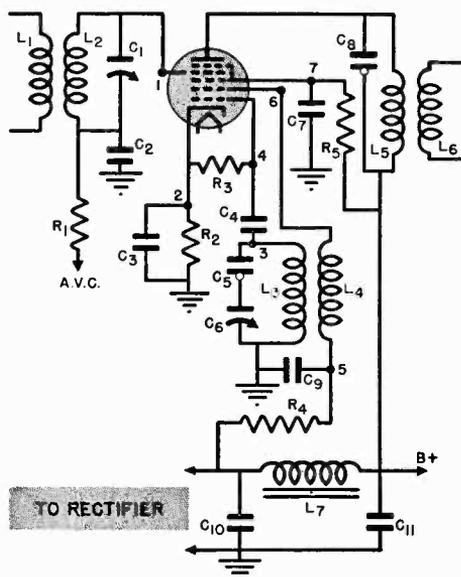


FIG. 14. There are several possible causes of modulation hum in the frequency converter stage.

dible output indicator, or you can make use of a signal generator.

► If you use a signal tracer, first tune in some signal which has the modulation hum. Then, start at the input of the receiver and move toward the output with the signal tracer, listening to each stage. As you move toward the output, the signal will be hum free until you pass through the stage where the hum is introduced. The signal tracer must be tuned to the frequency of the incoming signal until you pass the first detector, at which point you

should change to the intermediate frequency of the receiver.

If you use a signal generator, start at the second detector and move back toward the input, feeding the signal into the stage being tested and listening to the output of the receiver. Use the audio modulation of the signal generator to make sure you are tuned to the proper frequency, then turn the modulation off so you can hear the hum. As you move back toward the input with the *unmodulated* signal generator, you will not hear hum until you have passed through the defective stage—provided no stage is overloaded. Of course, the signal generator will be set at the i.f. value until you pass the first detector, moving back toward the input.

As you move toward the input, the signal strength increases; this may overload some stage previously passed through, causing another modulation hum to be set up. To prevent such overloading, reduce the output of the signal generator enough as you move back so that the output of the set, shown on the tuning indicator, will be approximately constant. Use a d.c. type v.t.v.m. across the a.v.c. leads if no indicator is on the set. An audio indicating device would not show up the overloading.

When you reach the input of the first detector, be sure to change the signal generator to the frequency for which the receiver dial is set.

**Defective Part Isolation.** Suppose your stage isolation tests show that the modulation hum arises in a frequency converter stage like the one shown in Fig. 14. Let's see how the defective part may be found.

The most logical causes of hum in this circuit are cathode-to-heater leakage in the tube or hum voltages in the oscillator circuit caused by a defective condenser  $C_6$ . (Notice that  $C_6$  and  $R_4$  act as a filter.) It is also

possible for condenser  $C_7$  to be open, permitting hum voltage to get into the screen grid circuit; usually, however, an open in this condenser would cause lower than normal volume. This open would cause oscillation if it occurred in an r.f. or i.f. stage, but might not do so in a converter as the output is not tuned to the same frequency as the input.

First, check the tube and try other condensers across the suspected ones. If tests show these parts are normal, next use an *audio* signal tracer to listen in the grid circuit between terminal 1 and ground, in the cathode circuit, and in the other element circuits, to determine just which one is introducing the hum. A station should not be tuned in, since you are

looking for an audible hum voltage. Of course, the hum level may be too low to be heard directly; if so, the easiest procedure is to substitute good parts for logical suspects.

► Hum does not often enter the grid circuit of a first detector or an r.f. stage, but it may get in if the grid circuit opens up—because, say, of a defective wave band switch arrangement—or if some hum-carrying wires develop leakage to the grid circuit. Capacity coupling between hum-carrying wires and the grid leads is usually kept at a minimum by the normal arrangement of the grid circuit. Similarly, stray hum fields from the power transformer or filter choke will not usually induce hum voltages in a radio frequency coil.

### COMMON CAUSES OF HUM

This table covers only the more usual causes, with the most common first. Use it as a guide or memory refresher. Make localizing tests first, unless you are led directly to one of these troubles by the symptoms.

Type	Location	Causes
Steady, 120 cycle (or 60 cycle, if set uses half-wave rectification)	1. Power Supply	Open, low capacity or high power factor filter condensers; leakage causing choke saturation; gassy rectifier tube.
	2. A. F. Stages	Open decoupler condenser; open grid circuit; reversed hum-bucking coil; poor connection.
Steady, 60 cycle	1. A. F. Stages	Cathode-to-heater leakage in tube; open grid circuit; hum adjuster unbalanced (where used); poor connections.
	2. Power Supply	Defective rectifier tube; open in half the high voltage secondary or in connection to tube. (Full-wave rectification only.)
Modulation Hum	1. R. F. Stages	Cathode-to-heater leakage in tube; open decoupler condenser; reduced screen grid or plate voltage; excess bias; poor connections.
	2. Outside Set	Poor Antenna or ground; power line modulation; defect at station.

# Residual Hum Problems

Residual hum problems may arise if the original receiver design was poor or overall aging of the receiver has caused an increase in the normal hum level. Assuming you have checked for all the usual causes of abnormal hum, your next step will be to see what changes may be made in the design of the set to decrease the hum level, provided the receiver owner is willing to foot the bill for the time required.

Before beginning such a procedure, be sure to find out if the receiver was repaired just before the hum was first noticed. If so, the hum level may have been raised by the repairs—or you may have run into an owner who has become critical of the normal

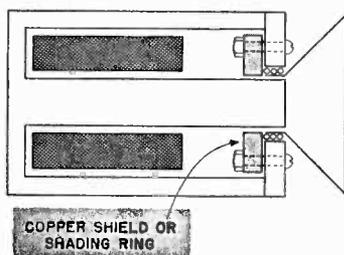


FIG. 15. Eddy currents induced in a heavy copper ring cut down the hum induction from the field into the voice coil.

hum level of the receiver, which was called to his attention by the previous trouble.

Remember that the design of the receiver was carried only to the point where the hum level was considered acceptable. This usually means that not everything possible was done to reduce the hum, so going over the circuit carefully will probably prove helpful.

## HUM-BUCKING

The simplest initial step is to try hum-bucking in the circuit if the nor-

mal hum-eliminating procedures are not helpful.

First, find out if the loudspeaker is adding to the residual hum level. Short the primary winding of the output transformer with a test lead; if you hear hum, it comes from the speaker. Next, see if the loudspeaker uses a hum-bucking coil (Fig. 8). If not, see if a shading ring is used, like the one in Fig. 15. (This ring is between the front pole plate and the field coil, so it can be seen only in speakers having an open "pot" field enclosure unless you take the speaker apart.) This copper ring uses eddy currents to reduce the hum induced in the voice coil.

If the speaker has neither a shading ring nor a hum-bucking coil, replacing it with one having a hum-bucking coil will usually reduce the hum level considerably. Some reduction may be obtained by such a replacement even if the old speaker has a shading ring. Be sure the replacement speaker matches the voice-coil impedance and the field resistance of the original speaker.

► Fig. 16 shows two ways to buck or cancel out hum by deliberately introducing an out-of-phase hum into the hum-producing circuit.

The method shown in Fig. 16A involves feeding a hum voltage from some source (such as the input of the filter in the power supply) into the cathode bias resistor of the hum-producing stage. If the cathode bias resistor is normally bypassed, the bypass should be removed. Then, the resistor  $R_2$  is added and varied until hum is minimized. The .5-mfd. condenser  $C_3$  is used in series with resistor  $R_2$  so there will be no d.c. current flow to upset the bias.

A somewhat similar scheme is shown in Fig. 16B, which can be used in any screen-grid or pentode tube stage producing hum. Condenser  $C_3$  is connected across the screen-grid voltage-dropping resistor  $R_2$ , forming a hum voltage divider with condenser  $C_2$ . Any hum which is in the plate supply will then be fed to the screen grid, where it will be out of phase with that in the normal plate circuit or grid circuit of the tube. By choosing the size of  $C_3$  properly, the circuit can be balanced and hum minimized.

► Of course, these methods of hum-bucking should never be used if there is any actual defect in the radio. Such methods should be employed only when you are attempting to reduce a normal residual hum level. If the

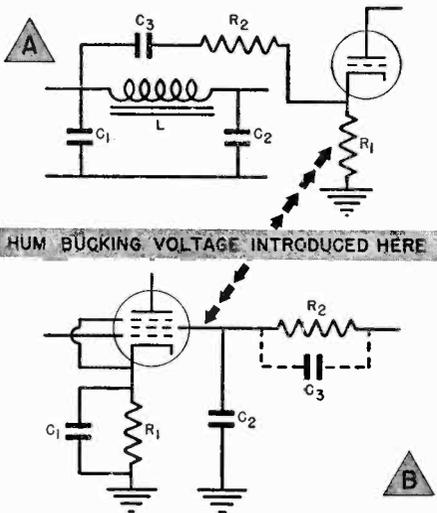


FIG. 16. Two methods of hum bucking.

residual hum level appears abnormally high, almost always some defect exists which you can—and should—find and correct by the basic methods given in the earlier part of this lesson.

### REDUCING RESIDUAL HUM

As you carry out the following hum-reducing steps, you may find

each one individually results in very slight reduction in the hum level. In fact, you may have to use a sensitive output indicator to determine whether you are getting any drop at all. Make measurements with care, because it will take a number of small reductions to decrease the hum level appreciably.

► Increasing the filter condenser capacities will frequently work wonders,

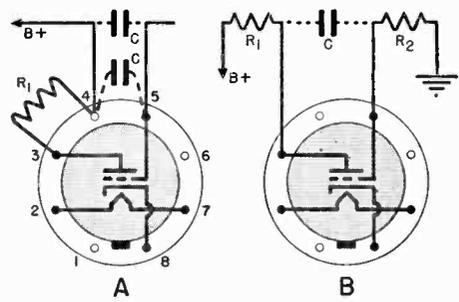


FIG. 17. Parts and lead placement contribute to the residual hum problem by providing capacity coupling.

particularly on some of the older receivers. Values as high as 16 to 30 mfd. can be used satisfactorily.

► As you learned earlier in this lesson, inductive and capacitive coupling between wires often causes hum, especially if the wiring has been changed about. It is important that plate and grid wires, particularly in the first audio stage, be kept well away from supply leads.

Fig. 17 shows two examples of how the mounting of parts may cause stray hum pick-up. It is common practice for manufacturers to wire up circuits in the easiest possible manner, using blank socket lugs for mounting. The plate resistor  $R_1$  in Fig. 17A may be mounted as shown, between socket terminals 3 and 4. Using terminal 4 brings one end of this resistor and its supply lead close to grid terminal 5, so that capacity coupling can exist between the grid lead and both this

supply lead and the end of the resistor. Moving the resistor so it is between terminals 3 and 1 brings the B+ lead to terminal 1, which separates the resistor and supply lead from the grid and reduces the coupling.

In Fig. 17B, the resistors  $R_1$  and  $R_2$  may be lined up on a terminal strip. Here, moving either of the resistors or reversing the connections to resistor  $R_2$  so as to move the grid lead away from the plate resistor may prove helpful.

► Stray hum currents in the chassis add to the residual hum level, par-

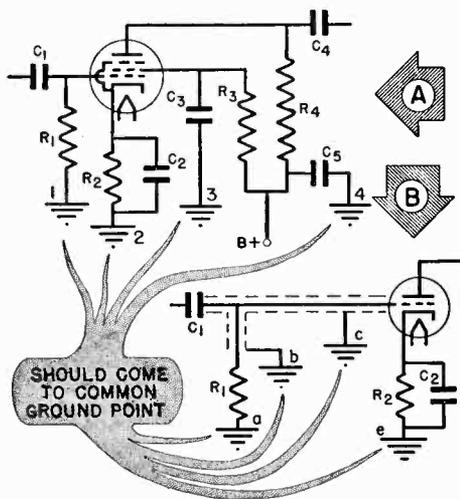


FIG. 18. Using a common ground point in a stage prevents hum currents in the chassis from producing hum voltages in the stage.

ticularly if the manufacturer has again followed the easiest procedure and grounded parts to different points on the chassis. For example, in Fig. 18A, terminal 1 and terminal 2 may not be grounded to the same point on the chassis. If so, any stray hum currents between these points will produce a voltage drop in the chassis, which is effectively in series with the grid circuit and can cause hum.

There are four ground symbols shown in Fig. 18A; it is desirable to

have all of these grounded parts in a single stage come to a single ground terminal. This sort of connection will remove the effects of stray chassis currents. Of course, these four terminals must each make good contact to the ground point, without any common lead, so as to avoid the situation pictured in Fig. 9. (In Fig. 9, the resistor  $R$  should be brought to the ground lug  $G$  directly instead of to terminal 7.)

Three separate grounds may be used in a circuit like that shown in Fig. 18B. If hum is traced to such a circuit, grounding terminals  $a$  and  $b$  to the same point will prove helpful. If the hum comes from stray chassis current, removing ground  $c$  altogether may help.

► You might try shielding wires subject to hum pick-up to reduce the amount picked up. Shielding must be used judiciously, however, as there

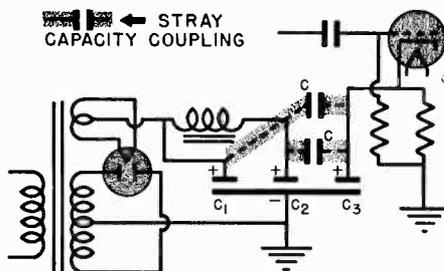


FIG. 19. Condensers in a common block or case will have capacity coupling between leads and sections which can contribute to the hum level.

is a considerable capacity between a wire and its shield. This capacity shunts the circuit and will lower the high frequency response of an audio amplifier considerably. Therefore, use no more shielding than absolutely necessary and use low-capacity cable if space permits. (This type of cable has a large amount of insulating material between the inner wire and the shield, thus reducing the wire-to-

shield capacity by increasing the spacing between them.) Sometimes shielding can be avoided altogether by re-routing wires.

Stray capacity coupling may also prove annoying if you have filter condensers and a cathode bypass condenser in the same electrolytic condenser block, as in Fig. 19. The cathode bypass condenser normally will be about 10 mfd., and so will have

an impedance of about 130 ohms to 120-cycle ripple. However, if there is sufficient capacity coupling between the filter condenser leads and this condenser, an appreciable hum level may appear. This will certainly be true if the condenser loses capacity or develops a high power factor. In such cases, using separate condensers will help considerably in lowering the hum level.

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## Oscillations, Squeals and Motorboating

A certain amount of feedback takes place in practically all radio tube stages. This feedback may be in phase with the input signal, causing regeneration, or out of phase, causing degeneration. The amount of feedback is one of the limiting factors in radio design. A certain amount helps to obtain desired characteristics. Regeneration increases the sensitivity, while degeneration in an audio amplifier increases the fidelity by flattening the response characteristics. However, if either kind of feedback gets out of hand, undesirable effects result. Excessive degeneration lowers gain, while excessive regeneration makes the receiver unstable.

As regeneration increases, the set becomes much more critical in its operation. Sensitivity and selectivity increase abnormally and the receiver response becomes very erratic. Small changes in humidity (affecting circuit  $Q$  slightly) will have such great effects on the response that the receiver will seldom act the same from day to day. When regeneration is carried too far, oscillation occurs. (There is so much feedback that the stage sustains oscillation by itself). Then radio reception is blocked entirely, or is accompanied by squeals, whistles, rush-

ing noises, or motorboating sounds.

Before we learn how to localize oscillation, let us see just how feedback can occur.

### FEEDBACK PATHS

Fig. 20 shows a typical i.f. amplifier stage. As in all radio stages, there is a certain amount of grid-plate capacity. This can act as a feedback path—even in modern pentode tubes, in which the screen grid and the suppressor grid both tend to reduce this grid-plate capacity to a very small value. In addition, there is capacity coupling between grid and plate leads, as well as possible inductive coupling between the input and output tank circuits.

Since it has a path for feedback, and resonant grid and plate circuits, you can see that this circuit contains all the elements of a tuned grid-tuned plate oscillator. If the plate tank circuit is tuned so that the plate load is inductive (the tank circuit is tuned above resonance), then the feedback will be in phase with the input signal, will aid it, and may cause oscillation.

Of course, oscillation will develop only if there is enough feedback. This will depend on both the amount of

voltage in the plate circuit available for feedback, and upon the effect the feedback has in the grid circuit. If the stage has high gain, and if the plate tank circuit has a high Q factor (thus acting as a high impedance load), there will of course be more voltage in the plate circuit available for feedback than if the stage had low gain or the tank circuit a low Q factor.

The Q factor is also important in the grid circuit; for the same amount of feedback, a grid circuit with a high Q factor will have more voltage across it than one with a low Q factor. (The reason is that the feedback capacity or inductance is in series with the grid impedance, forming our old friend, the voltage divider. The feedback voltage must divide between them; naturally, the higher the impedance of the grid circuit, the greater the percentage of the feedback voltage that will appear across it.)

Thus, problems caused by feedback will always be most frequent in high gain stages where tuning circuits of high Q are used. The more sensitive the receiver, the more you can expect oscillation troubles.

► Incidentally, proper alignment may clear up oscillation in a circuit like that in Fig. 20. If the tank circuit  $C_1-L_1$  is far off resonance, the reflected effects on  $C_2-L_2$  are reduced, so the Q factor of the latter circuit will rise. Similarly, if tank circuit  $L_4-C_4$  is far off resonance, the impedance of tank circuit  $C_3-L_3$  will rise. Either condition may produce instability and oscillation. When the circuits are all correctly aligned, the reflected effects will load the tuned circuits in the offending stage and reduce their Q. Further, if the plate tank circuits are carefully tuned slightly below resonance (using an increased trimmer condenser capacity setting), then there is little chance

for feedback to produce oscillation.

► To sum up, oscillation can occur only if there is: 1, a feedback path; 2, feedback of the proper phase to aid oscillation (regeneration); and 3, the strength of the feedback is sufficient. Usually you will try to cure oscillation by blocking or removing the feedback path or by reducing the amount of feedback; changing the phase of feedback is not usually possible unless the feedback is caused by circuits whose alignment may be corrected.

**Feedback Examples.** Fig. 21 shows several more examples of circuits in which feedback can occur. In 21A the plate load is inductive (unless the reflected effects from the tuned circuit  $L_4-C_2$  cancel all inductance in the plate circuit), so feedback from it will be in phase with the input signal and so may cause oscillation. In addition to the usual feedback paths, coupling may also exist between the tuned circuit  $L_4-C_2$  and the antenna lead—especially if an excess length of antenna wire has been tucked into the radio. Feedback along this path may cause oscillation if it has the proper phase and is sufficiently large.

In an audio stage like that shown in Fig. 21B, oscillation might occur but usually does not. The effectiveness of the capacity coupling is relatively small, as the capacitive reactance of such small capacities is very high at audio frequencies, so most of the feedback energy is dropped in the coupling. In addition, the audio transformers usually have appreciable d.c. resistance. Thus, although they resonate with their distributed capacities, the Q is very low, so rather large feedback voltages are necessary to cause oscillation.

Of course, if another stage is added, the increased amplification makes it possible for the feedback voltage to be considerably higher; if sufficient coupling exists between the input and

output of the two stages, there may be enough feedback to cause oscillation.

Also, inductive coupling between transformers may cause trouble unless they are placed so minimum feedback can occur.

► Since each of the resistance-coupled stages of Fig. 21C inverts the signal  $180^\circ$ , two of them cause a  $360^\circ$

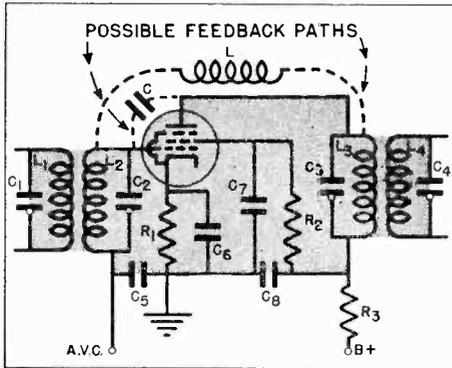


FIG. 20. This i. f. stage can have either inductive or capacitive feedback paths between grid and plate circuits.

change. Therefore, if there is any stray coupling between the grid circuit of  $VT_1$  and the plate circuit of  $VT_2$ , the feedback voltage will have the proper phase to cause oscillation. Whether or not oscillation will occur then depends on the amount of coupling, on the amplification (which determines the amount of feedback voltage) and on the size of the grid resistance (which determines the proportion of feedback dropped across the grid circuit). The larger the grid resistance  $R_1$ , the greater the drop across it and the greater the likelihood of oscillation.

**Low-Impedance Paths.** In each of the foregoing examples, the feedback voltage is part of the rather high voltage developed across a high-impedance circuit or part. It is also possible for a voltage developed across

a low impedance to cause feedback. For example, the a.c. plate circuit of the output tube in Fig. 22 traces from the plate, through the primary of transformer  $T$ , through condenser  $C_5$ , and then through  $C_6$  back to the cathode. Therefore, audio variations in this plate circuit will create a voltage drop across  $C_5$ , the amount depending on the strength of the variations and on the reactance of  $C_5$ . Since  $C_6$  is the output filter condenser of the power supply, this audio variation is impressed on other tube plate circuits.

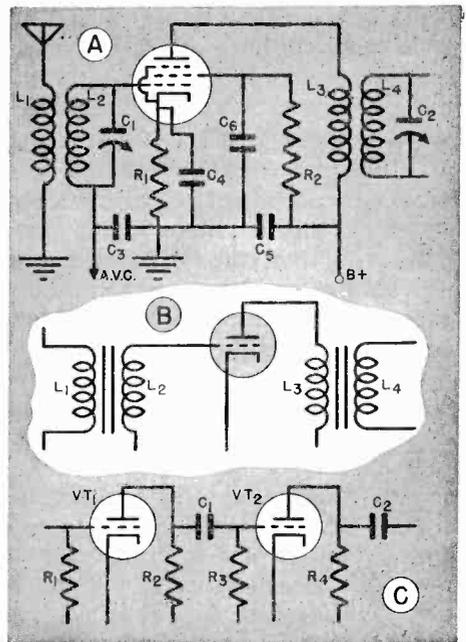


FIG. 21. Examples of stages where feedback can cause oscillation.

The plate supply of tube  $VT_1$ , for example, may thus be varied at an audio rate.

Such variations introduced in the plate supply of tube  $VT_1$  are passed through the intervening stages and applied to the grid of tube  $VT_2$ . If there are enough stages, these variations may arrive back at the grid circuit of  $VT_2$  in phase with the plate

variation and thus provide regenerative feedback and oscillation. On the other hand, if  $VT_1$  is coupled directly to  $VT_2$ , the feedback will be degenerative. Therefore, such oscillation will usually occur only if there are three stages in the audio amplifier. This kind of low-frequency oscillation is usually called "motorboating", because it has a "put-put-put" sound.

If condenser  $C_s$  loses capacity, its reactance will increase, and the amount of feedback will be greater. Motorboating can therefore be minimized by replacing  $C_s$  with a higher-capacity condenser. Incidentally, most receivers have additional filtering (represented by  $R_s-C_s$ ) in the plate circuit of the first tube, which tends to remove such audio feedback and thus prevent motorboating.

► If either condenser  $C_2$  or  $C_s$  in Fig. 22 opens, the variations in the plate current in that stage will develop an a.c. voltage drop across the cathode resistor. This drop will be introduced in the grid circuit; it will be out of phase with the grid voltage, and so will be a degenerative feedback. Such degeneration does not cause oscillation—in fact, it suppresses it. Manufacturers sometimes deliberately introduce this degeneration both to control oscillations and to flatten out the response characteristics of the amplifier. This must be done carefully, because too much will cause an excessive reduction in gain.

**Parasitic Oscillation.** Any unwanted (or unintended) self-sustained oscillation is a parasitic oscillation, because it "lives off" the stage. However, servicemen and technicians usually use this term only to describe oscillations which occur at some frequency to which the circuit is not tuned or which is outside the normal frequency band of the offending stage.

In radio receivers, this trouble is usually limited to the output audio

stage, because this is usually the only stage where enough power is available to sustain the oscillations. A pentode output stage is particularly subject to these oscillations, because such a stage has high gain, has a tube with a relatively coarse screen grid structure (so that the inter-electrode capacity is high), and uses circuit elements which readily permit parasitic oscillations. This trouble is most common when the output stage is run as a class AB or class B push-pull amplifier, where a low-resistance input transformer must be used. A typical circuit is shown in Fig. 23.

The oscillation occurs at frequencies where the leakage inductance and distributed capacity of the transformers form resonant circuits, or where the transformer capacities remove the inductance effects, leaving the grid and plate leads to act as transmission lines due to their distributed inductance and capacity. The circuits act as tuned-grid tuned-plate oscillators which may produce oscillations well up in the short waves.

This parasitic oscillation does not occur in every circuit, of course. It may occur, however, in any circuit in which enough power is available, in which enough feedback exists, and in which the grid inductance and capacity can form a resonant circuit.

Parasitic oscillation causes severe distortion, weak reception, and perhaps a rushing noise or exceedingly high frequency whistle. The large amount of power consumed lowers all operating voltages. The output tubes may glow blue or even get so hot their elements melt. The rectifier tube, filter choke, power transformer and output transformer will be passing excessive current, so will overheat.

The condition may be cured either by introducing suppression, making the plate circuit bypass condenser more effective, or else by shortening

the effective length of the grid leads so much that the tube will not be able to oscillate.

► As you have learned, a low Q factor tends to suppress oscillation. Therefore, one of the most effective cures

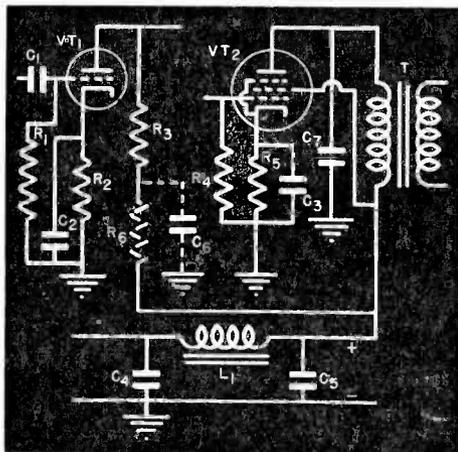


FIG. 22. Motorboating usually occurs in an audio amplifier which does not have decouplers  $R_6$ - $C_6$ , or where this filter is ineffective.

for parasitics is to insert suppressor resistors right at the grid terminals of each tube, as shown in Fig. 24. Resistors  $R_2$  and  $R_3$  should be between 100 and 1000 ohms. Use the smallest size which will eliminate oscillation.

In a Class B output stage, grid current will cause distortion if values above 500 ohms are used. Hence, if larger suppressors are needed to stop oscillations, use about 200 ohms and consider the following procedures.

► Manufacturers generally use condensers such as  $C_2$  and  $C_3$  of Fig. 23 in circuits where parasitics may develop, to make the plate load more capacitive. Such condensers should be right at the tube socket, a position which makes the effective length of the plate leads shorter, reducing further the inductance effects as well as the ability to feed back.

In the circuit shown in Fig. 23, the

bypass paths from the plates run through condensers  $C_2$  and  $C_3$  to  $B+$ , from  $B+$  to  $B-$  by way of the output filter condenser, and then to the cathode through condenser  $C_1$ . Bringing  $C_2$  and  $C_3$  directly to the cathode, as shown in Fig. 24, eliminates a great deal of this path, and so makes parasitic oscillations less likely. If  $C_2$  and  $C_3$  are used in this manner, they must have voltage ratings of 600 volts or higher.

► As an additional measure, grid condensers of about .0005 mfd. each can be installed to change the grid resonant frequency, as shown by  $C_4$  and  $C_5$  in Fig. 24. (Sometimes they are installed by the manufacturer.) If used,  $C_4$  and  $C_5$  should return to the same cathode point as  $C_2$  and  $C_3$ .  
► Remember that an electrolytic condenser makes a very poor r.f. bypass condenser because its inductance

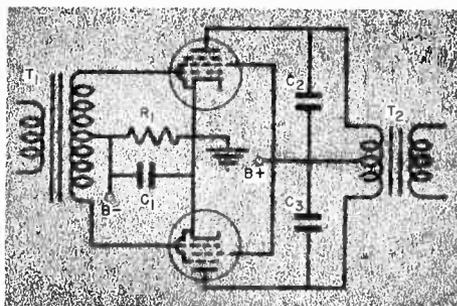


FIG. 23. A typical push-pull output stage.

ive winding limits its effectiveness at high frequencies. For this reason, a paper bypass condenser is connected across the output filter condenser in many receivers; sets without this feature frequently prove unstable. Never assume that an electrolytic condenser is an adequate bypass for all frequencies which may be present.

## EFFECT-TO-CAUSE REASONING

As you have learned, oscillation may occur in a single stage of a radio

receiver or may be the result of feedback across several stages. Your first problem is to localize the trouble, then you must cure it, generally by reducing the gain of the stage, by restoring bypass facilities to normal, or by blocking feedback paths.

► Oscillations may be audible or inaudible, and may occur anywhere in the radio. *If the oscillations are audible, whether or not signals are tuned in, they are audio signals and in all probability originate in the audio amplifier.* The only usual exception occurs when an a.v.c. controlled stage is oscillating and block-

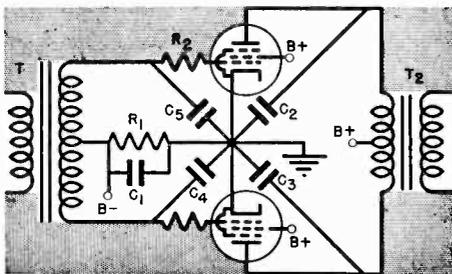


FIG. 24. Here are pictured the cures for parasitic oscillation in an output stage.

ing simultaneously—that is, the circuit starts to oscillate, but sufficient grid bias is developed (particularly across a.v.c. resistors) to block oscillation at an audio rate. Motorboating can be caused once in a while by this particular combination of conditions.

To tell whether a steadily produced whistle or “put-put” is in the audio system or in the last i.f. stage, turn the volume control to zero output. In most modern receivers the volume control is at the input of the audio amplifier. If a volume control adjustment will stop the sound coming from the speaker, then it is originating ahead of the control and is probably in the i.f. amplifier. On the

other hand, the oscillation is probably originating in the audio amplifier if moving the control does not affect the oscillation.

If the feedback occurs *in or through* the circuit containing the volume control, varying the volume control will vary the amount of feedback and throw this particular circuit in and out of oscillation. If the set has a tuning indicator, notice whether the volume control adjustment stops the tuning indicator deflections as well as the sound output. If the tuning indicator continues to deflect in step with the oscillations, even when the volume control is at zero, then one of the r.f.-i.f. stages is oscillating.

If no indicator is used, pull out the last i.f. tube. If this stops the sounds, the trouble is in the r.f. section; otherwise, an audio trouble is present.

► If oscillations occur only when a station is tuned in, then they are undoubtedly starting in the r.f. or i.f. stages. In a set with this trouble, you will notice a loud whistle or squeal as you tune in a station; then as you tune slowly through the proper dial setting, the squeal will first drop to zero frequency and then increase in pitch again on the other side of the correct dial point.

If the ability to oscillate appears greater at one end of the dial than the other, an r.f. stage is probably to blame. In this case, the variation is caused by the fact that different frequencies will vary the amount of feedback and may, because of differences in alignment, vary the input and output impedances of the offending stage.

Be careful in making this check for suspected oscillation not to confuse these noises with whistles due to oscillator harmonic interferences, second harmonics of the i.f. amplifier, and similar causes, which have been discussed elsewhere. These last inter-

ferences will show up only at certain particular spots on the dial or only on certain stations, rather than over the entire dial or a large portion of it.

► Parasitic oscillations are not usually audible except as a sort of rushing noise. To help yourself diagnose them, remember that the great amount of power taken when these oscillations occur lowers the voltages throughout the receiver, so that weak or badly distorted reception, or possibly a dead receiver, may result.

## LOCATING THE DEFECTIVE STAGE

The most common sources of trouble are missing or poorly connected shielding, and defective condensers in supply circuits common to more than one stage. The condensers most likely to cause oscillation, arranged in order with the most common troubles first, are:

1. Open screen grid bypass.
2. Open or low-capacity output filter.
3. Open or low-capacity plate decoupler.
4. Open cathode bypass.
5. Open or shorted grid decoupler (a shorted one removes bias).
6. Open bias supply bypass.

To make a quick check, first inspect the shielding visually, then tune in a signal and try a good bypass condenser across those listed in 1 and 2 above. If the oscillations continue, it is best to localize the defective stage.

A simple way to localize the trouble is to tune in a signal, and bring your hand near one tube at a time. As you approach the offending stage, you will change the pitch of the whistle or squeal.

► If you have a signal tracer, use it on each stage in turn (starting at the input and moving toward the output) with no signal tuned in. Since oscillations produce r.f. voltages, you will find an r.f. voltage present when you

enter the defective stage. Sometimes this procedure will not work satisfactorily, because connecting the signal tracer or bringing your hand near the offending stage may stop it from oscillating. You will then probably pass over the offender, find the signal in the next stage, and believe that it is the defective one.

► In the audio stages, you can test each stage with an audio signal tracer, starting at the input and moving toward the output. Here again, you may stop the oscillation when you connect the tracer.

► The fact that your tests may stop the oscillation temporarily makes it rather difficult to do more than locate the defective stage with test equipment. If an open bypass condenser is suspected, you will have to hold a good condenser across each section you want to test. Remember, particularly when working with r.f. stages, that long leads cannot be used—you must hold the replacement condenser directly across the suspected one. Bend the test condenser leads so you can touch the proper points.

If the oscillation stops when you make such a condenser test, remember that it may do so because of the presence of your hand, not because of the condenser. If this happens, you will find it out quickly, because oscillation will usually start again as soon as you finish putting in the new condenser and remove your hand. It is usually a good idea to temporarily solder the trial condenser across the suspected one, then take your hands away to see if oscillation has stopped, before making a permanent replacement.

You should, of course, check an oscillating set to make sure that all tube shields for which provisions are made are actually used. But don't assume that the shielding is satisfactory just because they are all present

—remember, a poor contact between a shield and the chassis may make the shield ineffective. Try grounding such shields to the chassis with a short test lead or screwdriver blade. If this helps, sandpaper the edges of the shield and tighten the screws or rivets used to ground it. Since rivets sometimes corrode, replacing them with screws may prove helpful.

► Oscillation in an r.f. stage may be caused by a poor contact to the tuning condenser rotor shaft if the resistance of the contact provides a common coupling between condenser sections. Be sure to clean and tighten wiping contacts if trouble is localized in this section of the set.

► Remember that abnormal feedback must occur in the set to cause oscillation. To cure the condition, you must either block the feedback path or make circuit adjustments to cut down the amount of feedback. If you can find nothing out of the ordinary which could provide a feedback path, see if the stage has exceptional gain.

Since this can easily be caused by higher-than-normal screen grid voltage or by lack of proper bias voltage, a short-circuited bias supply or an open screen bleeder may be the cause of the oscillation.

► Misplaced wires are a frequent cause of oscillation. Always look for them if an oscillating set shows evidence of a previous repair. If you find that moving a wire causes a change in the pitch of the oscillation, move both this wire and its neighbors carefully until you find a position which will give minimum feedback.

The manufacturer's service information is frequently helpful in finding this position. In all sets in which regeneration is allowed to exist to a considerable degree—small a.c.-d.c. receivers, for example—misplaced leads can easily cause oscillation. The manufacturer of such sets will frequently give information on "lead dressing," that is, how to position leads with respect to each other and to the chassis to prevent oscillation.

## COMMON CAUSES OF OSCILLATION

This table covers only the more usual causes, with the most common first. Use it as a guide and memory refresher. Localize the trouble first, then check for these probable causes.

Condition	Location	Causes
Audible at all times.	Audio Stages	Defective filter or bypass condensers.
Squeals only when station is tuned in. Occurs on all stations.	I. F. Stages	Open bypass condensers; shielding missing or making poor contact; alignment off; excess screen grid voltage; low bias; leads improperly placed.
Squeals only when station is tuned in. Occurs mostly at one end of tuning band.	R. F. Stages	Open bypass condensers; shielding missing or making poor contact; poor contact at tuning condenser rotor shaft; excessive screen grid voltage; low bias; alignment off; leads improperly placed.

# Lesson Questions

Be sure to number your Answer Sheet 41RH-2.

Place your Student Number on *every* Answer Sheet.

*Send in your set of answers for this lesson immediately after you finish them, as instructed in the Study Schedule. This will give you the greatest possible benefit from our speedy personal grading service.*

1. Will a shorted filter condenser cause hum?
2. What quick check can be used to determine if a suspected condenser is open or has high power factor?
3. Excluding the power supply, what stage in a radio is the one most likely to be the source of hum?
4. Is the temperature of a speaker field, used as a choke, a reliable indication of excess current flow?
5. If the complaint is a loud 120-cycle hum, with a loss of low-frequency response and oscillation, what filter part would you suspect?
6. A 60-cycle hum is heard from a receiver using full-wave rectification. Which *two* of the following can cause this hum: 1, shorted filter choke; 2, cathode-to-heater leakage in a tube; 3, open output filter condenser; 4, open in one of the rectifier tube plate circuits; 5, reversed hum-bucking coil.
7. Does capacity coupling to a grid circuit cause more hum in: 1, a high-impedance grid circuit; or 2, a low-impedance grid circuit?
8. What three conditions are necessary for oscillation to occur?
9. Suppose motorboating is heard and the tuning eye deflects in step with the oscillations. Turning the volume control to zero output stops the sound from the speaker but the tuning eye continues to deflect. What section would you suspect is the source of the motorboating?
10. Name three procedures used to stop parasitic oscillations in an audio output stage.

Be sure to fill out a Lesson Label and send it along with your answers.



## DOING THE IMPOSSIBLE

To a vast number of human beings, the term “impossible” is like a closed gate, barring the path that leads to a more satisfying life. They turn away from the “impossible” without the slightest attempt to find out just *why* a particular accomplishment has this awful word attached to it.

“Impossible” is applied in the majority of cases simply because *nobody has done that particular thing before*. The history of science is full of thrilling examples of men doing the “impossible”—the steamboat, the airplane, the radio, and many, many more. Naturally, brilliant men were responsible for doing these things, but more important is the fact that they refused to believe in the “impossible.” They were determined to find out *really* why it had not been done and *how* it might be done.

This is the approach to the “impossible” that works not only in science and invention, but in bringing to all of us the satisfactions we seek in life. When you believe the “impossible” is not a closed gate but a challenge to you to achieve and move forward into new fields, then you are a man marked for success.

*J. E. Smith*

**HOW TO ELIMINATE  
DISTORTION**  

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**SERVICING LOUDSPEAKERS**

42RH-1



**NATIONAL RADIO INSTITUTE**

**WASHINGTON, D. C.**

**ESTABLISHED 1914**

# STUDY SCHEDULE NO. 42

For each study step, read the assigned pages first at your usual speed, then reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind. Study each other step in this same way.

- 1. Types of Distortion . . . . . Pages 1-5  
Although there are three kinds of distortion, only one type is of great importance to radio servicemen. You are shown that a non-linearity is responsible for amplitude distortion and that improper operating voltages on tubes is a very common cause of this condition.
- 2. Defects Producing Amplitude Distortion . . . . . Pages 5-12  
Here you study in detail the circuit defects that can cause a tube or iron-core device to produce distortion. Then troubles in push-pull stages are covered. This section serves to show you what to look for when you are localizing the trouble.
- 3. Defects Producing Frequency Distortion . . . . . Pages 13-14  
This short section gives a listing of the few defects which may cause unusual amounts of frequency distortion. These defects are localized just like those producing amplitude distortion.
- 4. Localizing Distortion . . . . . Pages 15-21  
Now that you have studied the causes of distortion, you can confidently proceed to localize the trouble to the defective section, stage, circuit and part. This portion of the lesson explains the methods used.
- 5. Loudspeaker Defects . . . . . Pages 22-30  
As a loudspeaker is a mechanical device, it has many mechanical troubles as well as the usual electrical opens and shorts. These can be recognized by the distortion or noise caused. Here you learn all about them and why they occur.
- 6. Speaker Repair and Replacement . . . . . Pages 31-36  
After finding the speaker to be at fault, you will have to remove the cone or field and install a replacement. This is a simple procedure in many cases, once the method of installation is seen. You can also send the speaker away for repairs if you desire, and in some instances this is necessary.
- 7. Answer Lesson Questions and Mail your Answers to N.R.I.
- 8. Start studying the Next Lesson.

# HOW TO ELIMINATE DISTORTION

## SERVICING LOUDSPEAKERS

### Types of Distortion

**W**HEN a customer tells you that his receiver doesn't sound right, you have a case of distortion to correct. He may say the set sounds as though the person talking had a "mouthful of mush," or the receiver sounds "tinny or boomy," or he may not be able to describe just what he finds objectionable.

Complete elimination of distortion from an amplifier or reproducer is, of course, an impossibility, for the receiver without any trace of distortion has not yet been made. Your task is to reduce the distortion below the level that causes customer dissatisfaction.

Therefore, when a customer complains about his receiver, be sure you find exactly what he dislikes about its performance. You can't depend on your hearing; even when the receiver is working at its best, you may consider its output distorted, or the customer may complain about distortion which you find unnoticeable. Much depends on what the customer has been used to in the past and the exact quality of his hearing. Remember that the untrained human ear can stand reasonable amounts of distortion indefinitely. Some people even prefer it—for example, those who set the tone control for a greater than normal bass response.

There are three types of distortion: *Frequency*, *phase* and *amplitude*. Let us see just exactly what they are, and which you'll be called on to correct most frequently in service work.

#### FREQUENCY DISTORTION

Frequency distortion occurs in a receiver or an amplifier which does not pass all signal frequencies in the audio range equally. Assuming that audible sound ranges from 20 to 20,000 cycles, ideal transmitting and receiving equipment should transmit and reproduce these frequencies with their original relative amplitudes. Fortunately, intelligible and entertaining programs may be had with a smaller frequency range. The usual inexpensive table model radio receiver will have a frequency range of 100 to 3,500 c.p.s.; high-fidelity receivers reproduce from 30 to 15,000 c.p.s. Both receivers will receive programs from a high-fidelity station, but only the one with the wide frequency range can do them full justice.

You cannot tell whether a set has good, bad or indifferent frequency response by listening to a program, as the quality of the program itself is unknown. Instead, you must take frequency response curves to determine the frequency range of the equipment.\* Since this is usually a long, tedious procedure, the serviceman generally neglects the frequency range of the receiver except when dealing with high-fidelity receivers, and then only if there is some possibility of a change in response caused by new parts.

\* The procedure used in taking frequency response curves is explained elsewhere in the Course.

## PHASE DISTORTION

Since radio equipment employs coils and condensers, phase shifts occur in the transmission of different frequencies. Thus, several simultaneously transmitted frequencies may not end up in the same phase relationship, but may be advanced or retarded in time. Phase distortion is usually of no importance in sound reproduction, so we will not consider the matter here; you will meet it again in your studies of television.

## AMPLITUDE DISTORTION

When a radio produces signal frequencies that originally did not exist, amplitude distortion is produced—so called because it is the result of departure from the original amplitude wave shape. Since the added signals are usually harmonics of the original, this distortion is sometimes called harmonic distortion.

Now let us see just what causes amplitude distortion. A graphical presentation will show you most clearly.

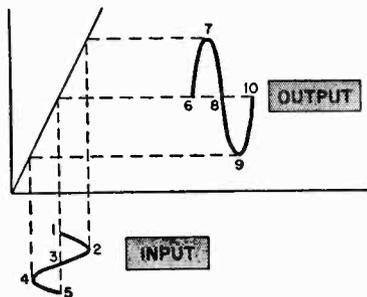


FIG. 1. A straight line characteristic curve will give linear, distortionless output.

No distortion occurs in a device with a straight-line operating characteristic like Fig. 1. Here, the sine wave signal 1-2-3-4-5 produces an enlarged but otherwise exact replica, 6-7-8-9-10. Any device or amplifier with a straight-

line characteristic like this is said to be linear, and will faithfully reproduce the input signal.

As you know, the  $E_g-I_p$  character-

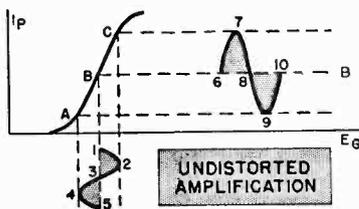


FIG. 2. The actual tube characteristic is never straight, but there is a center portion which will give relatively undistorted amplification.

istic of a vacuum tube is never a straight line but is curved, as shown in Fig. 2. Part of the curve is sufficiently straight so that operation on it will not produce appreciable distortion. In Fig. 2 the applied grid signal 1-2-3-4-5 swings the plate current over the almost straight section A-B-C of the

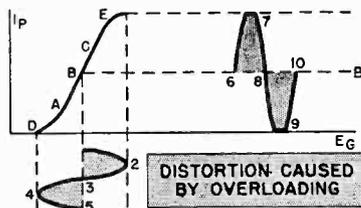


FIG. 3. Too high an input signal swings the operation into the curved regions so distortion occurs.

curve; plate current wave 6-7-8-9-10 has approximately the same shape as the input signal voltage, and no appreciable distortion exists.

**Overloading Effects.** Now suppose we have the same grid bias and plate voltage as for Fig. 2, but apply a larger signal to the grid so that we work into the upper and lower bends of the characteristic (Fig. 3). Here, the signal 1-2-3-4-5 swings the tube operation over range D-B-E, producing the out-

put wave 6-7-8-9-10. The output wave is highly distorted, as the peaks at 7 and 9 are flat. This type of distortion is the result of overloading (too much signal voltage).

The output wave shape in Fig. 3 is still symmetrical, since the upper and lower halves are identical. As you

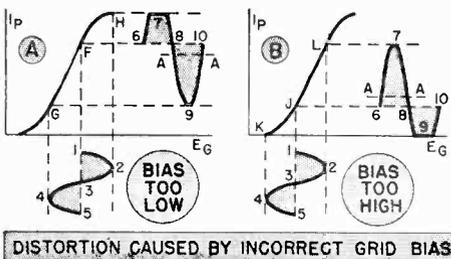


FIG. 4. Even with a normal signal input, too high or too low bias will shift the operating point so distortion occurs.

know, this means that odd harmonics, such as the third, fifth and seventh, have been added to the fundamental. Such distortion is quite noticeable to the human ear.

**Bias Changes.** Suppose that instead of overloading the stage with too much signal, we apply a normal signal and allow the bias voltage to shift. In Fig. 4A, the tube operating curve is the same as in Fig. 2, but the operating point is now point *F*. The bias voltage has been reduced, allowing a higher average plate current to flow.

Now when we apply grid signal 1-2-3-4-5, we will operate over the curve in the region between *G* and *H*. The output wave 6-7-8-9-10 has a flat portion at 7, and the swing from 6 to 7 is much shorter than the swing from 8 to 9, so the upper half of this wave is quite distorted compared to the lower half. Since the two halves of the wave are unsymmetrical, we have an even harmonic distortion.

Similarly, we can get distortion at the lower bend of the curve (Fig. 4B). Here, the bias voltage has been in-

creased so *J* is the operating point. The input signal swings over the operating region *K-L*, producing an output wave 6-7-8-9-10. The lower half of the wave is squared off and shorter than the upper half. This is the same distortion as that in Fig. 4A, except that it occurs on the other half of the wave. Since the human ear can't recognize phase shifts of 180°, the two distortions sound alike.

**Plate Voltage Changes.** Distortion like that in Fig. 4 will also occur if the plate voltage changes and the grid bias remains fixed. Fig. 5 shows how.

Suppose the plate voltage is 250 volts and the bias is adjusted to give class A amplification. Then, on curve 1 (the middle operating curve) the operating point will be *B* and the operating range *A-C*, so the output curve *M* is produced—a duplicate of the grid signal.

With the same bias, suppose we increase the plate voltage to 300 volts.

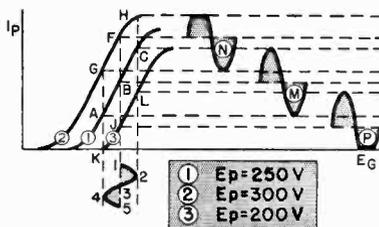


FIG. 5. Incorrect plate voltage can also move the operating point to where distortion becomes excessive.

We will then operate on curve 2, over the upper curve in the *G-H* region, and get output curve *N*. Notice that this is the same type of curve as that in Fig. 4A.

If we drop the plate voltage to 200 volts, we use curve 3 and operate in the *K-L* region of the curve, getting output curve *P*. This curve is like Fig. 4B.

Thus, overloading the tube, changing the grid bias or changing the plate

voltage may shift the operating point on the tube curve enough to cause distortion.

**Self-Biased Stages.** In a self-biased tube stage, we normally get changes in both the grid voltage and the plate voltage when any defect affects either. Since these two changes tend to compensate for each other, distortion in a self-biased stage is not as great as that shown by our curves so far. However, a self-biased stage cannot compensate fully for changes in operating voltages; some distortion occurs when any change is made, but considerably less than occurs in a fixed-bias stage for the same change.

Remember that changes in screen voltage will have much the same effect as improper plate voltage. The distortion shown in Fig. 5 will occur if screen grid voltage changes make the tube operate over a curved part of its characteristic curve.

Low emission in a tube, caused by a worn-out cathode or by low filament voltage, can also change the operating point of an  $E_g-I_p$  curve so that distortion occurs.

## PLATE CURRENT SHIFTS

In Fig. 2, the average of the plate current is the line  $B-B$ . When the signal is applied, the increase in plate current from 6 to 7 of Fig. 2 is equalled by the decrease from 8 to 9, so there is no change in the average value of the plate current. Thus, you should notice no change in the plate current when a signal voltage is applied to a class A stage.

In Fig. 3, the average plate current is the line  $B-B$ . Again the change from 6 to 7 is equal to the change from 8 to 9, so the average plate current will not change. Therefore, the plate current will not tell us when a stage is over-

loading, if the overload causes an output like that in Fig. 3.

However, in Fig. 4A we find something very interesting. The average normal plate current is represented by the line  $F$ . When we apply a signal, the plate current rise from 6 to 7 is not nearly as great as the plate current drop from 8 to 9. Therefore, the new average caused by this signal variation is somewhere near the point represented by the line  $A-A$ . Thus when a signal voltage is applied, the plate current average drops from the  $F$  value to the  $A-A$  value—so a drop in plate current when signals are applied shows you the stage is operating at or near the upper bend in the tube characteristic.

In Fig. 4B, the plate current changes from the average value  $J$  to the value  $A-A$ —increasing when a signal voltage is applied. Therefore, an increase in the plate current when a signal is applied shows the stage is operating on the lower bend of the tube characteristic.

In Fig. 3, we chose an operating point exactly at the middle of the curve. If the actual operating point is higher or lower on the curve, an overloaded input will cause unequal half cycles and, again, a plate current shift. The direction of the shift depends upon whether we are higher or lower on the curve, as in Fig. 4.

*In general, there should be a steady plate current in a class A amplifier, whether or not a signal is applied. A plate current increase or decrease means some distortion is occurring in that stage. The direction of the change shows whether the stage is operating closer to the upper or the lower bend of the tube characteristic. This test applies only to a class A amplifier, like an r.f. stage with no a.v.c. or an audio amplifier—not to a detector or a class B amplifier, where a plate current change always occurs during nor-*

mal operation when a signal is applied. As you will learn later, this gives you

a highly useful test for distortion in a class A amplifier.

## Defects Producing Amplitude Distortion

In this section of the lesson we are going to concentrate on amplitude distortion, the type you will most often be called upon to correct. Let us repeat, amplitude distortion exists when the output wave shape no longer resembles the wave shape at the input of the stage or section in which the distortion occurs; the change in wave shape indicates harmonics which were not an original part of the signal.

### DEFECTS CAUSING TUBES TO PRODUCE DISTORTION

Let us see what can happen in each stage of a radio to produce distortion, starting with defects which make tubes distort.

Generally, stages ahead of the first a.f. are free of amplitude distortion, so you can usually concentrate on the a.f. stages, power supply and speaker. Let's see why.

Suppose the r.f. tube  $VT_1$  in Fig.

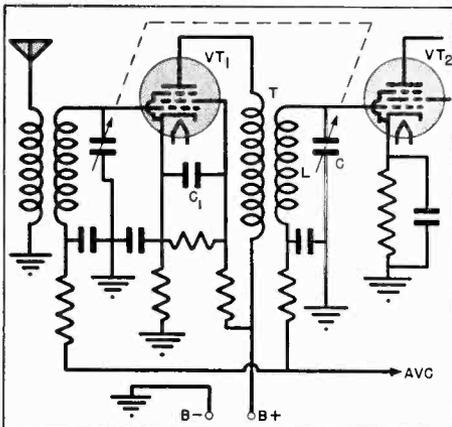


FIG. 6. A typical r.f. stage.

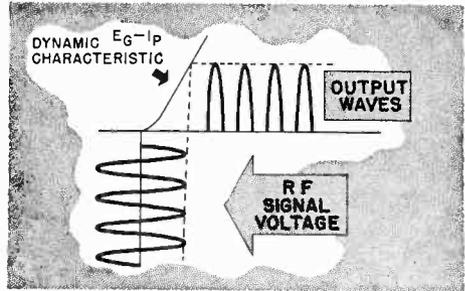


FIG. 7. Operating an r.f. stage at cut-off, the output wave is a series of pulses, but they are reconstructed by the following resonant circuit.

$\theta$  is forced to operate on the curved portion of its characteristic (Fig. 7) by, say, excess bias and low screen voltage caused by leakage in condenser  $C_1$ .

This practically cuts off the lower half of the plate current wave form. However, when we feed the output signal through the coupling transformer  $T$  to the following resonant circuit, the latter's flywheel action\* restores the lower half of the signal and a replica of the original input signal voltage appears across  $C$  for application to tube  $VT_2$ . Thus the distortion

\* When pulses of energy are fed into a resonant circuit, the condenser charges up on the pulses and then discharges through the coil between pulses. This stores energy in the coil, which in turn charges the condenser with the opposite polarity. When the coil energy is used up, the condenser again discharges. As a result, we have both positive and negative halves of a wave across the condenser if the charging pulses have the same frequency as the resonant frequency of the L-C circuit. This continuing action of the resonant circuit, called the flywheel effect, restores the missing parts of the applied signal.

has been corrected, but the customer will probably complain of low volume or low sensitivity; you'll service the set for that complaint rather than for distortion.

► Similarly, the first detector is designed to operate at the plate current cut-off point just like any other detector. It must cut off half the wave (Fig. 7) to give the necessary mixing action. If a defect occurs *the detector may work on the straight portion of its curve* and become more of an amplifier than a detector. Some mixing of

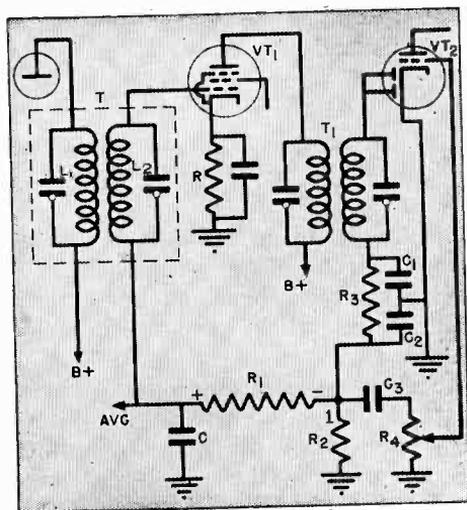


FIG. 8. An i.f. amplifier stage.

the incoming carrier and the locally generated oscillator signal will still occur, but the i.f. output of the detector will be low or circuit noise will arise. Weak reception or noise—rather than distortion—will be the customer's complaint.

Incorrect electrode voltages in the oscillator circuit may make the oscillator block and thus chop up reception. Again, the customer will not recognize this as distortion.

► However, an i.f. stage like that in Fig. 8 may cause distortion when overloaded. Suppose that a.v.c. filter con-

denser  $C$  breaks down, removing the a.v.c. voltage from the i.f. and preceding tubes.

Then on strong signals, the sensitivity of the r.f. section of the receiver will not be reduced by the a.v.c., so the signal voltage developed across the secondary of i.f. transformer  $T$  will become considerably greater than the d.c. bias across resistor  $R$ . This will allow the tube to operate on both the bends of its characteristic (as in Fig. 3), so both positive and negative signal peaks will be distorted just as if a distorted a.f. signal had been modulated on the carrier at the transmitter. This distorted signal will pass through the second i.f. transformer  $T_1$ , be rectified by the diode detector, and ultimately produce a distorted loudspeaker output. The following resonant circuits cannot correct for this as *both* halves of the wave have been affected instead of just one half.

This overloading can also be caused if  $VT_1$  or some other a.v.c.-controlled tube is gassy and draws grid current. This current will set up a voltage across  $R_1$  with the polarity shown. As a result, the a.v.c. voltage will be opposed by this drop so is effectively decreased, and may allow the r.f. gain to increase enough to produce a signal which will overload  $VT_1$ .

► Trouble may occur in the second detector  $VT_2$ . For example, suppose the r.f. by-pass condenser  $C_2$  opens. This will allow so much r.f. energy to be fed into the first a.f. stage, that it may produce overloading and distortion.

► If resistor  $R_2$  is incorrectly replaced with a resistor of too high an ohmic value, the diode detector **may be cut off** for short periods because of the grid leak and condenser action of  $R_2$ - $C_2$ . If the charge stored in  $C_2$  can't leak off rapidly enough, weak signal pulses

will be cut off, thus producing a distorted output.

**Audio Stages.** Amplitude distortion is most commonly produced in the audio stages. Distortion caused by too much grid bias sometimes occurs in the first a.f. amplifier, which may be the triode section of  $VT_2$  in Fig. 8. During rectification, a d.c. voltage is built up across diode load resistor  $R_2$ , with point 1 negative with respect to ground. If coupling condenser  $C_3$  becomes leaky, the d.c. voltage across  $R_2$  will also appear across the volume control  $R_4$ . Then, as the slider arm on the volume control is moved towards

tive. The self-biasing feature of tube  $VT_2$  in Fig. 9 will compensate for small voltages across  $R_2$ , but usually the coupling condenser leakage progresses to a point where the self bias can't keep up. The tube then operates on the upper half of its  $E_g-I_p$  curve, and a very annoying type of distortion is produced.\*

You can easily tell if distortion is caused by a leaky coupling condenser—simply check with a high-resistance voltmeter, or better still a vacuum tube voltmeter, to see if voltage exists across  $R_2$ . Here's how:

With the aid of a tube chart, locate

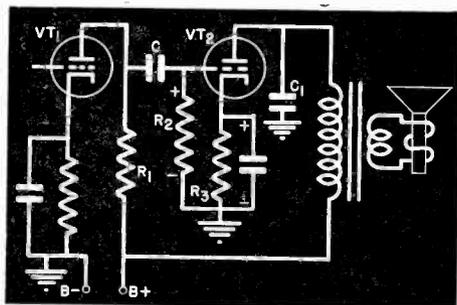


FIG. 9. Most distortion complaints are due to troubles in audio stages like this one.

the point of maximum volume, an increasingly negative bias will be applied to the triode control grid of tube  $VT_2$ , making the triode work on the lower bend of its  $E_g-I_p$  characteristic curve, and so producing distortion. The higher the volume control is turned up, the greater the distortion becomes.

► In a resistance-coupled amplifier, the most common cause of amplitude distortion is incorrect grid bias caused by a leaky coupling condenser. Fig. 9 shows a typical circuit. If condenser  $C$  leaks,  $R_1$ ,  $C$  and  $R_2$  form a voltage divider across the B supply. The voltage developed across  $R_2$  has the polarity shown, making the grid of the tube less negative than normal, or even posi-

the control grid terminal on the socket of tube  $VT_2$ . Place the positive voltmeter probe there, as one end of  $R_2$  is also connected to this point. Touch the negative probe of your voltmeter to the other end of  $R_2$  (or, in this case, to the chassis). Use a high voltmeter range at first, dropping to a lower range if needed. If  $C$  is leaky, you will get an up-scale reading on the voltmeter. If no voltage is present across  $R_2$ ,  $C$  is cleared of suspicion.

\* The next time you are servicing a standard a.c. receiver with a resistance-coupled stage like this, shunt the coupling condenser with a resistance of about 75,000 ohms. This will give the effect of leakage in the coupling condenser and you can become familiar with the distortion produced.

Voltage across  $R_2$ , however, is not definite proof that  $C$  is faulty. Grid current drawn by a gassy tube will also produce a voltage across  $R_2$  with the same polarity as that caused by a leaky coupling condenser. Both defects produce the same type of distortion. To find whether the voltage across  $R_2$  is caused by gas or by a leaky coupling condenser, simply pull the tube out of its socket. If the voltage across  $R_2$  disappears, the tube is gassy; if the voltage remains,  $C$  is leaky. When testing a.c.-d.c. receivers or battery sets—where the test might be upset or damage caused by pulling a tube—unsolder one lead of  $C$  and notice the effect on the voltage across  $R_2$ . If the voltage disappears with  $C$  disconnected,  $C$  is leaky. If the voltage is still present, it is caused by gas in the tube.

**Fixed Bias Troubles.** Fig. 10 shows the audio and power supply circuits of a typical a.c. radio using fixed bias. The bias for the 42 tube is developed across resistors 57 and 61, while the voltage drop across 61 also biases the 75 tube. This circuit is widely used; when distortion occurs in it, there are a number of points to watch.

Leakage in coupling condenser 45 will reduce the bias on the 42 tube and increase its plate current. Since this current flows through resistors 57 and 61, the voltage across them also increases. This in turn increases the bias on the 42 tube, partially compensating for the leakage, so distortion in the 42 tube circuit may not occur until the leakage is so bad the self-biasing action of resistors 57 and 61 can no longer compensate for it. However, only a slight leakage in condenser 45 produces distortion in the 75 tube circuit, because the increase in voltage across resistor 61 is applied directly between the control grid and cathode of this tube. Excess bias on the 75 tube prac-

tically cuts off plate current and causes a choked-up sound, since only the positive peaks of the signal can then be amplified by the tube. The action is the same as that shown in Fig. 4B.

Many servicemen, untrained in effect-to-cause reasoning, stumble on the fact that the distortion does not sound so bad if resistor 61 is shorted. This, of course, removes all bias from the 75 tube except that caused by con-

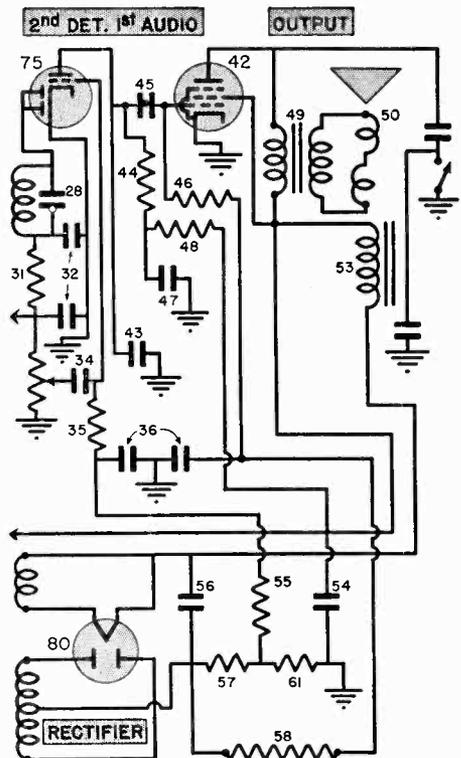


FIG. 10. A good example of trouble in one stage affecting another stage.

vection current through resistors 55, and so allows reception without too much distortion. However, it does not correct the real trouble, which lies in coupling condenser 45.

If resistor 61 is shorted out, the plate current of the 42 output tube will increase even more, and the tube will

soon wear out. Also, since leakage in condensers is progressive, condenser 45 will become completely shorted before long, so the set will come back for service.

We have gone into detail about this incorrect "repair" for two reasons, first, to impress you with the necessity of finding the real cause of the trouble. *Don't attempt to cure distortion by making changes in design unless you know the effect of the change and the original cause of the distortion.* Second, the fact that shorting out the bias clears up the distortion points to an easy way of finding whether excess bias is causing trouble. In receivers using this popular circuit, simply touch the top cap of the 75 and the chassis with the fingers of one hand. Since your hand is a relatively low resistance, this practically shorts out the bias. If the distortion clears up when you make this check, you can be sure it is caused by excess bias. A voltage check across resistor 46 will show leakage in coupling condenser 45.

This distortion is prevalent in three-way (a.c. - d.c. - battery) portables where the output tube plate current may furnish bias for other tubes. A gassy output tube would increase bias while a weak tube decreases bias.

► As another example, suppose there is no voltage across resistor 46, and the grid-cathode voltage of the 75 tube is normal as measured across resistor 61, yet distortion clears up when you touch the top cap of the 75 tube and the chassis. This, you have learned, shows excess bias—even though the bias appears to be correct. But remember, Fig. 5 showed that the correct bias depends on the plate voltage. Evidently, then, the plate voltage has decreased, so what appears to be a correct bias is actually excessive.

Naturally, you should eliminate the

defect that has reduced the plate voltage rather than adjust the bias. If the plate voltage of only the 75 tube has been reduced, leakage in decoupling condenser 47 is probably the cause. A voltmeter or an ohmmeter will quickly show if this part is faulty.

## DISTORTION IN IRON-CORE DEVICES

Distortion may occur in a transformer-coupled circuit like that in Fig. 11 if the d.c. plate current of tube  $VT_1$  increases because of lowered grid bias, increased plate voltage or any other reason.

The reason is that the d.c. plate current, as well as the a.c. signal current, flows through winding  $L_1$  of the trans-

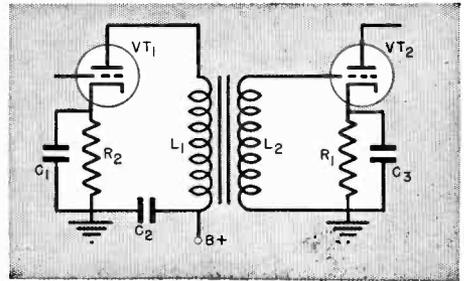


FIG. 11. A transformer-coupled a.f. stage.

former. Audio transformers used in circuits like this are designed to carry a normal amount of d.c. without ill effects. But if the plate current becomes abnormally high, the transformer core may become saturated.

This saturation will make the transformer non-linear in response. A plate current increase will produce practically no increase in flux—and therefore practically no secondary voltage—while a plate current decrease will produce a flux change and a secondary voltage. Thus, if saturation occurs, one-half the signal current in the  $VT_1$  plate circuit will be wiped out by the non-linear transformer action. The secondary output will then be dis-

torted in much the same fashion as is the output of a tube operating with improper bias.

### AMPLITUDE DISTORTION IN PUSH-PULL STAGES

As you know, even harmonics are cancelled in a push-pull stage like that in Fig. 12, but odd harmonics are not. Thus, distortion produced in one-half the wave at each tube (second harmonic distortion) will be removed by the stage. Distortion in both halves of the input wave (third harmonic distortion) will not be removed, but this distortion is not usually appreciable except at high output levels. This relative freedom from distortion is one rea-

be considered satisfactorily balanced.

Such a test is not always conclusive, however. Tubes may check satisfactorily at the reduced voltages of the tube tester, but may not be matched under actual operating voltages. For this reason, several tubes should be tried to find a pair most nearly matched. You will frequently find high-power amplifiers have terminals provided for connecting a meter to check the plate currents individually, and have individual bias supplies so the bias can be adjusted to give equal plate currents.

► Even if the tubes are perfectly matched, distortion may still be passed on to the loudspeaker if different

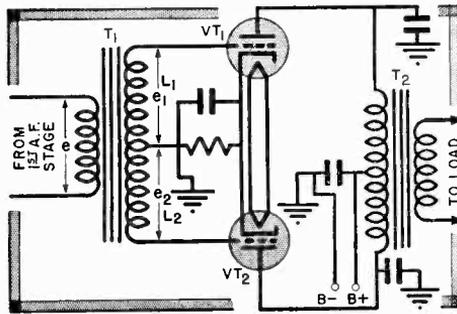


FIG. 12. A transformer-coupled push-pull stage.

son push-pull output stages are so popular.

However, second harmonics will be cancelled by a push-pull stage only if the plate currents of the two tubes are just about equal at all times. This means that the two tubes must have identical characteristics, and be fed the same amounts of signal voltage, for the stage to be free from distortion.

Your first step, therefore, in checking a push-pull stage should be to test the emission of both tubes. When tested on a tube tester having a 0-100 scale, the tubes should give readings within a few points of one another to

amounts of signal voltage are fed into them. In Fig. 12, for example, signal voltages  $e_1$  and  $e_2$  must be equal to prevent distortion. They may not be if transformer  $T_1$  does not deliver equal voltages at all frequencies, if there is a ground in either  $L_1$  or  $L_2$ , or if turns in either winding are shorted.

Replacement is the only cure for a faulty transformer, but you should make sure the transformer is defective before installing a new one. To do so, disconnect both primary leads from the circuit and feed a sine wave signal into the primary of  $T_1$ , preferably a frequency of 400 to 1,000 cycles from an a.f. signal generator, although 6

volts from a power transformer secondary can be used in a pinch. (The low 60-cycle frequency may not produce a balanced signal in less-expensive transformers.) Then compare the output voltages across  $L_1$  and  $L_2$ . You can use an a.c. type v.t.v.m. or a copper-oxide rectifier type a.c. voltmeter to make the comparison. While the latter is not very accurate on high frequencies, such as may be obtained from an a.f. signal generator, it is perfectly all right for comparative readings.

If the voltages are off by more than 10%, trouble is present. You should then disconnect the three secondary leads and check between one of them and the transformer core with a high-range ohmmeter. Normally the leakage should be greater than 20 megohms. A relatively low resistance reading (1 or 2 megohms) shows the presence of leakage between the core and either  $L_1$  or  $L_2$ . No repairs are possible—install a new transformer.

If the leakage test does not disclose a fault, check between each outside lead and the center tap with an ohmmeter. The readings should be within a hundred ohms or so of each other. A much greater difference shows shorted turns on one winding. (The readings will not be the same because, while there are the same number of turns in the outside section as in the section next to the core, the outside turns are larger in diameter. Thus they use more wire and have greater resistance.)

If you are still in doubt, apply a low a.c. voltage to the primary and recompare the voltages across  $L_1$  and  $L_2$  with all leads disconnected from the receiver. If the inequality still exists, the transformer is definitely unsatisfactory. A new one should be installed.

► To check the output transformer, disconnect all leads and apply a low

voltage (1 volt will do) to the secondary. The two primary voltages should be approximately equal. Since a large voltage step-up will occur, use the highest a.c. voltage range of your tester first, then switch to a lower range if necessary.

Shorts or leakage in input and output transformers are not common, however. It is seldom that a properly designed transformer does not provide equal secondary voltages. The usual transformer defect is an open winding.

**Phase Inverters.** Not all receivers use costly and bulky transformers to provide equal input signals to the push-pull tubes. We know that the signals must be of equal intensity and  $180^\circ$

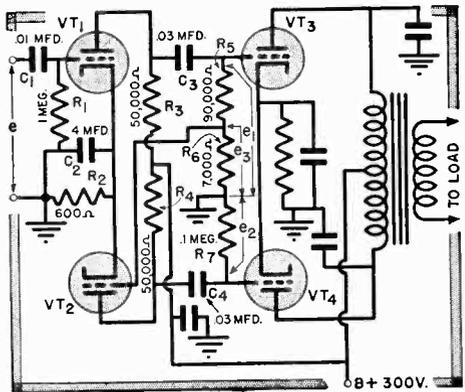


FIG. 13. Using a phase inverter to get the phase reversal necessary for push-pull operation.

out of phase. A tube will shift a signal  $180^\circ$ , so an extra tube and a coupling and supply network may be used to replace an input push-pull transformer.

One typical circuit is shown in Fig. 13. Here the original signal  $e$  is applied to the input of tube  $VT_1$ , amplified by this tube, and applied through coupling condenser  $C_3$  across  $R_5$  and  $R_6$  to the input of tube  $VT_3$  as  $e_1$ . The portion of the signal across  $R_6$  ( $e_2$ ) is fed to the grid of the phase inverter

tube  $VT_2$ . The resulting amplified signal  $e_2$  across  $R_7$ , which is  $180^\circ$  out of phase with signal  $e_1$ , is fed to the other push-pull tube  $VT_4$ . Thus,  $VT_1$  drives both output tubes while  $VT_2$  acts only as a phase inverter. Pulling out  $VT_1$  will block all signals, while pulling out  $VT_2$  will only block half of the push-pull stage.

Since  $VT_1$  and  $VT_2$  are used in similar circuits, they should provide the same gain. Therefore, if the tubes are fed the same amount of signal, the signal voltages  $e_1$  and  $e_2$  applied to  $VT_3$  and  $VT_4$  will be equal as well as being out of phase. By choosing the values of  $R_5$  and  $R_6$  properly, the voltage across  $R_6$  ( $e_3$ ) fed to  $VT_2$  can be made equal to the voltage  $e$  applied to the input of tube  $VT_1$ . Then, with both tubes amplifying these signals the same amount,  $e_1$  will equal  $e_2$ , and distortion produced by even harmonics in output tubes  $VT_3$  and  $VT_4$  will be cancelled by this push-pull action.

Sometimes aging of  $VT_2$  will reduce the phase inverter gain, causing unbalance and distortion. If tubes  $VT_1$  and  $VT_2$  cannot be matched,  $R_5$  and  $R_6$  should be adjusted to produce equal  $e_1$  and  $e_2$  signals. This may be done by measuring voltages  $e_1$  and  $e_2$  with a vacuum tube voltmeter while  $R_6$  is varied in value; when  $e_1$  equals  $e_2$ ,  $R_6$  has the right ohmic value. A convenient way to make this adjustment is to substitute a 10,000-ohm linear-tapered rheostat for  $R_6$ , adjust it until  $e_1$  equals  $e_2$ , then measure the rheostat resistance with an ohmmeter and install a fixed resistor of the same value.

In very bad cases of distortion, remove tube  $VT_2$  and notice the effect

on the tone quality. If there is no change, no signal is passing through  $VT_4$ . You should then check tube  $VT_2$ , tube  $VT_4$ , resistor  $R_4$ , coupling condenser  $C_4$  and the supply voltages to  $VT_4$ .

**Loudspeakers.** As long as the cone motion of a loudspeaker is proportional to the signal currents flowing in the voice coil, no distortion will occur in the speaker. However, the voice coil motion may be non-linear, due to warpage, obstructions in the air gap, etc. We will go into speaker defects, repairs and adjustments in detail later in this lesson. Right now, just remember the speaker can be non-linear too.

**Summary.** You have seen how incorrect voltages will make a tube non-linear in operation, and thus create distortion. This is the case you will generally encounter. Usually, defective parts (except for loudspeakers, which will be taken up later in this lesson) do not cause distortion themselves; rather, the distortion is produced by the changes these defective parts cause in the operating conditions of tubes or iron-core devices.

For example, a condenser may become leaky or a resistor may change in ohmic value, yet signals may pass through them without excessive amplitude distortion being introduced. However, these defects may change the operating voltages of a tube enough to cause serious distortion, or may allow so much current flow through an a.f. transformer primary winding or a.f. choke that the core is saturated, decreasing the reactance so much that distortion occurs.

# Defects Producing Frequency Distortion

Frequency distortion occurs when some frequencies are amplified more than others. Whenever you have a combination of inductance and resistance, capacity and resistance, or inductance and capacity, frequency distortion will always exist, although its amount may be limited by proper initial design. Therefore, there is always some frequency distortion in any audio amplifier, causing losses at both the low- and high-frequency ends of the audio band.

Usually you will correct frequency

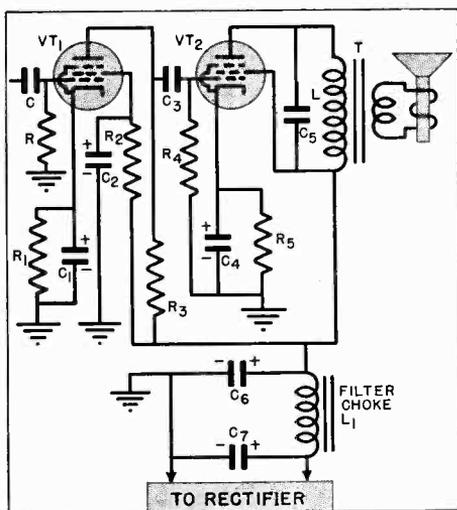


FIG. 14. A typical two-stage audio amplifier.

distortion only when some defect has caused a radical change in the frequency range. Let's see what this defect may be, starting with the r.f. stages.

**Side-Band Cutting.** Some high audio frequency loss will be caused in the r.f. stages by side-band cutting, unless the receiver has a band-pass response. When the tuned circuits resonate sharply, the side bands are

attenuated and the higher audio frequencies in the side bands will not be passed completely (If only one side band is attenuated, amplitude distortion occurs.) This condition may be caused by the design or by too sharp peak alignment. The alignment can be quickly corrected with a frequency-modulated signal generator and an oscilloscope, by staggering adjustments slightly so as to broaden the peak.

Severe side-band cutting can also be caused by regeneration, although the trouble causing regeneration usually causes oscillation before a serviceman is called in. Treat regeneration just as you would oscillation complaints.

## FREQUENCY DISTORTION IN A TYPICAL AUDIO CIRCUIT

Fig. 14 shows a typical two-stage audio amplifier. Let's see how various troubles may cause frequency distortion.

If either coupling condenser  $C$  or  $C_3$  opens, the reduction in capacity will cause severe frequency distortion. The volume will be cut down tremendously, but enough residual capacity will be present in the wiring and between the broken ends of the condenser leads to allow very high-frequency audio notes to pass. This defect can easily be distinguished by the great reduction in volume.

► Condenser  $C_5$  is used in the output circuit to prevent possible oscillation. If this condenser opens, there will be no effect on low audio frequencies, but more high audio frequencies (normally by-passed by  $C_5$ ) will be passed by the stage to the speaker.

► An open in condenser  $C_1$  or condenser  $C_4$  removes the by-passing across the bias resistors, allowing de-

generation to occur. This will reduce amplification somewhat, but will also flatten the frequency response and give better fidelity. However, if small amounts of capacity remain in these condensers, the higher frequencies will be by-passed. There will then be less degeneration and a rising response at these frequencies.

► Signals flowing in the plate circuit of  $VT_2$  normally pass from the plate through  $L$ ,  $C_6$  and  $C_4$ , back to the cathode. The primary  $L$  acts as the plate load across which the amplified audio voltage appears.

If  $C_6$  loses capacity, the external plate-to-cathode path for *low audio frequencies* will be  $L$ ,  $L_1$ ,  $C_7$  and  $C_4$ . The plate load will be  $L + L_1$ ; the amplified signal voltage will divide between  $L$  and  $L_1$ , most of it appearing across filter choke  $L_1$  because its inductance is much higher than that of  $L$ . The *higher audio frequencies* still go through the remaining capacity in  $C_6$  or find a fairly low reactance path through the distributed capacity of  $L_1$ . As a result, most low-frequency voltages are developed across  $L_1$ , most high-frequency voltages across  $L$ . Since audio signals developed across  $L_1$  cannot be transmitted to the loudspeaker,

there is a marked loss of low and medium frequency notes.

This action will not occur in a push-pull output circuit. But, if the schematic shows a single-ended\* output stage and you encounter a loud, high-pitched response with a marked absence of bass notes, accompanied by a loud hum, check the output filter condenser. This is easily done by shunting another condenser of about the same capacity (working voltage at least as high as the original) across the condenser you suspect is open. If you are right the tone will at once clear up. When making this test you must observe the polarity markings on the test condenser. In this circuit, touch the negative lead of the condenser to the chassis and the positive lead to the + terminal of  $C_6$ . If the latter is not readily accessible, use the screen of the output tube (since it is directly connected to the positive terminal of  $C_6$ ).

Frequency distortion in high-fidelity receivers may be caused by a slight change in a part value or by misalignment. In ordinary receivers, defective parts are usually responsible, as we shall see later in this book.

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\* Only one tube—not push-pull.

# Localizing Distortion

The causes of distortion may be found by using the same general procedures used for other defects: confirming the complaint, looking for surface defects, effect-to-cause reasoning, then localization to a section, stage, circuit and part.

When confirming the complaint, listen carefully to the receiver. Different defects produce somewhat different distortion sounds, and a little ear training will often help you go right to the trouble.

In the less obvious cases, particularly when dealing with a high-fidelity receiver, be sure to get all the clues the customer can give you. For example, find out whether the distortion is noticeable as soon as the set is turned on. A distortion which occurs only after the set has warmed up a half hour or so is usually caused by gassy tubes or by heat warping the speaker frame. Many other worth-while clues like this can be picked up by careful questioning.

**Interference Problems.** Before going into the receiver, be sure the "distortion" is not produced by external causes. The customer will frequently confuse distortion with interference.

Perhaps one of the most annoying types of interference is chopped-up reception caused by picking up two or more stations at the same point on the dial. If the customer's radio is band-passed to 10 kc. or more, picking up two stations 10 kc. apart and modulated with a 5-kc. sound causes garbled reproduction which sounds so much like "monkey chatter" it is referred to by this term. Nothing can be done with an ordinary receiver except to reduce fidelity by peaking the tuning system. In a high-fidelity receiver with variable band width, compressing the band

width will help.

Sometimes when interfering stations are distant, a change in antenna direction will favor one station more than the other and allow satisfactory reception of the favored station. This may even help when the stations are on the same frequency, but in general it is best for the customer simply to tune elsewhere on his dial. Interference of this type is more prevalent at night and during the winter months, because then better long-distance reception is possible. In the summer time reception may be entirely satisfactory, for then not as many distant interfering stations will be picked up.

Now, let us see what can be done about a real case of distortion.

**Surface Defects.** An inspection for surface defects is not very revealing, since few troubles causing distortion are visible to the eye.

Of course you should first test all tubes and replace any bad ones. The speaker cone should be examined to see if it is torn or crushed. You should also look for a blue glow between the electrodes\* of glass tubes, which indicates either the presence of gas or excessive plate current caused either by a tube defect or by incorrect operating voltages.

Look for corroded connections at the power supply terminals of battery receivers; these might lower operating voltages and so cause distortion by shifting the operating points of tubes.

Sometimes the wrong tube will be placed in a stage. Always check on this when a diagram or tube layout is available.

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\* Pay no attention to a glow on the glass envelope as this is natural; also a glow should appear between the electrodes of a gaseous type rectifier such as a type 82 or OZ4.

A worn volume control which makes a poor contact between the slider and the resistance element can chop up the signal (the customer will call this a distortion sometimes) or cause intermittent reception, noise or hum. If you notice any noise as the control is adjusted, replace the control.

► After making the usual check for circuit defects, you can then resort to effect-to-cause reasoning. However, there are many causes of distortion; not until you distinguish them by their sounds is effect-to-cause reasoning greatly helpful. Since the most common sources of distortion in the average receiver are leaky coupling condensers or gassy tubes, you might check these points, but it is usually best to resort to section and stage isolation procedures.

### DEFECTIVE SECTION ISOLATION

There are two ways of making a section isolation. For convenience, we divide a receiver into: 1. The a.f. section, which includes everything between the output of the second detector and loudspeaker; and 2, the r.f. section, which includes everything between the antenna and the first a.f. stage. You can isolate the defective section either by clearing one of them of any fault or by finding the one which is at fault.

First, you can tune in a signal from a broadcast station and listen at the output of the second detector, either with a signal-tracing device having an audible output or with a pair of phones—or you might even feed from the second detector into an audio amplifier-speaker combination which is known to be in good condition. If there is no distortion at the output of the second detector, the trouble must be in the a.f. amplifier.

To check the audio amplifier, feed

an audio signal through it from another receiver or from a phonograph record. If the receiver is a phono-radio combination, be sure to use a phonograph record. If the output from the record is undistorted, the trouble must be in the r.f. amplifier. If the output from the record is distorted, as well as that from the radio program, the trouble is in the audio amplifier.

Many servicemen keep a phonograph record player in their shops for checking receivers for distortion. Naturally, a good quality pick-up must be used, as well as a symphonic recording with a wide tonal range and plenty of high- and low-frequency notes, in order to judge the fidelity of the receiver.

You can also check the audio amplifier by using an a.f. signal generator and a c.r.o. A comparison of the input and output wave forms on the c.r.o. screen will show if distortion exists. Details of this test will be given later. Incidentally, an audio generator alone can't be used to localize distortion, because your ear won't notice distortion in a single tone.

### LOCALIZING THE DEFECTIVE A.F. STAGE

Let us assume you've traced the trouble to the a.f. section of the receiver. Now, let's see how you can localize the trouble to a particular defective stage. There are two methods: you can take voltage and current measurements, or you can use some form of signal tracing.

**Voltage and Current Readings.** As you recall, amplitude distortion is usually caused by abnormal tube operating voltages. By taking voltage measurements and comparing them with those given by the manufacturer, you can determine when the grid bias, plate voltage or screen voltage is not normal. If you do not have the manu-

facturer's information, you can tell from experience if the measurements are unusual.

► Current measurements quickly spot class A stages in which distortion exists. Just connect a current meter in the plate circuit, as in Fig. 15, and notice whether the reading changes when signals are applied. You learned from Figs. 4 and 5 that distortion causes a change in the average plate current when signals are applied.

The direction in which the current

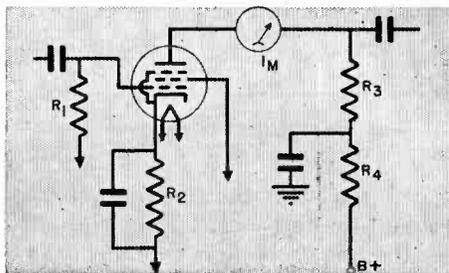


FIG. 15. Distortion is occurring if the plate current changes when signals are applied to a class A stage.

changes is important. The table in Fig. 16 shows probable causes of either an increase or decrease in the average plate current when a signal is applied to the input.

It's easier to make this check if you measure the plate current indirectly, instead of breaking the plate or cathode circuits for a direct measurement. Notice the plate current of the tube in Fig. 15 flows through resistors  $R_2$ ,  $R_3$  and  $R_4$ . If the plate current changes, the voltage across these resistors will also change, so all you need do is connect a voltmeter across one resistor and notice whether the reading changes when a signal is applied. An increase in voltage means an increase in plate current, while a decrease in voltage means the plate current has decreased.

This method is practical where you have just one or two stages to work

with, as in most modern receiver audio amplifiers. If you are dealing with an amplifier having a number of stages, some means of further isolation will probably find the trouble faster. Let us now consider some of the signal-tracing methods.

**Signal Tracing.** Fig. 17 shows a typical audio amplifier. Underneath the diagram are the various pieces of equipment which can be used to localize the stage producing distortion: *A*, a signal tracer with an audible output indicator; *B*, a pair of phones with a protective condenser in series; *C*, a phonograph pick-up; *D*, an a.f. signal generator; and *E*, a cathode ray oscilloscope.

When using the signal tracer or the phones, you can use either the output of a second detector or a phonograph record as the audio signal source, feeding it into terminals 1 and 2 of the amplifier. First, tune in a station (or play a record) and notice the distortion level of the loudspeaker output. Next, it's a good idea to mute the set loudspeaker so you won't hear it while

MILLIAMMETER DEFLECTION	GRID BIAS	PLATE VOLTAGE	SCREEN VOLTAGE
UP	TOO HIGH	TOO LOW	TOO LOW
DOWN	TOO LOW	TOO HIGH	TOO HIGH

FIG. 16. This chart shows what may be wrong to produce a plate current change up or down.

you're trying to listen to the audio output of your signal tracer. To do so, disconnect the voice coil of the speaker and connect a 10-ohm, 5-watt resistor in its place across the secondary of the output transformer.

Then, using either the signal tracer or the phones, start at the input of the amplifier and move to the output, looking for the point where the distur-

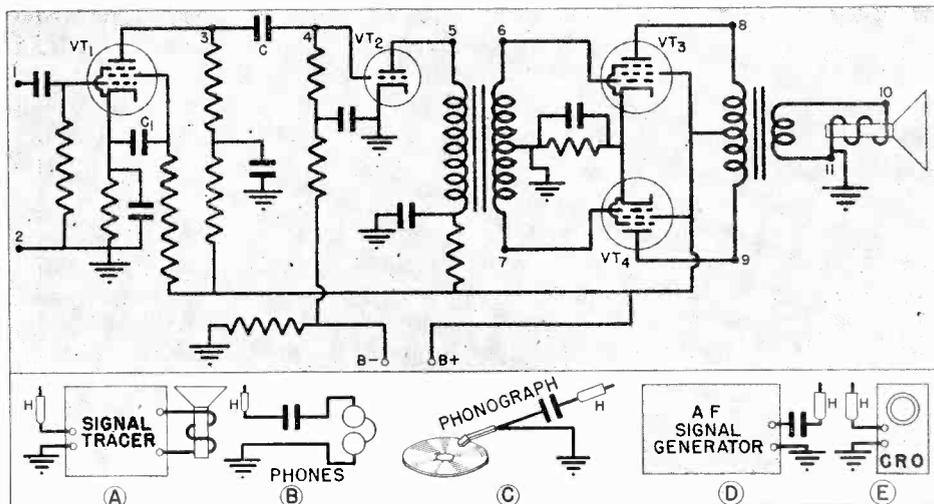


FIG. 17. Here are the devices that can be used to localize the distortion in an audio amplifier like the one shown above.

tion originates. The signal must be kept at a low level when using phones, to avoid overloading them.

Start by sampling the signal fed into the a.f. system. To do so, clip one terminal of the test instrument to the receiver chassis, and connect the hot probe (marked *H*) to point 1. The signal should be clear and free from distortion at this point—if not, the r.f. section rather than the a.f. section is at fault. Next, move the hot probe to terminal 3 to see if tube *VT*<sub>1</sub> introduces distortion. If it does, you have isolated the trouble to this stage; check the tube, the operating voltages and the parts to find the exact cause.

No trouble can occur between points 3 and 4 that will result in amplitude distortion appearing at point 4. If coupling condenser *C* is leaky the signal at 4 will remain undistorted, although the change in the grid voltage of *VT*<sub>2</sub> will distort its output. An open coupling condenser will cause weak reception and frequency distortion, not amplitude distortion.

Thus, the next place to sample the signal after point 3 is at 5. If it is all

right here, check the signal applied to output tubes *VT*<sub>3</sub> and *VT*<sub>4</sub> by connecting the hot probe first to point 6 and then to point 7. (A signal tracer, which indicates the signal level, will show if the output tubes are receiving equal signals better than a headphone.) The signal should then be checked at point 10. (If one terminal of the voice coil is not grounded as shown, unclip the test lead from the chassis and connect it to point 11 while the hot probe is on point 10.) If the signal is clear at point 10 but the loudspeaker reproduces distortion when reconnected, the speaker itself is at fault.

**Using a C.R.O.** If a c.r.o. is used to sample the signal, the signal source should produce a single frequency, preferably having a sine wave form. This means an a.f. signal generator must be used. Then the signal samples gathered at strategic points in the amplifier can be directly compared to the source signal; in this way the nature of the distortion and the stage in which it originates can be accurately determined.

Fig. 18 shows in block form how the a.f. signal generator and c.r.o. are con-

nected to the amplifier under test. The hot lead of the signal generator is connected to the amplifier input (point 1 in Fig. 17). The hot vertical amplifier lead of the c.r.o. is connected to the point under test through a single-pole, double-throw switch SW. When this switch is thrown to point a, the c.r.o. "sees" the signal fed into the amplifier. By throwing the switch to point b, the signal being sampled can be seen and mentally compared to the original signal. Of course, the amplitude of the sample signal will usually be greater than that of the original, but if the wave shapes of the two signals are the same, no distortion exists.

in Fig. 17, touch the hot c.r.o. test lead to points 1, 3, 5, 6, 7, 8, 9 and 10. When the pattern first changes shape, you have just passed through the defective stage; you can then find the actual defect with your voltmeter and ohmmeter.

As the c.r.o. probe is moved from 1 to 9, the signal strength increases, so you should reduce the c.r.o. vertical gain to keep the pattern from becoming larger than the original. When moving from point 9 to point 10, however, there is a sharp reduction in signal voltage; here the vertical gain must be increased to bring the pattern up to normal size. Since the c.r.o. shows

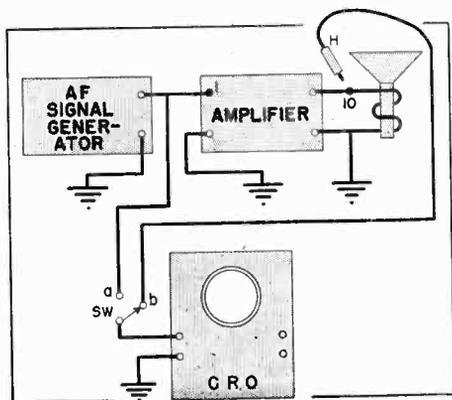


FIG. 18. The connections for a c.r.o. when tracing for distortion.

Switch SW can be eliminated if you connect the hot c.r.o. lead to the hot signal generator lead, then, with a piece of onion skin paper, trace over the sine wave signal produced by the signal generator on the c.r.o. screen. You can then compare this trace with the signal at other points.

When you have the a.f. signal generator and c.r.o. connected as described above, set the c.r.o. timing and synchronizing adjustments to produce 2 or 3 cycles of the input signal on its screen. You are then ready to compare signals. To do this in a circuit like that

signal strength as well as wave form, you can use it to determine whether the signals fed the push-pull output tubes at points 6 and 7 are equal in magnitude.

► A number of typical c.r.o. patterns and their probable causes are shown in Fig. 19. These patterns are for a c.r.o. with negative input polarity;\* if the input polarity of your c.r.o. is positive, the pattern will be inverted. The actual causes, of course, are found when

\* The method of determining c.r.o. input polarity is described in another lesson.

the stage itself is analyzed. If you obtain a c.r.o., you will quickly learn to interpret distortion patterns in terms of the defect producing them.

**Signal Injection.** Instead of sampling the signal at various points in the amplifier, you can feed a signal into some point to see if the distortion occurs between this point and the loudspeaker. (Of course, you should not mute the speaker if you use this meth-

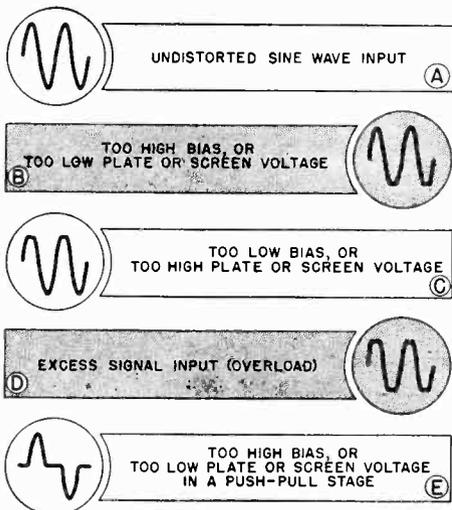


FIG. 19. These are the c.r.o. patterns which disclose conditions in an amplifier.

od.) The easiest signal source to use for this is a phonograph pick-up. However, a pick-up does not have sufficient output to work from points near the loudspeaker; for these points you should use a two-stage, hum-free test amplifier having minimum distortion (like the one in Fig. 20) with the pick-up. Adjust the test amplifier output by its volume control so that the sounds from the receiver loudspeaker approach normal room volume.

This amplifier is satisfactory for checking anywhere in the amplifier, except directly across the voice coil. Therefore, you will have to start testing at point 8 or point 9 of Fig. 17.

The output will tell you whether the output transformer and loudspeaker are in good condition.

Do not inject the signal at points 6 and 7, for this feeds only one output tube at a time and so itself produces distortion. Instead, feed the signal into point 5, then into point 4, and finally into point 1. As more stages are included, reduce the phono volume to prevent overloading the receiver amplifier. When you reach a point where distortion is produced, the defective stage has just been included in the stages between the phono output and the receiver loudspeaker.

Remember, any one of these isolation tests will locate the defective stage. Which one you use in your ser-

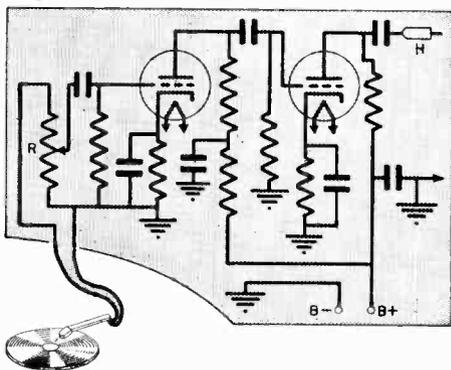


FIG. 20. A record player and amplifier combination like this can be used to feed a test signal into the amplifier being serviced.

vice work depends on your own inclination and the equipment available.

### ISOLATING THE DEFECTIVE R.F. STAGE

If your preliminary tests show the trouble is in the r.f. section, your first step should be to use effect-to-cause reasoning—remember that distortion in r.f. stages may be caused by overloading or by a defect in the second detector circuit.

If effect-to-cause reasoning fails,

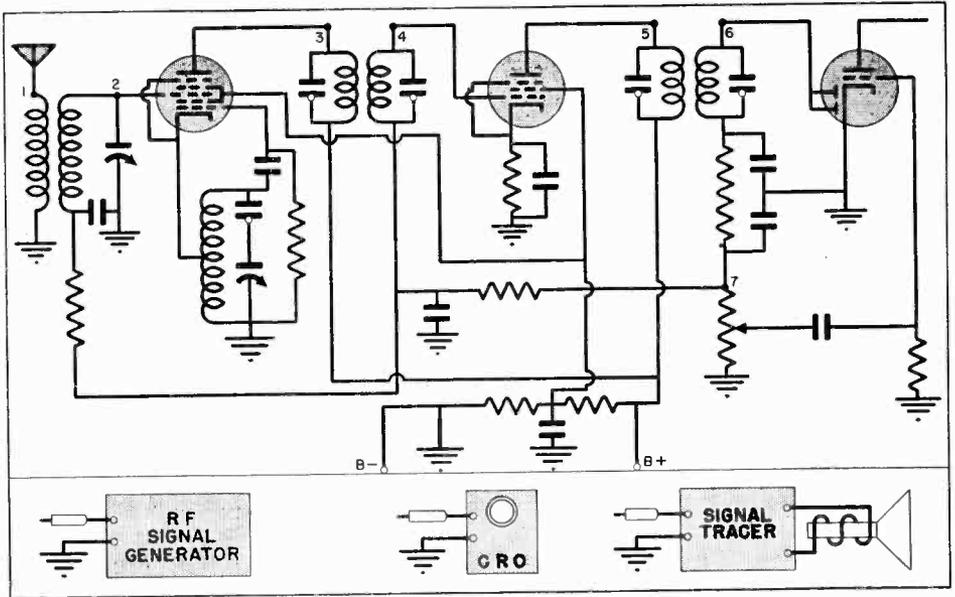


FIG. 21. Here are the devices that can be used to localize distortion in an r.f. amplifier like the one shown above.

you can isolate the defective stage with an r.f. signal tracer. To check an r.f. system like that in Fig. 21, tune in a station that causes the loudspeaker output to sound distorted. When you know what the distortion sounds like, substitute a resistor for the speaker as you did in checking the a.f. system. Connect the ground lead of the signal tracer to the chassis, then hold the signal tracer hot probe on point 1 and tune the tracer to the signal. Move the probe to point 2. If you find distortion at either of these points, the station output itself is distorted.

Next, pass to point 3 and tune the signal tracer to the i.f. of the receiver, then move the hot probe to points 4, 5 and 6 in turn. When you first hear distortion, the defective stage has just

been passed. If you find none, check at point 7 with the audio section of the tracer to learn whether detection has introduced distortion.

The c.r.o. is not very satisfactory for sampling the signal in r.f. circuits, since its sweep frequency is not high enough to give a useful picture of the carrier wave shape. However, the c.r.o. may be used with an r.f. signal tracer by connecting its vertical plates to the a.f. output of the tracer. When this is done the receiver should be fed from a sine wave modulated signal generator. Sample the signal with the tracer in the manner just described, and watch the sine wave modulation on the c.r.o. screen. When the wave shape becomes distorted, you have just passed the defective stage.

COMMON CAUSES OF AMPLITUDE DISTORTION			
R.F.	DET.	A.F.	SPEAKER
Gassy a.v.c.-controlled tube (including Magic Eye)	Incorrect voltages	Leaky coupling condenser	Voice coil rubs pole piece
_____	_____	_____	_____
Leaky a.v.c. filter condenser	Diode load resistor too large	Gassy tube	Spider cracked
_____	_____	_____	_____
_____	_____	Leaky or shorted by-pass condenser	Cone brittle or torn
_____	_____	_____	_____
_____	_____	Volume control worn	Cone unglued
_____	_____	_____	_____
_____	_____	Tubes in push-pull stage unbalanced	Speaker field open (when not part of voltage supply circuit)
_____	_____	_____	_____
_____	_____	Incorrect supply voltages	Armature of magnetic speaker not centered

## Loudspeaker Defects

Most loudspeaker difficulties are mechanical. Therefore, once effect-to-cause reasoning has localized the trouble to the loudspeaker, a visual inspection will generally lead you right to the cause of the trouble.

The speaker is likely to be involved only when the complaint is distortion, noise, weak reception, a dead set or hum. Speaker troubles rarely cause intermittent reception or oscillation, although hardened mounting supports may help cause microphonic howl. The hum level produced by the speaker itself (not that coming from the set) is above the design value only if the connections to a hum-bucking coil are reversed; this is cured by a simple reversal of the connections.

Distortion is one of the most common complaints caused by a defective speaker. Speaker distortion is of a peculiar nature, accompanied by rasp-

ing, scraping sounds; it can be localized, but experience will quickly teach you to identify it as soon as it is heard.

Let us now briefly review loudspeaker types, then take up speaker troubles. In the next section of the lesson, we will cover the practical problems of repair and replacement.

### SPEAKER TYPES

The most common loudspeaker today is the moving-coil type, in which a voice coil is mounted so it can move in a fixed magnetic field. The audio voltage fed to the voice coil produces a varying magnetic field around it. The interaction between the two fields makes the voice coil move back and forth. A cone or diaphragm fastened to the voice coil moves with it, creating sound waves in the surrounding air.

There are two general types of moving-coil loudspeakers. In one, the fixed field is produced by a field coil. This type is known as the electrodynamic or dynamic loudspeaker. The second type uses a powerful permanent magnet to establish the fixed field; it is called the permanent-magnet dynamic or the p.m. dynamic loudspeaker.

The only other kind of speaker used much today is the magnetic type, which we will describe later.

**Moving-Coil Systems.** Let us first see just how moving-coil speakers are constructed.

The paper or fabric cone is supported by a paper, cloth, or leather ring or rim, which is mounted on the speaker

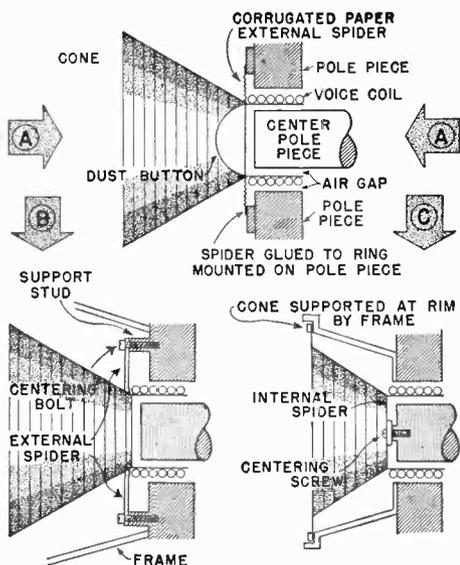


FIG. 22. The three types of loudspeaker spiders.

frame. The voice coil, which consists of several turns of wire on a thin fiber or bakelite tube, is fastened to the apex of the cone. To keep the voice coil properly positioned in a relatively small air gap between the center pole piece and the front pole piece of the loudspeaker, the apex of the cone is supported by a "spider"—a highly

flexible paper or fiber support, so constructed that it will normally hold the voice coil part in and part out of the air gap when no signals are applied. When signals are applied, the spider must not greatly impede the movement of the voice coil and cone but must act as a spring or restoring force, tending to move the coil back to its at-rest position.

► There are two types of spiders, one within the circumference of the voice coil form and the other outside the cir-

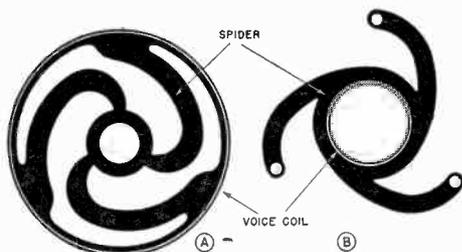


FIG. 23. A face view of an internal spider (A) and an external spider (B).

cumference, known respectively as the internal and external spiders. Fig. 22A shows an external spider made of a circular paper ring glued to supports mounted on the pole piece. In 22B is an external fiber spider, while in C an internal fiber spider is shown. The general shapes of these (fiber) spiders can be seen in Fig. 23. Notice that the fiber spiders have a curled construction so they will move in and out. The paper spiders have ridges or corrugations in the paper so that the spider can stretch readily under the influence of cone and voice coil movements.

► The cone-voice-coil assembly must be entirely free to move under the influence of the desired signal. The voice coil must not strike the pole pieces nor encounter any foreign objects in the air gap which might impede the back and forth motion. The cone must not vibrate at any frequency other than that contained in the original signal,

nor should it produce any unwanted noises. Let us now see just what can cause trouble in the moving system.

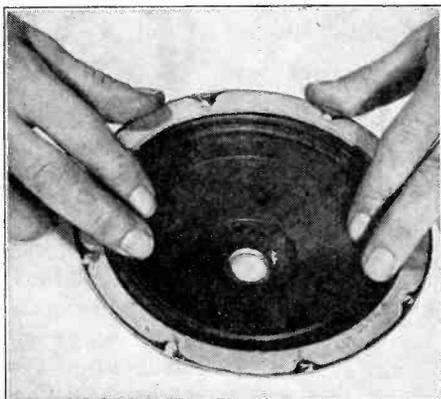
### VOICE COIL RUBBING

If the voice coil touches the pole pieces, free in-and-out motion of the cone is interfered with and signals will be distorted. High-frequency audio signals cause only a slight movement of the voice coil, but at low frequencies a large movement of the voice coil will take place. Since the greatest rubbing occurs during the greatest voice coil movement, signals composed of low

pressure, still keeping your finger tips in contact with the cone rim. If the voice coil hits the pole pieces you can generally hear a scraping sound, and you can often feel the scraping through the tips of your fingers.

► Let's see what can cause this trouble and what may be done to correct it.

If the speaker frame is bent, this will tilt the cone, and in turn tilt the voice coil so that the voice coil will rub against the pole pieces. To correct the trouble, bend the frame in the proper direction and, if necessary, recenter the voice coil.



Testing the centering of the voice coil by pressing on the cone rim and listening for rubbing sounds.

notes are more affected by the voice coil hitting the pole pieces than are signals composed of high notes. For example, male voices will be affected while female voices will be clear, so you can make a preliminary check for this trouble by tuning in a male voice and setting the tone control for maximum bass response.

If such a test indicates rubbing, remove the speaker from the cabinet. With the speaker on its back (cone up), press in on the cone rim lightly with your finger tips, then release the

The frame may be warped by heat from the speaker field. In most cases like this, there will be no distortion for a half hour or so, then distortion will begin and gradually become worse. Bending the frame back into shape or replacing the speaker is the cure. This trouble occurs most commonly with small, light-weight speakers such as those found in a.c.-d.c. receivers.

The voice coil is wound on a light-weight fiber or bakelite tube. This tube is perfectly circular when new, but heat or moisture may warp it out

of shape and allow it to rub against the pole pieces. The remedy is to install a new cone.

The cone itself may lose shape; this will bend the spider and throw the voice coil off center. If the cone is bent or crushed on one side, try to straighten it. Should this prove impossible or should the voice coil still rub, recenter the voice coil. In extreme cases, a new cone will be required.

Loose turns on the voice coil will allow rubbing, since the clearance between the voice coil and pole pieces is slight, especially in small speakers. If possible, remove the cone (instructions will be given later) and voice coil. When this has been done, see if the loose voice coil turns can be recemented to the voice coil form. Before cementing, put a cork in the voice coil form to prevent warping. Allow at least a half hour for the cement to set, then remove the cork and install the cone in accordance with the instructions given later for new cone installation. Removal of a cone for repair is a difficult procedure which is not always successful. If the cone is damaged during its removal, chalk it up to experience and install a new one.

► Cones and voice coils are not the only speaker parts which cause this trouble. The spider is often at fault, which means a new cone is necessary. The spider may fatigue and lose its ability to return the voice coil to its normal position. Then the voice coil will be too deep in the air gap, so at times the end of the voice coil may strike the bottom of the gap. A new cone should be installed, although sometimes a thin metal washer placed under the spider and the centering screw or bolts will lift the voice coil out of the air gap sufficiently to give fair results.

► Corrugated spiders which are fa-

tigued or which sag because of moisture absorption cannot be shimmed up. Sometimes an inexperienced serviceman makes temporary repairs by shoving a small wad of cotton batting between the cone and the speaker frame. This trick is also used at times when a warped or crushed cone lets the voice coil strike the pole pieces. The results after a short time are nil—the only satisfactory repair is to install a new cone.

### PARTICLES IN THE AIR GAP

Particles of dirt or iron filings in the air gap will also interfere with free voice coil movement. When you check for an off-center voice coil, you can feel the coil turns bump as they pass over these obstructions. Sometimes you can get out dirt particles if you feed 2.5 volts or less of 60-cycle a.c. to the voice coil (a filament winding on the power transformer can be used as the voltage source) and put the speaker face down on the workbench. The field must be energized so the voice coil will move back and forth. Non-ferrous particles will often work themselves out of the air gap. Lightly striking the back of the pot (field enclosure) with a wooden mallet or large screwdriver handle will help. (Don't try this on a p.m. speaker, as you may demagnetize the permanent magnet by striking the back of the center pole piece.) Compressed air blown into the air gap from a bellows or hand pump is also an effective means of removing dirt.

When ferrous (iron) particles are lodged in the air gap, the magnetism holds them fast and makes their removal difficult. If the speaker is a dynamic, disconnect the field and apply 110 volts a.c. from a wall outlet across the field with the set turned off. Place the speaker face down and rap it sharply on the back of the pot several

times. At some point in the a.c. cycle all flux disappears from the air gap and the iron particles can be jarred out just as if they were dirt.

To clean filings from a p.m. speaker, remove the cone and push electrical (or tire) tape down into the air gap with a piece of stiff wire. Metallic slivers will adhere to the tape and be withdrawn with it. Never disassemble a p.m. magnetic circuit to clean the air gap, as this will weaken the magnet.

Foreign material in the air gap is a serious matter. Dust buttons (see Fig.



Courtesy General Electric Co.

Applying speaker cement to the inner edge of the cone apex so as to fasten a dust button or dust cover.

22A) will prevent anything from getting into the air gap, but their use is by no means universal, particularly in older speakers.

► Most foreign objects enter the air gap when the speaker is removed for servicing and laid face up on the workbench. Dirt from the ceiling, particularly if you have a basement shop, may fall into the cone and work its way into the air gap. A piece of cloth over the speaker will prevent this, or you can place the speaker face down. Set it on

a piece of newspaper to prevent the field from drawing metallic slivers off the bench into the gap.

Whenever the cone is removed for any reason, cover the air gap with Scotch tape and remove the tape just before you install the new cone. This is particularly important with p.m. speakers, since their field strength is undiminished even with the set turned off.

## NOISES

A defective speaker may cause rattling and buzzing sounds. Each of these sounds will have its own characteristics, which can best be remembered after hearing them a few times. Let's see how some of these noises are produced.

If the cone is exposed to considerable heat in an enclosed cabinet, or if the field runs "hot," the cement holding the cone to the rim, the dust button to the cone, the cone to the voice coil, or the corrugated spider to the pole face may lose its holding properties. This causes sounds like those obtained by humming on a comb and a piece of paper. Speaker cement carefully applied will correct this trouble. Wait at least one half hour for the cement to dry before trying the speaker.

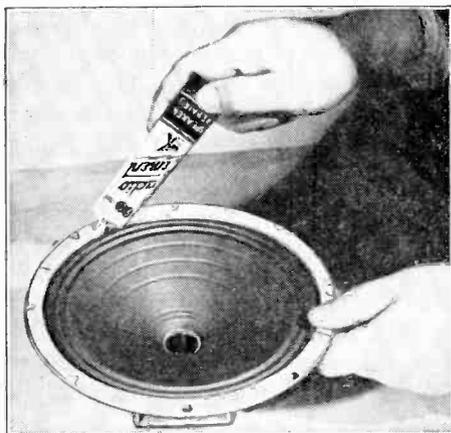
A spider leg may break under repeated movement; each time the broken pieces rub against each other the cone will produce noises. A replacement cone is the only solution.

Cone defects can also cause undesirable sounds. If the cone is torn, the torn sections may vibrate and produce sound waves by striking against each other. Scotch tape under and over the tear will often fix this. Holes poked in the cone can be repaired in the same way.

In some cases the cone material may dry out and become brittle; the cone

may then rattle when power is fed to the speaker. This calls for a new cone.

► The cones of speakers safely mounted in their cabinets do not become torn or have holes punched in them. A careless serviceman is usually responsible for such mishaps. Never put the speaker on top of the chassis when carrying the "works" to and from your car; if the speaker is face down it may slip and allow some sharp corner on the chassis to rip or tear through the cone. To carry a speaker properly in a car, place it face down on a seat, not on the floor. When you



Courtesy General Cement Co.

Applying speaker cement to a loose cone gasket ring.

reach for a speaker, don't just stick out your hand and grab—your extended fingers may go through the cone. If you must pick up a speaker with one hand, take hold of the pot—not the rim to which the cone is fastened, or the frame. If you treat speakers with care, you won't be likely to damage them.

► In a few cases you will find that signal-distorting rattles and buzzes are in sympathy with the cone vibration but are not caused by a speaker defect. Loose chassis and cabinet parts—

particularly celluloid dial shields on plastic cabinets—may be to blame. A little speaker cement will permanently anchor a dial shield in place. Speaker cement can also be used to fix loose push-buttons or anchor voice coil leads which whip against the cone.

Loose tube shields will sometimes buzz sympathetically. You can locate the offender by barely touching it with the back of a finger nail—you will feel the vibration and may change the tone of the buzz. To correct, bend the shield to a tighter fit.

► In phono-radio combinations, a very annoying buzz will frequently be set up when a signal is fed to the speaker. You will usually find one or more steel phonograph needles hanging onto the pole face near the cone, particularly with p.m. speakers. When the cone moves, it strikes the needles and a cone buzz results. Always pick off any needles adhering to a pot, even though they do not produce a buzz at the moment.

## EFFECTS OF AGING

Aging of the loudspeaker cone can also cause noise and distortion. A cone subjected to considerable heat will eventually dry out and become brittle, causing a loss of high-frequency response and perhaps a rattle. The only cure is to replace the cone.

When new, the cone is carefully balanced. It weighs just so much, and has a particular shape best suited to give a desired response. Dust collecting on the top surface of the cone will cause a weight unbalance, and distortion, in time. Too, the cone may absorb moisture and thus increase greatly in weight. This moisture may also soften the cone, resulting in a loss of both low and high frequencies.

The edge or rim of the cone undergoes considerable strain as the cone

vibrates back and forth. The ring between the actual cone and the speaker frame may break or become fatigued, particularly if it is leather. There may be very little physical evidence of aging. However, the tone quality of the receiver will not be all that might be expected from it. If a receiver is being pepped up, replacing a cone that is several years old is a good idea.

## ELECTRICAL TROUBLES

The most common electrical trouble in a speaker is an open field coil. However, there may also be an open voice coil, a high-resistance joint in the voice coil circuit, or shorted turns in the speaker field.

Resistance in series with the voice coil will, of course, make the signal energy divide between the resistance and the voice coil. This cuts down output. This "high-resistance connection" may mean that a connection has changed from practically zero ohms to just a few ohms—4 or 5 ohms is a high-resistance value in a voice coil circuit, where the impedance of the voice coil itself may be only 2 to 6 ohms. Weak reception will be the only complaint, unless the loading effects of the voice coil on the output tube are affected enough to cause distortion in the output stage.

An open voice coil will usually cause a dead set. An ohmmeter will show either an open voice coil or high resistance, after disconnecting the coil so as not to get a false reading through the output transformer secondary.

► The usual speaker field difficulty is an open, which can be found with an ohmmeter. As another easy test, hold a screwdriver near the magnetic air gap in which the voice coil moves. With the set turned off, see how much pull is caused on the screwdriver by

the residual magnetism in the field assembly. Then turn the set on. If the field is energized, the screwdriver will be attracted much more strongly than by the residual magnetism. If there is no increase in the pull, the field is open or is not being energized.

Incidentally, this is one way of magnetizing a screwdriver if you want such a device to pick up small nuts or bolts in inaccessible corners of a radio chassis. Just hold the screwdriver near the air gap while the field is energized and it will be magnetized.

► It is not safe to wear a wrist watch when working on loudspeakers—the watch may become magnetized. While a jeweler can demagnetize it for you easily, sooner or later a balance spring will have to be replaced if the watch is repeatedly magnetized and demagnetized.

► The effect of an open field depends on how the speaker field is connected in the circuit. If it is used as a choke coil in either the positive or negative side of the filter circuit, an open field will interrupt the B supply for the receiver, so the receiver will naturally be dead.

On the other hand, if the speaker field has its own excitation supply, or is connected between B+ and B— as in many a.c.-d.c. receivers, an open field will cause very weak reception (also distortion). Reception also becomes weaker and distortion develops when the magnetic level of a p.m. speaker goes down.

A break in the speaker field is almost always caused by electrolysis or by excess current flow through the field. Electrolysis corrodes the field winding. Excess current causes overheating which will eventually open the winding—usually near the inside of the field, where the heat is greatest. A break caused by electrolysis is very

frequently right at the field terminals where the winding itself, made of rather small wire, is connected to heavier leads which emerge from the field enclosure. Sometimes you can cut the paper or tape wrapped around the field and expose these terminals. If the break is right at a terminal, you can often make a new connection, but the

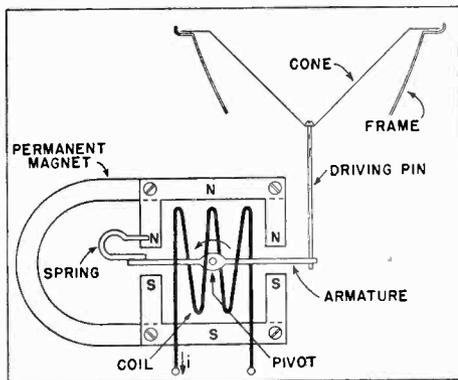


FIG. 24. The "works" of a magnetic speaker.

field must be replaced if the break is inside the winding.

► Weak reception and excessive field coil heat may indicate shorted turns. (Don't be surprised at the heat coming from the field coil even under normal conditions, however. Most loud-speaker fields dissipate about 8 watts of power, which will definitely make them uncomfortable to the touch after operating for a while.)

To check for possible shorted turns, allow the speaker to heat up for a half hour or longer, then measure the resistance of the winding while hot. Compare the measured value with the field resistance marked on the schematic diagram of the receiver. Usually such shorts develop between layers, so an appreciable change in resistance will be noticed.

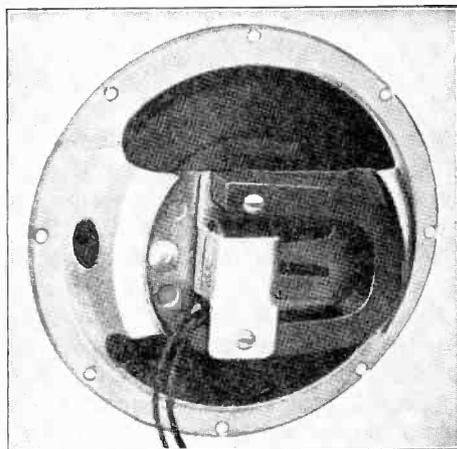
## MAGNETIC SPEAKERS

A picture and a diagram of a typical

magnetic speaker are shown in Fig. 24. These speakers are used in inexpensive a.c.-d.c. midsets, in some intercommunication systems, and as extension speakers.

When a signal is applied to the armature coil, the armature (which is pivoted in the center) swings back and forth in accordance with the shape of the signal wave form. The armature motion is transmitted through a driving pin to the cone. This pin usually passes through and is soldered to a metal cap fitted to the cone apex, as shown in Fig. 25. In some larger speakers, the solder may be replaced by a nut.

The power-handling ability of a magnetic speaker is slight; if excess signal is fed to it, the armature may



A picture of a magnetic speaker unit.

move enough to strike the pole pieces. Thus, noise can be created by overloading. Turning the volume down will clear this up.

► Trouble in magnetic speakers is generally caused by: Crushed cones; cones torn or fatigued around the metal apex cap; loose electrical connections; general cone fatigue, causing improper centering of the arma-

ture and allowing the armature to hit the pole pieces; iron filings in the air gap; loose driving rods; open armature coils; and loss of magnetism.

Damaged cones or loss of magnetism will usually cause weak, distorted reproduction; a new speaker is the best solution. Loss of magnetism can be identified by checking the output tube and the voltages applied to it. If everything seems to be normal, yet a circuit-disturbance test performed on the output tube gives only a weak thud or click, the speaker is at fault. If it has no electrical or mechanical defects, the only other possibility is loss of magnetism. The final test is, of course, to try another speaker.

Naturally, loose parts or connections should be tightened and filings in the air gap removed.

► Loud chattering sounds are caused by the armature hitting the pole pieces—recentering should be attempted. Some speakers have screws permitting armature adjustments, but most do not. If no adjustment is visible, see how the driving pin is attached to the cone. If a nut is used, screw it tighter; this may offset the sagging of a fatigued cone and recenter the armature between the pole pieces. If the pin is soldered to the cone cup, touch the tip of a soldering iron to the joint, taking care not to burn the cone. When the pin has loosened, the armature may move to its natural position. Remove the iron, let the joint set, and try the speaker.

If the armature has not recentered itself, slip shims (thin metal or paper

strips) between both sides of the armature and the pole pieces so the armature is blocked in the center of the gap. Then apply the soldering iron to the junction of the driver pin and the cone cup. When the solder melts, the cone will move to its normal resting position. Allow the joint to reset itself, remove the shims and test the centering—it should be correct.

If no shims are available, or cannot be introduced in the gap, notice the armature position and determine where the cone should be on the driving pin to center the armature. Apply the iron to the joint and move the cone manually to the proper position. Let

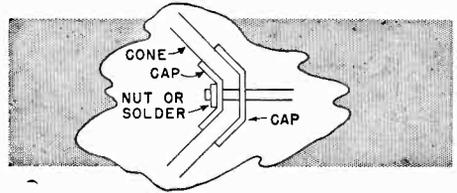


FIG. 25. How the magnetic driving unit is connected to the cone apex.

the joint set before testing the speaker. Repeat this procedure if necessary.

► Test the armature coil with an ohmmeter when an open or short is suspected. Watch for an erratic ohmmeter reading; this will indicate a partial open in the coil which will cause machine-gun-like bursts of noise when the system operates. Coils cannot be repaired unless the break is plainly visible. Since most jobbers do not stock replacement coils, the speaker should be sent off for an overhaul or a new speaker should be installed.

# Speaker Repair and Replacement

In dealing with speaker troubles, there are always four courses you can take. You can: 1, try to repair the defect; 2, install a replacement part; 3, send the speaker back to the manufacturer, take it to your local parts jobber, or send it to a firm specializing in speaker repairs;\* or 4, install a new speaker.

As a general rule, the cost of the cone plus your time is greater than the cost of a new speaker for midget a.c.-d.c. or three-way receivers. If you replace an open field coil, it is usually best to replace the cone at the same time, since cones deteriorate rather quickly. Thus, an open field in an inexpensive receiver requires a complete replacement speaker for best customer satisfaction.

With larger speakers this is not so true. Repairs or overhaul may be the best procedure. Of course, no repair should be attempted if the response of the speaker would be affected. And, as mentioned before, cones should be replaced rather than repaired if they are several years old.

Let us see how cones and fields can be repaired.

## REMOVING CONES

The first step is to examine the speaker carefully to see what kind of spider and rim mounting hold the cone in place.

In many older speakers, the cone rim is held by a metal ring bolted to the rim of the speaker frame. You can easily release the edge of the cone by unscrewing the bolts. The internal

spider shown in Fig. 22C was also commonly used in the earlier speakers. Detach the spider by unscrewing the screw holding it to the center pole piece, then disconnect the voice coil leads, and you can lift the cone and voice coil assembly out of the speaker frame.

► In most recent speakers, the edge of the cone is glued to the speaker frame. If the cone is to be removed and then replaced, paint the rim with a radio service solvent. This liquid softens the cement holding the cone and its spacing ring to the speaker frame so you can lift out the edge of the cone. If the cone is to be destroyed and a replacement used, you can run a knife around the edge, cutting the cone out entirely.

Again, you must disconnect the spider and the voice coil leads before the cone can be removed. If the spider is an external type secured by cap screws, remove them with an end wrench. Sometimes thin-nose pliers can be used, but the screws are in a rather awkward position between the cone and the frame supporting the cone. You thus have to work through holes in the frame which do not allow much room.

If the spider is a corrugated paper type, you can loosen it with a solvent or cut it off with a knife.

After removing the cone and voice coil from the speaker, carefully scrape off any paper adhering to the rim of the speaker frame or to the pole face. Clean the air gap thoroughly, using a small hand pump to blow out dirt and using a piece of tape to extract iron filings. Methods previously described can also be used.

► Do not destroy the heavy card-

\* If you cannot locate such a firm near you, your favorite radio supply house undoubtedly can get the speaker repaired for you.

board rings around the cone rim unless new rings are supplied with the replacement cone. These rings are used to position the cone so the voice coil will not be too far back in the air gap. Another ring is then used on top of the cone as a gasket. Sometimes this gasket ring is the only one used.

### INSTALLING CONES

Presuming you have the proper replacement cone, installing it is the reverse of removing the original cone.

If spacing rings are used, apply a coat of speaker cement on both sides of the bottom ring and both sides of the cone rim. Set the lower ring in place on the speaker frame rim and install the cone with the voice coil in the air gap. Be sure to position the

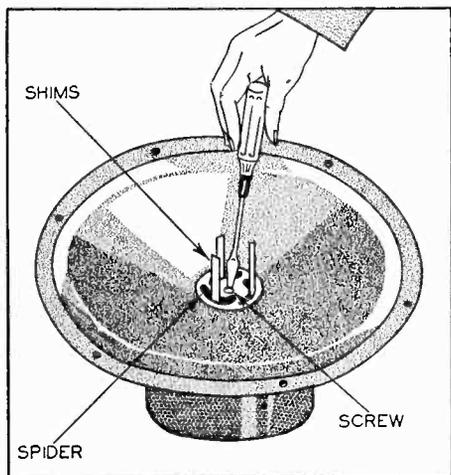


FIG. 26. How shims are used during adjustment of an internal spider.

cone so that the voice coil leads come near the speaker frame terminals. These leads must be looped away from the cone and directly back to the point of connection, not wrapped around the cone. If an external fiber spider is used, the spider legs must line up with the mounting supports.

Next, coat the outside ring or gasket

with cement and press it onto the cone, sandwiching the cone rim between the outer and inner rings. While the cement is still wet, adjust the position of the cone rim so as to center the voice coil approximately.

After the cement has set, center the voice coil with speaker shims or a cen-



Courtesy General Cement Co.

Withdrawing shims from a speaker having an external spider, after making the adjustments.

tering gauge. Speaker shims are thin strips or spacers (see Fig. 26) which are placed equal distances apart in the gap between the voice coil and the center pole piece, and thus hold the voice coil equally spaced from the center pole piece. Shims come in various thicknesses for different speakers.

A centering gauge is shown in Fig. 27A. These gauges are intended for specific speakers or voice coil spacings and cannot be used universally. Before the cone is installed, the gauge handle is passed through the voice coil and the gauge then pulled up within the coil. It spaces the voice coil from the center pole piece as shown in Fig. 27B.

With shims or a gauge holding the voice coil in the proper position, tighten the screws or bolts which anchor

the spider in place. If a corrugated paper spider is used, treat the edge of the spider with speaker cement and fasten it to the pole face of the speaker.

Allow the cement to set before removing the shims or gauge. Then solder the voice coil leads in place, positioning them so they will not whip against the cone.

To check the centering of the voice coil, remove the gauge or shims, then move the cone and voice coil assembly in and out by pressing against the cone rim with your fingers, applying equal pressure to both sides. You should hear no scraping sounds.

If you find the cone is properly installed, cement the dust button or dust cap in place, if one is used.

Next, connect the speaker leads to the receiver. Turn on the set but do not tune in a program. If the hum level is abnormally high, the voice coil may be connected backwards to the hum-bucking coil. Check by reversing the connections of the voice coil leads. If the hum is intensified, the original con-

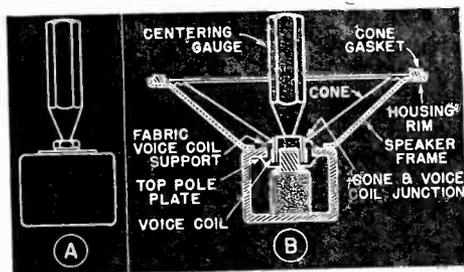


FIG. 27. Using a centering gauge.

nections were correct and the excess hum is caused by some chassis defect. However, if the hum is reduced, the last connection is the correct one.

Finally, try out the set to see if it sounds normal, with no scraping sounds or distortion.

► Sometimes the cone installation procedure will differ from the above

description, particularly where the cone is supplied separately from the voice coil. This type of construction is used chiefly with corrugated paper spiders, because it is difficult to glue down the spider with the speaker cone in the way. A typical replacement cone and voice coil assembly of this kind is shown in Fig. 28.

With this type replacement, cover the spider rim with cement and place the voice coil in the air gap, with the

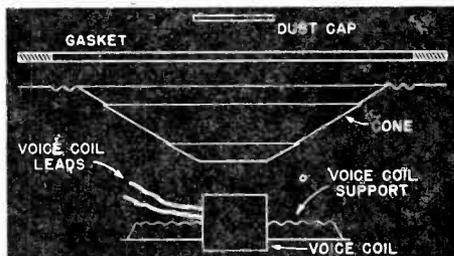


FIG. 28. A cone and voice-coil assembly which is put together during the installation of the replacement.

coil leads coming out toward the terminals on the frame. Use a centering gauge or speaker shims to center the voice coil properly. Next press the edge of the spider into place and let the assembly set long enough for the cement to harden.

At this point it is a good idea to solder the voice coil leads to the terminals on the speaker frame, as this is easier to do without the cone in place. Be sure to allow sufficient slack in the voice coil leads to permit free motion of the cone. Position the leads so they will be well away from the cone and speaker housing.

Apply a ring of cement around the rim of the speaker frame. (Some manufacturers also recommend coating the voice coil neck and cone apex with cement at this time.) Place the cone apex over the voice coil neck and press the cone rim tight to the speaker frame, using the voice coil as a guide. Allow

the cement to dry on the cone rim, then run a ring of cement around the junction of the cone and voice coil, being careful that the cement does not run inside the voice coil. After the cement has dried, remove the center gauge or shims. (The gauge should not be jerked out. Instead, turn it like a screw as you pull it out. Be careful not to apply such pressure that the voice coil is torn loose or the spider damaged.)

Cement the gasket in place around the cone rim. It's a good idea to turn the speaker upside down, so it presses

tion (Fig. 29) or a solid pot (Fig. 30). Notice that the yoke construction lets you see the field coil through the gaps at the sides of the yoke.

The speaker field fits around a center pole piece. This piece connects at the back either to the yoke or the pot, which forms a magnetic return path to the front pole plate of the speaker. Thus the magnetic field is concentrated in the voice coil air gap, between the front pole plate and the center pole piece.

To get the field coil out, you must

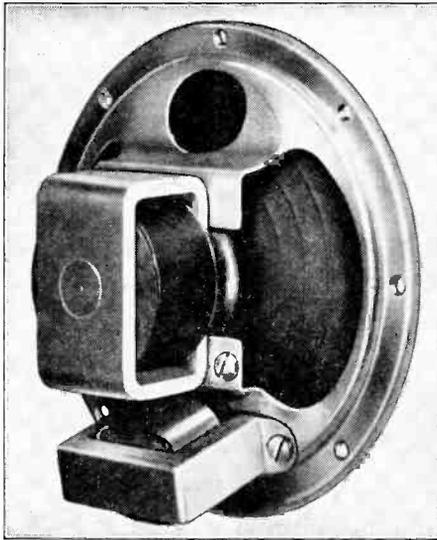


FIG. 29. This field enclosure is a yoke.

on the gasket while it is drying. Then cement the dust cap in place.

### SPEAKER FIELD REPLACEMENTS

In some speakers a replacement field can be installed without disturbing the cone-voice coil assembly. In others, the moving system must be removed before the field can be replaced.

The first thing to do is to examine the speaker carefully. The pot in which the field is housed may be made in either of two forms—a yoke construc-

tion either remove the center pole piece and slide the field coil out of the yoke or, if the speaker has a solid pot, separate the front pole plate and pot so the field can be lifted out of the pot assembly. (If the pole pieces and pot are welded together, as they are in some speakers, you must send the speaker back to the manufacturer or get a new one.)

► If the speaker uses yoke construction, examine the rear of the yoke. If there is a large nut at the back of the speaker, directly behind the center pole

piece, just remove this nut. You can then extract the pole piece from the front of the speaker assembly, unless an internal spider is used. If one is, you must remove the cone and voice coil assembly before the field coil.

If the core is not held in place by a nut, it will be either pressed or swaged into the yoke assembly. Swaging, a process similar to riveting, can be identified by hammer and chisel marks on the rear of the core. As this fastens the pole piece to the yoke permanently, you must remove the entire yoke from the front pole plate. If the yoke is folded, as in Fig. 29, the assembly must be returned to the factory.

► The method of extracting a pressed core is shown in Fig. 31A. Hold the

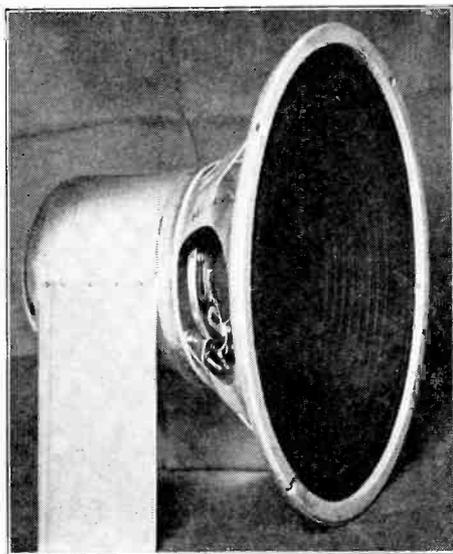


FIG. 30. The solid-pot type field enclosure.

speaker as shown, with the core over the opening between the jaws of a large vise (or over a board with a hole in it). Remove the dust button with a razor blade or sharp knife. Then, using a large piece of hardened drill rod or cold rolled steel, drive the core out the rear

of the speaker. Lift the field coil out of the opening in the yoke and install the replacement coil.

After replacing the field coil and putting all spacers and washers, the hum-bucking coil, and other parts in their original positions, reinsert the core through the voice coil opening and drive it into position as shown in Fig.

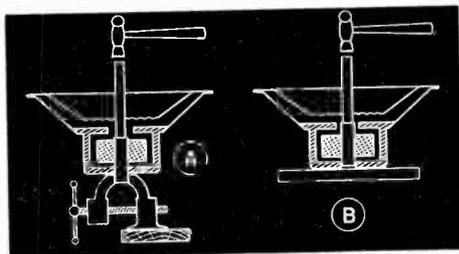


FIG. 31. How to drive out the core, and replace it, in the types having this feature.

31B. You must be very careful not to damage the voice coil. Drive the core straight down.

If the core is not centered in the voice coil opening, it may be centered by driving it from side to side, as necessary, with a center punch. A hole is provided in the front of the core for this purpose. Be sure to tap the center punch lightly, so as not to drive the core over too far and crush the voice coil form.

### TEST SPEAKERS

You can usually localize speaker defects by the methods given in this text. However, sometimes substitution of a test speaker speeds up the diagnosis. Regular test speakers mounted in cabinets for bench work are available and fill a real need in a large shop.

In general their construction is similar to that shown in Fig. 32. This outfit consists of an 8-inch p.m. speaker connected by a rotary tap switch to the secondary of a universal transformer. The transformer allows the voice coil

to be matched to any type or arrangement of output tubes.

To check a set which has push-pull output, connect terminals 1 and 3 to the plates of the power tubes and terminal 2 to B+. Disconnect the set output transformer, but leave its speaker field in the circuit. Turn the set on and adjust switch SW to give a match, indicated by least distortion. If results are satisfactory, you have definite proof that the receiver loud-speaker is defective.

For a single output tube, use terminals 1 and 3, following the same procedure.

If the field of the original speaker is open and you wish to check the receiver to see if anything other than the speaker is defective, substitute the 10-henry, 100-ma. choke and the voltage divider *R* for the field. To substitute for the low-resistance field of an a.c.-d.c. set, use terminals 6 and 7; for a tapped field, use terminals 7, 5 and 4; for a field with no tap, use terminals 7 and 4. Adjust the sliders on *R* so the correct resistance is added in the circuit.

Of course, the field substitution sec-

tion is not used if the set being tested has a p.m. speaker.

► Even if you have a test speaker in the shop, never leave the set speaker at the customer's home. The set speaker may be faulty, and you may be quite

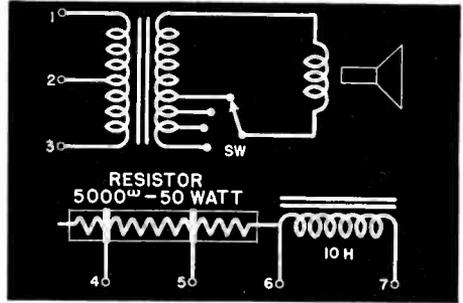


FIG. 32. A test speaker assembly.

embarrassed to find distortion, rattles or buzzes present after installing the repaired chassis.

Speaker troubles requiring the use of a test speaker are so rare that you are advised not to buy or build a test speaker until you have a large shop through which many receivers pass every day.

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# Lesson Questions

Be sure to number your Answer Sheet 42RH-1.

Place your Student Number on *every* Answer Sheet.

*Send in your set of answers for this lesson immediately after you finish them, as instructed in the Study Schedule. This will give you the greatest possible benefit from our speedy personal grading service.*

1. In a properly operating class A stage, does the plate current change when signals are applied to the grid?
2. Which TWO of the following operating voltage conditions will cause the tube to operate over the *lower bend* of its characteristic curve: 1, bias too low; 2, bias too high; 3, plate voltage low; 4, plate voltage high?
3. Why is an r.f. amplifier relatively free from amplitude distortion?
4. Suppose you have a resistance-coupled amplifier with a coupling condenser from the plate of one tube to the grid of the next. Explain briefly how leakage in this condenser causes distortion.
5. How can you determine whether a voltage found across a grid resistor is due to a gassy tube or to a leaky coupling condenser?
6. If tube  $VT_1$  in Fig. 13 burns out, will the symptom be: 1, hum; 2, distortion; 3, dead set; or 4, howling?
7. What defects would you suspect when distortion does not show up until the set has operated for at least a half hour?
8. Is voice coil rubbing more noticeable when *low-frequency* sounds or *high-frequency* sounds are being reproduced?
9. What is the best way of getting iron filings out of the air gap of p.m. speakers?
10. If a speaker field is used as a choke coil in the filter circuit, what will be the symptom produced by an open field?



## HOW STRONG IS THE CHAIN?

Let's suppose you are running a construction job and need a chain to carry a heavy and very valuable load. You order your blacksmith to forge a chain; you want it made of the best steel—each link must be perfect. Such a strong chain will carry the valuable cargo safely. But if just one link parts — **DISASTER!**

Life is like that. It is like a chain composed of ambition, perseverance, character—and **TRAINING**. Until your link of Training has been forged there is nothing to tie the Success chain together. And you need a strong link to hold it together.

You are using good, strong, time-tested material now. But don't skimp on material. Don't think you know a subject — **KNOW THAT YOU KNOW IT!** Don't try to cover a subject or a lesson in a day when common sense tells you it should require a week. Don't ruin your steel by allowing flaws to creep in which may eventually weaken your chain.

Put all of your N.R.I. Training into your success chain—make it as big and as powerful as possible.

*J.E. Smith*

**SERVICING NOISY  
AND  
INTERMITTENT RECEIVERS**

43RH-1



**NATIONAL RADIO INSTITUTE**

**WASHINGTON, D. C.**

**ESTABLISHED 1914**

# STUDY SCHEDULE No. 43

For each study step, read the assigned pages first at your usual speed, then reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind. Study each other step in this same way.

1. **How Noise is Produced** - - - - - **Pages 1-6**

Some noise is always present, due to atmospheric disturbances and normal circuit disturbances. However, the loud popping, crackling, sputtering sounds that result in a hurry-up call to a serviceman are caused by poor connections (which in turn produce arcing or partial open circuits) or by partial short circuits. Electrolysis, mechanical stresses, poor soldering and aging all produce these poor connections. A list of common parts defects is included.

2. **External Noise Sources** - - - - - **Pages 6-9**

The antenna system and man-made electrical disturbances can cause a great amount of noise. Hence, you must first determine whether the noise originates within the set or is external. If external and due to antenna system defects, you should go over the system and put it in good order. Certain household appliance defects can also be cured, but man-made interference is the subject of another lesson.

3. **Localizing Noise Within the Set** - - - - - **Pages 9-17**

Again we follow the standard service procedures to localize the defective stage, circuit and part. Stage-blocking or signal-tracing are the most useful techniques to use.

4. **Intermittent Reception** - - - - - **Pages 18-21**

The manner in which parts defects produce intermittent operation or intermittent hum, oscillation, noise, etc. is given here. You are shown that just certain parts defects are likely to cause intermittent trouble, which helps to narrow down the possibilities in all cases except intermittent operation or intermittent noise. This section shows how and why intermittent troubles occur.

5. **External Intermittent Troubles** - - - - - **Pages 21-24**

First, determine the exact nature of the complaint. In many instances, you will find that atmospheric conditions are the cause of the complaint. In cases of intermittent noise or fading, the antenna system may be at fault.

6. **Localizing Intermittent Troubles** - - - - - **Pages 24-36**

Having localized the trouble to the set, you are ready to locate the defect. Here is a case where wiggling parts and connections, thumping tubes and other parts, and pulling on wires may disclose the defect. However, if this fails, localization must be made. The signal tracer is the best device to use, as it breaks the receiver circuit up into smaller sections and makes it possible to find the trouble more quickly. However, other methods are given also.

7. **Answer Lesson Questions, and Mail Your Answers to N. R. I.**

8. **Start Studying the Next Lesson.**

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# SERVICING NOISY AND INTERMITTENT RECEIVERS

## How Noise Is Produced

**N**OISE is another of the annoyances which always exist to a certain extent in a radio receiver. The residual noise level may be very low in low-sensitivity midget receivers, but it is liable to be rather high in larger sets having more tubes and greater amplification.

What is noise? It may perhaps best be described as an unpleasing sound or sounds with irregularly-produced, sharp peaks—for example, popping, crashing, hissing, frying, clicking and scraping sounds. Whether a sound is a noise or not depends on the materials used in producing it. Striking a board with a hammer produces a noise, as the board structure and shape produce irregular sound waves. However, when a piano string is struck by the hammer in the piano action, a pleasing musical sound is produced, because the string vibration produces a regularly repeated wave form consisting of a sine wave fundamental plus harmonics. That is another characteristic of a noise—it is not a sine wave nor is it produced by a sine wave source. Thus, hum, squeals and whistles are not classed as noises and are therefore treated in other lessons.

► You can demonstrate one kind of noise to yourself with the simple circuit shown in Fig. 1. Place a pair of headphones in series with a flashlight cell, attach one side of the circuit to the file, then draw lead A across the file teeth. You will hear a clattering noise in the phones which will be quite similar to the noise caused by several radio difficulties.

► The irregular nature of noise is easily demonstrated by a c.r.o. With *no signal* tuned in on the receiver and the c.r.o. connected to indicate the audio output, you should get a straight line pattern similar to that shown in Fig. 2A. (You may get a small hum ripple.) Should you notice irregular jagged lines instead, as in Fig. 2B, then noise is present.

If a sine wave signal is fed through a noisy set, the noise will distort the pattern, as in Fig. 2C.

On the c.r.o. screen, a regular broadcast signal may look somewhat like a noise signal because it is continuously changing and is so crowded and complex. The noise pattern is more

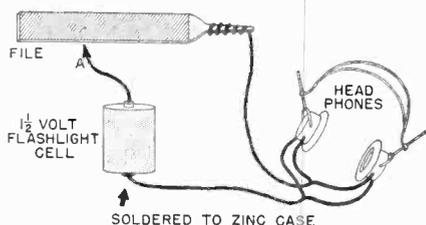


FIG. 1. While scratching the file with contact "A," a clattering, static-like noise will be heard in the phones.

broken up, but to avoid confusion, do not have a broadcast signal tuned in while looking for the noise source.

► The very sharp changes in noise signals mean that noise pulses contain frequencies ranging from low audio frequencies on up into the r.f. spectrum. If any noise pulse gets into a tuned circuit, some component of the pulse will probably be at the right frequency to pass through the tuned circuit just like a radio signal, or it

may "shock excite" the tuned circuit into momentary oscillation, producing a noise-modulated signal capable of being passed on through the radio. The tuned circuits do help keep out noise at other frequencies, however. We may pick up more noise at some frequencies than at others, but this is either caused by the noise source radiating better at those frequencies or by the receiver being more sensitive at those points.

### BACKGROUND NOISE

Before studying noises produced by defects, let's learn something about the kinds of noises which may be heard at any time from a sensitive receiver.

**Atmospheric Disturbances.** Practically any radio will pick up atmospheric noises at times. (F.M. receivers using limiters have the ability to wipe out the amplitude variations producing noises, and so do not reproduce atmospheric noise when an incoming signal has the limiter saturated.) These noises, caused by electrical disturbances in the atmosphere, are usually much worse as thunderstorms approach. The short waves are particularly subject to both these and man-made noises (to be described later.) You may be called in at some time to see if you can get rid of atmospheric noise on the short-wave bands. Since the amount of noise varies with the season and the wave bands used, suggest the trial of other bands. Sometimes a better aerial system, particularly the noise-reducing variety, will prove helpful by giving a better signal-to-noise ratio. However, since you have no control over atmospheric disturbances, you must explain the trouble to the customer instead of making any actual repair.

**Circuit Noise.** We usually consider

that each and every electron is perfectly controlled in ordinary circuit actions. Actually this is not true—there is always a certain amount of random electron motion. This random motion will have no effect as long as it is small compared to the electron movement we introduce as a signal. However, when we try to amplify signals so weak that they are not much larger than the random electron movement, we run into trouble with circuit noises.

► Similarly, tubes contribute greatly to the amount of noise. We think of the electrons emitted from the cathode as moving steadily to the plate. Actually, they tend to form small bunches and go over to the plate in shots or spurts. (This is called the "shot effect.") The *average* number of electrons emitted over a period of time is constant, but from instant to instant there are small variations in the number of electrons emitted. This effect is particularly noticeable in the first detector of superheterodynes. However, as long as these variations are small with respect to the signal, little noise is heard.

If a high gain radio receiver is tuned to a point on the band where no signals are picked up and the volume control is turned up, you will hear considerable amounts of noise. Then, as you tune in a signal, the a.v.c. circuits reduce the sensitivity of the set. If this signal is large enough, the amplification drops considerably. This reduces the noise level, but the strong signal is still reproduced at full volume. When this happens, the noise becomes unnoticeable, because the signal-to-noise ratio is so high. Therefore, you will always find receivers more noisy between stations, or when you attempt to pick up weak long-distance signals, than they are on strong local stations.

## BASIC CAUSES OF NOISE

Now let's turn to noise-producing defects. The sharp, ragged peaks and breaks in the signal produced by noise show that the noise is caused by a sharp and sudden change in current or voltage in a signal circuit or electrode supply circuit.

If we open and close a circuit rapidly, we make sharp changes in the circuit current and so cause noise. Similarly, if a short circuit across part of a circuit opens and

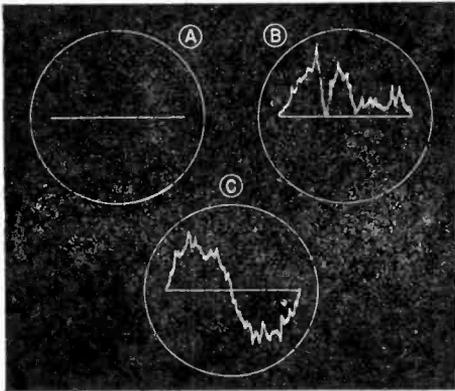


FIG. 2. The c.r.o. pattern for no noise (when no signal is tuned in and the hum level is low) is shown at A. A noise pattern is shown at B, while C shows noise mixed with a sine wave signal. Noise patterns resemble the rapidly changing patterns produced by voice or music modulations, except for being more irregular in shape and timing. To avoid confusion it is best not to have a broadcast signal tuned in while testing for noise.

closes rapidly, it will cause large changes in the current flow or in the signal voltage and will produce a noise.

Noise may also be produced by sparking or arcing at a poor contact. An arc is formed when two points at different potentials are so close together that a spark can jump between them. When this happens, the air in the gap ionizes and becomes semi-conductive. A current flow,

shown by a visible sparking, then starts through the air space. Since the resistance of the air gap varies rapidly, the circuit current is changed erratically and noise is produced. This arcing often occurs in circuits where oscillations may be set up, with the circuit in turn radiating electric and magnetic fields. This circuit then acts as a small transmitter, introducing its noise signal into other circuits.

*Thus, we can say any circuit condition producing an irregular current change will produce noise. Hence, a rapidly closing and opening contact of any kind can cause noise by producing either a partial open circuit or a partial short circuit. An arc also will produce noise.*

## HOW POOR CONTACTS ARE PRODUCED

Since some sort of faulty contact is normally responsible for noise, let's learn how these poor contacts develop so we will know where to look for them.

**Electrolysis.** One of the most prevalent sources of a partial open or arcing contact is a form of corrosion produced by electrolysis. This corrosion exists where moisture or chemical deposits on the surface of a wire provide a conductive path for current flow in and out of the wire. This flow sets up an electrochemical action, producing a form of oxidation or corrosion which eventually eats away the conductor. This action is hastened if the surrounding air is salty (near a seashore) or contains industrial fumes, or if there is a deposit of corrosive soldering flux on any of the circuit wiring. Tropical climates, which have high humidity, cause so much corrosion that sets for use in these climates must be specially insulated and moisture-proofed.

We can particularly expect corrosion to occur at exposed joints. The action will be retarded if the joint is wax-coated or is insulated by a compound or insulating tape which does not absorb and hold moisture and contains no active chemical ingredients itself.

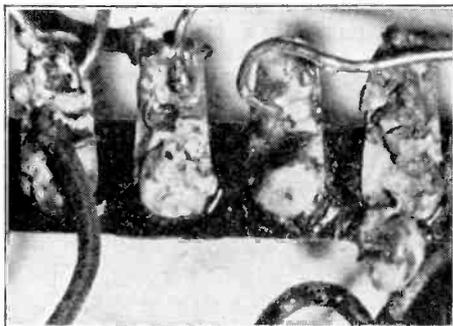
► Similarly, when coil windings are well protected, corrosion rarely develops on the winding itself. However, there is always the possibility of chemical deposits from the sweaty hands of workers, soldering fluxes and other material, being under the protective coating. In windings with many layers, such as audio transformers and speaker fields, electrolysis is liable to be set up between layers.

Remember, this electrolysis sets up an action which will eat through the wire itself. As the wire is eaten away, the reduced cross-sectional area produces resistance at this point. The resulting heat will burn away the remaining wire, providing a momentary open circuit. If the voltage is sufficiently high, an arc may form across the break. The heat of the arc may then melt enough of the copper to close the circuit again, and the action may be repeated over and over.

**Mechanical Stress.** Often, wires or parts are mounted so that contraction and expansion resulting from temperature changes may cause a break. Thus, you may find a resistor or condenser with its leads pulled so tightly between two mounting points that contraction caused by cooling will pull the leads loose from the part. Coil forms may expand when heated and thus break wires pulled tightly to the terminals. If the broken ends of the wire are held in place by wax or coil dope, an arc may form.

**Poor Soldering.** Again we must

emphasize the importance of good soldering. Usually the original soldered joints in a receiver are well made. However, servicemen all too frequently do sloppy soldering work. A cold-soldered joint almost invariably traps a considerable amount of soldering flux within the joint, where it may cause a high resistance between the wires being joined together. Always be suspicious of any



An example of poor soldering. The lumpy, cracked appearance shows too much solder was used and the joints were not heated enough.

joint with a large blob of solder on it. Upon heating such a joint, you will frequently find that soldering flux boils out of it.

Also, the average serviceman uses entirely too much solder. Great amounts of it often drip down between terminals. These may form a path between terminals and provide partial short circuits which can cause noise.

**Watch your own soldering.** Heat the joint thoroughly before applying solder, then use solder *sparingly*. If you have a receiver for repair and observe poor soldering, go over these joints with a hot soldering iron to remove excess solder. Be certain excess amounts of flux are completely removed.

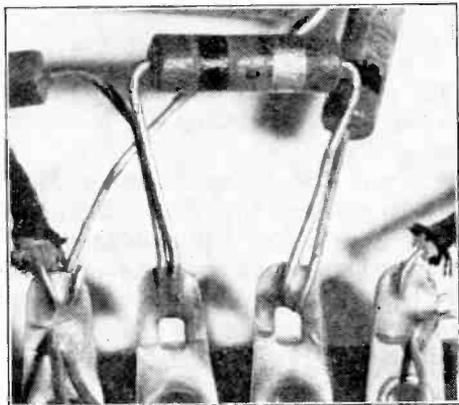
**Effects of Aging.** Insulation is greatly affected by aging. It will

dry out and crack, so that when wires are bundled together partial shorts can exist between them. If an arc occurs between terminals of a tube socket, the socket may carbonize and thereafter provide a semi-conductive path between terminals.

▶ Dust and dirt will inevitably collect on a radio receiver and will produce leakage paths between tube socket terminals, between parts mounted on terminal strips, and between the plates of tuning condensers.

▶ Wave-band switches and other contacts which depend on spring tension will eventually lose this tension with use and thus become noisy.

▶ The volume control is one of the most frequent sources of noise. Poor contacts may develop between the rotor arm and its contacting ring and



Good soldering produces a smooth, even coating of solder. The joints have been heated sufficiently to melt the solder and just enough solder has been used to coat the joint thinly.

between the rotor arm and the resistance element. The element itself wears away and becomes pitted, presenting differing contact resistances at different points.

### COMMON PARTS DEFECTS

The following list shows the more common sources of noise within the radio receiver. The items are ar-

anged in approximately the order in which they are most commonly found.

#### Tubes:

Internal shorts; poorly welded connections; gas.

#### Volume Controls:

Worn resistance strips; poor contact to resistance strip; poor contact between rotor and contacting ring.

#### Coils, Air or Powdered-Iron-Core:

R.F., oscillator or i.f. coils partially opened by corrosion; shorts between turns; leads snapped near lugs; coil shields making poor contact to chassis.

#### Coils, Iron-Core:

A.F. transformers, power transformers, chokes and speaker fields corroded where the winding and external leads join; shorted layers in windings; shorts between windings and core; loose laminations; poorly grounded shielding.

#### Tuning Condensers:

Dust and metal particles between plates; worn wiper contacts; plates bent or warped out of shape.

#### Resistors, Candohm (5- or 10-watt resistor mounted in a metal housing, used for voltage dividers):

Poor contacts between taps and resistor elements; internal arcing; shorts between turns on wire-wound resistor elements.

#### Switches:

Dirt and corrosion on contacts; possible loss of spring tension.

#### Carbon Resistors:

Cracks in the resistor element.

#### Condensers, Wet Electrolytics:

Internal arcing caused by sludge partly shorting the plates; scintillation (arcing) caused by high-voltage peaks breaking down the

dielectric film, poor contact to grounding bracket on chassis.

### **Leakage Paths:**

Leakage between terminals of a tube socket or along terminal strips, caused by dust, dirt, soldering flux, excess solder, or carbonization of the insulating material.

The preceding list gives just the

*more common* troubles in their approximate order. You may find other troubles, or you may find the order of troubles is different, depending on the kinds of receivers you work on and the type of climate you have. The more moist the climate (particularly if the percentage of salt or corrosive fumes is high), the greater the amount of trouble.

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## **External Noise Sources**

Noises are not always caused within the set. It is well to know what external sources can cause noise, because you will be called on to cure these troubles as well.

### **LOCALIZING NOISE TO SET**

As with other complaints, question the customer carefully. Find out if there is anything about the noise which will reveal its source. A noise that occurs at a certain definite time every day, and at no other time, is undoubtedly caused by some external source, perhaps some machinery in the neighborhood. If noise is heard every time the oil burner comes on, it is probably caused by the oil burner electrical system. Also, be sure the customer is not complaining about normal atmospheric disturbances.

If no clue leads you to the source of the noise, turn on the radio to confirm the complaint.

► If again no clue leads to the trouble, the next step is to disconnect the aerial and ground leads from the set, connect them together, and short the aerial and ground posts on the receiver with a small length of wire. (If the set has only an antenna, disconnect it and ignore the statement about shorting the aerial and ground terminals.)

With the antenna and ground leads

moved well away from the radio, turn the volume control all the way up and rotate the tuning dial. Adjust the other controls also. *If the noise is not present, the trouble is outside the set*, and is either in the antenna-ground system or is caused by an external noise source being picked up by the antenna.

On the other hand, *if the noise is still present, it is originating in the receiver or is coming in over the power line.*

► We must now localize further. Most servicemen carry a line noise filter. This is a complete plug-in unit containing chokes and by-pass condensers and is obtainable from any radio supply house. Plugging this filter into the outlet and the radio into the filter blocks noises from coming in over the power line. Hence, if the noise stops, it is coming in over the power line; if it continues, it is probably originating in the receiver.

Usually, if the noise is traveling over the power lines, disconnecting the antenna and ground will reduce its intensity (because the noise is probably being radiated from the power lines and introduced again via the antenna).

**Using a Test Set.** A test receiver—a three-way (a.c.-d.c.-battery) portable receiver with a built-in loop and

provision for an external antenna—is excellent for localizing the noise source. To use it, treat the customer's radio as directed in the previous section, and at the same time operate the test receiver on its batteries (not plugged into the power outlet). If noise is heard in both receivers, it is of external origin.

You can check the customer's antenna by connecting the test receiver (still operating on batteries) to the antenna, and you can plug the test receiver into the power line outlet and switch to power line operation to determine if noises are coming in over the power line.

► If the customer's receiver has a loop antenna, it is not practical to disconnect the loop. However, you can rotate the loop if it is adjustable, or rotate the entire cabinet if a fixed-position loop is used. If the noise varies in loudness when the loop is rotated, the loop is probably picking up the noise. This test will not always work, because the noise field may be so strong that it apparently has no direction. If so, using an extra receiver is the only sure test in the customer's home.

► Of course, you can always carry the customer's radio to your shop. If the set is quiet in the shop, the noise is probably originating in or near the home of the customer (unless jarring the set in transit has temporarily cured some internal trouble). However, if the same noise continues, it is definitely originating within the set.

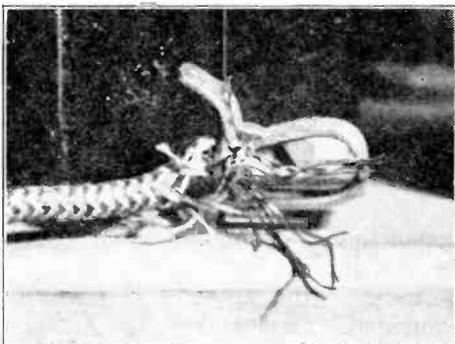
### OUTSIDE NOISE

There are many external sources of noise which you will often be expected to run down.

**Man-Made Interference.** Noises are caused by faulty switch contacts, arcing at motor brushes, and other faults of man-made apparatus. This

subject is so broad that complete details for running down and curing such interference are given in another lesson in your N.R.I. Course.

**Antenna System Troubles.** Many things may happen to make an antenna system a source of noise. You may find the antenna is down altogether, or is touching another wire, a tree, or some other object. Some one may have connected another lead-in to the same aerial wire. There are liable to be corroded contacts where the lead-in wire joins the antenna, as well as at any other point where the lead-in wire is not a continuous wire



Loose wire ends touching the chassis will produce noise. Be sure ALL strands of wire are twisted together and contact ONLY the terminal to which a connection is intended.

—at a lightning arrester or at a lead-in strip, for example.

You may find a loose connection at the binding post or Fahnestock clip where the antenna or ground leads fasten to the receiver—or even frayed ends of the lead-in wire shorting to the set chassis. Carefully examine all these items, even going all the way up to the antenna wire if it is a logical suspect. If any of these defects are found, repair the condition and try the receiver again.

► If nothing is visibly wrong with the antenna system, reconnect the antenna and ground leads to the re-

ceiver. Wiggle the wire from the receiver to the window strip to see if noise is produced. Also wiggle the strip and shake the lead-in wire itself. If there is a poor connection at any point, you will notice an increase in the noise level as the joint is moved.

Many modern all-wave antennas have a two-wire, twisted-pair lead-in. After these antennas have been up for several years, the insulation between the pair of wires may rot to such an extent that shorts can develop between these leads. Shaking the lead-in wire vigorously will usually show up the trouble by increasing the noise greatly.

If you suspect the lightning arrester is defective, disconnect it and see if the noise stops. Be sure the connections at the arrester are clean and tight.

Wiggle the ground lead to see if it is loose. If it is, tighten the ground connection or grounding clamp.

► You can use test equipment to determine if the antenna is partially grounded. First, be sure the antenna is not in any way involved with electrical wiring by checking with a voltmeter between the antenna lead-in and ground. (Use an a.c. meter in districts where the power lines are a.c. and a d.c. meter in a d.c. district.) You should get no voltage reading. If you do find voltage, be careful to clear up the short before reconnecting the antenna to the receiver.

If you find no voltage, check for grounds with an ohmmeter. Connect one ohmmeter probe to the antenna lead-in and the other to ground. Shake the lead-in and watch for an intermittent or continuous reading, showing a short between the antenna and some grounded object. Use the highest range of the ohmmeter.

► When the customer complains of noise that occurs during rainstorms or

when there is wind, watch out for an antenna erected so that it can sway in the wind and touch foreign objects (such as metal gutter piping, another antenna, or metal weather-stripping). You may find that rainstorms change the resistance to ground by providing leakage paths across insulators. Usually a high-range ohmmeter will show if any leakage exists. If you find any, the antenna system should be repaired before going further.

**House Plumbing and Wiring.** If the complaint is noise when any one walks around in the room, the resulting vibration may be jarring the radio (this condition will be discussed later), but probably, some water pipes or electric cables are rubbing together under the floor. This will cause a great amount of noise. It can be cleared up either by wiring the pipes together to make a good electrical connection between them or by wedging the pipes apart with a piece of wood so they cannot touch. Move electric cables apart and away from pipes.

An unusual case is one where static discharges are produced when walking across a thick rug. This annoyance will occur only under certain conditions of temperature and humidity. If these conditions cannot be controlled, moving the radio or the rug to another room is the only cure, although a more efficient antenna system may be helpful.

► Sometimes noise will be caused by loose connections in the electrical wiring system. Be sure to wiggle the receiver plug in its wall or floor outlet. If a great amount of noise occurs, bend the plug prongs to make better contact or advise installation of a new wall outlet.

Many people use cube taps (devices for connecting three or more fixtures to a single wall outlet). These devices quickly lose spring contact

tension and become noisy, so should be eliminated if possible. Remove all such extra appliances and plug the radio directly into the outlet to see if this clears up the noise.

Be on the lookout for a line cord with a frayed covering near the plug. Too many radio owners grab the cord to pull the plug out of a wall socket. This soon frays the insulation and loosens the connections between the cord leads and the plug.

**Electric Fixtures.** A great amount of noise is caused by light bulbs which are just ready to fail and by poor wiring in floor lamps and other electrical appliances about a home. Always be sure to ask the customer if the noise is noticed only when he turns on his floor lamps, the light in the dining room, or some other switch about the house. If so, recommend the fixture be examined by an electrician (or go over it yourself if you also repair lamps, etc.).

Noise impulses of this kind will usually travel to the receiver over the power line. They are particularly apt to occur where antenna leads run

parallel to power cords for any distance, or where the defective lamp is plugged into the same outlet as the radio.

### SYMPATHETIC VIBRATIONS

Sound waves from the receiver loudspeaker may cause mechanical noise by setting parts of the radio or objects in the room into vibration. The mounting bolts holding the receiver in the cabinet, and the glass or celluloid covering over the dial are common offenders. Receiver cabinet doors and handles, and loose ornaments on or near the set are other noise sources you may find. Careful listening, and touching objects with your hand to feel or stop the vibration will lead you to the offender. Obviously, the part or object must be fastened more securely or removed from the room to correct this condition.

You will probably be called in oftener than you expect to correct conditions like these. Customers often think such noises come directly from the speaker.

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## Localizing the Noise Within the Set

Once you know the noise source is within the radio, you must localize it to a particular section, stage, circuit and part.

Noises have certain characteristics that sometimes help and sometimes hinder localization steps. If noise acted only as a signal and followed the signal paths exactly, you could use the ordinary methods of localization. However, a source of noise will often radiate electric and magnetic waves which may be introduced at several points in the receiver. For example, arcing at a corroded fuse

clip, as in Fig. 3, will not only introduce a signal which may follow the power supply leads to all stages, but will also produce radiated fields which may enter any stage directly in spite of the receiver shielding.

Mechanical shocks caused by jarring the receiver, or electrical shocks caused by a sudden change in current (snapping the receiver or a lamp switch off and on) may make the noise stop or start. This fact puts noise in a class with intermittent reception and at times makes the source of trouble rather difficult to locate.

## EFFECT-TO-CAUSE REASONING

Once you decide the noise is in the set, pay careful attention to the effect of rotating the various controls. If turning the volume control to zero cuts the noise off altogether, and the set is a modern one with the volume control at the input of the audio amplifier, then the noise is developed in the r.f.-i.f. section of the receiver. If

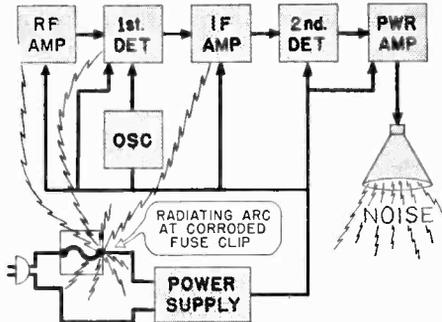


FIG. 3. Arcing produces a noise radiation which can travel directly to several stages at once.

the noise is just reduced, however, it is getting into more than one stage and will have to be localized by other means. On the other hand, if the noise remains as strong as ever, it must be in the audio amplifier or power supply.

► Notice whether the noise becomes much worse while the volume control is rotated. If so, the volume control itself is probably the source of the noise.

Watch for additional clues. If the volume control is defective, you will frequently find that noise is accompanied by erratic changes in volume and, in some cases, by hum.

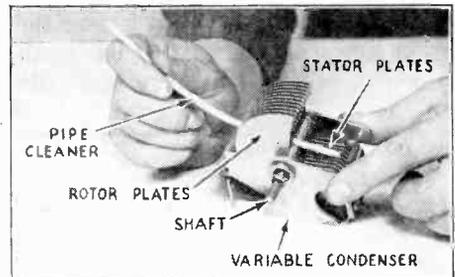
► Rotate the tone and sensitivity controls also. These are made just like volume controls and are frequently a source of noise. Obviously, replacing them will clear up the noise they cause.

► Throw the wave-band switch back

and forth a few times to see if this introduces unusual noise or clears up the noise temporarily. Either effect will indicate that the wave-band switch itself is the source of noise, provided you are not jarring associated parts or wires. If an examination leads you to the switch, clean it carefully, bend its contacts, or replace it—depending on the nature and extent of the trouble.

**Tuning Condenser Troubles.** Try tuning the receiver. Any loud noise occurring while the dial is being rotated indicates dirt or metal particles between plates, touching plates, or poor wiping contacts in the tuning condenser gang. Reception may be cut off altogether at low frequencies; this usually means shorted plates. Oscillations may occur when poor wiping contacts develop.

Bent or warped plates can sometimes be straightened. Some older types warp so badly they must be re-



How to clean between tuning condenser plates with a pipe cleaner. (You do not have to remove the condenser from the chassis to do this cleaning; this one is used for illustration only.)

placed. The plates may be cleaned of dust by running a pipe cleaner between them.

► Metal particles, or condensers which are difficult to reach, may require the more drastic treatment of burning out the dust and metal flakes with high voltage. Make up a circuit like that in Fig. 4, using a power transformer

with a protective lamp bulb (60 to 100 watts) in series with the primary. DISCONNECT THE R.F. COIL LEADS FROM THE TUNING CONDENSER BEFORE USING THIS DEVICE. Otherwise, the high voltage will burn out the coil. To use it, connect the high-voltage winding across the tuning condenser section, close switch SW, and rotate the condenser plates. Large sparks will occur as metal slivers are burned out. If the plates touch, the protective lamp will light; you should then open switch SW and clear up the short if possible. After the repair, remove the transformer connections and reconnect the r.f. coil leads.

### STAGE ISOLATION

If the tests so far have not localized the trouble, you can localize the defective stage either by using a stage blocking or stage interruption test, or by using a signal tracer.

**Stage Blocking.** The volume control test previously described is a form of stage blocking, since it prevents noise signals originating in the r.f. section from being passed on to the a.f. amplifier through normal channels. Of course, if the noise is being radiated, it may get around the volume control anyway.

In using the following methods, always remember that the noise may radiate around stages or travel through supply leads, as well as travel along the normal signal paths. However, the noise will always be diminished as long as you are blocking between the noise source and output, and will usually disappear entirely when the defective stage is blocked.

► There are several ways of blocking stages, all of which are intended to make it impossible for the stage to pass a signal.

One of the most effective ways is to

pull out tubes, working from the output back toward the input. The noise will stop or decrease greatly each time you remove a tube, as long as the noise originates further back toward the input. When you pull out a tube and find the noise remains as loud as ever, the trouble is probably in the last stage interrupted (that is, the next stage toward the loudspeaker). Both tubes must be pulled out simultaneously to block a push-pull stage. ► Of course, the tube-pulling technique cannot be used in a.c.-d.c. receivers where pulling out a tube would interrupt the entire filament circuit.

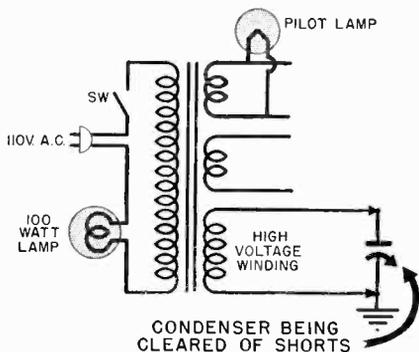


FIG. 4. A high voltage applied to the plates of a tuning condenser will burn out metal peelings from between the plates.

or in battery sets where removal of one tube might put excessive filament voltage on the others. In these cases, you must short the signal input circuit, stage by stage, by shorting the resistor or transformer secondary across which the signal normally appears. If one end of the input part connects to chassis or ground, you can hold a test lead between the control grid terminal and the chassis. However, if the return end of the part connects to some bias source, you must hold the test lead right across the part terminals.

Block each grid input in turn, moving the test lead along from grid circuit to grid circuit toward the an-

tenna. The noise will be blocked altogether or reduced greatly as long as the source is further back toward the antenna. When you come to a grid circuit where blocking does not affect the noise, however, the noise is either in that same stage plate circuit or in the coupling device to the next stage toward the loudspeaker. In other words, the defect is between the blocked grid circuit and the next grid circuit toward the loudspeaker.

It is possible to block the output device also, if you are careful to avoid short circuiting the plate supply. The

the volume control all the way up and rotate the tuning condenser gang. The noise will still continue, which shows it is originating in the receiver or coming in over the power line. A line filter does not affect the noise, so you know it is in the set.

Rotate the volume control to zero volume. This partially cuts out the noise, showing that it is in the r.f.-i.f. section of the receiver.

Turn the volume control back up again. If this is a standard a.c. radio, pull out the second detector tube. This will block the noise or reduce

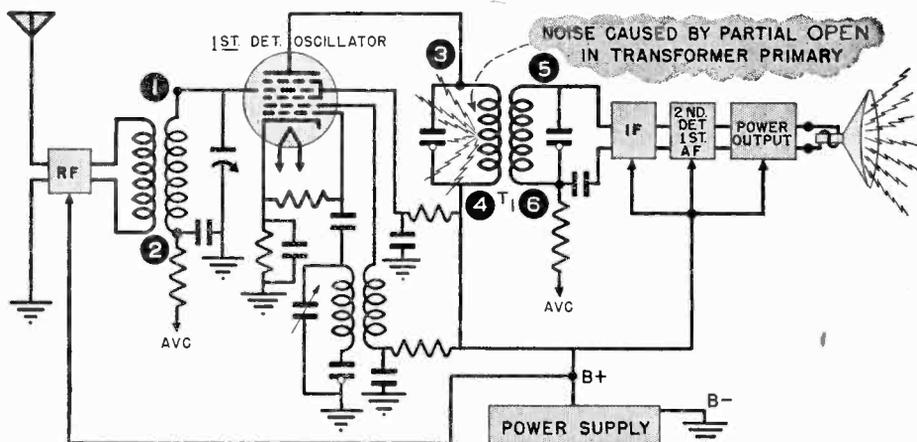


FIG. 5. A typical converter stage, with noise produced by a partial open in the i.f. transformer primary.

test lead must be held *across the load only*. The same procedure as for grid blocking is then used.

**Examples of Blocking.** Suppose you have the receiver shown in Fig. 5, in which a partial open in the primary of the first i.f. transformer  $T_1$  causes a noise-producing variation of the mixer plate current, and produces a rapid machine-gun-like crackling in the speaker. Here's how you could localize the trouble:

First disconnect the aerial and ground and short the binding posts together with a piece of wire. Turn

it to a very low value. Replace the second detector in its socket, then pull out the i.f. tube. The same blocking will occur. Replace the i.f. tube, then pull out the detector-oscillator tube. This time the noise will stop completely.

Replace the detector-oscillator tube, then pull out the r.f. tube. The noise will continue. This shows you that the noise source is in the detector-oscillator stage.

► Now suppose the receiver is an a.c.-d.c. set or is battery operated, so the tubes cannot be removed. You should

localize the noise to the set, then to the r.f. or a.f. section, as you did with an a.c. receiver. You can then find the defective stage by blocking the stage input or output circuits (or both), one at a time, moving from the output toward the input.

Fig. 6 shows how to use test leads for blocking the audio amplifier circuits. Positions 6 and 7 block the outputs of tubes  $VT_3$  and  $VT_4$  respectively. Positions 4 and 5 block the inputs of tubes  $VT_3$  and  $VT_4$  respectively. If only one of these positions is used, only one of the push-pull tubes will be blocked, and noises originating nearer the set input will pass through the tube which is not blocked. This is all right if you are trying to determine which output stage is defective, but trouble in another stage requires blocking both tubes in the push-pull circuit. Therefore, you should apply the test leads at both 6 and 7 or both 4 and 5 simultaneously. (Test leads with clips at each end should be used for this purpose), or use one test lead from plate-to-plate or from grid-to-grid.

If the tests at positions 6 and 7 or at 4 and 5 block the noise, you can move back either to position 3 at the output of tube  $VT_2$ , or to position 2 at the input of tube  $VT_2$ . Finally,

move back to position 1 or to the grid circuit of tube  $VT_1$ .

► Blocking the output as well as the input lets you determine just which circuit in the stage contains the defect. For example, if the test lead held at position 3 eliminates the noise but the noise continues when you short-circuit the grid input at position 2, the noise must originate in the plate circuit of tube  $VT_2$ . If, on the other hand, you shorted only input circuits, you would go from positions 4 and 5 back to position 2 and wouldn't know whether the trouble was in the plate circuit of  $VT_2$  or in the grid circuit of tube  $VT_3$  or  $VT_4$ . (Usually, however, the noise source is in a d.c. circuit, so the plate circuit would be the logical suspect.)

► One difficulty of shorting the output circuit of a stage is that you deal with a high-voltage circuit and must be very sure you get between the right points. *Never short from plate to chassis*; this shorts the plate supply and may damage some parts in the plate circuit.

► This shorting procedure can be applied to the r.f. circuits shown in Fig. 5. Here, if the i.f. transformer were faulty, shorting between terminals 5 and 6 would eliminate the noise, but shorting between terminals 1 and 2

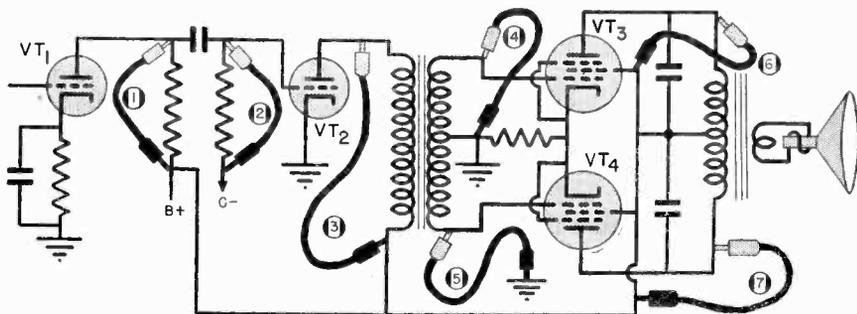


FIG. 6. Here are the positions for using test leads for the stage-blocking technique. Instead of blocking the output tubes individually, many servicemen block the entire push-pull stage with one test lead by holding it from plate to plate, or from grid to grid, of tubes  $VT_3$  and  $VT_4$ .

would not. This shows the trouble is in the detector-oscillator circuit.

Notice—you should short directly across the input part, not go from 5 to ground or from 1 to ground. There may possibly be a bias supply in the a.v.c. network somewhere; so going from 1 to ground or from 5 to ground would remove the bias and upset the circuit.

If you notice the noise diminishes as you block each stage but does not disappear, you can be fairly certain that the noise is arising in the power supply or is being fed through it.

**Signal Tracing.** Servicemen who have signal-tracing equipment generally use it instead of the signal blocking technique to localize noise. Remember that the noise is the signal, and in tracing to find its source you may move the signal tracer in either a forward or a reverse direction. If you move from the loudspeaker toward the antenna, the noise will decrease in intensity, since fewer stages of amplification are between the noise source and the signal tracer. When you pass through the defective stage, the noise will disappear entirely or become very weak.

If you move the signal tracer from the antenna toward the loudspeaker, no noise will be heard, or the noise will be at a very low level until the defective stage is reached. Then the noise will at once increase to a much higher level.

A visual indicator on the signal tracer, such as a magic eye or a meter, is not very satisfactory for noise tests. Use the audio output or feed the output of the signal tracer into the vertical plates of a c.r.o.

### **LOCATING THE DEFECTIVE CIRCUIT AND PART**

After localizing the noise to a particular stage, first test the tube or,

better yet, try another one. If the noise continues the tube is not at fault, so you must again use effect-to-cause reasoning. You will recall that noise produces erratic changes in current flow. Therefore, if there is no signal coming into the set but the noise is heard from the loudspeaker just the same, an operating current is changing. This means that the noise source is probably in a plate or screen grid supply circuit.

An intermittent open in the control grid circuit would also cause noise by removing bias and thus changing the plate current. However, this kind of trouble is far rarer than a defect in a current-carrying circuit. If in the control grid circuit, it is probably caused by a defective connection of such a nature that mechanical jarring makes it open and close rapidly. Arcing will not be the cause, for there is no current flow to form an arc.

► On the other hand, if the noise is heard only when a signal is tuned in, the trouble may be in the speaker or may be caused by speaker vibration of some part or connection. If jarring the set and speaker does not make the noise appear or disappear, the source is in a signal circuit (a grid circuit or a circuit isolated by blocking condensers so no supply currents flow through it).

► A defect in an i.f. transformer makes a different noise from that caused by, say, a defective volume control. The actual sound produced will be a valuable clue, when you have had sufficient practical experience to recognize the many different noises which may develop.

► Once you localize the noise to a particular stage by the stage blocking method, you can use several different tests to determine just which circuit contains the defect. If you pulled out tubes, you wouldn't know just

which circuit it might be, although it is logical to first suspect a current-carrying circuit such as the plate or screen grid. If you used a test lead and blocked the input circuit, you would usually be justified in assuming the noise source was in a plate circuit.

Make a careful examination of the parts in and about the suspected stage. Look for the characteristic green spots which mean corrosion. Corrosion, by the way, is of course far more likely to occur in a current-carrying circuit than in a signal circuit which carries no direct current. If you put out the lights on your work bench, you can frequently see arcing if it is present.

**Meter Tests.** Since noise means that the current is changing rapidly, there will be a voltage variation somewhere in the set. Using a voltmeter between B— and the various tube electrodes, you should notice a quiver in the meter needle each time a burst of noise occurs if you are in the circuit containing the defect. Remember, however, that both the plate current and the screen grid current flow through the cathode resistor; also, sometimes the voltage change in one circuit will produce a change in another. Therefore, a voltage variation test may or may not locate the defect.

A voltmeter is particularly useful when the plate circuit is suspected. Remove the tube from the socket, which will stop the noise. Now, connect the voltmeter between the plate terminal of the tube socket and B— or the set chassis. If an erratic meter reading results, and the noise appears when the meter is connected but disappears when the meter is disconnected, the noise source is in either the plate load or the supply circuit. The reason you can hear the noise

even with the tube out is that the meter draws sufficient current through the defective part to start the noise again.

Should the noise not appear, connect the meter between the plate and the cathode terminals of the tube socket. If noise now appears and the meter reading varies erratically, the trouble is in the cathode circuit for that stage.

If the noise does not appear in either case, it is in another circuit. The screen grid and control grid circuits are logical suspects.

► With the set turned off, you can

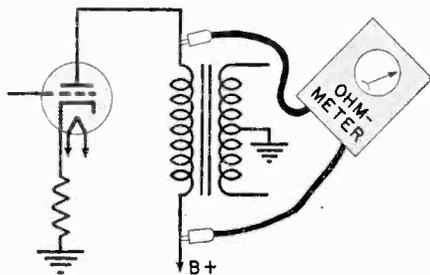


FIG. 7. How to connect an ohmmeter when looking for noise. The ohmmeter current flow is frequently sufficient to cause the defect to "act up," causing a wavering reading when the ohmmeter is connected across the defective part.

check with an ohmmeter directly across a suspected part (see Fig. 7). The ohmmeter will frequently provide the necessary current to cause the source of noise to act up. An erratic reading will show the part is defective.

**Shock Testing.** When the noise source is an i.f. or a.f. transformer in an a.c. set using a power transformer, you might short-circuit the plate terminal of the tube socket to the chassis momentarily with a screwdriver, while the set is turned on. The resulting high current will either burn out the defective spot altogether or will temporarily heal it. In either case, whether the noise disappears or

the set goes dead, you will have definitely localized the trouble to the plate load and can go ahead with a replacement.

You should replace the transformer even if this treatment temporarily heals it, for almost always the open will again occur or the part in question will have another corroded spot forming.

You don't have to worry about accidentally burning out a good part when making this test on an a.c. set with a power transformer, for a part in good condition can stand such a momentary overload without trouble.

source of noise will frequently let you pick it up; you can thus localize the source by determining where the noise sounds loudest.

If this fails, you can touch the signal tracer probe to various points in the suspected stage to find the point where the noise is most intense. For example, the imperfect joint in Fig. 8 will radiate noise and send noise pulses over the B+ wiring. As you move from point 1 to points 2 and 3 the noise will get louder; right at the defect, it will be loudest.

► You can use a c.r.o. as a signal tracer by connecting the grounded

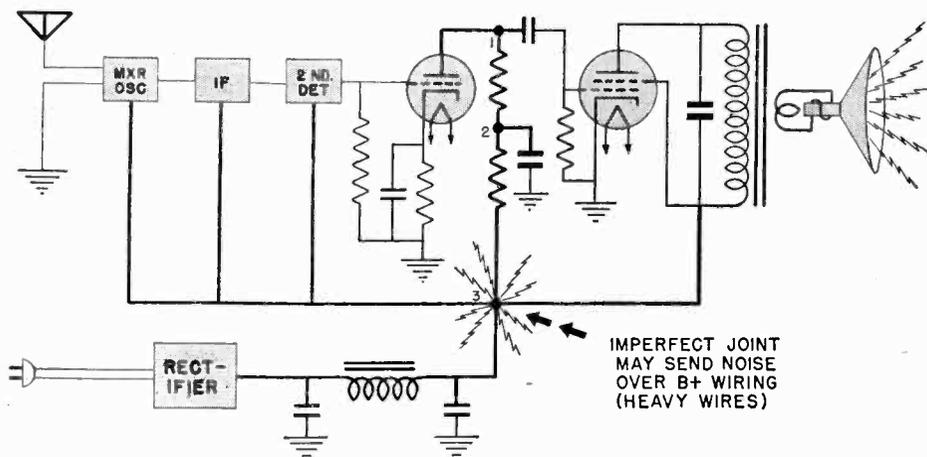


FIG. 8. How noises may travel over the wiring and thus get into several stages at once.

Any part which does burn out would fail sooner or later, anyway.

This shock test cannot be used safely on a.c.-d.c. or battery sets. In an a.c.-d.c. set, even a momentary short might ruin the rectifier tube, while in battery circuits there are no limiting factors on current flow and parts may be burned out or the batteries unnecessarily discharged.

**Signal Tracing.** Since noise is a complex pulse with components of many frequencies, you can often use the a.f. section of a signal tracer to locate noise in any part of the receiver. Bringing the probe near the

vertical plate terminal to the chassis or B- and the hot vertical terminal to a test probe. This probe is used exactly like the hot probe of a signal tracer. Turn on the c.r.o. vertical and horizontal amplifiers and turn up the gain controls. Keep the sweep frequency at some low frequency. The sweep will cause a line to be traced across the c.r.o. screen. When noise is picked up by the hot vertical lead, you will get a pattern similar to Fig. 2B, which will constantly shift.

If a sine wave is sent through an amplifier and the c.r.o. sweep is adjusted to the same frequency, any

noise picked up by the probe will cause sharp peaks, lines, and breaks on the sine wave (Fig. 2C).

**Mechanical Tests.** Many noises respond to mechanical shocks. If you find rapping on the chassis will make the noise appear, disappear, or become louder, it will frequently pay to search for the trouble by tapping suspected parts with a small wooden dowel or an insulated probe. The defective part will be the one which, when rapped, has most effect on the noise.

Be careful to rap lightly. Remember that a severe jar will be carried to the defective part through the chassis even if you are probing some distance away.

Shields should be shaken or twisted gently to see if they are making good electrical contact with the chassis. This test may also disclose a short between the shield and the device within it. If you wish, pull on the leads going to a suspected joint, or push the joint with a wooden stick—but going over all suspected joints with a hot soldering iron is perhaps the quickest procedure. Continue to operate the receiver for some time after thumping on parts or resoldering, as you may have only temporarily cleared up the trouble.

**Noise in Signal Circuits.** Occasionally the noise-producing defect

may be in a signal circuit—for example, a trimmer used to tune an i.f. transformer secondary may be partly shorted or may short at intervals. There is no current flow through such a circuit when signals are not being received, except that caused by between-station atmospheric and residual noises, so the defect usually will be heard only with a station tuned in. Thus, when you remove the ground and antenna leads, the noise may disappear. This may lead you to blame the antenna rather than a set defect.

A test receiver will keep you from making this error, since it will show that the set, rather than the antenna, is defective. If you have no test set but suspect the receiver, you might try another aerial. Better yet, feed the full *unmodulated* output of a signal generator (tuned to the receiver dial setting) into the aerial and ground posts. If noise is then heard, an r.f. defect is indicated; if modulation of a signal generator is necessary before the noise appears, the trouble may be in the a.f. section, or the sounds from the loudspeaker are vibrating some mechanically poor joint, which may be anywhere in the radio. You can then localize the noisy stage either by using a signal tracer or by moving the signal generator along from stage to stage.

# Intermittent Reception

A radio may be described as intermittent if it operates normally *part* of the time, but stops, cuts down in volume, squeals, gets noisy, distorts, or hums *some* of the time.

The intermittent receiver has a bad name among servicemen, for, at one time or another, most servicemen have been stumped by some stubborn case which refused to yield to their best efforts, or which consumed hours of time for which they couldn't collect.

Naturally there is always a definite cause for any intermittent action, but the difficulty lies in the fact that while the receiver is playing normally the defect does not exist, and no amount of testing will disclose it. Only when the receiver is in an intermittent condition is the defect present. At this time tests can lead to the defective part, but disturbing the receiver in any way, even by attempting to take voltage measurements,

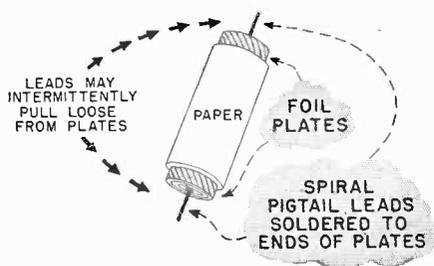


FIG. 9. Paper condensers are a common source of intermittent reception.

may cause it to snap back and play properly. You may then have to wait hours — even days — for the trouble to show up again.

Thanks to the modern methods of servicing, much of the uncertainty of dealing with intermittents has been eliminated. The most stubborn case of intermittency can be licked by a combination of searching for surface defects, effect-to-cause reasoning, and

section and stage isolation. Modern equipment even allows the serviceman to do other work while waiting for the intermittent action to make its appearance.

## INTERMITTENT DEFECTS

You know already the ways in which parts may become defective—how paper condensers may open or short, how electrolytics may open, leak, or develop a high power factor, how interelectrode leakage may take place in tubes, etc. Now we will see how some of these defects can occur intermittently.

**Paper Condensers.** The internal construction of a paper condenser is shown in Fig. 9. The leads of the condenser are ordinary bare wires whose ends have been curled into a loop. The loop is soldered to the foil if tinfoil is used. With aluminum foil, the loop is filled with solder, making a solder disc, and then pressed into the soft face of the exposed foil to give an electrical contact. The foil may be crimped around the solder disc. Both ends of the condenser are then dipped in a wax which, on hardening, holds the discs in contact with the foil plates. When we say that a condenser has opened, we mean that one of the discs has pulled away from its foil plate. The effect is the same as if the condenser lead were cut or a resistance were inserted between the disc and the condenser plate.

Now, suppose one of the flat discs is barely in contact with the foil, instead of being completely pulled away from it. Slight jars or electrical surges may make the connection open and close. For example, the disc may be in contact with the foil at a single point, and a signal surge may cause

enough current to flow to burn out the connection. Another voltage surge may start an arc or may cause the disc to again come in contact with the foil and the cycle of troublesome events may be repeated.

Another possibility is that the condenser may be open at the disc-foil connection, and a signal surge may

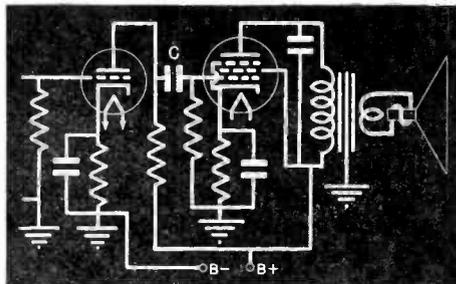


FIG. 10. A typical resistance-coupled audio stage.

increase the voltage between the foil and the disc enough to allow an arc to form between these two points. This arc will be self-sustaining and will complete the connection, thus restoring the condenser to operation. If the arc fails for any reason, the condenser will again be open.

► Notice—either of the conditions just described would cause noise, rather than intermittent reception, if it occurred frequently enough. For example, if one of the connections to coupling condenser *C* in Fig. 10 were to open, then close again in a few seconds, the signal level will jump up and down, and you would diagnose the condition as intermittent reception. But if exactly the same effect occurred much more frequently—say several hundred times a second—the set would be noisy. Similarly, other rapid open-and-closing actions will produce noise, while a slower tempo will cause another form of intermittent reception. For instance, if the screen grid by-pass opens and closes

at relatively long intervals of time, intermittent oscillation results; if the same action occurs rapidly, noise is produced.

Thus, you are not entering brand-new territory when you search for the sources of intermittent reception. The intermittent defect that produces intermittent noise, hum, oscillation, distortion, or decreases in volume would produce these same conditions continuously if the defect were continuous. You have already learned which defects may cause *continuous* noise, hum, etc.; now, all you need learn is which of these defects can occur *intermittently* and you will be prepared to solve the most puzzling cases of intermittent reception. Let's see, briefly, what these defects may be.

**Electrolytic Condensers.** Dry electrolytics normally do not open or short intermittently—in them, such defects are permanent. However, some dry electrolytics are anchored



Leakage through the case to the mounting bracket may cause intermittent hum.

to the chassis by a metal clamp around the condenser housing, and leakage sometimes develops between the condenser and the chassis through the paper housing. The trouble may be intermittent, disappearing when the paper dries out after several hours of operation, and reappearing when the receiver has not operated for some time or when the air is particularly humid.

► Wet electrolytics sometimes have intermittent defects. A conductive

sediment may form and settle to the bottom of the container, causing leakage. This deposit may dissolve, thus temporarily eliminating the leakage path, after receiver operation has heated the electrolyte. Suspect this condition if you see a white deposit around the vent of a wet electrolytic.

**Other Condensers.** Tuning condensers may intermittently short because of dust or metal particles between the plates. Midget-sized condensers may short due to bent plates, as even a small vibration can cause the plates to touch due to the close spacing. Watch for broken leads to the condenser gang, as it moves considerably during tuning. If the electrical connection is made to the stators via the bolts securing them, corrosion on the bolt threads may cause a varying resistance in the connection. Dirt or improper spring tension in wiping contacts may cause a varying resistance between the rotors and the condenser frame.

Fixed mica condensers do not ordinarily have intermittent defects. However, intermittent changes in capacity or short circuits may be caused in trimmer condensers by cracked mica or fatigue of the spring metal rotor plates.

**Volume Controls.** Volume controls frequently have carbon resistor strips which are pitted, or the carbon may have flaked off. Only a small portion of the slider may be in contact with the strip, so may be thrown out of contact by the slightest mechanical jar or by a current surge which burns out the section which the slider touches. Also, high-resistance joints between the slider contact and its terminal on the case may intermittently open the slider arm circuit. Troubles caused by burned or pitted strips are most apt to occur in controls through which d.c. flows, as in

the diode load circuit or in a grid circuit having grid currents.

**Tubes.** Tubes are one of the most prominent causes of intermittent reception. Quite often electrodes will expand with heat and touch other electrodes, thus causing intermittent shorts. Also, the filament may expand and break. This interrupts the current flow; the filament then cools, the broken ends come together, and current flow resumes. This is sure to cause intermittent reception.

Gas in a tube will often cause erratic operation, as will a faulty cathode. Sometimes particularly active emission spots appear on a cathode; as these lose their excess emission ability, the gain in the tube will vary.

**Resistors.** Wire-wound resistors may have intermittent opens at their terminal connections — particularly metal-clad “candohm” resistors. Heavy current through molded carbon resistors may cause uneven resistance distribution, resulting in varying resistance values and erratic current flow. This condition may not occur until the receiver has been in operation for some time and the resistor has become thoroughly heated. The carbon rod in a resistor of this type may sometimes break, causing an intermittent open.

**Coils.** Electrolysis in coil windings that carry d.c.—particularly the primary windings of i.f. transformers, a.f. transformers and oscillator coils—may cause intermittent opens. Frequently the form on which an r.f. coil is wound will expand with heat, snap a lead from the coil to its terminal lug, and so produce an intermittent contact.

**Connections.** Connections made to the chassis through rivets frequently work loose or corrode and so cause intermittent contacts. A poor contact may also develop between the chassis

and the can of a grounded electrolytic. On the other hand, if the condenser is insulated from the chassis by a fiber washer, leakage may develop across the washer and allow the condenser case (condenser negative terminal) to short to the chassis intermittently.

**Vibration.** Vibration is very apt to cause intermittent operation, particularly in car and portable receivers. Many parts in these sets may be mounted on rubber feet to reduce mechanical shock; even so, leads to such parts may be snapped and the broken ends may make intermittent contact.

**Mice.** Mice will frequently nest in radio receivers. They may drag in paper scraps, chew the insulation on wires, eat the cone of the loudspeaker, or wet insulation or some vital part and cause almost any effect on the receiver.

**General.** We have given the more common intermittent defects. There are, of course, many more. There are also some defects which almost never contribute to intermittent difficulties. A by-pass condenser is more commonly open intermittently than it is intermittently shorted, yet this last

possibility should be kept in mind. The power transformer rarely is intermittently defective, nor are properly soldered connections. (However, servicemen frequently introduce intermittent troubles by poor soldering; examine all connections carefully if you find a wire loose in a radio after the set has been serviced by some one else.)

Now for some general rules: All intermittents respond to some stimulus—a voltage surge, a mechanical shock or a thermostatic expansion or contraction—which opens and closes a circuit.

Tapping the radio will show whether mechanical shocks cause the trouble. Troubles due to heat are usually very regular; the radio is on and off at rather definite time intervals, depending on the part or circuit containing the defect. A short time interval usually indicates a defective tube or part carrying a high current. If the time period between cut-off intervals is rather long, you usually have trouble in a circuit carrying small amounts of current in which less heat develops, or trouble in some large slow-heating part.

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## External Intermittent Troubles

Relatively few parts can cause intermittent distortion, intermittent hum, etc. However, literally hundreds of things might cause an intermittently dead receiver or one in which the volume cuts to a low value and then snaps back. For such complaints, a professional means of localization is definitely required. We will deal with this kind of intermittent reception first.

### HOW TO DETERMINE THE COMPLAINT

When a customer complains of intermittent reception, you should satisfy yourself that trouble actually exists, and that it occurs with sufficient regularity, before you accept the job. Here are some highly important questions you should ask:

1. *Does the receiver fade only on distant stations or on all stations?*

2. *Does the fading occur only at night, only in the daytime or both?*

3. *How long must the receiver be on before the fading occurs?*

4. *Does the fading occur at regular or irregular intervals, and about how long a time elapses between the cut-on-cut-off intervals?*

► If the fading occurs on all stations, you can assume that something is wrong with the radio or the installation. Should the customer say that locals come in all right and that fading is on distant stations, the action is normal. It is caused by the sky wave (on which distant reception is obtained) fading in and out as the height of the Heaviside layer shifts. Complaints on this score are more prevalent during the fall and winter months when long-distance reception on the broadcast band improves in the northern hemisphere. Then, more distant stations can be heard during favorable periods, but they are subject to more fading.

You won't have many complaints of this kind with customers who have had their receivers a year or longer. However, many people buy new radios around Christmas. They are unfamiliar with the characteristics of the set and may happen to encounter a favorable night or two before the fading becomes noticeable. At once they become alarmed.

When the normal summertime drop-off in reception of distant stations occurs, you may again encounter some customer complaints which are caused entirely by normal conditions rather than by the receiver.

► If the fading takes place only at night, it is probably caused by fading of the sky wave, unless nearby locals fade. However, stations normally considered to be locals may fade at night if the transmitting antenna is more than ten miles away and if the sta-

tion operates near the high-frequency end of the broadcast band.

Fading in the daytime generally indicates a receiver defect, for then reception is by ground waves which are constant in intensity. Bear in mind, however, that if the receiver fades and the volume remains weak for definite periods of the day or night, line voltage fluctuations rather than a receiver defect may be the cause.

► Be careful to learn from the customer how long the receiver is actually operated. You may find that he rarely turns the set on for more than a few minutes in the daytime, and so may not notice that the fading does occur in the daytime as well as in the night, or vice versa.

**Line Voltage Fluctuations.** If you suspect line voltage variations (which are most apt to occur in industrial districts), attach a voltmeter to the power outlet and watch it for a short time. Also watch lights for blinking or varying brilliancy. If line voltage fluctuations occur at the same time as the fading, you have tracked down the source of trouble.

If the line voltage variations appear to be the source of trouble, you might take the matter up with the local power company as they will frequently cooperate by correcting their systems.

Line voltage regulators are available which will sometimes prove helpful. Essentially, these are resistances which change radically in value with changes in current flow. One of them in series with the receiver power cord will usually correct the rises in line voltage, but it cannot correct for large voltage drops.

**How Often?** Since some intermittent conditions will not start for a half hour or an hour after the receiver is turned on, to save time you should arrange to have such a set operated

long enough for the intermittent condition to occur before you arrive.

► Should the customer tell you that the receiver cuts off only once or twice a day or even more infrequently, and comes back to normal almost immediately, the trouble has not progressed far enough to repair. You should tell the customer to continue to use his receiver, calling you when the trouble becomes more frequent or when the set stops entirely. Explain that this will result in a less costly job and that the chances of anything else in the receiver being damaged are remote. Point out that it is a difficult, time-consuming process to repair the set in its present condition, so it is to his best interest to postpone repairs for a time.

Now, let's go through the servicing procedure you should follow if the receiver fades with sufficient regularity for you to accept the job.

### INSTALLATION DEFECTS

Your first step, of course, is to turn the receiver on and confirm the complaint. When the intermittent condition shows up, or while you are waiting for it to occur, you should make sure the defect is not in the installation. The installation can make the receiver go dead or fade intermittently, or produce intermittent noise or intermittent modulation hum. On the other hand, it cannot normally cause intermittent distortion, steady hum, or oscillation.

A poor antenna or ground is a common cause of intermittent reception. For example, if a set playing at a normal level suddenly becomes quite loud when some appliance or light is turned on, a poor ground on the receiver is to blame. The set is depending on the power line for a ground, and the chassis happens to be coupled to the ungrounded side of

the line. Connecting lights or other devices across the power line reduces the impedance of the ground path and so changes the level of the signal fed to the input of the receiver. On the other hand, should the signal level drop when an appliance or light is turned on, the set is depending on stray coupling to the ungrounded side of the line for an antenna. The light reduces the impedance to ground, thus reducing the signal level.

A.C. sets using a power transformer are more subject to this form of trouble than a.c.-d.c. sets, as a.c.-d.c. sets are connected directly to the line while a.c. sets are isolated from it by a transformer, so stray capacity or inductance provides the coupling. Often a cure may be effected by connecting .01-mfd., 600-volt condensers from *each side* of the power transformer primary to the chassis. However, a better antenna and a good ground are the best cures.

In these cases, notice that turning a switch on or off causes a change in volume, *but the volume then remains steady until a switch is again operated*. Should turning on or off switches cause an intermittent change, or should snapping a switch on and off clear up the trouble, the receiver is intermittent in such a way that it is subject to electrical shocks.

► Checking the installation further, inspect the antenna-ground system for mechanical failures which could cause either opens or shorts. Check it just as suggested earlier in this lesson for noise. Check the flat top to see if it touches anything or if it can sway in the wind and so short to some object. Shake the lead-in vigorously to see if this action causes the intermittent or any noise to appear. Frayed insulation on the lead-in will often permit a short to occur when wind moves the lead-in about.

Check the connections to the lightning arrester and lead-in strip. See if the latter can short on the window weather-stripping. Trace the lead-in right up to its point of connection to the receiver; at the set, look for loose strands of lead-in wire shorting to the chassis. Investigate all connections covered by tape to make sure none are bad. Any one of these little things may cause intermittent reception, and no amount of work on the chassis at your shop will locate the trouble.

The surest way to check the antenna-ground system is to connect your own test receiver to the system in place of the customer's radio. If the customer's complaint shows up in the test receiver while you make your inspection, you know that the installation rather than the customer's radio is at fault.

► If a shielded antenna lead is used, find out if rain always accompanies the trouble. If so, rain water may be

getting inside the shield and shorting the antenna wire lead.\* Since poor ground connections frequently cause trouble, check the ground clamp carefully—make sure a clamp is used if the ground connection is made to a pipe. A cold water pipe makes the best ground; make the connection near the point where the pipe enters the home if possible. Avoid gas pipes for two reasons—it is against fire regulations to use one as a ground, and the sealing compound in the joints has considerable resistance, thus reducing the effectiveness of the ground system. The hot water heating system does not always make a good ground either. ► Bad connections, loose appliances and loose wiring at outlet plates cause intermittents. Check for such conditions by connecting a low-wattage bulb to the radio outlet in place of the receiver and watching for flickering.

\* This trouble frequently occurs in auto radio installations.

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## Localizing Intermittent Troubles

Let us presume we still have a case where the set is intermittently dead or where the volume cuts to a low value and then snaps back. If nothing is wrong with the installation, you must go to work on the receiver. However, before removing the chassis and speaker from the cabinet, make a check for surface defects. Look in the back of the cabinet. Make sure that the tubes, shields and any chassis or speaker plugs used are firmly in place, and that the top cap connectors clasp the top cap studs on the tubes tightly. If the lead to the top cap of a shielded tube comes from the chassis, make sure the shield has not cut the lead insulation and created a short. If the lead comes through the

chassis inside the tube shield, see that it remains inside the shield; otherwise, keep it outside. Gently shake any cables or wires in the cabinet; any intermittent effect produced shows a loose connection or broken lead.

Test the tubes. Tap each tube as it is checked and watch for intermittent flashes of the short indicator or intermittent meter readings. Intermittent troubles won't always show up in a tube tester, so remember that this test does not eliminate the tubes as suspects.

► If you find no surface defects, *take the set to your shop*. Curing intermittent reception calls for tests which may appear quite aimless to the receiver owner, so it is psychologically

better to take the set away. Further, you need room to work comfortably, you should test the receiver carefully by operating for some time after the repair and, often, you'll need shop equipment to localize the trouble.

► Once you get the receiver on the workbench, set it up for normal operation. If the radio has a tuning indicator, see whether it is affected when intermittent reception develops. Should the tuning indicator change when the volume drops or the set goes dead, you can be fairly sure that the trouble is between the antenna terminal and the second detector (although occasionally it may be in the power supply). On the other hand, if the tuning indicator does not vary, the audio section of the receiver is faulty.

Next, notice whether the residual or background noise level increases when fading occurs. An increase in the residual noise level usually means a defect in the oscillator, the r.f. stage or the input circuit, which reduces the input signal so the first detector noise level is more prominent.

### BRUTE-FORCE SERVICING

Intermittent reception is one of the few troubles where the "brute-force" method pays dividends. Remember, it may take some time for the set to act up; anything you can do which will produce defective operation and localize the trouble at the same time is definitely worth while. However, don't let your methods fall into a "try everything," aimless probing and testing. Limit yourself to no more than five minutes of probing.

Now, let's see how you can use "brute-force" and effect-to-cause reasoning at the same time.

You have already learned the general causes of intermittent action. The most common is the joint which opens

and closes under mechanical or electrical stress, and here's the point on which this first five minutes of testing depends: *If you can disturb the bad joint so that you can make the set cut off and on at will, your problem is solved.*

How can you disturb defective joints? Easily—just pull on leads, wiggle parts, thump tubes and resistors, or apply mechanical pressure to joints. Let's see how this should be done.

**Checking Controls.** First, check the various controls. The volume control can be checked by rotating it over its full range. If noise is produced, the control is badly worn and is a possible cause of the intermittent condition.

Intermittency caused by poor contacts in wave-band, manual push-button or tone-control switches can be located by wiggling the switch knob or button back and forth (but not moving it enough to change from one position to another). The switch is defective if you can make the intermittency appear and disappear at will. Cleaning the contacts and bending them to make better contact will often clear up this trouble. Sometimes you must install a new switch.

Noise when the tuning condenser gang is rotated indicates shorting particles between the plates; perhaps dust or dirt is lodged there or plating has peeled off the plates. Poor wiping contacts may also be to blame. Tighten them by bending the springs. ► While the set is right-side up, check the tube socket contacts by wiggling each tube to make it move around a slight amount in its socket. If a tube is too hot to handle, hold it with a heavy cloth or a kitchen pot-holder.

Tubes may also have internal shorts and intermittent contacts at the points where the electrodes are welded to

their leads. These may be found by snapping the tube vigorously with your finger. When any doubt exists, try another tube.

Of course, you must be careful not to jar surrounding parts when wiggling or thumping tubes. A mechanical shock may travel through the chassis to another part, cause the intermittent to act up, and lead you to believe the tube is faulty. If a new tube acts just like the old one, the trouble is probably somewhere else.

► Now let's check underneath the radio. You will need a pair of long-nose pliers and some sort of an insulated probing tool. An ordinary test lead can be used in a pinch. Some servicemen use a pencil; however, since pencil lead is conductive and so may permit you to be shocked, we suggest you make up a tool similar to the one shown in Fig. 11 instead.

This can be a hard rubber or bakelite rod or a wooden dowel, from  $\frac{1}{4}$  to  $\frac{3}{8}$  inch in diameter and about 8 inches long. Cut one end off square and file the other end to a point resembling a screwdriver blade. Cut a notch in the square end. If a wooden dowel is used, it should be varnished or shellacked (not painted) so it will not absorb moisture.

With a tool of this kind, you can pull and push on parts at will without any danger of shocks.

**Avoiding Shocks.** While we are talking about shocks, let's take a moment to see how you can avoid getting them when you work on a live receiver.

To get a shock, you must touch *two* points which are at different potentials. Therefore, use only one hand, if possible, when you probe in a live receiver; this will make touching two points less likely. Further, don't lean on the radio, and make sure it is firmly supported. You will uncon-

sciously grab for a receiver that is falling over and may get "bit."

► Sometimes, especially when working on an a.c.-d.c. receiver, you will get a shock when you touch the chassis. This may happen if the receiver power plug is inserted in such a way as to connect the chassis to the ungrounded side of the power line (there is often a direct connection between the chassis and one power cord lead). Then, 110 volts a.c. will exist between the chassis and earth, and you'll get a shock if you touch the chassis while you're grounded. If you reverse the power plug at the wall outlet, the chassis will be at earth potential and no shock will result from touching it. However, it's best not to be grounded, since you can't tell whether the chassis is "hot" just by looking at it.

Also, in some a.c. receivers, a condenser is connected from one side of the power transformer primary to the chassis. If the power plug is inserted into its socket in such a way as to connect the chassis to the ungrounded side of the line through the condenser, you'll get a small shock if you are grounded and touch the chassis. If the chassis is connected to an earth ground with a ground wire, you can touch it without fear of being shocked. (There may be a spark when you attach the ground wire to the chassis, but it will not harm you.)

► Your best single precaution against shocks is to insulate yourself from the earth. Don't stand on a concrete floor (an excellent ground) unless you are wearing dry rubber-soled shoes. The best insulation is dry wood—so sit on a wooden stool or, if you must work in a place with a cement floor, put a wooden flooring in around your workbench. Don't just lay boards down on the concrete—they will absorb moisture and soon become con-

ductive. Space the floor up from the concrete by strips of wood.

► You can insulate your tools with rubber tape, or buy properly insulated electrician's tools. However, tape will eventually rot and get sticky, while factory-insulated tools may make you careless — which means you'll be almost sure to get a heavy shock some day.

► Be careful how you touch the case of a metal-clad electrolytic condenser mounted on top of a chassis. Remember that the can is the negative terminal of the electrolytic condenser and, if the can is insulated from the chassis, 100 volts or more may exist between

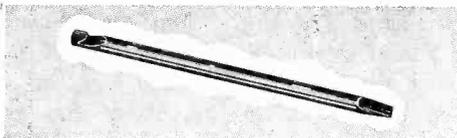


FIG. 11. This home-made tool is helpful when pulling on wires and moving parts.

the can and chassis. If you see a fiber washer between the base of the condenser and the chassis, don't touch the condenser case and chassis at the same time. If the condenser case is in electrical contact with the chassis, both can be touched at the same time without danger of getting a shock.

The important thing to remember is that to get shocked you must make your body part of a *complete* electrical circuit. Remember this always and you need never worry about getting shocked.

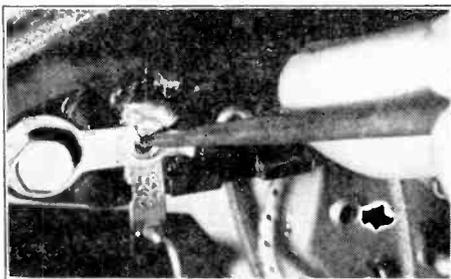
**Under-Chassis Defects.** If you have localized the trouble to the audio or the r.f. end of the receiver by watching a tuning meter, tune the set to a station and gently wiggle the paper by-pass condensers and the coupling condensers in the suspected section with a pair of long-nose pliers. If the intermittency occurs when you do this to one of the condensers, this

condenser is at fault and should be replaced. Be careful not to short terminals together with the long-nose pliers.

Rather than waste a lot of time trying to identify the condensers when they are mounted on terminal strips, wiggle all of them. Be sure you wiggle them *gently*. If you are too vigorous, you are liable to pull loose a good condenser.

If the condensers are mounted so that you cannot grasp them easily with the pliers, use the insulated probing tool to move them.

► One manufacturer encloses his condensers in black bakelite cases which are bolted to the chassis. The condenser leads are brought up through eyelets and soldered to mounting lugs. Check the connections to the plates of such a condenser by probing through the eyelet with a sharp probe and gently lifting up on the wire connected to the lug. (The hooked end of a small crochet needle is ideal for this purpose.) If movement of this wire cuts out the receiver, you have found the defective part. Remember,



Probing in the eyelet of a bakelite-enclosed condenser to see if disturbing the lead will cause the intermittent reception condition to appear.

be gentle—internal condenser connections are so delicate that undue stress may ruin a good condenser.

► Check connections by pulling on leads. If the receiver cuts on and off when you do this, applying a hot soldering iron to the joint in question

should clear up the trouble. Make certain that excess solder on the joint is not shorting to the chassis or to an adjacent connection.

► Ordinary resistors seldom cause trouble, but should be tapped lightly with your probing tool to see if intermittent action can be caused. The metal-clad candohm resistor, which is usually bolted to the chassis wall, frequently develops poor connections at its taps and ends. Check for this source of trouble by wiggling all terminals with a pair of pliers.

Tap exposed coil forms with the probing tool and move their lugs back and forth a *slight* amount with long-nose pliers. This will check for broken wires near the terminal lugs.

► Sometimes jarring shielded parts mounted on the chassis will show a partial open or short in the enclosed part. This is particularly true of i.f. transformers and, to a lesser extent, of a.f. transformers and chokes.

► The leads on loudspeaker voice coils sometimes become loose at their terminals, perhaps only during maximum movements of the cone. With the set turned on but with no signals tuned in, check for this by moving the cone with your hands, listening for a characteristic "pop" which will be heard if tension on the voice coil leads opens the circuit.

► It should not take more than five minutes to make these tests, and they will solve about 75% of the intermittent cases you encounter. However, effect-to-cause reasoning and signal-tracing equipment—and often plenty of patience!—are necessary to solve the remaining 25% of really obscure cases.

## SECTION AND STAGE ISOLATION

We have already mentioned how the receiver tuning indicator may localize the defective section. If,

when fading occurs, the indicator shows a loss in the strength of the signal delivered to the second detector (or a.v.c. diode), the trouble is either between this point and the antenna or in the power supply. If the tuning indicator is unaffected, the trouble must be between the second detector (or a.v.c. stage) and the loudspeaker cone. These observations at once tell which section is at fault.

If the receiver does not have a tuning indicator, you can connect a high-sensitivity d.c. voltmeter or a d.c. type vacuum tube voltmeter across the a.v.c. circuit and get the same indication.

**Signal Tracing.** Should you have a signal tracer, however, the problem is vastly simplified. The average signal tracer contains a section, known as an r.f.-i.f. channel, which is capable of tuning over the i.f. and the broadcast band. It may also contain a less sensitive section, called an oscillator channel, which will cover the oscillator and broadcast range of the receiver, also an a.f. channel, and perhaps a d.c. vacuum tube voltmeter. A typical instrument of this type is shown in Fig. 12. The advantage of this type of signal tracer is the fact that the signal progress can be observed at a number of points simultaneously. Let us see how you can use an outfit of this sort to check the typical a.c.-d.c. receiver shown in Fig. 13.

► First, let's make connections so as to use all the channels at one time. We will need the *r.f.-i.f. channel* for the i.f. stages, so let's connect the *oscillator channel* to point 2 to check the constancy of the r.f. signal delivered by the 12SK7 tube.

The *r.f.-i.f. channel* can be connected to point 4 to check the mixer-oscillator.

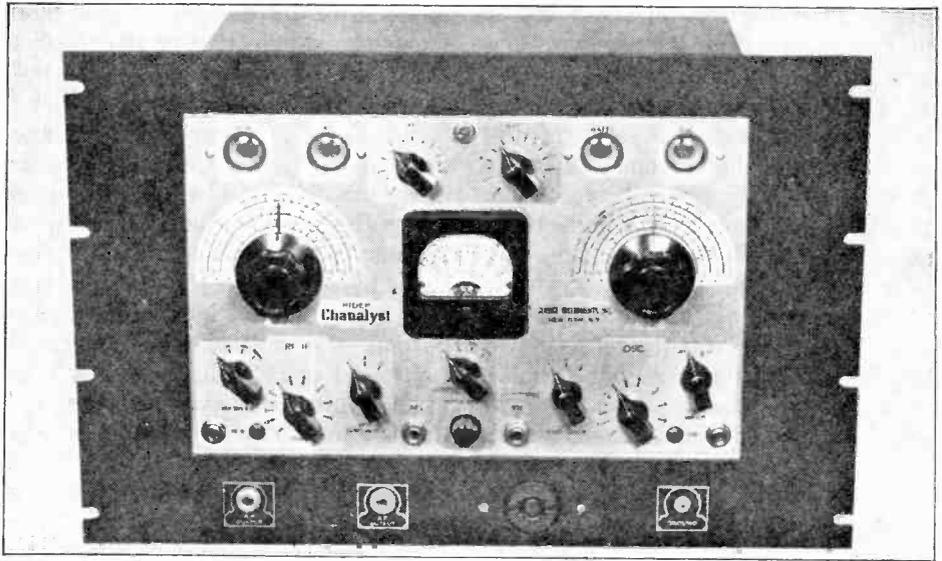


FIG. 12. A multi-channel signal tracer. Tuning eyec tubes act as indicators, except for the d.c. type v.t.v.m., which uses the meter. This instrument has a tunable r.f.-i.f. channel; a tunable r.f. channel (called the "oscillator" channel); an audio channel; the d.c. type v.t.v.m., and a wattage indicator. By using all the channels (except the wattage indicator) at one time, a radio can be broken into four sections, making it possible to localize the defect quickly.

The *d.c. type v.t.v.m.* can be connected to test either the a.v.c. or the power supply. The a.v.c. can be checked at point 12 (at the volume control) if the meter does not by-pass the audio signal too much; otherwise, point 13 is preferable. The power supply can be checked at point 14.

The *a.f. channel* can be connected to point 9 (the 35L6 input) to check the progress of the signal from the volume control to the input of the 35L6 power tube. (The loudspeaker output would indicate conditions from point 9 onward, as any change not indicated by the channel at 9 must be due to an output stage or speaker defect.)

► The test probes usually end in peewee clips which may be clamped onto a terminal or tube socket lug. However, the probes generally fall off if you try to turn the chassis right-side up. If it is necessary to turn the

chassis over, you'll save time by soldering wire leads to the points where you are checking. You can then clip the signal tracer probes to the ends of these wires and turn the chassis over without disturbing the connections.

► You must make a ground return connection to the receiver. In standard a.c. sets, the ground lead of your signal tracer can be clipped to the set chassis. However, in a.c.-d.c. receivers (like that in Fig. 13) where there is no electrical connection to the chassis, you must find the B-circuit, particularly to measure a.v.c. and power supply voltages. Point 15, or any other point on the B— bus or lead, may be used for this connection. Many servicemen use the set side of the ON-OFF switch as a convenient connecting point.

Here's the general procedure to follow to isolate the defective section or

stage. First, connect the signal tracer channels properly to the receiver, then turn on the set and the test instrument. Tune in a program on the receiver and adjust the gains of the channels until their visual indicators show some easily remembered value. (If they are magic eye indicators, adjust them so the eyes are just closed.) Then let the set play while you take care of your other bench work. Sooner or later the receiver will become intermittent, and one or more of the signal tracer indicators will show a changed signal level. Let's take a few practical examples to see how this will point out which section or stage is defective.

✓ **Example 1.** Let's start with connections at points 2, 4, 9 and 14. Suppose that the signal level remains normal at points 2 and 4, and that the power supply at 14 remains constant, while the signal level at point 9 decreases. This is definite proof that the cause of the trouble lies between points 12 and 9.

How can you be sure that the trouble is between points 12 and 9 and not between points 4 and 12? The steady readings at points 2 and 4 are proof. You see, a defect between points 4 and 12 would drop the signal input to the second detector. The a.v.c. voltage would then drop, permitting an increase in the gain of the r.f. and converter tubes, and the channels connected at points 2 and 4 would show a larger signal. Since, in our example, the signal level remains constant at points 2 and 4, no defect exists between 4 and 12.

To localize further, remove the a.f. probe from point 9 and touch it to point 8. If the signal indicated here is much greater than at point 9,  $C_{21}$  is open.

If the signal is still weak, move the probe to the junction of  $C_{20}$  and  $R_{10}$ .

A weak or intermittent signal here shows  $C_{20}$  is defective or the slider is not making a good contact on the volume control  $R_9$ . On the other hand, if the signal is as strong or stronger than that at point 8, the 12SQ7 tube is not amplifying. In the latter case, either the tube or plate load resistor  $R_{11}$  may be defective. Check the resistor by measuring the plate voltage. If it is normal or slightly higher than normal, the resistor is probably all right, and you should try a new tube. If the intermittency continues with the new tube, substitute a new resistor for  $R_{11}$ .

It is entirely possible that the receiver may snap back to full volume while you are changing the signal tracer connection or while some other test is being made. If this happens, make the new signal tracer connection to the point where the probe was at the time the volume snapped back, readjust the channel gain for the new point, and go about your other business until the intermittent appears again, then continue to move back, step by step. In this way you can gradually narrow your search until part substitution becomes feasible.

**Example 2.** Suppose the signal level indicated at points 2 and 4 increases, while that at point 9 decreases, and the supply voltage (point 14) remains constant. As you learned in the first example, this indicates trouble between terminals 4 and 12. Shift the d.c. type v.t.v.m. from point 14 to either 12 or 13; if you find the a.v.c. voltage decreases when the trouble occurs, you have confirmed this diagnosis.

► Now remove the r.f.-i.f. probe from terminal 4 and move it to point 5, the control grid of the i.f. tube. The signal level should be at least half the value it was at point 4. If it is considerably lower, either transformer  $T_2$

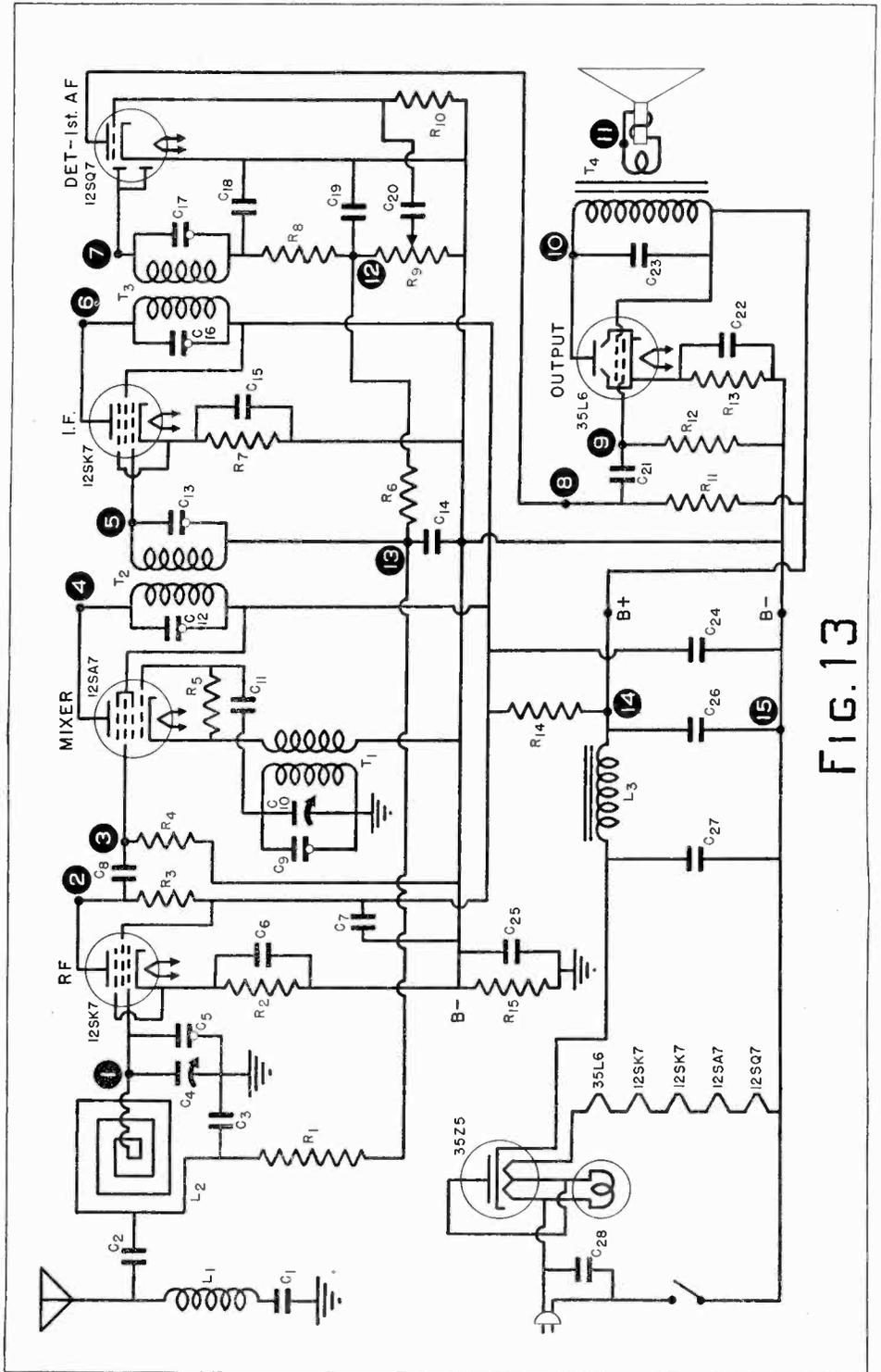


FIG. 13

or condenser  $C_{14}$  is defective. Hold the probe across  $C_{14}$ . You should *not* get a signal here if the condenser is working.

If you prefer, shunt  $C_{14}$  with a test condenser. The signal will jump up to normal at point 5 if  $C_{14}$  is defective. However, remember that charging this condenser may provide an electrical shock, restoring operation anyway. Leave it temporarily connected and see if the trouble recurs before making a permanent repair.

► Should the signal at the i.f. grid be normal, move to the plate of the i.f. tube, point 6. If there is no gain in signal, the i.f. tube is defective, the trimmer  $C_{16}$  is shorted, the primary of  $T_3$  is defective, or possibly condenser  $C_{15}$  has opened. Try another tube. If this does not clear up the trouble, move your signal tracer probe to the cathode of the i.f. tube. You should *not* be able to find an i.f. signal across  $R_7$ ; if you do, condenser  $C_{15}$  is defective and should be replaced. Should this condenser be normal,  $C_{16}$  is the remaining suspect. Disconnect it and test it with an ohmmeter.

In an a.c. radio using a power transformer, the screen grid may be supplied through a series resistor and will be by-passed to ground or chassis by a condenser. A defect in either of these parts could easily cause a loss of gain in the affected stage. Of course, there are no such parts in this a.c.-d.c. set.

Should you find the signal normal at the plate of the i.f. tube, move to the second detector diode plate, point 7. A large drop in signal (to less than one half) from the level found at point 6 indicates something wrong with the i.f. transformer  $T_3$  or with condenser  $C_{18}$ .

You can check  $C_{18}$  by taking a reading directly across it with the signal tracer probe at the junction of

$R_8$  and  $C_{18}$ . You should get little r.f. signal at this point. If you find a large signal, replace  $C_{18}$ .

► A normal signal at 7 leaves the second detector and its circuit as the remaining sources of trouble, so the defective stage is localized.

**Example 3.** Now let's see what to do if the signal fades at points 4 and 9, the supply voltage at 14 remains constant (or if the v.t.v.m. is connected to 13, the a.v.c. voltage drops), but the signal remains constant (or increases) at 2. This shows trouble in the 12SA7 stage; the mixer stage may be at fault or the oscillator may have failed. Check the input to the mixer by moving the oscillator channel from 2 to 3. If the signal is considerably less than at 2, condenser  $C_8$  is open; if not, try another 12SA7 tube.

► If the set still does not perform, the oscillator circuit is at fault. Check it by placing the oscillator channel probe on the stator of the oscillator tuning condenser and tuning the channel to a frequency equal to the receiver dial setting plus the amount of the i.f. The oscillator signal, if present, will be picked up and indicated by this channel.

If there is no oscillator signal, check the oscillator coil for continuity, go over all connections in the oscillator circuit with a hot soldering iron, check the tuning condenser and its trimmer, try another resistor in place of  $R_5$ , replace  $C_{11}$ , and finally, install a new oscillator coil if nothing else has remedied the trouble.

**Example 4.** Returning to our connections at 2, 4 and 9, suppose all the readings drop simultaneously—what is indicated? The answer may depend on whether the d.c. type v.t.v.m. is connected to the power supply (14) or to the a.v.c. network (13). If the d.c. meter is connected to 14 and its reading goes down simultaneously

with the others when the fade occurs, there is obviously a defect in the power supply (which, of course, affects every stage in the radio).

But if the meter is connected to the a.v.c. (13) and all readings drop, you can't tell whether the defect is in the radio input or in the power supply. In such a case, you must take a reading at 14 also. If this reading remains constant and the other three drop, the trouble is probably between the antenna and the plate of the r.f. tube (2). The next example will discuss this.

Of course, sometimes only a single power supply circuit is defective—for example, a separate screen grid supply or the plate supply to an r.f. or i.f. stage. This may have so little effect on the main power supply that the reading at 14 won't be appreciably lowered. Such defects will be run down, however, as you check the signal levels in the various stages.

**Example 5.** If the signal tracer channels at all points except 14 show a drop, thus indicating an input circuit defect, you should first move the probe at point 2 to point 1 (the grid of the 12SK7 r.f. tube). If the signal level does not change when the set snaps back, the trouble is in the r.f. tube or its voltage supply circuits. Try a new tube. If the trouble continues, replace  $C_6$ ,  $C_7$ ,  $R_3$  and  $R_2$ , one at a time. In sets which have the plate circuit decoupled by an R-C filter in the B supply, the filter may be defective. Check this by connecting the d.c. meter across the decoupling condenser; if a radical change in voltage occurs when the fade or cut-off takes place, the bypass condenser is leaky or shorted or the decoupling resistor is open.

If the signal level at the input of the r.f. tube rises when the receiver snaps back, the defect may be in the

r.f. tuning condenser, in its trimmer  $C_5$ , in  $C_3$ , or in the loop. If an outside antenna is used while the receiver is under test, then  $C_2$ ,  $L_1$  and  $C_1$  are possible suspects.

**Other Methods.** You can see from these examples that a signal tracer can be quite valuable in isolating the cause of intermittent reception. Exactly how valuable the instrument is depends on how complete it is. A tracer with all the channels mentioned in our examples is the most useful, but less complete instruments can often be very helpful.

For instance, types which have only a single r.f.-i.f. channel can be used to trace any defect between the antenna and second detector, by moving the probe along until you find the point where the signal level decreases. The types with audio channels can be used similarly to check the audio amplifier.

If you do not have a d.c. type vacuum tube voltmeter, you can use any d.c. voltmeter to measure supply voltages and any high-sensitivity meter to get a rough check of the a.v.c. voltage. Similarly, you can use a high-sensitivity d.c. voltmeter across resistor  $R_5$  to determine if the oscillator is functioning.

Thus, using a signal tracer is not the *only* way to localize the trouble. It is, however, one of the best ways.

### MAKING THE SET FADE

Sets are often exasperatingly slow to cut off when you have them on the bench. To save time and to avoid tying up your test equipment any longer than necessary, you're wise to speed up the process as much as you can. Here are some practical ways to do so.

► You can make a thermal intermittent appear much more quickly by covering the chassis with a heavy

piece of canvas or with a large cardboard box to trap the natural chassis heat.

► Increasing the line voltage may make the intermittent occur, or may permanently break down the defective part (which you can then quickly locate by routine tests). The only other parts which may break down are those that are near the ends of their useful lives and would soon have to be replaced anyway.

You can vary the line voltage most readily by inserting a tapped or variable transformer between the line and the set. The schematic of a simple device you can make yourself with a toy train transformer is shown in Fig. 14. This arrangement will vary the applied voltage from about 85 volts to 130 volts. (Never apply more than 130 volts to an a.c. receiver.) The device will give you voltages higher than the line voltage with the double-pole, double-throw (d.p.d.t.) switch  $SW_1$  in one position, and voltages lower than the line voltage with the switch in its other position.

Lower-than-normal supply voltages are useful if you suspect the oscillator is cutting out. In three-way receivers using low-voltage filament type tubes, even a slight decrease in oscillator filament voltage will stop the oscillator. So, if you encounter a three-way set which fades on power line operation but not on battery operation, try reducing the line voltage to see if you can thus reproduce the observed intermittency.

Sometimes it may be necessary to operate the set for several hours with higher-than-normal voltage. This will not cause you any difficulty, since most toy transformers can supply as much as 5 amperes continuously without overheating.

► Many sets fade in and out when a light switch is thrown. Make sure

this is not caused by a poor ground system at the customer's home. If it is not, duplicate the on and off clicks at the shop by inserting a flasher button (the type used with Christmas tree lamps will do) in series with a 100-watt lamp across power line near the receiver plug. The thermostatic flasher will regularly cut the lamp off and on so surges will travel to the radio. When the set stops, cut off the flasher and start trouble-shooting on the chassis. If the set snaps back on, use the flasher again until the cut-off recurs.

**Testing Parts.** While substitution is sometimes the only way to learn whether or not a suspected part is

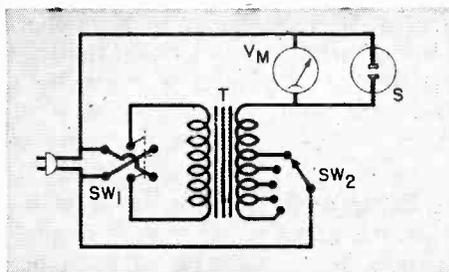


FIG. 14. How to connect a toy transformer to step up or step down the line voltage.

actually defective, often a very careful examination combined with mechanical wiggling will give definite proof.

An ohmmeter will sometimes show up an internal defect in coil windings by giving a varying meter reading.

You can check for tuning condenser short circuits by unsoldering the coil leads and connecting an ohmmeter between the stator and rotor. If the ohmmeter gives a reading or flicker when you rotate the condenser gang, a short exists. To remove it, you must clean between the plates, burn out metal peelings in the manner described in the noise section of this lesson, or bend the plates.

Trimmers can be disassembled and examined under a magnifying glass. This will often let you find cracked mica dielectrics.

## OTHER INTERMITTENT COMPLAINTS

Intermittent fading and cutting off are not the only intermittent troubles you will encounter as a serviceman. On occasion you will deal with receivers which intermittently squeal, hum, distort, and become noisy.

These troubles are easier to conquer than the intermittent fade or cut-off. A hundred and one things might cause a set to cut off or fade, while the other symptoms can be caused by only a limited number of defects. You can usually check these parts readily and isolate the defect in short order.

► For example, suppose a set exhibits intermittent distortion. Leaky coupling condensers are the first logical suspects. You can test for this (as explained elsewhere) while the distortion is present, or simply try new condensers. Gassy tubes are the next possibility—again, you can test or substitute. There are very few other distortion-producing defects which can appear intermittently. However, if you do happen to strike the unusual case, a signal tracer will quickly point out the defect.

► Intermittent oscillation may be caused by a faulty by-pass condenser or by corroded shield contacts.

► In the case of intermittent hum, check the electrolytic filter condensers and the plate and screen by-pass condensers for opens. If an electrolytic in a cardboard case is secured to the chassis by a metal strap, suspect leakage between the strap and condenser through the cardboard cover. (You can eliminate this by removing the strap and allowing the condenser to be self-supporting in the chassis.)

Always be suspicious of tubes—intermittent heater-to-cathode leakage, which produces hum, is fairly common.

► Intermittent noise is more difficult as it can be caused by many of the part defects listed earlier in this lesson. In fact, constant noise will seldom be encountered, but it usually lasts long enough to allow the source to be located. Tubes and transformers are most likely to be intermittently noisy. Jarring these parts will usually show up the offender.

## A CHECK LIST

Always remember that an intermittent, like any other radio defect, has a plain, everyday cause which you can find if you're patient and careful enough. When you get a tough job that seems to defy every rule of servicing, run over the following list of causes of intermittents to see if you have forgotten something. You'll be surprised how often this will solve your problem.

### *Antenna Circuit:*

Corroded joint between lead-in and antenna; poor contact at rivets in lead-in strip; poor contact between wire and clip on lead-in strip; faulty contact between ground clamp and ground; antenna lead shorted to chassis; antenna rubbing against conductive or semi-conductive material.

### *Oscillator Circuit:*

Tube checks okay but does not oscillate; tuning condenser plates peeling and partially shorted; electrical breakdown in padding condenser mica.

### *Power Supply:*

Faulty contacts within filter condenser; improper contact between slider and voltage divider; broken

turn on divider, not apparent before a complete separation occurs; leakage across insulation between two divider taps; high-resistance contact in ON-OFF switch, leakage between wires.

*Speaker:*

Poor connection at voice coil leads; scratch across voice coil winding (intermittently shorting the coil or turns on the coil); voice coil which opens when cone moves too far.

*Power Line:*

Arcing contact at fuse block or at wall plug; leak between transformer winding and core; fused outlet contacts (contacts rough and pitted because of arcing).

*Volume Control:*

Loose terminal wire at inside lug; dirt and wear under slider arm; loose wire in resistance element; increase in ohmic value.

*Tubes:*

Faulty weld at element support (affected by heat); interelement short; cold joint between element lead wire and prong; twisted glass envelope; shorted leads; emission coating drops

between elements after flaking off; poor socket contact to prong; heater-to-cathode leakage; gas in tube.

*By-Pass Condenser:*

Open; shorted; leak between sections.

*Resistors:*

Cracked carbon rod; loose terminal connection; contact with other set parts; in wire-wound type, partial shorts between turns; overheating from overload.

*Tuning Condensers:*

Rubbing or dirty plates; poor connection between rotor and chassis (faulty wiping contacts); poor connection to stator plates; cracked mica in trimmers.

*A.F. Transformers:*

Inter-winding leak; imperfect insulation between winding and core.

*R.F. Transformers:*

Cold soldered joint at lug; shorts between windings because of crossing of wire leads or deterioration of enamel insulation.

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THE N. R. I. COURSE PREPARES YOU TO BECOME A  
**RADIOTRICIAN & TELETRICIAN**  
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# Lesson Questions

Be sure to number your Answer Sheet 43RH-1.

Place your Student Number on every Answer Sheet.

*Send in your set of answers for this lesson immediately after you finish them, as instructed in the Study Schedule. This will give you the greatest possible benefit from our speedy personal grading service.*

1. Where is the source of noise, if the noise is still heard after removing the antenna and ground connections and installing a line noise filter?
2. Suppose the receiver is noisy when jarred, but the noise disappears when the volume control is turned to the minimum volume position. In which section of the set is the noise source located?
3. The noise stops when a tube is removed, but reappears while the plate-chassis voltage is being measured with a voltmeter. In which circuit of this stage is the noise source located?
4. Name *three* mechanical defects you would look for if noise is heard only as the station selector (tuning) knob is turned.
5. When following the stage-blocking method of pulling out tubes one at a time, you find the noise is decreased but not completely blocked. In which section of the receiver is the noise source located?
6. If the complaint is intermittent hum, which of the following intermittent defects would you suspect: *1, open coupling condenser; 2, open screen grid bleeder resistor; 3, open output filter condenser; 4, open plate supply resistor?*
7. What is the brute force technique (used on intermittent receivers)?
8. Suppose an audio signal tracer shows an intermittent change in volume when at the junction of  $C_{20}$  and  $R_{10}$  in Fig. 13, but a steady signal exists at point 12. Which two parts are the most logical suspects?
9. Suppose the loudspeaker volume varies, but a signal tracer connected to point 9 of Fig. 13 shows a normal steady signal. Where is the trouble?
10. Where in Fig. 13 would you connect a high ohms-per-volt meter to determine whether the intermittent trouble is in the a.f. section or in the r.f. section?



## HOW AND WHY

There is an exceptionally fine quotation I want to pass on to you. I do not recall the name of the author, but the truth in the quotation makes it unforgettable. Here it is:

*"The man who knows HOW will always have a job—the man who knows WHY will be his boss."*

Your N.R.I. training fits this thought perfectly. You are being taught HOW to service radio receivers, so you can be sure of plenty of work in this, your chosen field. Also, your fundamental training gives you the background of WHY radios function as they do, WHY they break down, and WHY the particular servicing methods we recommend lead directly to the source of trouble.

We not only want you to be a successful serviceman—we want you to have an assured future, with every opportunity to advance to the top. You will need only the touchstone of a little practical experience to weld your "HOW and WHY" training into a single, compact unit of knowledge which can lead you wherever you wish to go.

*J.E. Smith*

**INSTALLATION AND SERVICE  
OF AUTO RADIOS AND  
FARM RECEIVERS**

44RH-1



**NATIONAL RADIO INSTITUTE**

**WASHINGTON, D. C.**

**ESTABLISHED 1914**

# STUDY SCHEDULE NO. 44

For each study step, read the assigned pages first at your usual speed, then reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind. Study each other step in this same way.

1. Radio Operation in Automobiles - - - - - Pages 1-4

Auto sets must be designed particularly for operation under the conditions found in cars. They must have high sensitivity, be compact and sturdy, and must be carefully shielded and filtered to keep down the interference level. A discussion of the sources of interference is given, as this has an important bearing on the installation and repair of auto sets.

2. Installing Auto Sets - - - - - Pages 4-15

The various types of auto sets, aeriels and controls are described, then the proper installation and connection steps are described.

3. Eliminating Ignition Interference - - - - - Pages 16-25

All eliminating steps are covered, in the order of their importance. Bypass condensers, suppressor resistors, bonding and shielding are all used for interference reduction. A step-by-step localization procedure is given to assist in running down the noise.

4. Servicing Auto Radios - - - - - Pages 25-34

How to service auto sets in the car and on the workbench. Vibrators and vibrator circuit troubles, as well as auto set peculiarities, are covered here.

5. Farm Radio Receivers - - - - - Pages 34-36

A description of a typical vibrator-powered farm radio and service hints for this kind of set.

6. Answer the Lesson Questions and Mail your Answers to N. R. I.

7. Start Studying the Next Lesson.

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# INSTALLATION AND SERVICE OF AUTO RADIOS AND FARM RECEIVERS

## Radio Operation in Automobiles

AS FAR as their wiring diagrams show, the use of a vibrator power supply and special filters in the power leads is all that distinguishes the average automobile receiver from the average home radio. But actually, because of the unusual interference and sensitivity conditions it must meet, and because of the physical requirements imposed by use in an auto, the car receiver is a far different set. You must take these special conditions into consideration when you install or service an automobile radio—so let's see, briefly, what they are.

**Sensitivity Requirements.** The antenna of an automobile radio must, of course, be small and is quite close to the ground. As the antenna effectiveness varies directly with size and height, this means the amount of signal pickup is small, so the sensitivity of the set must be high.

Also, as one drives on the roads between cities, it is desirable to be able to pick up signals from favorite stations as long as possible, or to be able to pick up the stations in the next town as those in the last one fade out. (This is particularly important to truck drivers and salesmen who depend on the radio to relieve the monotony of driving.) Again, high sensitivity is required.

Hence, auto sets use high Q coils and high-gain circuits. Some even introduce a certain amount of regeneration to increase the sensitivity further. Most auto sets have an r.f. stage ahead of the first detector to raise the weak signal input to the point

where first detector noise is less of a problem.

► Further, as the car is driven around, the receiver may be at one moment in an area of good reception—the next instant in a section where signal strength is tremendously reduced by some shield (the steel framework of a building or a bridge, overhead trolley wires, etc.)—and, a few seconds later, back in a good reception area. The set must have sufficient reserve sensitivity and a fast-acting a.v.c. network (one with a short time constant) so it can minimize the effect of variations in signal level.

**Physical Requirements.** Besides having high sensitivity, an auto receiver must be compact, so it can be installed out of the way of the driver and car occupants, and must be rugged to withstand road shocks and bumps. Its controls must be easy to reach and use. And, since the set must operate from an already heavily-loaded car battery, it must draw as little current as possible.

**Interference.** An auto set must work in the middle of a "hot-bed" of the worst kind of radio interference, caused by the automobile ignition system. Special shielding and filter devices must be used to produce good reception without interfering greatly with the operation of the car or the radio.

► After the auto radio set is designed, there remain three problems: 1, its installation; 2, suppression of interference; and 3, radio servicing. The interference problem affects the other

two, as the set must be installed so as to pick up a minimum of interference, and you must recognize the differences between interference and radio troubles for proper servicing. Therefore, as an important first step in your study of auto receivers, let's learn just what can cause radio interference in cars.

### LOW-VOLTAGE CIRCUITS

The electrical parts of an automobile, including the starter motor, the ignition system, lights, electrical gauges, heater, etc., all operate from the car battery.

The battery is a 6-volt storage battery in all modern pleasure cars and in most trucks. This storage battery is rather heavily loaded, particularly by the starter motor. As the battery is essential for the car's operation, it must be kept charged, so a generator driven by the car engine is used for this purpose.

Fig. 1 shows what is called the "low-voltage circuit" of a car, including an ammeter  $M$  (mounted on the instrument panel) that keeps track of conditions in the circuit. The meter has zero at the center of its scale and can deflect either to the left or to the right, depending on the direction of current flow. One side of the scale is called the "discharge" range, the other the "charge" range. A meter reading in the discharge range indicates that the battery is supplying more current to the car's electrical components than it is receiving from the generator, while a reading in the charge range shows that the generator is supplying all the current required by the electrical circuits plus an extra amount that is being stored in the battery.

**The Cutout.** Before the car engine starts to run, the generator must be disconnected from the battery to prevent the battery from discharging

through the generator winding. The cut-out (see Fig. 1) does this. The cut-out is a relay which remains open until the voltage across its operating coil  $L_1$  (that is, the generator voltage) rises to a little more than 6 volts. When it does, the relay closes its contacts and connects the generator across the battery, making the connections through coil  $L_2$ . Since the generator voltage is higher than the battery voltage when this connection is made, current flows from the generator into the battery through coil  $L_2$ . If the speed of the engine slows down enough so that the generator voltage falls below that of the battery voltage, the battery will start discharging through the generator and the direction of the current flow through coil  $L_2$  will be reversed. This coil then forces the cut-out to open, and it will remain open until the voltage of the generator rises enough for coil  $L_1$  to close it again. Thus, the generator is connected to the battery only when the generator voltage is higher than the battery voltage.

Many cars also have voltage-regulating relays, which close only when the generator gets to the right voltage and will put a resistance in series with the circuit if the generator voltage exceeds a safe amount.

**Interference Sources.** The generator causes most of the radio interference produced by the car's low-voltage circuit, because of the arcing and sparking which is bound to occur between its brushes and its commutator. As dust and oil collect on the commutator, the sparking becomes worse.

The cut-out may cause a clicking sound as it operates, but this is not usually troublesome. The oil, gas and other indicating circuits are usually resistive in nature and rarely cause trouble unless poor contacts develop in their circuits. However, if

the heater has a motor-driven fan, the fan motor commutator and brushes may cause disturbances. Of course, a poor contact anywhere in the low-voltage circuit may cause arcing and so produce radio interference.

When the storage battery is well charged and in good condition, it acts like an extremely high-capacity condenser across this circuit. In other

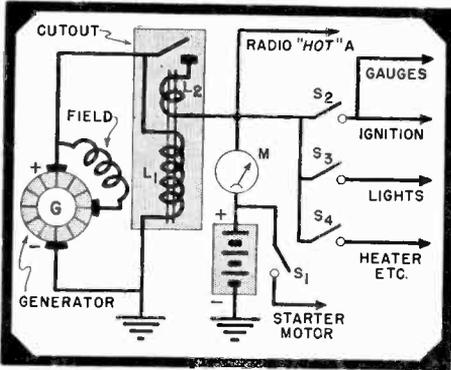


FIG. 1. The low-voltage wiring system of an automobile. The return circuit is through the car chassis, as shown by the ground symbols.

words, it tends to bypass noise ripples by holding the circuit voltage steadily at the battery voltage value. If a poor connection develops at the battery, or if the battery is run down or defective, this action is lost and the amount of interference will rise to rather high levels.

### THE IGNITION CIRCUIT

The amount of interference caused by the low-voltage circuit is relatively immaterial compared to that which comes from the ignition circuit. A typical ignition circuit is shown in Fig. 2.

As you probably know, the spark plugs which ignite the gasoline-air mixture in the cylinders of a gasoline engine must be fed rather high voltages to produce the hot, high-intensity sparks needed to cause proper com-

bustion. Voltages up to 5000 volts are common.

A cam-driven switch — usually called a “breaker”—and a voltage step-up transformer are used to obtain such high voltages from the 6-volt battery. The switch ( $S_2$  in Fig. 2) is opened very quickly and closed somewhat more slowly, by a cam driven mechanically by the car engine. When switch  $S_1$  (the car ignition switch) is closed and the engine is running, the high-speed opening of switch  $S_2$  causes abrupt changes in the current flowing in the primary circuit of transformer  $T$  and so induces a high voltage in the step-up secondary winding of the transformer.

Since the spark plugs in the various cylinders must be fired or ignited in a definite sequence to produce proper engine performance, the secondary of transformer  $T$  is connected to the

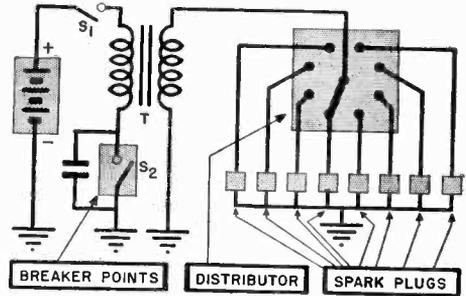


FIG. 2. The high-voltage ignition circuit.

spark plugs by a rotating switch, called a distributor, which is geared to the engine crankshaft. The distributor arm and the primary breaker are geared together so that each time the primary circuit is interrupted, inducing a voltage in the secondary, the distributor rotor arm is at the proper contact to deliver the high-voltage surge to one of the spark plugs.

► Thus, there are three arc sources in the ignition system which will produce noise surges whenever the engine

is running—the breaker contacts, the distributor contacts and, of course, the spark plugs themselves. (A condenser is connected across  $S_2$  to reduce the amount of arcing here, but even so the primary current surges are sharp, noise-producing pulses.) These surges are fed directly to the radio through its power supply lead, since both the radio and the ignition circuit are connected in parallel across the battery. In addition, and even worse, these sharp pulses and those produced in the secondary circuit cause radiation of noise energy from the ignition wiring itself.

In modern cars the low-voltage wiring is separated from the ignition wiring as much as possible, to prevent disturbances radiated by the ignition wiring from being picked up by the low-voltage wiring and carried about in the car. Some cars even have the ignition wiring in shielding compartments.

In addition, the modern car body is

all metal and if the hood over the engine is in good contact with the car body, the engine compartment is rather well shielded. This helps greatly in cutting down the amount of radiated interference.

### STATIC DISCHARGES

In spite of the best efforts of the car manufacturer, poor contacts will develop eventually between the car body and chassis and between various pieces of metal in the car. As the car runs, friction between various parts builds up charges of static electricity, which may discharge around these poor contacts (or to the road) and cause interference radiation. We will take up this subject in more detail in a later section of this lesson, where you will learn the various ways of eliminating interference. Right now, since you know how interference can be developed, let us go on to the subject of installing the receiver in a car.

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## Installing Auto Sets

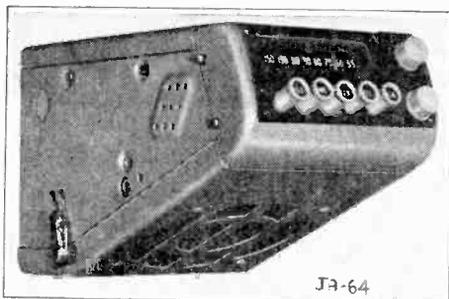
Installing auto radios tends to be a rather specialized branch of radio work which many servicemen do not attempt. Whether you should take it up depends on the number of probable jobs, your mechanical inclinations, and the facilities available. However, even if you do not intend to install car radios, you must know how they are installed and connected so you can get them in and out of the car for service—so let's run through the procedure.

► First, you need a place to work and a few tools. Most auto radio specialists either work for car dealers or operate from a garage. "Drive in" facilities help you work conveniently and safely on the car's ignition system

or on the job of installing or servicing the radio.

► For tools, you will need an electric drill or a breast drill of good quality for drilling holes to mount the antenna and receiver. You should use high-speed, good-quality steel bits, and you will need a center punch. An adjustable end wrench or a set of assorted end wrenches in sizes up to  $\frac{3}{4}$ -inch will complete your collection of special tools.

► As the installation must not interfere in any manner with proper operation of the auto, you'll be wise to have an auto mechanic help you on the first few jobs if you are not too skillful about cars yourself. However, the instructions furnished with



*Courtesy General Electric*

FIG. 3. A standard single-unit auto radio.

new sets are quite detailed and can be readily followed. The next section of this lesson gives general information which will help you get started properly.

### SET TYPES

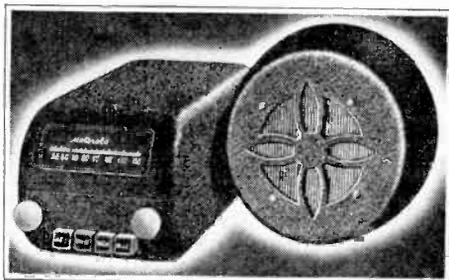
Modern automobile receivers divide automatically into two general types. One type is a general-purpose radio (available from radio supply houses or distributors) which can be mounted in almost any automobile by providing the proper mounting facilities. The other type is a custom-built set, made for a particular make and model of car, and usually sold by the distributor for that car. The majority of both types are single unit radios like that shown in Fig. 3. All early models, and some present ones, have separate speakers (see Fig. 4). Better tone quality is claimed for these latter by their manufacturers. However, space limitations often force the choice of a model with a built-in speaker.

The exact method of mounting will depend on the type of set and the provisions in the car for the radio controls.

Many custom-built receivers are compact models designed to mount in a space behind the dash or instrument panel itself, so that their controls protrude through an aperture in the instrument panel. In such cases

the controls are a part of the radio itself.

If the receiver is not custom-built, it will probably be mounted high behind the instrument panel. It may have extended controls which come out underneath the instrument panel at a convenient location, or may be provided with a remote control system. Usually, in the latter case, the control head will be mounted in an opening in the instrument panel, or even mounted on the steering wheel column, and flexible cables will be used to operate the receiver from this control head. Commonly, such remote controls will operate the receiver



*Courtesy Motorola*

FIG. 4. An auto set with a separate speaker.

tuning condensers, on-off switch and volume control. Sometimes the tone control will also be operated from the control head, but more often it will be right on the receiver cabinet.

Remote control heads designed to be mounted as a part of the instrument panel of a car are usually available in several styles to allow selection of a head which will blend nicely with the other instruments of the car involved and will fit the space allotted for this purpose. Therefore, anyone buying an auto set should be sure to specify the make, model and year of the car in which it is to be used, so he will be furnished with the proper matching control head.

Whenever you undertake to install

an auto set, be very certain to read carefully all instructions furnished by the manufacturer. The receiver manufacturers furnish very complete installation data and, if you follow them carefully, you will usually experience little trouble. Before starting, check the material you received to be sure you have the correct control head for the particular automobile.

## POLARITY

Before installing the radio, check the set diagram and read the manufacturer's instructions carefully to see if polarity matters. If the set is modern and has a *tube rectifier*, polarity will not matter at all. On the other hand, if the set uses a *synchronous vibrator* (which acts as a mechanical rectifier), *polarity is important*. The following thus *applies only to sets using a synchronous vibrator*.

The storage battery must be connected to the car generator with such a polarity that current will flow in the proper direction for charging. Therefore, the polarity of the generator (which may be different in different cars) determines which terminal of the storage battery can be grounded to the car frame, and determines the connections of the electrical equipment. Hence, the storage battery polarity will differ in different cars, having the negative grounded in some cases and the positive grounded in others.

► If the receiver is designed for a particular make and model car, it probably will be already adjusted for the polarity found in that model. However, if the receiver is a universal type which can be used in any car, you may have to make this adjustment. To do so, determine from the manufacturer's instructions the polarity used in the car in which you are

interested. If the car is not listed in your service information, check the polarity by measuring with a voltmeter between the car frame and one of the terminals on the ammeter. Reverse the test probes if necessary, until you get an up-scale deflection of about 6 volts. If the negative probe of the voltmeter goes to the car frame, the negative battery terminal is grounded (and vice versa). With that information, you can proceed to make the required adjustments on the receiver.

## SET INSTALLATION

Before making an installation, try the receiver on your workbench—because even a new receiver can be defective. (Details on how to do this will be given when we take up receiver servicing.) If the receiver operates properly, and you have the proper control head for the particular make and model car, you can now proceed with the actual installation.

Study the recommended positions for mounting the receiver. Usually there will be several possible positions, but your choice may be limited by the fact that a heater or some other accessory has been mounted so it will interfere with one or two of them. Other things being equal, you should, of course, choose the position in which installation is easiest.

The exact mounting of the receiver will depend on its type. The kinds which mount up behind the instrument panel are frequently rather easily mounted on a bracket assembly furnished with the receiver. Another type has bolts on the back of the receiver which are used to secure it to the fire-wall (the partition separating the engine compartment from the driver—it is also sometimes called a bulkhead or dash). With still another type, a mounting plate is mounted on

the fire-wall and the receiver is hung on the plate.

► If the radio is to be mounted on the bulkhead, a drilling template will usually be furnished. This consists of a piece of cardboard about the size of the back of the radio, marked with the location of the holes to be drilled. If a template is not included, make one. To do this, get a piece of cardboard about the size of the back of the radio. Place the radio with the bolts facing upward, put the cardboard on top, and press it against the bolt ends. Then draw small circles on the cardboard around the end of each bolt, thus showing the bolt size and its location.

Move the template to the recommended position in the car and use it as a guide for making the mounting holes in the fire-wall. If dimensions for mounting are given, use a ruler to check the accuracy of your positioning of the template. Then, as a further check, open the hood and examine the fire-wall on the engine side to be sure no ignition coil mounting, vacuum tank or similar object is in the way of the installation. You should also make sure there is enough room for the receiver in the position chosen, particularly if it is one of the bulkier sets.

Mark the location of each hole by driving a steel center punch through the center of the bolt-locating circles on the template. The punch will both mark the steel fire-wall satisfactorily and provide a slight indentation which will help prevent the drill point from slipping. Since the fire-wall is usually rather tough steel, it will be hard to drill unless you use high-grade drills which are in good condition.

► If the speaker is separate, its location should be determined in a similar manner. Usually the speaker is held by a single bolt, so only one hole is

necessary. Be careful to choose a position where the speaker will be well out of the way of the feet of passengers in the front of the car, and check again on the engine side of the fire-wall to be sure you will not drill into some vital part of the car.

After the bolt holes are drilled, sandpaper, scrape, or file the area around them on the engine side of the fire-wall so the washers or nuts will make a good ground to the fire-wall.



Two men working together make installation easier, particularly when mounting the set.

► If the mounting bolts are on the back of the radio, you will need a helper. One of you must hold the set in position and force the bolts through the fire-wall, while the other must place the washers and start the nuts on the bolts after they have come through. Tighten these nuts, with an end-wrench or heavy pliers, enough so they will not loosen even when jarred by the car travelling over rough roads. Be very sure that a good contact is made to the fire-wall, because the

electrical circuit from the grounded side of the storage battery must be completed through the frame of the car and through the fire-wall to the radio.

If a mounting plate is used, hold it in position and force the bolts through the holes. This plate will usually stay in place while the nuts are started. Draw the plate tightly into position by tightening the nuts, then hang the receiver on the hooks provided on the mounting plate.

Remember, complete instructions are furnished with all new receivers. *Read them carefully.*

### ANTENNA TYPES

Once the receiver is mounted, the next step is usually installation of the

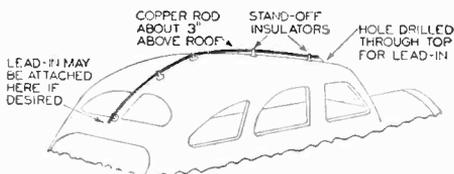


FIG. 5. An antenna mounted on top the car.

proper aerial. The kind of aerial used depends upon the recommendations of the receiver manufacturer and the desires of the set owner. Many automobiles have provisions for certain special types of antennas.

**Top Types.** A very satisfactory aerial could be had in early cars by connecting to the wire mesh in the fabric top of the car. The all-steel tops of modern cars has made this impossible but, since the roof is an ideal location—as it lets the antenna be fairly high and well away from most interference sources—a number of different antennas have been developed to go on top of the car. A typical one is shown in Fig. 5. Several more decorative types have

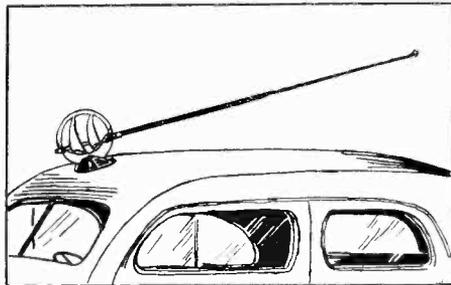


FIG. 6. A typical modern top-type antenna.

been developed, like those shown in Figs. 6 and 7.

The antenna shown in Fig. 7 may be rotated by a knob within the car so that it is in a downward position in front of the windshield. This more or less removes it from sight and safeguards it from low obstructions, such as overhanging tree limbs. The pick-up in this position is sufficient for local conditions. When increased pickup is desired, the knob is turned to bring the antenna to an upright position above the top of the car. Often, antennas of this type are extensible (can be made longer, with one section telescoping within

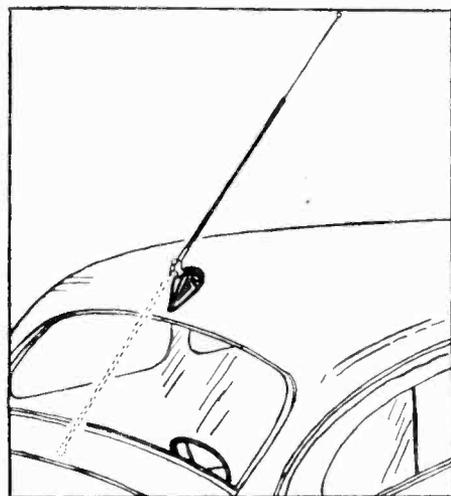


FIG. 7. This antenna is raised for distance reception. It can be turned down to the position shown by the dotted lines if only locals are desired.

another). Some have to be pulled out by hand, while others operate from the knob within the car.

You may wonder why antennas are made so that their lengths can be changed. In cities, the signal strength is usually high enough so that a relatively short antenna will give sufficient input energy for the modern high-sensitivity auto set, so you can make the antenna practically invisible when driving about town. However, as you go out on the road and get

tenna in this location. A more thorough interference elimination procedure is always required for such antennas. In some cars, it is impossible to eliminate interference enough to use them. However, they have proved quite popular with car owners who do not want the antenna visible.

► Some car manufacturers, realizing the desire of the customer to conceal the antenna as much as possible developed cars with insulated running boards which could be used as an aerial in a manner similar to the regular running board antenna. In

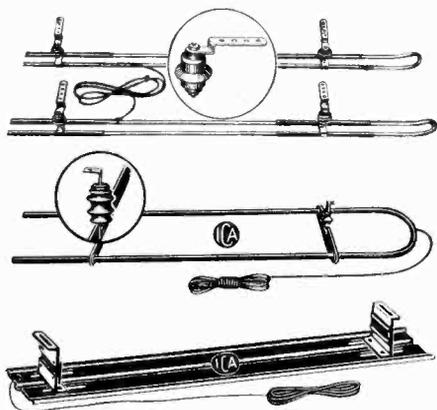


FIG. 8. Several typical antennas intended to mount under the car's running board.

away from stations, increased pickup is necessary. Here, a longer antenna is quite helpful.

**Under-Car Types.** Fig. 8 shows several different types of under-car antennas which are designed to be mounted under the running board. A number of others have been used which run between the axles.

These under-car antennas are subject to breakage when the car runs against the curb or strikes obstacles in the road; they speedily become fouled with dirt, and they are in a poor location insofar as noise pickup is concerned. Brake and wheel static, and interference carried by the low-voltage wiring under the car, are picked up more readily with the an-

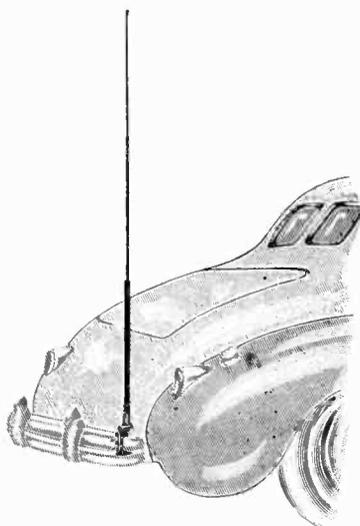


FIG. 9. A rear bumper whip-type antenna.

some of these cars, all you need do is attach a shielded lead-in to the running board. Such an aerial has mechanical advantages over the other under-car types, but it is still subject to considerable interference, especially when out-of-town reception is desired.

**Bumper and Cowl Types.** The whip antenna, which mounts on the rear bumper of a car (Fig. 9) can be extended to be six or eight feet long and gives rather good pickup. Its length and appearance do not add to its popularity, however.

Today, the antenna mounted on the side of the cowl is by far the most popular. There are many types, one of which is illustrated in Fig. 10. Such an antenna gives a reasonable amount of signal pickup and is so positioned that only a relatively short shielded lead-in wire is necessary from the antenna to the receiver. The antenna

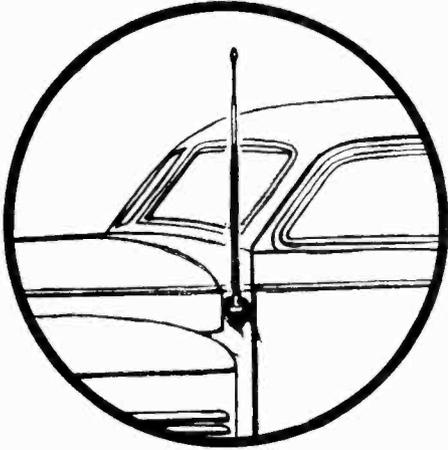


FIG. 10. A side-cowl mounting.

is made from hollow sections which fit inside each other and can be pulled out to increase the effective length.

Some cowl antennas are mounted on brackets which are placed under the edge of the hood, while mounting others requires the drilling of two holes. Typical examples are shown in Figs. 11 and 12.

► Some antennas are made even more inconspicuous by being mounted so they disappear into the cowl or fender. Fig. 13 shows an installation in which a hole is drilled through the cowl and the antenna is mounted in a tube inside the car. Some antennas of this type are manually extended, while others are operated by compressed air obtained from the engine. Fig. 14 shows a fender-mounted type.

These antennas are popular because they are out of sight except when extended and are "gadgets." They

must be mounted carefully, however, to prevent a leak from developing around the seal.

► You have probably noticed a knob or ball on the end of all auto radio antennas. This knob is not just a decoration. Any pointed object which can collect static electricity, such as a pointed antenna, will have a large discharge from its end and this discharge will cause noise. The amount of discharge is reduced if a ball is put at the end of the antenna. Remember this, because a missing ball

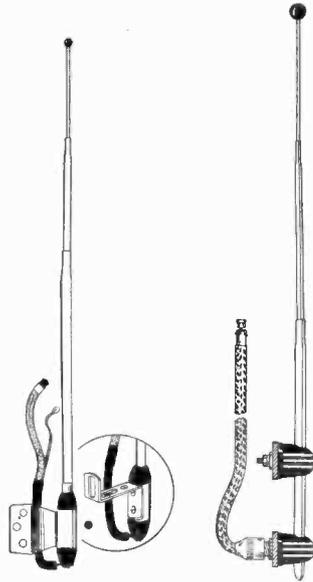


FIG. 11.

FIG. 12.

Two more side-cowl types. One uses a replaceable bracket mounting, while the other requires two holes for mounting.

may cause an unusual static-like noise, heard only when driving around.

**Lead-in.** All auto antennas require a shielded lead-in wire, which may come with the antenna or may have to be purchased separately. *Always keep the lead-in short.* Run it as directly as possible to the radio,

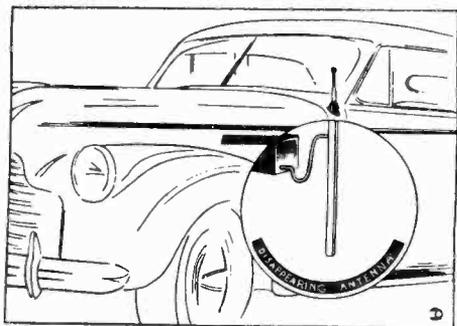


FIG. 13. A disappearing side-cowl type, which can be extended for distance reception.

but keep it away from the auto electrical wiring as much as you can, and of course keep it away from brake pedals or control rods so as not to interfere with the car operation. Remember—even a shielded lead-in will pick up interference if it passes through a high-noise zone.

The shield will ground through the coupling at the radio, but the other end must also be grounded *as close as possible* to the antenna. You may have to drill an extra hole, clean an area about it, and use a bolt or self-tapping screw to provide a good ground connection. Since this ground is sometimes hard to make, careless installation men frequently omit it—and so make the interference problem worse. Remember to check for a poorly made shield ground if the antenna or lead-in picks up interference.

### CONTROLS AND CONNECTIONS

After the receiver and antenna have been mounted, you must mount the controls of the receiver. As you've learned, these controls may be a part of the receiver and require no further attention. On the other hand, you may have a remote control which mounts on or underneath the instrument panel of the car, or on the steering column. In addition, some models have separate sensitivity or tone controls which have to be mounted.

The correct method of mounting the control head is usually obvious, but you should again follow the instructions furnished with the head.

Once the controls are mounted, you are ready to connect the radio. You will have the "hot" A lead, antenna connection, control head cables, and possibly speaker connections, to make. Usually a picture like Fig. 15 or 16 will be included in the installation instructions, so there is little chance for error in making the connections if you follow the instructions carefully.

**Control Cables.** If the set has a control head, it will have flexible cables to drive the tuning condenser

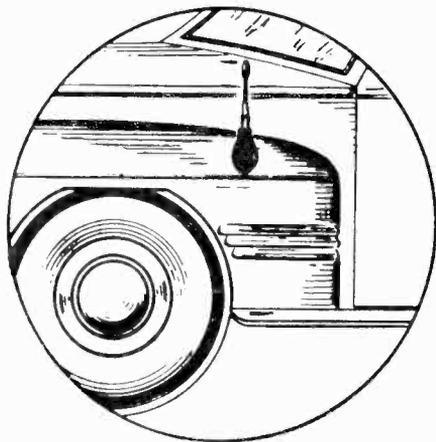


FIG. 14. A disappearing fender-mounting antenna.

and volume control from the remote position. The correct plug-in terminal for each of the control cables must be located by studying the instructions.

The control cables from the control head must be run to the receiver with a minimum of bending or kinking of the cable. Allow the cable to make wide radius turns when it must go around a corner. If you make a sharp bend, the cable will bind inside its housing and will not operate properly.

Of course, these cables must not get in the way of the driver or interfere with any of the control rods coming to the instrument panel.

The control cables plug into sockets on the case of the set. Some are locked in by turning the cable housing a quarter or half turn to engage locking ears in the case; others are anchored by set screws. Before locking into position, be sure the cable it-

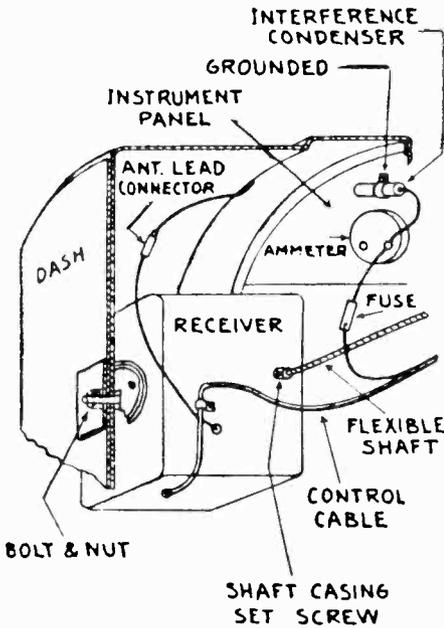


FIG. 15. Typical mounting and connecting diagrams, such as you can expect to be with a new auto set.

self is engaged properly. Most have slotted end fittings which slip inside a sleeve and fit over a projection as in Fig. 17. The control knob should be rotated until the fittings connect together before locking in the cable housing.

► When the control cables have been engaged with the tuning condenser and volume control mechanisms, you must synchronize the tuning dial with the actual rotation of the tuning con-

denser gang. Some models have a slip mechanism. With these, rotating the tuning control first to one end of the tuning condenser range, then to the other end of the range, will "slip" the dial pointer so that it will indicate correctly. On other models, you must release the dial mechanism by turning a set screw, rotate the shaft until the tuning condenser is at one end of its rotation, set the dial pointer to the proper end of the range, and tighten the set screw. Consult the manufacturer's instructions if you have any doubt as to the method used.

► In addition to the control cables, some control heads will have pilot light leads or other connections which have to be identified from the manufacturer's instructions.

**Speaker Connections.** If a separate speaker is used, plug in its connections at this point in the installation. There will be a terminal board or a socket-and-plug arrangement for this.

**Antenna Connections.** Some receivers have sockets for plugging in the antenna lead-in, while others use a short lead and a coupler. Be careful about the latter—it may look like the storage battery lead. However, the antenna will *always* be shielded, and the connector is usually a short affair, frequently with a fastening clamp like the one in Fig. 18.

Some receivers have two plug-in receptacles for the antenna lead, or a special reversible antenna connector. This is provided to compensate for the widely different capacities of auto radio antennas.

Antennas which mount close to the car body have a very high capacity between themselves and the frame of the car, so the antenna transformer must be of special design. Other antennas which are held away from the body of the car do not have such high capacities. Therefore, so that the re-

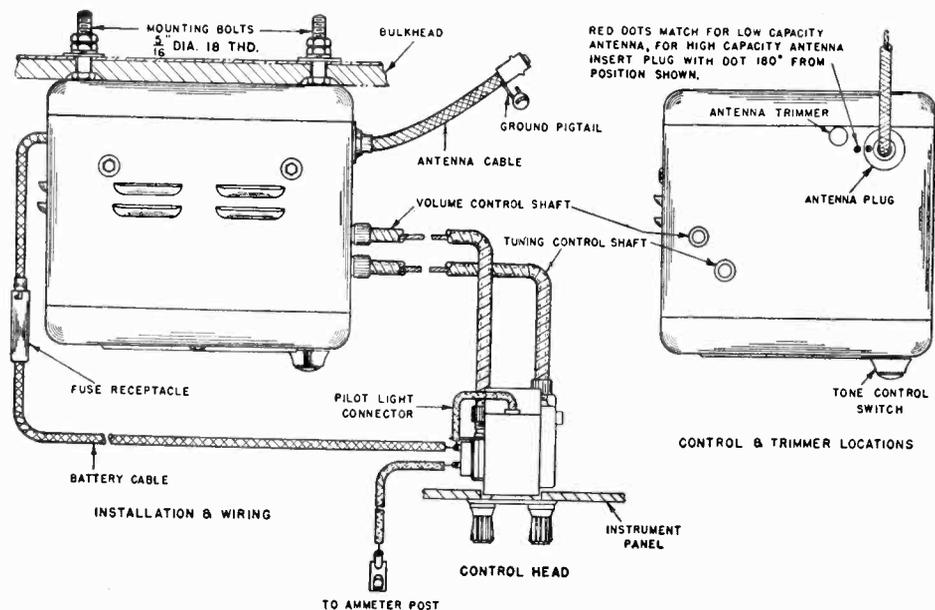


FIG. 16. Another example of mounting and connecting instructions. Notice the figure at the right shows how the antenna capacity is adjusted.

*Courtesy General Electric*

ceiver can be used with either type with equal satisfaction, you will sometimes find both high-capacity and low-capacity terminals provided on the set. Follow the manufacturer's instructions as to which terminal or method of plug-in should be used. Note that Fig. 16 shows one arrangement for adjusting the antenna capacity.

One manufacturer uses a tubular condenser which fits in the antenna connector to reduce the capacity of a high-capacity antenna. Be sure not to confuse the antenna connector for this type with a fuse holder, which it closely resembles.

**Battery Lead.** To provide power, the battery lead (called the "hot" A lead) of the receiver must be connected somewhere to the ungrounded side of the storage battery circuit. (The other half of the power circuit is completed through the set mounting bolts to the frame of the car and thus to the battery.) The terminals

of the car ammeter (which is mounted on the instrument panel) provide a convenient mounting point for this A lead from the receiver. You should connect to the ammeter terminal which does not connect directly to the battery, so the discharge produced by the radio will appear in the ammeter indication.

Sometimes connecting at the ammeter will allow a considerable

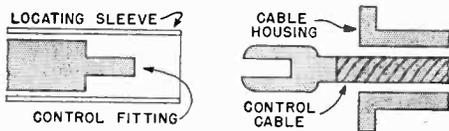


FIG. 17. How the control cable is engaged within the radio.

amount of interference to be fed into the receiver. If so, run the A lead from the receiver directly to the storage battery. As you've learned, a storage battery in good condition acts as a large condenser, so a minimum of

interference will be found directly across the storage battery itself. Once in a while you'll find interference can be reduced by running a ground lead from the radio case to the grounded terminal of the storage battery.

The battery circuit is completed by placing a fuse in the holder (Fig. 19). Use a 10- or 15-amp. auto fuse. Some servicemen use a 20- or 25-amp. fuse

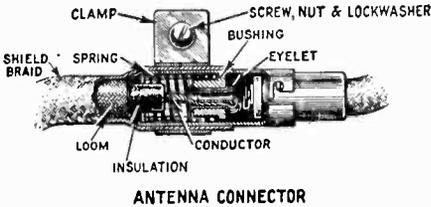


FIG. 18. The antenna connector is longer in diameter than a fuse holder and is in the shielded antenna lead.

because they happen to have it, but this is bad practice—it does not sufficiently protect the radio.

Tuck the A lead and lead-in up behind the instrument panel, out of sight and out of the way. Fasten them with tape if there is any chance of their falling down in the way.

### CHECKING THE RECEIVER

Now, with the receiver installed, you are ready to try it and make the necessary adjustments for best operation.

Since you have not yet suppressed the interference (we will take this up later in this lesson), do not have the engine of the car running when you try the set. Turn on the receiver and tune to the frequency of some local station. Check the operation and calibration of the tuning dial, and make sure the volume and tone controls work properly.

After assuring yourself that the set is working, locate the antenna trimmer from the manufacturer's instruc-

tions. In practically all cases, this trimmer is so placed that it can be adjusted on the outside of the radio to compensate for the effects of the particular aerial used. In some instances a small cover may have to be snapped out of the side of the set, or unscrewed, to reach the trimmer. Once you've located it, tune in a signal at the frequency indicated in the manufacturer's instructions and adjust the antenna trimmer for maximum output. Some are adjusted for maximum at the low-frequency end of the band while others are adjusted at the high-frequency end.

**Push-Buttons.** Many modern auto sets have push-buttons, which may be mechanical, electromechanical, or electrical. In general, the method of

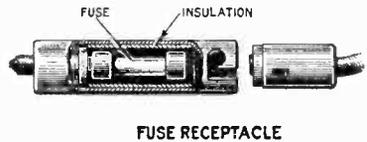
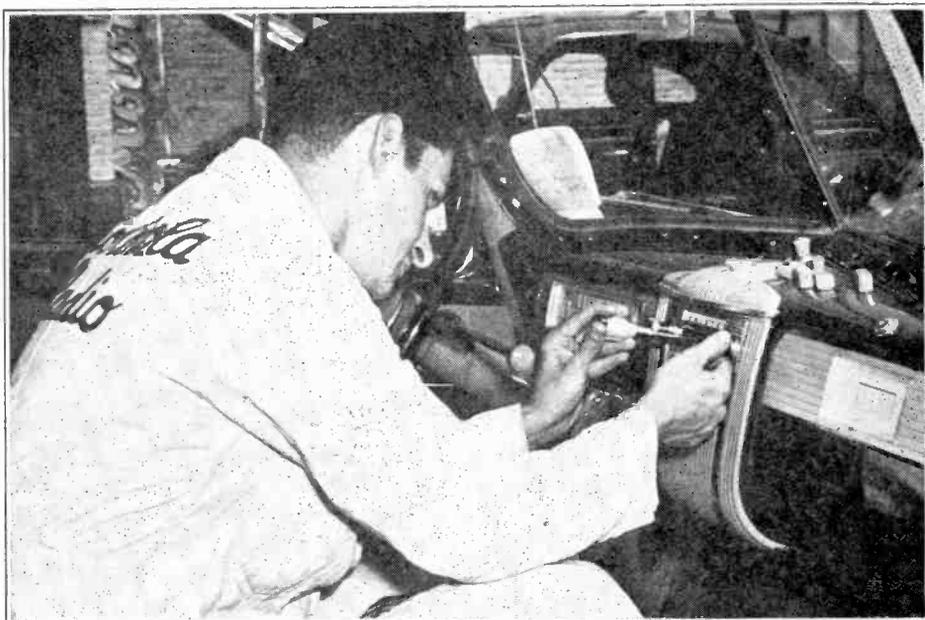


FIG. 19. The fuse holder can be recognized by its shape and by being in the A lead.

setting the push-buttons will be the same as that for a home receiver (given elsewhere in this Course). Of course, you should refer to the manufacturer's instructions to make this adjustment in the quickest possible manner.

### ADJUSTING THE GENERATOR CHARGING RATE

The radio represents an additional drain on the storage battery of the car. Since the battery is already heavily loaded in many cars by defrosters, lighters, lights, electric oil and gas gauges, heaters, clocks, etc., it is sometimes necessary to advance the charging rate of the generator. This is particularly true if a great deal of night driving is done—especially in wintertime, when the engine



Setting mechanical push-buttons after the installation.

is hard to start, and the consequent longer use of the starter motor puts a severe extra drain on the battery.

The charging rate should not be made so high that the battery is overcharged, however; the usual range is up to 15 or 20 amperes. Here again, the advice of a good auto mechanic will be helpful.

Some cars have automatic means of adjusting the charging rate, but in most cases it is necessary to adjust the third brush on the generator. If you remove the commutator cover, you will see this third brush located between the two main brushes—in contact with the commutator, of course. You can usually move this

brush holder by loosening a screw or clamp. Moving the holder *in the direction of armature rotation* will increase the charging rate, while a reverse movement will cause a decrease. After you've made the adjustment, run the automobile engine and watch the car ammeter to make sure the charging rate has increased, but has not increased too much.

It is well to caution the receiver owner that the condition of the battery should be checked frequently with a hydrometer. If the battery is somewhat low, it should be brought up to full charge by a service station and then frequently checked to see that it stays reasonably well charged.

# Eliminating Ignition Interference

An important job in any installation is cutting down the interference generated by the car's electrical system. This is always necessary in a new installation and, because of loosening of chassis bolts in the car and general deterioration in the ignition circuit, additional elimination steps may become necessary after six months or a year of operation.

You've learned that interference comes from the ignition circuit, the generator and low-voltage wiring, wheel and tire static, electric gauges and indicators, and from special appliances such as heaters, electric lighters, etc. Regardless of the source, the interference must enter the set through the antenna and lead-in, the A lead, the control cables, or by direct chassis pickup. To some extent, the method of entry depends upon the installation. For example, an antenna on top the car will be much less subject to wheel static than one underneath or on the side of the car.

There are two general methods of reducing this interference—we can suppress it at the source, or we can introduce shielding between the source and the point of entrance into the radio.

Suppression is usually the easier method, but since we must not affect the operation of the automobile adversely, there are definite limits on the kinds of suppression which can be used.

For this reason, we usually take advantage of the fact that the engine of the car is more or less completely enclosed in metal which can be made to act as a shield if all of it is properly bonded (electrically connected together). For example, if we bond the radiator, engine hood, oil pan and

fire-wall together, we cut down very effectively the amount of radiation from the engine compartment. This is not a perfect shield, however; some radiation may still escape through various openings in this compartment, in addition to the interference which is conducted out over the low-voltage wiring and through control rods, gear shift levers, brake and clutch pedals, steering gear column, etc.

Almost never can you take out every last bit of interference. When tuned between stations, it will almost always be possible to hear some noise. However, you can consider the job is finished if you can tune to relatively weak stations and hear programs without noticeable interference.

Now, let's take up the interference eliminating procedures. We'll start first with the basic elimination procedures—the steps that are practically always taken regardless of the car's make and model.

## BASIC ELIMINATION PROCEDURES

Practically all the equipment necessary for eliminating or minimizing auto radio interference is readily available from the wholesale supply houses. In most cases, the apparatus is made specifically for auto use—some parts being made for specific makes and models of automobiles. You won't have to worry about condenser capacity or resistor values either; you just purchase the parts needed according to their location and the car model. Thus, a distributor suppressor has a different resistance than a spark plug suppressor, but they are sold for their particular use and look different so you won't have any trouble telling them apart.

The bypass condensers used are metal-clad. This metal case keeps

heat from affecting the condenser, protects it from dirt and moisture, and also acts as a shield.

► Installing bypass condensers is the first step to take. A bypass condenser should be installed at the point where the receiver A lead connects to the ammeter. Typical installations are shown in Figs. 15 and 20.

Next, a condenser should be installed at the generator to reduce the interference produced by the commutator. Regardless of the generator type, you will always find a convenient screw for mounting the condenser so that the lead can be properly connected to the hot or ungrounded generator terminal. Several examples of such connections are given in Fig 21.

These two condensers minimize the interference produced in the low-voltage circuit and also take out much of that which is radiated to the low-voltage wires from the ignition circuit.

► Practically all installations will also require some sort of suppression in the ignition system high-voltage cir-

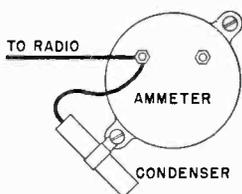


FIG. 20. The condenser should be installed right at the ammeter and must connect to the same terminal as the radio A lead.

cuit. The most effective device is a distributor suppressor which damps out or dissipates the sharp current changes (noise impulses) in the spark discharges. For this reason, a resistor or an r.f. choke coil is always placed in series with the distributor arm. The impulse energy developed in the ignition circuit must flow through this resistor or choke coil, and much of it is dissipated.

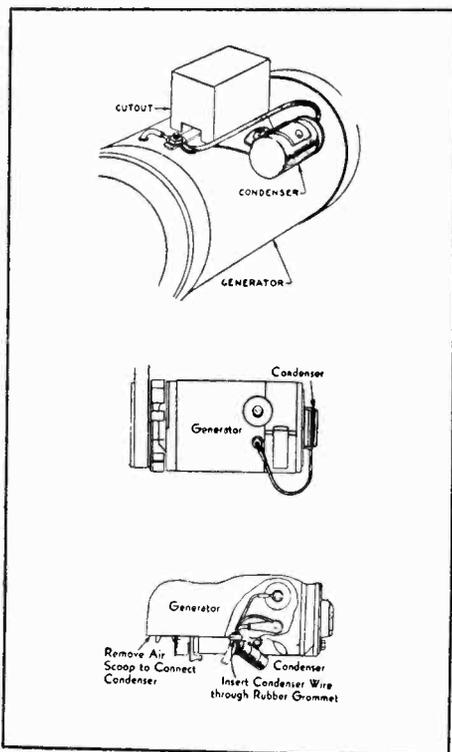


FIG. 21. Examples of installations of condensers at the generator.

Fig. 22 shows some of the many kinds of distributor suppressors available. On the usual distributor head, the high-tension wire from the coil comes to the center terminal. If the high-tension wire plugs into the distributor head, it is unplugged and the special resistor inserted as shown in Figs. 22A and B. If it cannot be unplugged, the wire itself is cut and a special screwtype suppressor is put in series with the wire, as shown in Figs. 22C and D. Several others are shown in the remaining illustrations.

► The three steps we have just described—installing a condenser at the ammeter, installing a condenser at the generator, and installing a suppressor in the distributor circuit—must be taken on practically every automobile. These might be called the basic suppression steps. You will often find

that these three are the only steps necessary, particularly if the car is new and the receiver is a better quality set with built-in interference eliminating features.

## SECONDARY PROCEDURES

If the basic procedures do not reduce interference sufficiently, however, then several secondary steps must be taken.

**Spark Plug Suppressors.** When auto sets first came out, most servicemen installed suppressors at each

spark, but this cannot be carried too far. If the spacing is made too small, the initial voltage may cause a spark to jump more quickly than normal, thus upsetting the timing and again interfering with the operation of the car. You should make every effort to

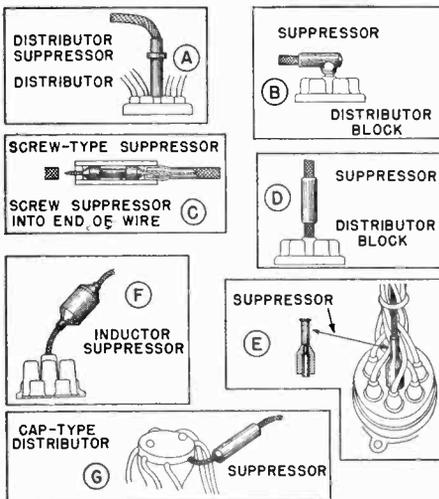


FIG. 22. Types of distributor suppressors.

spark plug. A typical suppressor is shown in Fig. 23 and a typical installation in Fig. 24.

Today, however, spark plug suppressors are not installed unless absolutely necessary, because they tend to interfere with the efficiency of the engine. When the spark jumps the gap, the current flow through the resistor reduces the voltage at the spark gap to such an extent that the spark is no longer as hot as it should be for best combustion. A trained mechanic can reduce the spacing between the spark plug points and so produce a hotter

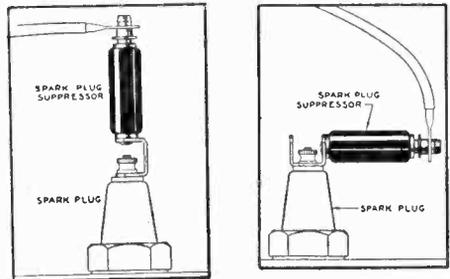
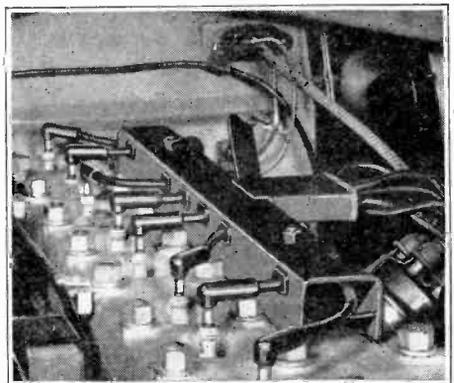


FIG. 23. Typical spark-plug suppressors.

exhaust other possibilities before trying spark plug suppressors.

**Other Steps.** The manufacturer's literature on the radio will tell you steps which may have to be taken for various makes and models of cars. Further information can usually be obtained from the firm selling the automobile. Make full use of this kind of information, for many of the suppression steps may not occur to you otherwise.

Bypass condensers are generally necessary on the leads of the oil pres-



Courtesy Erie Resistor Corp.  
FIG. 24. Spark-plug suppressors installed.

sure, gasoline, and water temperature gauges. In some installations, it is even necessary to bypass the electric clock. Always use the location suggested in the service notes for the bypass condenser, as sometimes it is better to put the condenser at the end of the cable which comes to the instrument panel, while in other instances it is better to bypass the other end of the cable. Condensers with mounting brackets specially designed for specific cars are available; using them will make installation much simpler. Several typical examples of such condensers are shown in Fig. 25.

### BONDING

An important secondary step necessary in a great number of installations is bonding—making good electrical connections between various parts of the car. In most cars, for example, the engine is mounted on rubber blocks so its vibration will not be transmitted to the frame. This means that the only electrical contacts between the engine and the frame are those obtained through the wiring and through such grounds as the car manufacturer may have felt necessary. These are rarely sufficient—so one of the most common bonds needed is between the cylinder head and the firewall.

You should first make the bonds directed in the installation data, then, if interference still persists, check to see if further bonding is necessary. For test purposes, use a length of copper bonding or shielding braid (Fig. 26) about two feet long, with large battery clips soldered to each end. The braid should be the wide, heavy type, like that used to ground the car battery. In fact, these grounding straps can be purchased and used for bonding. Make tests by starting the car and turning on the radio, then clipping together various metal parts of

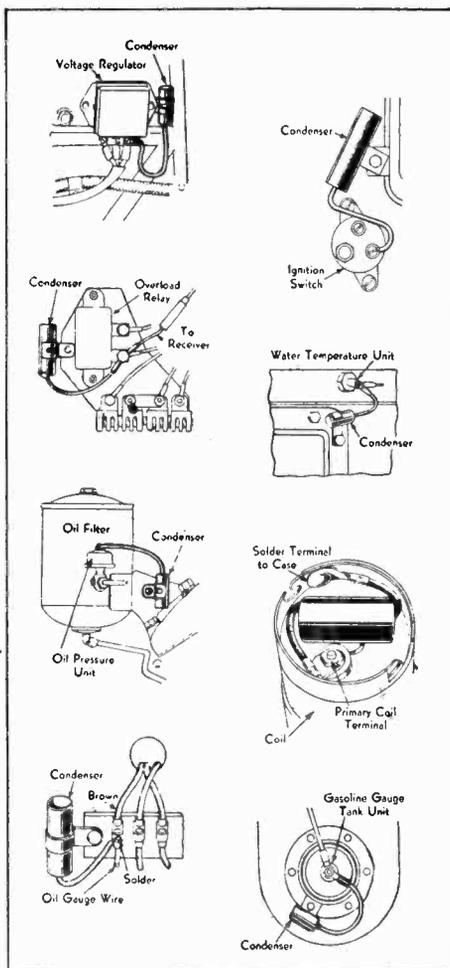


FIG. 25. Several examples of condenser installations for noise suppression. Diagrams like these come with receivers or are obtainable from auto dealers or from manufacturers of suppressors, showing the proper installations for the particular make and model receiver and car.

the car with the braid to see if interference can be reduced. Check particularly between the car body and the chassis, instrument panel and body, chrome trim (decorative grills) and body, transmission housing and chassis, steel floor boards and chassis, brake rods and chassis.

When you find points where your bond helps, either clean away paint and grease from under the bolts hold-

ing these parts together or install a piece of the bonding braid between these points. Be careful to make good contacts and use short bonds, but leave enough slack so vibration does not break the braid.

► Fig. 27 shows several examples of bonding. To make the cylinder-head-to-fire-wall connection, remove one of the bolts in the cylinder head and carefully clean around the bolt hole. Punch a hole in the braid, then slip a washer and the braid on the bolt, reinstall the bolt, and tighten it. Fasten the other end of the braid similarly to some convenient point on the frame or fire-wall of the car. Be sure enough slack is allowed for engine vibration, but don't use excessively long pieces of braid—they may get in the way of controls or have



*Courtesy Lenz Electric Mfg. Co.*

FIG. 26. A copper braid, useful for bonding or for shielding wires.

sufficient resistance to be ineffective.

► Fig. 27 also shows how to bond control cables with flexible braid. You should fasten one end of the braid to the fire-wall by drilling a hole in the wall and passing a self-tapping screw through the braid into the hole. Solder the other end of the flexible braid to the control cable. Be careful to allow enough slack so that the control cable can be moved throughout its range without interference.

► It is frequently necessary to bond the hood of the car. The hood is separated from the frame of the car by a strip of felt or fabric to prevent the rattling that a metal-to-metal contact would cause. Therefore, the only grounding of the hood is that obtained through the catch holding the hood closed and through its hinges. Usually, a great amount of interfer-

ence can be eliminated by using little spring contactors to improve the hood grounding. These contactors are flat strips of spring brass, made with a roughened, jagged surface which will make a good contact through grease, oil, and paint. To install one, loosen one of the screws holding the felt in place, and insert the metal strip *under* the felt so the strip is in contact with the cowl of the car. Put the screw back through the felt and through the metal strip to hold them in place, then bend the metal strip back over the top of the felt, so that the roughened metal surface is outermost. Now, when you close the hood, it will be firmly grounded to the car cowl and frame through the U-shaped strip. You should install one of these on each side of the cowl.

### SHIELDING

Ordinarily, you should avoid installing shielding on the car electrical system unless you are sure that it will not affect the operation of the car, and unless the manufacturer recommends it. Usually, it will be all right to shield low-voltage wiring, but shielding on the ignition circuit is to be avoided unless absolutely necessary. It is true that many modern automobiles are now manufactured with shielding over the ignition wiring, but this has been carefully placed by the manufacturer; it will not cause trouble by reducing the efficiency of the ignition system, nor will it trap heat and result in the ignition wiring being destroyed. This is an important consideration, as it becomes quite hot in the engine compartment of a car.

► It may be necessary to shield water hoses going to heaters. Remember that signal noise energy will flow over any path that is even semi-conductive. Some cars have heaters under the front seat with long hose connections from the engine. These water

paths carry ignition interference inside the car. A shield may be necessary (see Fig. 28) over such water hoses.

► Sometimes screen wire must be installed over the flooring of the car between the front seat and the fire-wall. Put this screen under the floor mat, cutting it to fit around the brake and

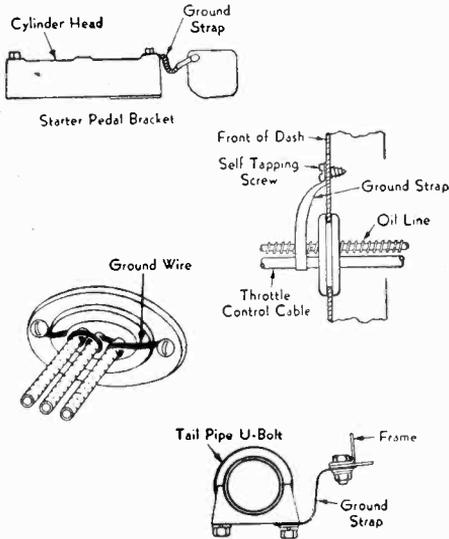


FIG. 27. Examples of bonding.

clutch pedals, etc. It must be bonded to the fire-wall and fastened to the car body at the sides.

### STATIC DISCHARGES

Annoying amounts of interference can be caused by discharges of static electricity developed in the brake system and about the wheels.

Brake static is commonly caused by metallic particles in the brake lining touching the revolving drum connected to the wheel, by dragging brakes (where the brake lining is partially in contact with the brake drum), or by improper adjustment of the brakes.

Usually, brake static can be cleared up by having a mechanic check over

and readjust the brakes. Sometimes bonding the brake control rods will prevent this interference from being carried to other parts of the car.

► Wheel and tire static are interesting examples of conditions which can arise when poor contacts develop between metal parts. The sounds produced may be an intermittent rasping or clicking noise, with the time intervals varying with the speed of the car, or may be a steady hissing developed after the car reaches a certain speed. The trouble will occur only when the auto is in motion and will usually be worse when going over asphalt or cement pavement, although it may be noticed in some cases on brick pavements or on dry gravel roads. It will frequently disappear during periods of high humidity or when the pavement or wheels are wet but will reappear as the road or wheels dry out. Driving off the

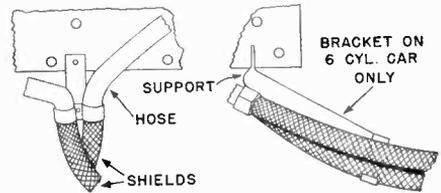


FIG. 28. Shields must sometimes be placed over the hose going to heaters located under the seat.

pavement onto the dirt shoulder or on to dirt side roads will usually stop the noise.

Apparently this noise is produced by friction between dry pavement and the rubber tires, which generates static electricity. This electricity collects on isolated conductive substances in the tires or on the metal wheels (which may be insulated from the body of the auto by grease or oil). The charge builds up, then discharges to the car body or road bed from time to time. If the application of brakes

causes the noise to disappear, we attribute the noise to wheel static; otherwise, it is tire static.

► Since wheel static means metal portions of the wheel are insulated from the car body by grease and oil, the obvious solution is to apply some contacting means. Special static collectors, which are installed under the decorative hub caps, are made for this purpose. There are several types, and the recommended type for the particular car wheels should be used. Typical installations are shown in Fig. 29.

► Tire static is harder to explain. Apparently it is sometimes caused by metallic paint spots inside the tires. (Some manufacturers use such paint inside tires to balance them.) In such cases, cleaning the inside of the casing with a wire brush and then wiping the casing thoroughly with benzine and a cloth to remove this paint will clear up the trouble. Sometimes a special paint available from the maker of the tire can be used to coat the inside wall of the tire.

The difficulty may also be caused by vulcanized patches if a metallic-base glue has been used on the tube or casing.

Often (particularly if the wheel has wooden spokes) the trouble can be minimized by running a bonding wire between the metal rim and the hub of the wheel to make better contact. We refer, of course, to the removable metal rim, which paint and rust may have insulated from the rest of the wheel.

### SPECIAL TROUBLES

With older cars, you can expect deterioration to raise the interference level higher than normal. A crack on the spark coil or on the distributor head will provide a leakage path which will affect the operation of the engine somewhat and will create a

terrific amount of interference. Also, dirty and worn spark plugs, cracked spark plugs, leaky ignition wiring, and loosened body bolts will all raise the interference level. The services of a good mechanic may be required to put the ignition system in first-class order.

► Frequently excessive noise will be produced if the rotor arm of the distributor is spaced too far from its contacts. Because it turns at high speed, this rotor arm must not strike any of the contacts, but it should be equally spaced from all of them and should just barely miss them as it rotates. A good mechanic, if he finds the spacing too great between the end of the rotor arm and the contacts, can take out the rotor and (with a peening hammer) flatten out the contacting tip to make it somewhat longer. Then, he will reshape it and make a check to be sure it does not actually strike any of the contacts.

If the spacing between the contacts and arm is erratic, a new distributor head should be obtained for the car.

### DETERMINING THE SOURCE OF INTERFERING NOISES

It is usually rather easy to determine whether noise originates within the receiver or is caused by ignition interference. Here are some logical step-by-step procedures which will identify the source by the sound of the noise and the conditions of operation.

**Case 1.** *First try the receiver with the car standing still and with the engine not running.* This condition eliminates mechanical vibration from the engine, the ignition circuits are not working, the generator is not working, and brake or tire static is not a problem. Any noises heard must be caused by defects within the receiver, by a loose connection or partial ground in the antenna circuit,

or by a poor connection to the low-voltage circuit in the car.

If the set is noisy, or striking the set sharply with your hand causes noise, you can be sure the noise source is within the receiver. (We will go into internal receiver noises later.) Wiggling the antenna lead will show if the antenna circuit is faulty.

Listen carefully to see if the noise actually comes from the speaker. Often the speaker will be vibrating a loose plate, ashtray, or some control cable from the instrument panel.

► While the car is standing still, try operating the brakes. You may hear a loud click when the stop-light (at the rear of the car) turns on as the brakes are applied. A condenser between the stop-light lead and the car chassis will eliminate this.

If you find a noise is not present under these conditions, go on to Case 2.

**Case 2.** Turn on the engine and let it idle or run slowly. The car should remain standing still. You will now get electrical interference from the ignition system and also a certain amount of mechanical jarring from the engine. Frying or crackling noises are very likely ignition noises. Presuming you have already carried out the basic elimination procedure (or you find this has been done if someone else installed the receiver), check to see that all the manufacturer's recommendations have been carried out. It is quite possible that the basic procedures were sufficient when the automobile and the receiver were new, but that aging has increased the interference level.

► Next, disconnect the antenna lead-in from the receiver. If the noises cease, the antenna or its lead-in is picking them up. Be certain the antenna lead-in is thoroughly shielded and that the shield has a good ground where it first enters the car.

In some instances it is advisable to try other aerials at different locations, to see if the position of the aerial is poorly chosen. For example, an under-car aerial is always subject to more interference than one on top of the car.

► On the other hand, if the noise continues with the antenna lead-in disconnected, the noise is coming in over the A battery lead to the set, or the set is picking the noise up directly, or the mechanical jarring of the engine is producing the noise in the radio.

Usually, in the last case, jarring the set with your hand will change the noise level.

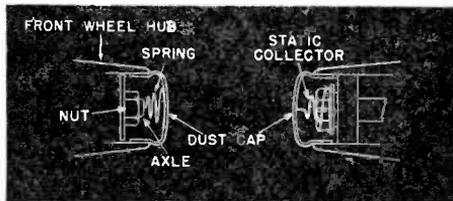


FIG. 29. Wheel static collectors.

Interference coming in over the A battery lead sometimes indicates that a bypass condenser was not used where the A battery connects to the car ammeter. Sometimes it is necessary to run the A battery lead directly to the storage battery to help clear up this interference.

In other instances the noise will enter the radio set directly, probably because some control rod or cable going to the instrument panel of the car happens to run close to some opening on the radio container. Ventilating holes are necessary on the radio, and, when the speaker is built in, holes must be provided to eliminate back pressure on the speaker. If control leads run close to these holes, interference may feed through them. Try moving such cables about to see if

this affects the noise level. If so, move them to a position of minimum interference which does not interfere with their normal operation.

**Case 3.** *After the preceding two tests have been made, speed up the engine but leave the car standing still.* Run the engine fast enough to show a charge on the ammeter. You will now get an increased amount of vibration, generator noises will be more common, and some ignition conditions may show up which were not apparent at lower engine speeds.

► A whining noise, increasing as the speed of the engine increases and decreasing as the engine is slowed down, usually indicates that the generator condenser is improperly installed or ineffective, or that the commutator needs cleaning. The commutator on the generator can be cleaned by holding against it a strip of sandpaper or a lint-free rag moistened with carbon tetrachloride. Sparking, with consequent interference, may also be present if the commutator or the brushes are badly worn or if the springs holding the brushes in contact with the armature have lost their springiness. It is usually best to have these conditions corrected by an experienced mechanic.

► A rattling noise usually indicates something loose inside the car or about the radio. First determine if the sound is coming from the speaker or seems to come from somewhere else. You may frequently find control heads or rods which vibrate. If the noise stops when you touch the vibrating part, the part must be anchored so that it cannot vibrate.

► Again, a buzzing or crackling noise may indicate ignition noise, which is worse at high speeds.

► On some automobiles, the coil is mounted behind the instrument panel instead of in the engine compartment. You may find interference in such cars

until the high-tension wire from the coil to the distributor is shielded. Sometimes it is even necessary to shield the entire coil. Use a piece of shielding braid for covering the wire.

The vibration produced by the engine at higher speeds may also cause noises by shaking some defective contact.

**Case 4.** If the noise complained of does not show up with the car standing still, *drive the car or have it driven, so you can determine the effects of motion.* Any noises which develop with the car in motion, but which are not heard with the car standing still, are usually caused by frictional electricity generated in the brakes or wheels. To prove this condition exists, *coast down a hill with the engine turned off.* When the engine is not running, the ignition circuits are obviously not functioning and there is no engine vibration, so noises must almost certainly be caused by brake or wheel static.

Complete cures for wheel and brake static have been given earlier in this lesson.

## SUMMARY

We have given here the general procedures applicable to any car and radio installation. As we have repeatedly mentioned, the manufacturer's instructions accompanying the receiver should be read carefully. If you intend to go into auto radio work, make a habit of collecting tips on various installations. Very good information is frequently available from the automobile distributors. These tips will always save a great amount of time by leading you directly into the proper first steps. You will have trouble only in those older automobiles where the interference level is high and where little information is available. For these, you may have to spend a great amount of time test-

ing interference elimination procedures before you hit upon the combination which does the most good. Remember also, you can't get rid of

all the noise heard between stations. Try to eliminate as much as you can, checking the level when tuned to local stations.

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## Servicing Auto Radios

Fundamentally, servicing auto receivers is no different from servicing any other kind of radio; exactly the same principles of effect-to-cause reasoning and localization can be applied as can be used on an a.c. home radio having a power transformer. Except for its extreme compactness, it presents only the special problems of the vibrator power supply.

Before taking up the service procedures, let's see what equipment is required and review vibrator systems.

### WORKBENCH EQUIPMENT

There are a few items you will have to provide at your workbench if you intend to do auto radio servicing. One will be a source of power to operate the set. The best device for this purpose is an auto storage battery.

If you intend to do any large amount of auto radio servicing, get a good battery and keep it in good condition at all times. It must be kept filled to the proper level with distilled water and must be kept charged. Therefore, you should have a charger in your shop.

Because of the trouble of taking care of a battery, some service shops use special eliminators designed for auto radio work. They are available from wholesale supply houses and can furnish 6 to 8 volts at currents of from 12 to 25 amperes, depending on whether you get a light-duty or a heavy-duty type.

This high current capacity is necessary. An auto radio in good condition

will draw anywhere from 4 to 10 amperes, depending on its type and size. When defective it will draw even more.

► To operate an auto set in your shop, connect one lead of your battery or A power pack to the case of the receiver and the other to the A battery lead. Polarity is of no importance unless the set has a synchronous or self-rectifying vibrator, in which case you must use the same polarity of connections as were used in the car from which the receiver was obtained.

Many servicemen forget that an auto set is extremely sensitive, and make the mistake of connecting a rather long piece of wire or a regular aerial to the set on the workbench. Even an auto set which has lost most of its sensitivity will operate wonderfully from this long aerial, but may be absolutely dead when installed in the car, where it must operate from a small antenna. Use a very short piece of wire or a standard auto aerial mounted on the workbench to try out auto sets properly.

### VIBRATOR POWER SUPPLIES

Practically all modern auto sets operate from a vibrator type power supply. Police receivers and a few of the older types may use motor generator sets, but these are not often encountered in regular service work.

Since the vibrator and its power supply systems are common sources of trouble, let's review the basic vibrator types briefly.

**Non-Synchronous Vibrator.** The circuit of a typical vibrator system is shown in Fig. 30. When switch *S* is closed, the circuit is completed from the storage battery to the tube filaments through choke *L*<sub>2</sub>. (Tubes having 6.3-volt filaments are used, with the filaments in parallel.) The vibrator supply and rectifier tube filament current flows through *L*<sub>3</sub>.

The vibrator contains a flexible reed *R* which can be made to touch contacts *A* and *B* alternately. When the switch *S* is turned on, current flows through *L*<sub>1</sub>, *L*<sub>3</sub>, *P*<sub>2</sub> and *L*<sub>6</sub>. This energizes the electromagnet *L*<sub>6</sub>, which

transformer *T* which are relatively sharp and square, with a great many harmonics—shaped like noise pulses, in fact. Of course, the voltage induced in the secondary is similar in shape and would have extremely high, sharp peaks except for the buffer condenser *C*<sub>5</sub>. This condenser tends to smooth out the pulses and prevent very high peaks.

Even so, the output of the rectifier tube has considerable “hash” in it. An r.f. filter is normally used in the cathode lead of the rectifier, before the power is fed to the filter, to eliminate some of this “hash” (which

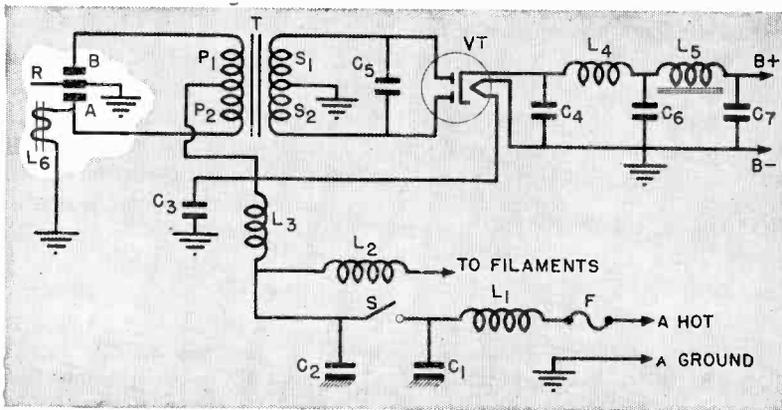


FIG. 30. A typical non-synchronous vibrator type power supply.

pulls the reed down, making contact to *A*. This permits full current to flow through *P*<sub>2</sub> and also shorts *L*<sub>6</sub>, causing it to release the reed. The reed then flies back, striking contact *B* and completing the circuit through *P*<sub>1</sub>. Then, coil *L*<sub>6</sub> again attracts the reed, repeating the cycle.

The pulsing current flow first through *P*<sub>2</sub> and then *P*<sub>1</sub> induces an a.c. voltage in the secondary of transformer *T*. This voltage is then rectified by *VT* and passed on to the filter.

► The rapid circuit openings and closings caused by the vibrator produce current pulses in the primary of

would cause interference). R.F. choke *L*<sub>4</sub>, condenser *C*<sub>4</sub> and the electrolytic *C*<sub>6</sub> make up this filter.

Similarly, coil *L*<sub>3</sub>, condenser *C*<sub>3</sub> and the *C*<sub>1</sub>-*C*<sub>2</sub> combination form a filter in the low-voltage circuit to prevent vibrator interference from feeding back into the filament supply, which is further filtered by *L*<sub>2</sub>.

Coil *L*<sub>1</sub> and condensers *C*<sub>1</sub> and *C*<sub>2</sub> act as a filter to prevent interference from coming in over the “*A*” battery lead; so do *L*<sub>3</sub>-*C*<sub>3</sub> and *L*<sub>2</sub>.

It would seem that when the switch is closed, a single condenser would do in place of *C*<sub>1</sub> and *C*<sub>2</sub>. However, the long leads to the switch are inductive,

so the pair of condensers is required for adequate filtering.

As you can see, a vibrator source requires far more filtering than do other power sources, because the interference effects of the vibrator itself must be eliminated. The vibrator must be well shielded; it is often encased in a special compartment with

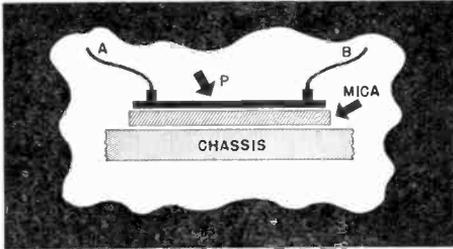


FIG. 31. How a "spark-plate" condenser is made.

the transformer, rectifier tube and the r.f. filter to isolate them from the remainder of the radio and so reduce the amount of noise further.

► Condensers  $C_1$  and  $C_2$  are called "spark plate" condensers and deserve some special mention. The particular symbol used indicates that one plate of each of these condensers is the receiver chassis. As Fig. 31 shows, the other plate  $P$  is insulated from the

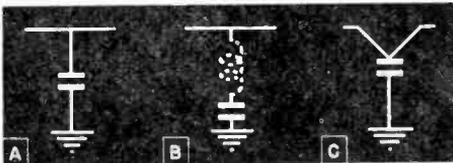


FIG. 32. The inductance of the wire lead is eliminated by the double-wire construction.

chassis by a sheet of mica. It may be held in position by insulated clamps or insulated rivets.

► Watch for connections like that in Fig. 31, where wire  $A$  ends at the condenser plate and wire  $B$  completes the circuit to the succeeding parts. In other words, although wire  $A$  ends

at the plate, the circuit continues. The reason for this peculiar connection is shown in Fig. 32.

Ordinarily we would expect a bypass condenser to be connected as shown in Fig. 32A. Actually, however, there will be a certain amount of inductance in the wire going into the condenser (Fig. 32B). This inductance is therefore between the circuit and the condenser, and limits the effectiveness of the condenser. By bringing the circuit wire up to the condenser and continuing on as in Figs. 31 and 32C, there is no common

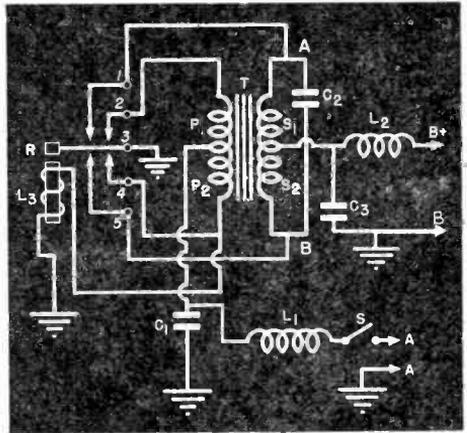


FIG. 33. A synchronous vibrator power supply.

lead to the condenser, so whatever inductance exists is that already in the circuit. Using the chassis as the ground plate of the condenser similarly eliminates the lead to ground, making the condenser far more effective at high frequencies.

**Synchronous Vibrators.** Some auto sets use synchronous vibrators to eliminate the rectifier tube filament current drain. In these, the vibrator both interrupts the current flow to make the pulses in the primary and mechanically rectifies the output of the secondary. The operation of the vibrator and primary circuit (see Fig.

33) is similar to that described for Fig. 30. Current flows through choke  $L_1$ , through primary winding  $P_2$ , and through coil  $L_3$ . Coil  $L_3$  attracts the vibrating reed  $R$ . When the moving reed touches the contact going to 4, coil  $L_3$  is short-circuited and the reed flies back to the other side.

In moving back and forth, the reed first touches the contacts going to terminals 4 and 5, and then those going to terminals 2 and 1. Touching the contacts going to 2 or 4 completes the primary circuit, while touching those going to 1 or 5 completes the secondary circuit, through the grounded reed connection.

► If the wires from the primary and secondary to the vibrator are correctly connected, the  $B$  supply will always deliver voltage with the proper polarity, as marked. On the other hand, if the connections are incorrect, the  $B$  supply polarity will be reversed and the set will not operate, as the tube plates will be negative instead of positive. If this circuit is operated for any length of time with the reversed polarity, the electrolytic filter condensers will be ruined.

For this reason, you must watch polarity when you connect a receiver with a synchronous vibrator to a car battery. If the vibrator circuit is wired to work properly in a car in which the negative terminal of the car battery is grounded, it will deliver the wrong polarity in a car in which the positive battery terminal is grounded—and vice versa. Therefore, if the circuit is wired to be used with the negative side of the battery grounded, but you want to put it in a car in which the positive side of the battery is grounded (or vice versa), it will be necessary to reverse *either* the leads to the primary or the leads to the secondary of the power supply transformer. It makes no difference whether you reverse the primary leads

or the secondary leads, but you must be sure to get a *pair* from the same winding—not one lead from each winding. For example, you can take the leads  $A$  and  $B$  from the secondary of transformer  $T$  and interchange them at the vibrator, so that  $A$  goes to 5 and  $B$  to 1, or you can reverse the primary by interchanging the leads so as to connect  $P_1$  to 4 and  $P_2$  to 2. Don't make the mistake of reversing *both* pairs, as this leaves things the same as before.

Often this change can be made merely by unplugging the synchronous vibrator unit, turning it half a revolution ( $180^\circ$ ), and plugging it back in again. This can be done if the socket contacts are arranged as in Fig. 34, so that reversing the vibrator this way automatically reverses the connections to one set of the terminals. (This arrangement is found *only* on some of the 5- and 6-prong synchronous vibrators. The *non-synchronous* vibrator and *other* synchronous types will plug in their sockets *in only one way*.)

Some sets have terminal boards with leads which either plug in or can be easily unsoldered for making this reversal.

► Always connect your bench power supply or battery to the set with the same "grounded" battery terminal connected to the chassis as will be connected to it in the car the radio is to work in. (You may not know this polarity in all cases.) As a check to see that the proper connections are made, connect a d.c. voltmeter between  $B+$  and the chassis so the meter  $+$  goes to  $B+$  and the meter  $-$  to the chassis. Turn the set on and watch the meter. If it reads up-scale, the proper battery polarity connections have been made.

If the meter deflects down-scale, turn the set off quickly to protect the filter condensers. If the set was working in a car and is to go back in the

same car, reverse the connections to your battery or supply (since the polarity of the set is obviously correct for the car). On the other hand, if the set is just being installed, connect it to your battery according to the known polarity of the car and if necessary, reverse the connections of the vibrator for this polarity.

It is important to remember that *this polarity question does not come up at all for a non-synchronous vibrator using a tube rectifier*, as the tube will automatically pass current from whichever plate happens to be positive at any moment and therefore is always correctly connected.

### VIBRATOR CIRCUIT DEFECTS

As we have shown, the vibrator is a noise source and requires considerable shielding and filtering, even when working normally. When it becomes erratic in performance, the amount of interference can assume terrific proportions. Furthermore, like any other mechanical device, a vibrator will eventually wear out. Arcing makes the contacts worn and pitted, and the reed loses its springiness. Thus, the vibrator has a definite life period just like a tube and must be replaced from time to time.

In addition, any short or leakage in the B supply circuits will draw excessive current through the vibrator, producing increased sparking and greater contact wear. For longest vibrator life, the receiver must be kept in tip-top condition.

In particular, the buffer condenser should always be checked whenever the vibrator is found defective. The buffer condenser must withstand high-voltage surges; even though they are usually rated at 1200 to 2000 volts, they still break down from time to time. Of course, a defect here would short-circuit the secondary and thus overload the vibrator heavily.

The size of the buffer condenser is rather important. The vibrator has a definite rate of operation (which is today around 115 cycles per second), and the vibrator, transformer, and buffer condenser are designed as a unit to suit this rate. However, different vibrator rates—and correspondingly different condensers—have been used. Should you have to replace a buffer condenser, be sure to get one with a similar or higher voltage rating than the original and *with the same capacity as the original*.

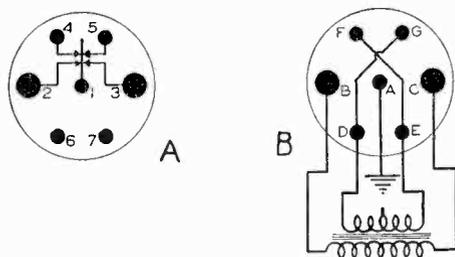


FIG. 34. The prong arrangement for a reversible synchronous vibrator is shown at A, while the socket connections are shown at B. When plugged in one way, the contacts 1, 2, 3, 4 and 5 make contact to socket terminals A, B, C, D and E. When the vibrator is removed, turned through a half circle and plugged in again, the vibrator contacts 1 to 5 inclusive now make contact to socket terminals A, B, C, F and G. This automatically reverses the connections to one of the windings on the transformer. The vibrator will fit the socket in only one of these two ways, because prongs 2 and 3 are considerably larger than the others, so no other plug-in arrangement is possible.

► Some sets use a cold cathode rectifier, which depends on gas ionization for conduction. The advantage of this tube is that it eliminates the filament current drain of the normal rectifier tube, so that the radio draws slightly less current from the storage battery. However, circuits using cold cathode tubes are subject to oscillation, which in turn produces noise.

If you find such a tube is causing interference, and if the automobile battery is not too heavily loaded by gadgets, substitute a regular heater

type rectifier. Usually the only change necessary is to wire the filament circuit (and even this may already have been done by the set manufacturer).

**Testing Vibrators.** How can we determine if the vibrator is defective? At one time, vibrator testers were available. These were similar to tube testers in their operation. The vibrator was plugged into the proper socket, and its worth was determined. However, it is so easy to try a replacement that there was no real demand for these testers, so they have practically disappeared. Hence, the serviceman first relies on an indication of efficiency; the vibrator should deliver its rated voltage with a minimum of current drain. The output of an aging vibrator will drop off slowly but the current consumed will increase rapidly. Therefore, the first indication of a defective vibrator is a higher-than-normal current drain.

If the receiver draws an abnormally high current, which may be shown by the car ammeter or by blowing fuses repeatedly, suspect the vibrator, but remember it may just be overloaded by some fault within the set. Hence, if you find a radio is drawing excessive current and you do not have a vibrator tester, you should first check the B supply circuit with an ohmmeter to be sure there are no leaky condensers or shorts. If there are none, try another vibrator to note the performance of the receiver. Improvement indicates the original vibrator was faulty.

► Suspect the vibrator if it makes an erratic sound and the B voltage is lower than normal. (The vibrator should produce a steady buzzing sound which does not vary in pitch after the set has warmed up.) Also, defective vibrators frequently will not

start operation until the set is kicked or jarred or the ON-OFF switch is snapped off and on several times.

► Defective vibrators should normally be replaced. In emergencies, it may be possible to get a few more weeks of operation by cleaning and filing the contacts and bending the vibrator reed. The contacts can be cleaned by using a thin, flat ignition or jeweler's file. After filing, the gap between the contacts should be adjusted to about .003 to .006 inch (about the thickness of this page). Then, try the vibrator and make further bending adjustments, if necessary, to obtain steady, smooth operation with a minimum of sparking.

Even at best, it will be impossible to approach the accuracy of the factory adjustment. Once contact arcing and spring fatigue have begun, no adjustment will hold for more than a few weeks at most. It is far better to make a replacement if at all possible.

### CHECKING THE SET IN THE CAR

Because of the trouble involved in taking the radio out of the car, it is best to make as many tests as possible with the set installed. Usually, taking off a cover will let you remove the tubes for testing. Most modern vibrators are of the plug-in type, so they can easily be removed for checking or to try another one.

Should the owner complain of unusually short tube and vibrator life, be sure to check the voltage coming from the low-voltage supply *with the car engine running*. In many such cases, a poor connection exists at the car storage battery terminals or the battery is over-discharged so it has a high internal resistance. This extra resistance in series with the battery prevents the battery from "pulling down" the generator voltage, because the battery does not draw the normal

charging current. This means the generator may deliver as much as 8 or 9 volts to the low-voltage circuit. This increased voltage is applied to the tube filaments, and also will be stepped up in proportion, producing as much as a 100-volt rise in B voltage. Both conditions produce an excessive tube current and shorten tube life.

Oscillation, or a remarkable sensitivity rise when the generator is charging, is a clue to this voltage rise. If you find the voltage between the radio A lead and chassis is above normal when the generator is charging, check the battery and its connections. In some cases it is best to connect the radio A lead directly to the battery and to run another lead from the grounded battery terminal to the radio chassis. The voltage then can never rise above the battery voltage as long as the battery is in good condition.

► If the set is dead, check the fuse in the A lead. Auto radios should use fuses rated at no more than 15 amperes. If this fuse blows, you can be fairly certain that the vibrator contacts were stuck together at least momentarily. This may occur once in a great while without there being any real trouble, but it usually means that the vibrator contacts are quite worn, or that the vibrator has been overloaded. If a new fuse blows out at once, there is definitely something wrong in the power supply of the radio. Of course, you should check the set rather than just replace the vibrator, as there may be an overload.

► When the complaint is weak reception or a dead set, try a short piece of wire as an aerial in place of the car antenna. If the wire used as an aerial restores reception to normal, be sure to check (with an ohmmeter) for possible shorts between the lead-in and its shielding, and between the antenna and the car body. Remember—the car body is the ground system, and

the antenna must be insulated from it. ► If your initial check shows that the trouble is within the radio, not caused by the installation or by interference, it will be necessary to take the radio out.

► Be very careful to notice where all control cables and leads fasten to the receiver when you take it out. Make a sketch of the cables and their location. This is particularly important if you do not have the installation instructions for the particular radio at hand, as of course you will leave the control head and antenna on the car and must connect the controls and cables properly when the set is re-installed.

It is a good idea to keep installation instructions on any set you install, so you will have them available for your service work later, on similar models or the same set.

### **WORKING ON THE SET AT THE WORKBENCH**

The equipment needed at the workbench has already been described. With such equipment available, you can connect the set on your bench and work on it exactly as you would on any other radio having a similar complaint. You can use the same test methods and the same test equipment as you would use on an a.c. receiver having a power transformer.

► The extreme compactness of an auto set will make it slightly more difficult to trace circuits and locate parts than in the standard receiver. Unless you have a layout diagram of the set, you will have to use the tube socket terminals and an ohmmeter for identifying connections and parts.

When using an ohmmeter to check the B supply circuit of a set using a synchronous vibrator, remember you may get false readings to chassis through the vibrator. Unplug the vibrator, if possible, or disconnect it

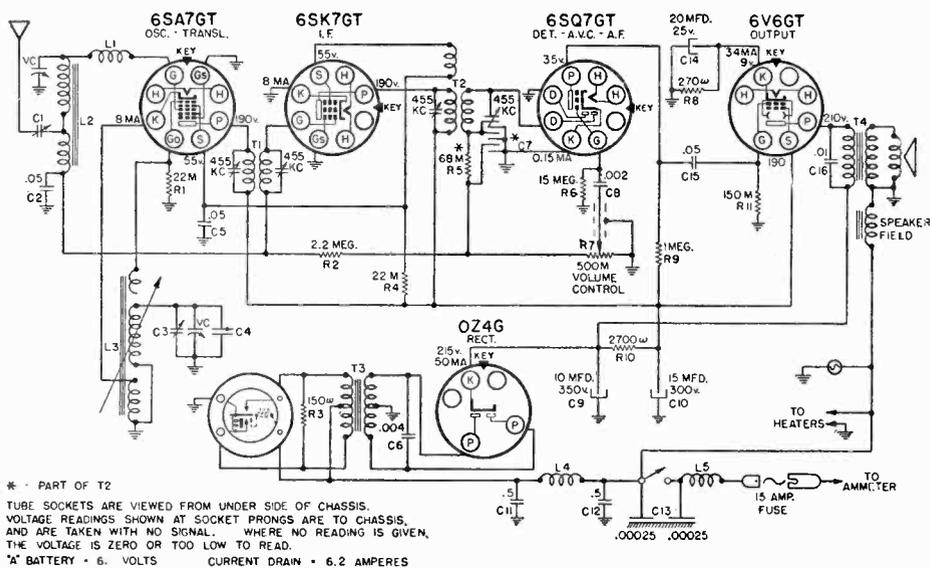


FIG. 35. Wiring diagram of the Silvertone Model 7091, having features found in many auto sets.

from the power transformer when making these tests.

► The metal case about an auto set acts as a shield for the receiver to keep down some of the interference. There will be an unusual number of screws holding lids to the case, and usually spring wiping contactors will be employed between the lids and the case to assure electrical continuity and a tight fit, thus providing an effective shield. After removing the necessary screws, you will frequently have to pry the lid off to get inside. Be sure to replace all these screws when you have finished the servicing, to prevent the possibility of an increased amount of interference.

As has been mentioned, most auto sets have a removable lid over the tubes and the above-chassis parts. After opening this lid, a careful examination of the case will show what is necessary to get underneath the chassis of the receiver. In some instances, the opposite side of the case is a lid which is held on by screws and can be removed. In other in-

stances, the receiver chassis itself comes out of the box.

On some of the older models, you may find that the receiver chassis comes out of the box or case, but some of the filtering equipment in the A battery leads and in the antenna leads remains in the case. In other words, a plug-in arrangement will be used between this filter equipment and the receiver chassis itself. Service information on such receivers is very desirable so the proper terminals in the receiver can all be identified when it is taken from the case.

► A typical auto set diagram is shown in Fig. 35, and the parts layout is shown in Fig. 36. This particular receiver does not use an r.f. stage at the input of the receiver, but does have regeneration introduced in the i.f. stage to increase the sensitivity. The screen grid lead is coupled by a small coil to transformer  $T_2$  in such a manner that regeneration occurs.

Condenser  $C_7$  is worthy of note. This condenser is actually three condensers in a single unit, as it composes

the trimmer for the secondary of  $T_2$ , a by-pass condenser between the secondary return of this transformer and chassis, as well as another by-pass condenser from the end of  $R_5$  to set chassis. The condenser is made up of alternate layers of metal plates and mica insulating strips, and appears to be just a part of the trimmer condenser for the i.f. winding. When you open a transformer of this kind, you will find that this trimmer condenser apparently has four terminals instead of the usual two, but the extra terminals actually go to the other condenser sections which are underneath the actual trimmer.

The resistor  $R_3$ , connected across the primary of the power transformer, helps to keep down the high-voltage surges common in vibrator operation. Notice that this set does not use an r.f. filter in the B supply lead from the cathode of the rectifier tube. This is rather unusual—most auto sets have this feature.

► In addition to having the same troubles as a standard a.c. receiver, auto sets have a few extra troubles caused by the nature of the installation. Better-grade tube sockets must be used, having tight gripping contacts, to prevent the tubes from working loose. Even so, a poor contact can develop at a tube prong due to the excessive vibration to which the set is subjected. This same vibration can jar loose parts which are not firmly mounted and may pull terminals loose if the mounting leads have been pulled too tightly in the original installation of the part in the set.

### SERVICE HINTS

Here are a few additional facts about auto receivers which will prove helpful in servicing these receivers.

► Poor connections must be avoided at all costs. It is extremely important that good soldering be done in auto receivers to prevent terminals from breaking loose or working loose.

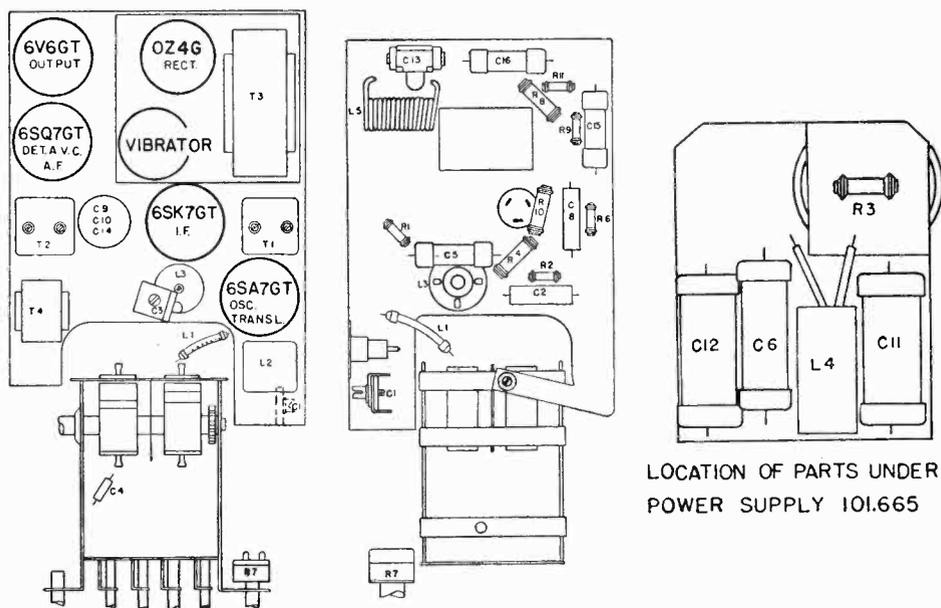


FIG. 36. Parts layout for the set shown in Fig. 35. These layouts are helpful on auto sets, where the crowded layout makes parts identification somewhat more difficult.

► Don't be afraid to jar an auto set when noise is the complaint. Lift it up an inch or so and drop it on your workbench. It will get shocks at least this severe in normal operation in the car, and frequently you will have to duplicate road conditions to localize the trouble.

The normal vibrator noise, which should be audible directly from the vibrator, is a steady hum or buzz. If this changes in pitch or stops and starts, the vibrator is not running steadily and may be either defective or overloaded.

A hissing noise, particularly if reception is weak, usually indicates a defect in the r.f. stage or antenna. The noise is the usual first-detector tube noise. Be sure to check the contacting device to which the antenna cable connects. It may be corroded or the springs may be weakened so that it is not making good contact.

► Oscillation in auto sets is due to the usual causes. However, since these receivers are highly sensitive, very slight defects may cause trouble. The positioning of leads may be quite critical and the alignment will have to be checked carefully. A poor connection somewhere can easily cause oscillation. In those models having regenerative circuits, changes in the tube characteristics may cause excessive regeneration. Another tube may

cure the trouble, or it may be necessary to adjust the position of the feedback coil if such an adjustment is possible.

► Hum in an auto set will not be 60-cycle hum. The frequency of the vibrator is usually around 115 cycles per second (although some vibrators have frequencies of from 85 to 165 cycles per second). Therefore, full-wave operation will give a hum frequency of 230 cycles. This hum frequency is easier to filter than one of a lower frequency, so you will frequently find a resistor used instead of a choke coil in the filter circuit. Incidentally, practically all speaker fields on auto sets are 6-volt fields, and operate in parallel with the tube filaments from the storage battery. This type of field cannot be used as a choke coil.

From this, the hum will be 115 cycles if caused by a faulty rectifier or by grid pick-up, or will be 230 cycles if caused by defective filter condensers.

Cathode-to-heater leakage in the audio tubes will not normally cause hum unless there is some unusual stray-field condition. The filament supply is d.c., not a.c., so leakage will cause bias voltage upsets and thus distortion rather than hum in auto sets.

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## Farm Radio Receivers

Farm radio receivers are not only sources of entertainment—they are a means of getting up-to-the-minute knowledge about weather and crop information and prices, which are vital to the farmer. A simple, reliable radio is desired for this service, so the average farm receiver is relatively straightforward in its design. Some

have short-wave bands, although most are just standard broadcast band receivers.

Of course, the receiver must be sensitive, as the farm may be quite some distance from broadcast stations. It must also have reasonably good selectivity, because there will probably be no strong local stations to over-ride

interference from more distant stations.

► As the average farm is at some distance from broadcast stations, you will usually find a good aerial is needed. A standard antenna of the inverted L type, or one of the modern noise-reducing varieties, should be used—erected as high as possible. Since the aerial is relatively in the open, it must be rugged enough to withstand storms and high winds. Springs, or a weight and pulley arrangement, should be employed if the antenna is to be supported by trees, for trees sway a great deal in bad weather and might easily snap a rigidly-connected aerial. If a wind-charger is used, keep it and the antenna as far apart as possible to minimize interfering noises.

► A good lightning arrester is necessary on a farm radio. A heavy wire should be run from the lightning arrester to an earth ground. This wire should run in a straight line to earth if possible, and the arrester should be installed at the point where the antenna lead-in enters the home. Not only is this good practice—it is a requirement in many communities for fire insurance.

► These farm radio receivers are serviced in the same manner as are other receivers, but their rural location will make some differences in your working practices. For example, you will find it necessary to carry more equipment with you, as it is not economical to go long distances, pick up a radio, bring it back to your shop for some simple repair, and then return it. Ordinary repairs should be made on the spot.

A serviceman working on farm receivers will usually carry with him a multimeter, signal generator, and tube tester. If the receiver requires use of signal-tracing equipment, it is more economical to bring the set to your

shop. To handle such work, servicemen frequently travel over a route, picking up the set one week and delivering it another when they come back over the same territory.

Since power lines will not always be available, your test equipment should be battery-operated, or your service truck must furnish power for its operation. Tube testers represent the greatest problem, as there are few battery-operated types available. However, some 110-volt a.c. tube testers can be converted to use, or are furnished with, a vibrator type converter which operates from a 6-volt storage battery and furnishes 110 volts a.c. for operation of the tube tester.

Many farms have power lines and will use standard a.c. receivers. However, others must depend on battery-operated sets. Many of these will be vibrator powered. Let's consider these types briefly.

► Vibrator-powered farm radios may operate from 32-volt farm lighting systems or from a 6-volt storage battery. The 32-volt systems include a gasoline-engine-driven generator for charging the 32-volt battery, but the 6-volt storage battery must be charged by some other means. Wind chargers have been developed which work reasonably well in most sections of the country. These consist of a wind-mill system driving a generator which is used to charge the 6-volt battery.

The erratic nature of such a charging system means that the farm radio must be rather economical in its power consumption so as not to run down the battery too rapidly. Synchronous vibrators are commonly used, as shown in Fig. 37, to save the filament current required by a rectifier tube. As this figure also shows, economical low-voltage tubes, with filaments in

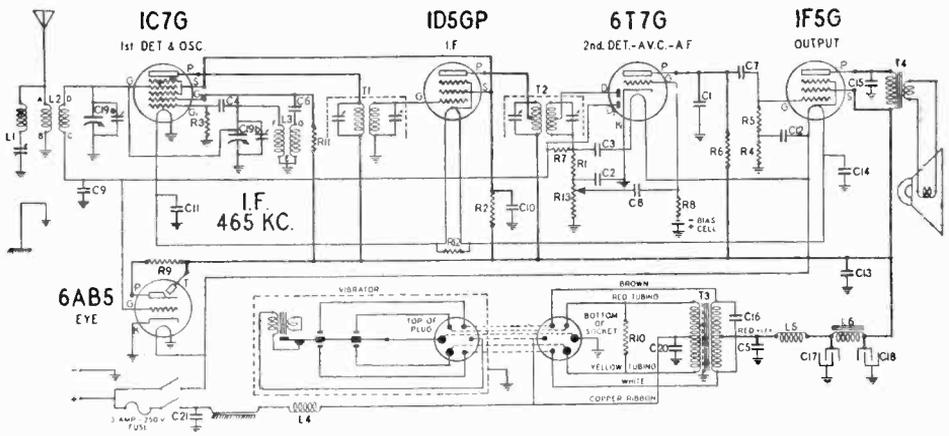


FIG. 37. A typical 6-volt vibrator-powered farm radio.

series, are frequently used to reduce filament power requirements.

► Now for a few service hints:

In a receiver like the one shown in Fig. 37, some of the tube filaments are arranged in series, so you cannot pull out any of the series tubes for circuit-disturbance testing. The set is much like an a.c.-d.c. receiver in this respect. Sets which operate on 32-volt systems also usually have some series-parallel arrangement of tubes. In fact, about the only farm radios which do not have some series-connected tubes are 6-volt sets which use 6-volt tubes throughout. For this reason, these receivers usually require

circuit-disturbance tests very similar to those given a.c.-d.c. receivers.

In using an ohmmeter to check the B supply of any kind of receiver equipped with a synchronous vibrator, you may get puzzling readings caused by connections made through the vibrator. Unplug the vibrator, if possible, or disconnect it from the power transformer when you make ohmmeter measurements on the B supply circuit in such a receiver.

The use of a synchronous vibrator means the storage battery *must* be connected with the correct polarity, so be sure to follow the markings on the leads.

THE N. R. I. COURSE PREPARES YOU TO BECOME A  
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# Lesson Questions

Be sure to number your Answer Sheet 44RH-1.

Place your Student Number on every Answer Sheet.

*Send in your set of answers for this lesson immediately after you finish them, as instructed in the Study Schedule. This will give you the greatest possible benefit from our speedy personal grading service.*

1. What three things are always done to eliminate electrical interference when installing an auto receiver?
2. How would you bond the engine to the fire-wall?
3. What could you do to eliminate a click, heard only when the brakes are applied, even when the car is standing still?
4. Suppose you have to connect an auto set using a synchronous vibrator to a storage battery on your work bench. What test can you make with a voltmeter to determine that the connections are made with the proper polarity?
5. What frequency would you expect hum in an auto set to be: 60 cycles; 115 cycles; 120 cycles; or 230 cycles?
6. Would cathode-to-heater leakage in an audio tube be likely to cause hum in an auto set?
7. Which of the following fuse sizes can be used for an auto set: 1-amp.; 5-amp.; 10-amp.; 15-amp.; 20-amp.; 25-amp.?
8. When ordering a replacement buffer condenser (like  $C_s$  in Fig. 30), what must be specified in order to get the proper replacement?
9. Is the field coil of an electrodynamic auto-set speaker used as a choke coil in the B supply?
10. What precaution is necessary when taking an ohmmeter reading in the plate circuit of a farm receiver using a synchronous vibrator?



## GETTING AHEAD

We have all heard the old proverb, "If wishes were horses, beggars would ride." This is just another way of saying that if wishing could bring success, all men would be successful.

You need only to look around you to see that wishing is not enough. The world is full of failures. But our common sense tells us that these men are not failures because they did not *wish* to succeed. So what *is* the secret of success?

It is *what we do about our wishes* that makes all the difference. The secret is ACTION. Take two men with equal ability and the one who works harder will get ahead faster than the other man.

If one of the two men has less ability than the other, less education, fewer opportunities—but is energetic, active—does something about his problems—he will be more successful than the man who does nothing but wish for success. The men who get to the top and stay there are men of ACTION.

*J. E. Smith*

# RECEIVER REVITALIZATION

## TUBE TESTERS

45RH-1



**NATIONAL RADIO INSTITUTE**

**WASHINGTON, D. C.**

**ESTABLISHED 1914**

# STUDY SCHEDULE No. 45

For each study step, read the assigned pages first at your usual speed, then reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind. Study each other step in this same way.

- 1. Revitalization and Radio Measurements . . . . .Pages 1-3  
The meaning of revitalization and the need for certain measurements for most accurate results are the subjects of this section.
- 2. Sensitivity Measurements . . . . .Pages 3-12  
The most used revitalization steps are those to restore sensitivity. Here are the methods of checking sensitivity to determine what it is, and to determine how much improvement has been made. Laboratory procedures and methods using a signal tracer are covered.
- 3. Selectivity and Fidelity Measurements . . . . .Pages 13-18  
Both laboratory and service methods of measurement are covered. The short-cuts used by servicemen are particularly stressed.
- 4. Miscellaneous Measurements . . . . .Pages 19-21  
How to check: the hum level; power consumption; frequency shift; for dead spots; image rejection; and noise.
- 5. Receiver and Part Revitalization . . . . .Pages 22-27  
The methods of overhauling a radio, with particular attention to the steps which pep up receivers.
- 6. Tube Testers . . . . .Pages 28-36  
This section covers the practical types of testers developed for servicemen. The methods of testing for tube quality, the variations permitted, special tests, and the proper tube-testing routine are all covered.
- 7. Answer Lesson Questions, and Mail your answers to N. R. I.
- 8. Start Studying the Next Lesson.

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# RECEIVER REVITALIZATION

## TUBE TESTERS

### Revitalization and Radio Measurements

**R**EVITALIZATION is the process of bringing a receiver back as nearly as possible to the condition it was in when new. Strictly speaking, any repair procedure could be called a revitalization, but the term has come to mean the process of overhauling a radio receiver.

If a receiver is normal except for some one particular defect (excessive hum, oscillation, etc.), you would of course concentrate on finding and remedying the cause of the defective operation. In most cases, troubles of this kind will be easily localized by the normal servicing procedures given elsewhere in your Course. Then, at some time during the service job, you usually will carry out two steps which are revitalization procedures, in that the receiver operation is being improved. These steps are: 1, test the tubes and recommend the replacement of weak ones; and 2, check the alignment and realign if necessary. These revitalizing steps are so much a part of the service job that most servicemen do not look upon them as revitalization. Instead, they consider revitalization as the procedure followed when servicing a radio which is generally run down and insensitive, but which exhibits no specific defect in its operation.

Using this meaning of revitalization, there are two methods of approach to the problem. One is a brute force method of repairing or replacing every suspected part. While

this will usually improve the over-all receiver performance and will probably result in the replacement of any defective parts, it is generally a lengthy and expensive process, and it is not certain to give the results you want. A more scientific approach is to take measurements to determine just what the receiver characteristics actually are. Comparing the results of these measurements with the ratings of the receiver will give you a fairly good idea of the actual condition of the set and indicate what you must do to improve it. Then, after making each repair or adjustment, you can check your progress by repeating the measurements to see how much improvement has been made.

Obviously, you'll be more certain of results if you follow this second method which we are going to study in detail in this lesson. But before we take up the actual procedures of revitalization, let's learn something about the measurements which can be made on radio receivers.

#### THE IMPORTANCE OF RADIO MEASUREMENTS

Measurements of radio characteristics were at one time limited entirely to laboratories. The expensive equipment and the exact, time-consuming, scientific methods needed caused radio servicemen to "steer clear." Today, however, the more widespread use of vacuum tube volt-

meters, the cathode ray oscilloscope, signal tracers, etc., has placed the necessary equipment in the larger service shops. Further, set manufacturers have begun to release measurement data made with such service equipment, so the problem of getting a useful measure of sensitivity, selectivity, fidelity, etc., has been greatly simplified. As a result, many servicemen are using these measurements as a means of determining radio characteristics more exactly.

► Ordinarily, the only guides a serviceman has to the performance a particular radio receiver should exhibit are data which may be a part of the service information on the set and the statements of the set owner. Of course, it is possible to judge the performance of some particular model after having worked on several identical receivers. However, this experience may not be too helpful when you meet a different model, for different types of radios vary considerably in their characteristics.

Contrary to popular opinion, the number of tubes in a receiver shows little about its performance. It is quite possible, for example, for a 6- or 7-tube receiver to have far greater sensitivity than a receiver with 12 or 15 tubes. Even receivers with the same number of tubes may be of different design and therefore have far different performances. Thus, a 5-tube a.c. operated receiver with a power transformer may have a far greater output than a 5-tube a.c.-d.c. or a 5-tube battery operated set, while these receivers may have better selectivity or better sensitivity characteristics than the a.c. receiver.

► Today, it is possible for the radio engineer to build into the receiver any desired amount of selectivity, sensitivity, and fidelity, within the limits of cost and the necessary compromises between these factors. The engineer,

in designing a set, takes into consideration the use to which the receiver is to be put and the sales features desired. A communications set, for instance, must have a high sensitivity and must be extremely selective, but it does not need much in the way of fidelity. (Frequencies above 3500 cycles are not needed for voice or code transmission, and cutting them off gives less noise and better selectivity.) On the other hand, a high-fidelity receiver is usually made relatively insensitive, with rather poor selectivity, but with very good fidelity or tone quality. The average broadcast set is some compromise in between these two extremes.

The fact that such wide variations exist in the designed performances of radios make it desirable to use fairly accurate measurements if you are going to revitalize receivers which have lost their pep. You must have some clear way of comparing the actual performance of a set with the performance it was designed to have—otherwise, you may waste a great deal of time attempting to give a receiver characteristics which were never built into it and which it cannot have without extensive modifications. Only actual measurements of performance will let you make such a comparison.

Comparison of actual performance with the rated value by means of measurements will be of particular help to you when a set owner complains about the lack of selectivity, fidelity, or sensitivity in his radio. Measurements will show you if the complaint is really justified and if the condition can be corrected at reasonable expense—or if it would be better (and cheaper) for the set owner to buy a new radio. And, finally, measurements are extremely useful as checks on your work, since they will show definitely how successful you have been in your attempts to bring

the set back to "good as new" condition.

Naturally, as a serviceman, you are not going to have the equipment or the time to make laboratory-type measurements. In recognition of this fact, the modern practice is for the set manufacturer to take laboratory measurements in a standardized manner so as to compare receiver characteristics, then to give service data measurements taken with service-type instruments. However, even though you may never make laboratory-type measurements yourself, it is desirable for you to know how they are made, what they mean, and how close service-type measurements are to them in results. Then you will be able to interpret more clearly the results you get from measurements made with service instruments.

► In this lesson we shall give a brief description of the manner in which the set manufacturers make their measurements, so you can understand the exact meaning of the ratings you may find in radio service data. Then,

we shall show how simplified measurements can be made with service equipment, which will permit an approximation of these results and give you some means of comparison.

Of course, these measurements will not be needed so very often in your service work, as by far the greatest number of troubles will be straightforward cases, easily and directly solvable by the professional servicing techniques you have already studied. Furthermore, as your experience grows, you will develop the knack of judging receiver response, so you will need to make measurements only in those comparatively rare cases where ordinary methods fall down.

Among the measurable response characteristics of a receiver are: sensitivity; selectivity; fidelity; hum level; noise level; frequency shifts; image rejection; dead spots; and receiver power consumption. Let's see just how the measurements of these characteristics can be made.

---

## Sensitivity Measurements

The receiver sensitivity is a measure of its ability to pick up and properly reproduce weak signals from low powered or distant stations. A loss of sensitivity first shows up as an inability to pick up weak distant stations, and then progresses to weak reception even from locals.

We could measure the sensitivity of a receiver by determining the weakest possible signal which would give even the slightest output from the receiver, if we wished to do so. However, a more useful value is found by measuring the input signal which will give some definite rated output. This lat-

ter measurement is far more valuable because it gives some idea of how weak a signal can be and still be amplified enough by the set to give enjoyable reception.

### LABORATORY SENSITIVITY MEASUREMENTS

Let us first consider receivers designed to operate from regular antennas (not loop aerials). To measure the sensitivity of the receiver, we must be able to measure both the signal voltage fed into it and the output voltage of the receiver.

It is not practical to radiate a sig-

nal and depend on an antenna for pickup, as there is a possibility of the set picking up the signal directly. Just feeding from the signal generator directly into the antenna-ground terminals will upset the input circuit of the receiver, due to the signal generator loading effects. The usual way of solving this problem is to feed a measured signal voltage into the radio receiver through what is called a dummy antenna. This dummy antenna is a combination of parts which

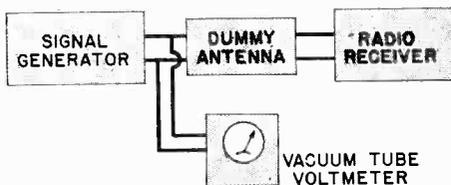


FIG. 1. The connections for making a sensitivity check.

simulates the inductance, capacity, and resistance of an ordinary receiving antenna. When the measured signal is fed through this device, the receiver "feels" an antenna at its input and will react as if it were connected to an average antenna.

The input signal must be measured with accuracy if the sensitivity measurement is to mean anything. One method of making this measurement is to use a sensitive vacuum tube voltmeter, as shown in Fig. 1. However, modern standard\* signal generators having a controlled output are used in most laboratories. In these, a built-in vacuum tube voltmeter is used to adjust the signal level fed into an attenuating network. The controls of the attenuator are marked directly in microvolts output. (A microvolt is one-millionth of a volt.)

\* Signal generators of the "standard" type are so named because they are made to laboratory specifications and have extremely accurate and reliable output signals.

Usually, there will be two controls, one reading from zero to 10, while the other is a switch type control varying the output in steps of ten. (It reads 1, 10, 100, 1000, and 10,000, and the output is found by multiplying the two dial readings.) This permits the output to be varied from a fraction of 1 microvolt up to 100,000 microvolts on most standard signal generators.

Fig. 2 shows the connections for a standard signal generator, and also shows the components of a standard dummy antenna. As we said, this particular arrangement of parts has been chosen to simulate the effects of an antenna on the receiver input, so the measurement will be in terms of signal strength fed into an average antenna.

**The Dummy Load.** To measure the output, the loudspeaker voice coil is disconnected and a resistor is used in its place, as shown in Fig. 2. The resistor  $R$  is called a dummy load, and is chosen to have the same ohmic

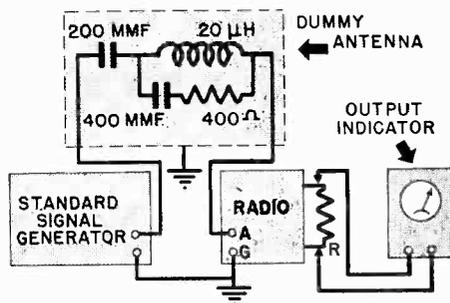
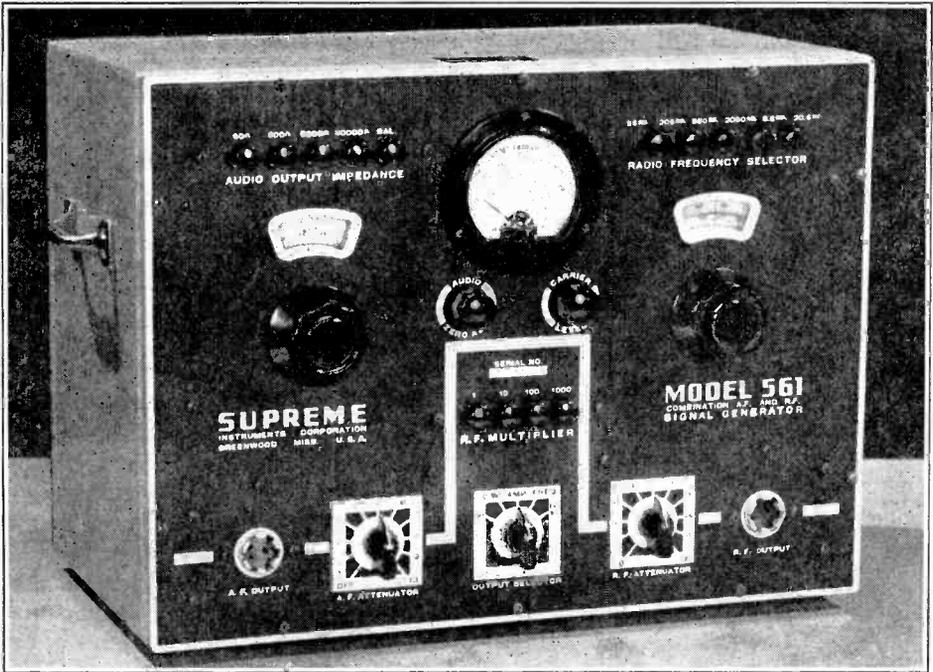


FIG. 2. Standard signal generators have built-in level indicators, so they do not need an external v.t.v.m. for measuring the receiver input. The parts values for a dummy antenna are given here.

value as the voice coil impedance. The voltage across this resistor is measured with an accurate a.c. voltmeter. Once we know the voltage and the resistance, we can determine the power output. ( $P = E^2 \div R$ )

► An equally acceptable way to measure the output is to place a



*Courtesy Supreme Inst. Corp.*

A standard signal generator intended for the serviceman. Frequency ranges are selected by push-buttons at the upper right. When the carrier level control is adjusted to give an indication at the point marked on the meter, the output in microvolts will be indicated by the output control setting. The r.f. attenuator reading is multiplied by the r.f. multiplier push-button being used. There is a built-in variable frequency audio oscillator, and the percentage of modulation is indicated on the meter.

resistance in the plate circuit of the output tube or tubes, as shown in Fig. 3. Leaving the secondary winding of the output transformer open makes the primary impedance become practically infinite, so the dummy load resistor  $R$  is chosen to give the proper load impedance for the tube used. The advantage of using the circuit shown in Fig. 3 lies in the fact that the voltmeter need not be so sensitive. The resistance value of  $R$  in Fig. 3 is much higher than that of  $R$  in Fig. 2, and a higher voltage is developed across it for the same power output.

**The Standard Output.** The standard output power used for sensitivity measurements is .05 watt (50 milliwatts) for a set which has an undistorted output power of 1 watt or

less, and is .5 watt for a receiver with an undistorted output power of 1 watt or more.

Thus, on a low-powered set, we adjust the input signal until the output indicator used with the set shows .05 watt output; similarly, with a high-powered set, we adjust the input until an output of .5 watt is indicated. We can then tell from the amount of input necessary to produce this standard output just how sensitive the set is.

Assuming we are using the 50 milliwatt output level, which is the one most often used, you can see that the voltage across a low resistance will be small. (This voltage is found from the formula,  $E = \sqrt{P \times R}$ .) Thus, if the resistance has a value of 5 ohms and the power is .05 watt, the voltage

is only .5 volt. If the resistance is higher, the voltage will be higher, so the method illustrated in Fig. 3 is somewhat more desirable.

Voltages equivalent to standard output (.05 watt) for various load resistance values are given in Fig. 4.

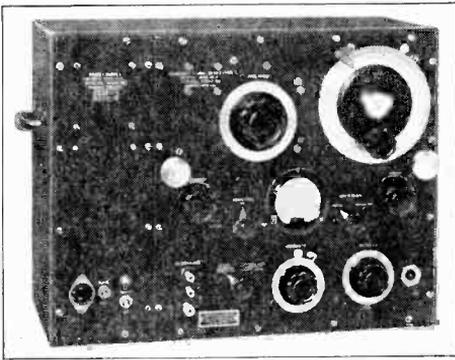
**Sensitivity Variations.** When we make measurements with the standard set-up of Fig. 2, what frequency should we use? You probably know from tuning your radio receiver that

in sensitivity over the broadcast band is given in Fig. 5.

Thus, to make a really complete study of the sensitivity of a receiver, it is necessary to start at one end of the band, tune the receiver and generator to the same frequency, and note the microvolts input necessary to produce the standard output. The measurement must then be repeated at other points throughout the frequency range of the set.

► Of course, for comparison purposes, we do not need to measure the sensitivity over the entire band as long as we know what the sensitivity should be at some particular frequency and make our measurements at that same frequency.

**Signal Generator Modulation.** As the output voltage is an audio voltage, the signal generator must be modulated a certain exact amount at a certain frequency for the output measurements to be reliable. Standard measurements are made with the signal generator modulated 30% by a

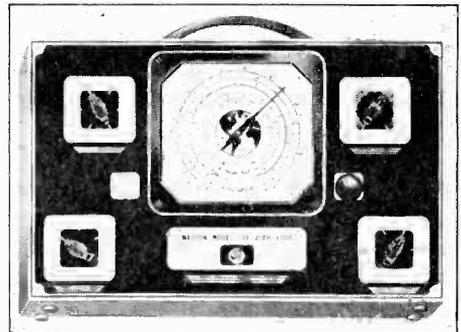


*Courtesy General Radio Co.*

A standard laboratory type signal generator. The output is calibrated in microvolts and the percentage of modulation is variable and indicated on the meter.

programs from different stations on the dial do not all come in with exactly the same volume and power. Partly this is caused by differences in the distances between the stations and the receiver and differences in the powers of the stations, but another important cause is the fact that the sensitivity of a receiver is not the same at all frequencies. Therefore, as we take measurements over a band of frequencies, we will find that different amounts of input will produce the standard output.

For this reason, we cannot say that a receiver has a certain sensitivity without specifying the frequency at which the sensitivity was measured. A typical curve showing the variation



*Courtesy Weston*

Another service-type signal generator. The output controls are calibrated in microvolts, and the output level is held constant by a built-in automatic level control circuit. The audio modulation is a 400-cycle signal, and the percentage of modulation is fixed at 50%. This is above the 30% value used in standard sensitivity measurements. Radios would have an apparent sensitivity better than they actually have if this signal generator is used. Otherwise, this is an excellent general purpose signal generator.

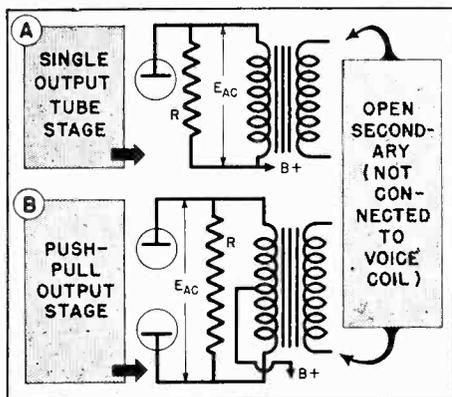


FIG. 3. These connections are used when the dummy load resistor is connected in the plate circuit of single-ended (A) and push-pull (B) output stages.

400-cycle audio signal. The modulation percentage must be accurately adjusted. Most laboratory-type standard signal generators have a means of varying modulation percentage and use a meter to indicate the exact percentage used.

**Loop Aerials.** When the receiver has a loop aerial, it is not possible to substitute a dummy antenna for the loop. When measurements are made on such a receiver, one method is to use a carefully designed radiating loop antenna, connected to the signal generator and then arranged a certain distance from the receiver loop, as shown in Fig. 6A. From the dimensions of the two loop antennas and their distance apart, it is possible to calculate the energy fed into the receiver loop and, from this, the input to the receiver.

► Another way of accomplishing the same result is shown in Fig. 6B. A small resistor is inserted in series with the receiver loop aerial, and the signal generator output is applied across it. The current flow through the resistor is measured by the meter  $M$ . From this, the voltage drop across the resistor—that is, the signal volt-

age introduced into the loop circuit—is calculated.

## SENSITIVITY TESTS WITH SERVICE EQUIPMENT

The serviceman with a great amount of practical experience rarely needs to make sensitivity measurements. He may put a set on the test bench and simply try tuning in a number of stations on the band. From experience, he knows what other sets of various types will do in his particular location on the same antenna, and he judges the performance of the radio on which he is working by the way it acts. If he knows that certain distant stations can be tuned in only by a receiver having good sensitivity, and finds he can pick up these stations on the set being checked, he knows that the sensitivity of the radio he is testing is pretty good.

He may also judge the sensitivity by the action obtained when he tunes off resonance. Normally, he would

VOLTAGES FOR STANDARD OUTPUT OF .05 WATT			
Dummy Load	Volts	Dummy Load	Volts
1	.22	1000	7.1
2	.32	1500	8.7
3	.39	2000	10.0
4	.45	2500	11.2
5	.50	3000	12.3
6	.55	3500	13.2
7	.59	4000	14.1
8	.63	4500	15.0
9	.67	5000	15.8
10	.70	5500	16.6
11	.74	6000	17.3
12	.78	6500	18.0
13	.81	7000	18.7
14	.84	7500	19.4
15	.87	8000	20.0

FIG. 4. This table gives the voltages that indicate a .05-watt output across various load resistance values.

expect the noise level to increase greatly as the set is tuned away from a station signal and the a.v.c. action increases the set sensitivity to maximum. If he finds that he doesn't get much of a "roar" as he tunes off resonance, he knows the set does not have very much sensitivity. If the receiver is a type which should be reasonably sensitive, he would then go to work to discover the reasons for the lack of sensitivity.

A serviceman with less experience is unable to judge radio receivers this way, and even an expert can be fooled occasionally, so a more exact procedure is desirable.

Better and better equipment is be-

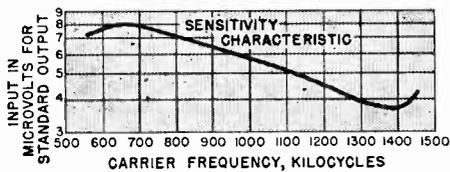


FIG. 5. The sensitivity varies over the tuning band, as shown here. The maximum sensitivity is indicated by the least input which will give the standard output, so this particular set is most sensitive near 1400 kc. Sometimes the readings near the center of the tuning range are given as the average over-all sensitivity value.

ing made available to servicemen, so today many have standard signal generators which they use in their service shops for alignment purposes. If you have such a standard signal generator (with an output accurately calibrated in microvolts), you can measure receiver sensitivity by the same method used by the set manufacturer.

### SENSITIVITY TESTS WITH SIGNAL TRACERS

Checking the set sensitivity in the manner just described will give the actual over-all sensitivity of the receiver. Comparing this with the manufacturer's ratings gives valu-

able information about the over-all gain of the receiver. If there is any wide difference between the measured sensitivity of the receiver and its proper value, trouble in the radio is indicated. Unfortunately this indication does not tell which *stage* the trouble is in, so the entire receiver must be carefully checked, using the methods given elsewhere.

The development of signal tracing equipment has, to a great extent, permitted quick localization of the defective stage. A signal tracer cannot be depended on to indicate accurately the *number* of microvolts of signal at any point in the radio. However, it can be used to measure the *ratio* of the output to the input of any desired stage. The ratio of the output to the input is a measure of the gain or amplification of the stage and is accurately determined by the signal tracer, for any errors caused by the tracer will be in both measurements and will cancel each other. Since the receiver sensitivity depends on the amplification obtained in each stage, this measurement of stage gain is an indirect measure of sensitivity. Even more important, if you know the gain which should exist in each stage, you can localize the defective stage (or stages) by making stage-by-stage measurements with the signal tracer.

As a check of the over-all gain only indicates that a defect exists but does not localize the trouble, you can see that the signal tracer, with its ability to localize the defective stage, is more valuable in service work.

**Manufacturers' Gain Values.** Before stage gain readings can mean much, we must know what to expect from each particular stage. In recent years, receiver manufacturers have cooperated in furnishing this data to servicemen. Some give the gain data directly on the diagram, while others

show it in a table apart from the diagram.

Fig. 7 is an example of the diagram system. Notice the gain values given above the schematic. As you can see, the input gain is figured at 600 kc. At that frequency, the gain from the antenna to the first tube grid is 2. If we have, for example, a 30-microvolt 600 kilocycle signal between the antenna and ground (chassis ground) the circuit should give us two times this signal level (a 60-microvolt sig-

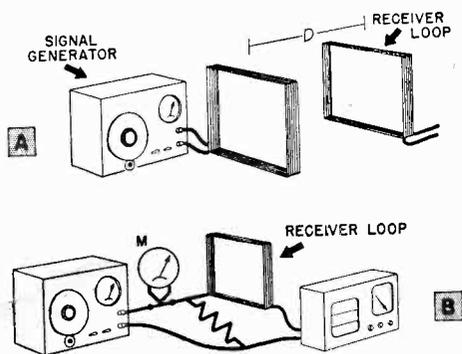


FIG. 6. Two ways of feeding a signal into a loop antenna. The practical serviceman would use the first method, winding a loop of 5 or 10 turns to connect to his signal generator. He won't know what the induced voltage is, so cannot determine the  $Q$  of the loop, but using a signal tracer will permit measurements from the loop onward.

nal) between the first tube grid and chassis ground, if everything is normal.

With the 600-kilocycle signal still tuned in, there is a conversion gain of 60 in the first detector-oscillator tube. This means that if we measure the 600-kilocycle signal fed to the grid of the 12SA7 tube, then measure the 455 kc. i.f. signal developed between this tube plate and the chassis ground, we should have an increase of about 60. Thus, if we continue with our example and use a 60-microvolt signal as the input to the 12SA7 tube, we should have 60 times 60 or 3600 microvolts of

i.f. signal between the plate of this tube and ground.

► Notice, however, that the gain of this tube will vary with the grid bias, which is controlled by the a.v.c. circuit. If we allow the normal a.v.c. action to occur, strong signals will increase the bias and hence reduce the gain. For the gain measurement to mean anything, we must block the a.v.c. action.

Therefore, when any measurements are made in the r.f. and i.f. sections of a receiver, it is necessary to disconnect the a.v.c. circuit from the source of a.v.c. voltage and substitute in its place a certain value of fixed bias. The fixed bias value must be that recommended by the set manufacturer—usually 3 volts. To make measurements on the receiver shown in Fig. 7, you should disconnect resistor  $R_2$  (either end) and place a 3-volt bias from a battery between the grid return lead and chassis as shown by the dotted lines in Fig. 7. Of course, you must observe the proper polarity in connecting this battery.

► Continuing with our example, the next gain measurement is taken from the plate of the 12SA7 tube to the grid of the 12SK7 tube. This measures the gain of the i.f. transformer, which is given as .5 (the same as  $\frac{1}{2}$ ). This means that the signal in the secondary of the transformer is only half as strong as that in the primary; in other words, we actually get a loss in this transformer. Thus, if we have 3600 microvolts between the plate of the 12SA7 and the chassis, we can get only one-half this, or 1800 microvolts, at the grid of the 12SK7 tube.

In the next stage, there is a gain of 100 between the grid and the plate of the i.f. amplifier, then a loss of one-half in the second i.f. transformer.

There is a loss (not shown on the diagram) in the second detector too.

The audio output of the detector depends on the percentage of modulation of the input signal.

Following the audio signal further, we find a gain of 40 in the triode section of the 12SQ7 tube. (Notice that in making the measurement on the audio tube stage we are comparing the signal levels found at its grid and plate terminals, and are measuring the 400-cycle audio modulation signal from the signal generator.) Finally, there is an audio gain of 10 between the grid and plate of the 50L6 output tube.

**Using Signal Tracers.** In making gain measurements with a signal tracer, you do not worry about the actual *amount* of microvolts. Instead, you are interested in the *ratio* between the two points where you make your measurement. You can make your measurements either with a signal tracer which gives a meter reading or with one which uses a magic-eye indicator.

If you use the second type of tracer, you will determine the gain ratio from the control settings. First, connect the signal tracer probe to the input side of the stage, feed in a signal, and adjust the controls until the eye closes. Make a note of the control settings which close the eye. Then, move the probe to the output side of the stage and adjust the controls until the eye is again closed. Dividing the second gain control reading by the first will give the relative increase in gain.

If your signal tracer indicates the signal input level on a meter, measure the level at both the input and output of the stage, keeping the tracer gain control at the same point. Dividing the second meter reading by the first will give the gain of the stage.

A vacuum tube voltmeter could be used in the same manner, but there is considerable chance of the vacuum

tube voltmeter reading stray voltages instead of the signal. For this reason, a tuned signal tracer, which can be made to select the proper signal, is the more desirable instrument.

► With either signal tracer, a signal generator is used as the signal source. It does not have to have a calibrated output voltage control as you depend on the signal tracer for the signal ratios. A dummy antenna should be used, however, when checking the gain of the preselector. If the set uses a loop antenna, just wind a four- or five-turn loop of wire and connect it to the output of the signal generator. Bring this loop near the set loop. You can't measure the preselector gain, but all other gain values can be obtained.

**Gain Variations.** The readings given by the manufacturer in his service information are average readings for the particular set type. These readings will not necessarily apply to any other receiver put out by the same manufacturer. When the gain data is given on the diagram, you can expect variations of as much as 20% in either direction. Variation in parts values can easily cause this much change.

If readings vary by more than 20%, then there is probably some defect in the set. If all readings are below normal, the set may be completely out of alignment, or the over-all sensitivity may be low.

More generally, a defect in a single circuit will be the cause of below-normal gain. In this case, you can expect most of the readings to be normal except in the one affected stage, where the reading will be very low.

► Don't be surprised at readings somewhat above normal, for better-than-average tubes, or coils with exceptionally high Q factors, will give increased gain. On the other hand, if the gain is very high and the re-



ceiver has a tendency to go into oscillation, look for circuit faults causing regeneration.

## GAIN DATA, AVERAGE VALUES

R. F. SECTIONS	Gain	
	Min.	Max.
Ant. to 1st grid.....	2	10
Ant. to 1st grid, auto sets.	10	50
R. F. Amp., supers, broadcast.....	10	40
R. F. Amp., t. r. f., broadcast.....	40	100
R. F. Amp., supers, short-wave.....	5	25
<b>MIXER SECTION</b>		
Converter grid to 1st i. f. grid (single i. f. stage)...	30	60
Converter grid to 1st i. f. grid (2-stage i. f. amp.)...	5	30
<b>I. F. AMPLIFIER SECTION</b>		
I. F. stage (single i. f.)....	40	180
I. F. stage (2-stage i. f.) (per stage).....	5	30
<b>DETECTOR SECTION</b>		
Biased det., 57, 6J7, 6C6, etc. (depends on % modulation).....	5	40
Grid leak det., square law.	5	50
Diode detectors (a loss) (depends upon % modulation).....	.2	.5
<b>AUDIO AMPLIFIERS</b>		
Triodes (low gain).....	5	14
Triodes (high gain).....	22	50
Pentodes.....	50	150
<b>POWER OUTPUT TUBES</b>		
Triodes.....	2	3
Pentodes and beam.....	6	20

FIG. 8. Average amounts of gain found in radio receivers.

**Average Gain Values.** When gain data on the receiver on which you are working is not available, you must use tables of average gain values. The gain to be expected from any stage depends on the type of receiver and on the number of stages.

The table in Fig. 8 gives the maximum and minimum values found by analyzing the information furnished by a number of manufacturers. It is quite possible that some receivers may have gain values above or below these averages, but in general you can expect most receivers to fall within these limits.

The possible variations here are quite wide. For example, the gain in the r.f. section from the antenna to the first grid in auto sets may be anywhere from 10 to 50. If you get a reading near the minimum value, you won't know whether this is natural for the receiver or whether the gain for this particular section should be near the maximum and is actually far below normal. You will have to be guided in cases like this by the results you obtain in the rest of your gain measurements and by the performance of the receiver.

If the receiver has less sensitivity than you would expect, yet all the readings are within the average limits, probably the loss of sensitivity is an over-all condition and the stage gains throughout are below normal. On the other hand, if the gain of some one stage is low but the other stages all give gain near the maximum, it would be logical to suspect the stage where the readings are low.

In general, allow a wider variation in the readings of the gain for the r.f., mixer, and detector stages than for the other stages.

# Selectivity and Fidelity Measurements

We will treat selectivity and fidelity together, since the selectivity of a receiver has a bearing on its over-all fidelity.

Selectivity is a measure of the ability of a radio to select the desired signal and to reject others on adjacent channels. If the set is not selective, interference is certain to exist between stations on adjacent channels if the stations are powerful enough. On the other hand, if the set is too sharply selective, the higher side band frequencies of the desired channel will be cut out, and the fidelity of the receiver will be affected.

## LABORATORY SELECTIVITY MEASUREMENTS

It is possible to indicate selectivity by plotting resonance curves which show the amount of amplification or gain at the resonant frequency and the gain at frequencies off resonance. However, since the gain of different receivers is not the same, it is difficult to compare receivers by using curves of this sort. As we are interested in how much *better* the response is at resonance compared with the response off resonance, it is standard practice to draw curves showing this ratio. Direct comparisons between receivers can be made from such curves.

To find the data needed to plot the curve, laboratory engineers use the same set-up used for sensitivity measurements in Fig. 2. Of course, the manner of using the equipment is somewhat different. The receiver dial and the signal generator are tuned to some frequency (say 600 kc.), and the microvolts input needed to give the standard output is determined. Let us suppose this input is 10 microvolts.

Then, the receiver dial is left at the same setting but the signal generator dial is rotated 10 kc. away from resonance. The signal generator output is turned up until the receiver output meter indicates the same output power as before. Since the receiver is not tuned to the same frequency as the generator, a greater input is necessary to force the signal through the radio.

Suppose it now takes 5000 microvolts input to give the standard output. Dividing the microvolts input for the off-resonance frequency by the microvolts input at the resonant frequency gives us a ratio number (called the "signal ratio") that is a measure of selectivity. ( $5000 \div 10 = 500$  in this case.) For comparison, engineers consider a set to have good selectivity if it takes 1000 times as much voltage to force a signal 10 kc. off-resonance through the radio as it does a signal to which the radio is tuned. Signal ratios of 100 to 1000 are considered fair, while a ratio of 10,000 represents excellent selectivity.

► To complete his data, the engineer continues in 10-kc. steps on each side of resonance, carrying out the readings over a range of 30 to 50 kc. on each side of resonance, and computing the signal ratio for each frequency. The ratio at each frequency is then plotted to form what is called a selectivity curve.

Fig. 9 shows several typical selectivity curves. The curve marked A-A shows fair selectivity. The broad dotted curve B-B is an example of poor selectivity, as the signal ratio for signals 10 kc. off resonance is less than 10. The curve C-C represents excellent selectivity. However, the extreme sharpness of this curve near the resonant frequency-setting indi-

cates that the fidelity of the receiver will be poor.

► The ideal curve for both good selectivity and good fidelity is shaped like the shaded area of Fig. 9. It has the selectivity of the curve *C-C* at the off-resonant frequencies, but has a broad flat "nose" around the resonant frequency, so that approximately equal amplification will be given the resonant frequency and its adjacent side-bands. Comparing this area with the other curves, you can see that the curve *B-B* indicates reasonably good fidelity, as the discrimination shown by this curve against frequencies 5 or 6 kilocycles away from resonance is not great. However, as you just learned, curve *B-B* also indicates very poor selectivity. Thus, selectivity and high fidelity are not apt to be found in the same receiver, unless the receiver uses some form of band-pass tuning to give a response like the relatively square-shaped ideal.

Usually, selectivity curves are taken at several frequencies over the band. Like sensitivity, selectivity varies at different frequencies. A typical curve showing the selectivity at 600 kilocycles compared to that at 1400 kilocycles is shown in Fig. 10.

### SERVICE TESTS OF SELECTIVITY

As far as the serviceman is concerned, the only way in which selectivity measurements actually can be taken is by the laboratory method just described. Hence, only servicemen with standard type signal generators are able to check selectivity accurately.

However, extremely accurate selectivity measurements are rarely necessary. Anything which affects the selectivity of a receiver greatly will also affect the sensitivity and other

characteristics at the same time, and the trouble can be run down by investigating these other effects. If the receiver is badly out of alignment, the selectivity is reduced and the sensitivity will be very low. If the receiver is too sharply aligned, the fidelity will suffer and the sensitivity will be above normal. (The selectivity may also go up because of regeneration.) Of course, after you have quite a bit of practical experience, you will become adept at judging the selectivity. The practical serviceman tunes in some medium distant station, and then tunes away from the resonant point to determine how many dial degrees can be tuned through before the signal fades out. The "spread" on the dial is a rough indication of the selectivity of the set. If the station is tuned out by a small dial movement, the selectivity is usually good.

### LABORATORY FIDELITY MEASUREMENTS

The fidelity of a receiver is a measure of its ability to reproduce exactly the modulation transmitted by the broadcast station. The ideal receiver would amplify all desired frequencies equally and would not introduce wave-distorting harmonics.

The audio amplifier is primarily responsible for the receiver fidelity characteristics, although the tuned circuits in the r.f. and i.f. stages may affect the high-frequency response. Theoretically, we need equal amplification of all frequencies from 30 cycles to perhaps 15,000 cycles to reproduce music with high fidelity. The response range of the average receiver is far more limited than this—a reasonably flat response from 150 cycles to 4000 or 5000 cycles is about all we will usually find.

To measure the over-all fidelity re-

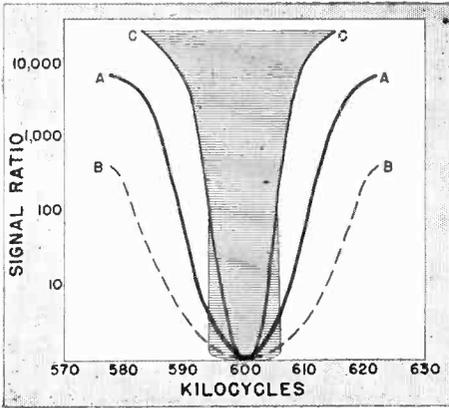


FIG. 9. Typical selectivity curves, showing excellent, fair, and poor selectivity. The shaded area shows the ideal, representing excellent selectivity without sideband cutting. The signal ratio is the ratio of the input off resonance to that at resonance, both giving the standard output.

response of the receiver, the same basic set-up as that shown in Fig. 2 is used. The only additional equipment needed is a variable audio oscillator, which is used to modulate the standard signal generator.

To get the response characteristic, the audio oscillator is first adjusted to produce a signal of 400 cycles, and the modulation percentage is adjusted to 30%. Then, with the receiver dial and the signal generator tuned to, say, 600 kilocycles, the signal generator output is adjusted to give some convenient output indicator reading. (This need not be the standard output value—just some convenient reading.)

This reading, obtained with a modulation frequency of 400 cycles, is our reference value. The audio signal generator is now varied to other audio frequencies, such as 30, 40, 50, 100, 1000, 3000, 5000, 7000, 10,000, 12,000 and 15,000 cycles. The exact frequencies at which the readings are taken do not greatly matter, as long as points over the complete range of the receiver are used.

At each of the new audio modula-

tion frequencies, the percentage of modulation is adjusted to 30%, but the signal generator output controls are left alone, as the same r.f. frequency is being used. The new output meter reading is noted at each of these frequencies; then the ratio between the actual output voltage at this new frequency and the output at 400 cycles is computed. A curve similar to Fig. 11 then is made up by plotting frequencies against the ratio of output at each frequency to the output at 400 cycles.

As shown by Fig. 11, the high frequency response depends on the selectivity. Another set of readings may be taken with the receiver and signal generator tuned to, say, 1400 kc. (Of course, the r.f. output must be adjusted to the same value as was used at 600 kc.)

The curves in Fig. 11 show the over-all fidelity of the receiver, excluding the loudspeaker and its response. Naturally, the loudspeaker and baffle assembly are going to

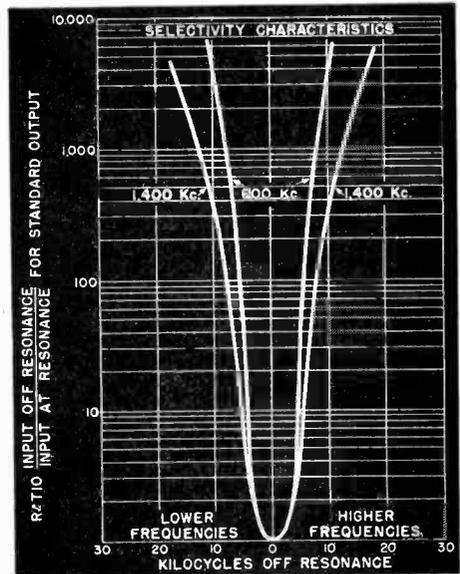
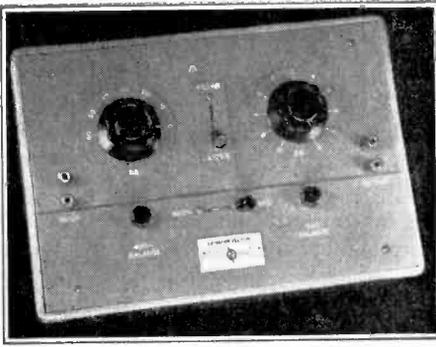


FIG. 10. How the selectivity varies at different points in the tuning band.



Courtesy Hewlett-Packard Co.

This distortion analyzer checks distortion at two frequencies, 400 cycles and 5000 cycles. It contains a tuned filter and an attenuator, so arranged that either one may be switched into use. A c.r.o. is used as an output indicator. The operating steps are: first, the tuned filter eliminates the fundamental frequency voltage, which leaves only the amplitude of the harmonic voltages indicated on the c.r.o. screen. Then the filter is switched out and the attenuator is used to reduce the c.r.o. indication to the same value as that obtained with the filter. The attenuator calibration then gives the harmonic distortion in db below the fundamental level. It is necessary that the audio source produce fundamentals of 400 cycles and 5000 cycles without any distortion.

affect the fidelity of the output to a great degree. However, we are now interested in getting the response of the receiver itself, and these curves give it.

**Distortion Measurements.** After plotting the frequency response curves, measurements are made to determine the harmonic distortion. Any of several laboratory procedures may be used for this. For example, the set-up shown in Fig. 12 has a distortion meter or wave analyzer at the output instead of an output meter. With this equipment, it is possible to determine the percentage of harmonics

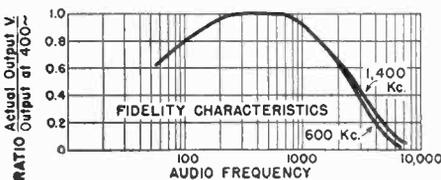
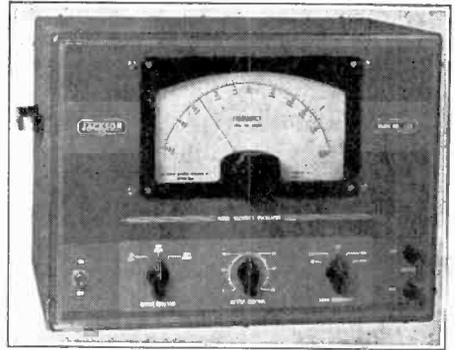


FIG. 11. Over-all fidelity curves.

introduced as a result of amplitude distortion in the amplifier.

The maximum undistorted power output can be determined by starting with a low input which is gradually increased until distortion is shown by the distortion meter or wave analyzer. An output meter is used with the distortion indicator, so that the power output level at which distortion first occurs can be measured.

**Audio Amplifier Response.** It is frequently desirable to check the characteristics of the audio amplifier alone. This is particularly true if



Courtesy Jackson

A typical audio signal generator designed for servicemen.

you are working on a public address or an electric phono system, either of which normally contains nothing but an audio amplifier.

The set-up is shown in Fig. 13. A variable audio signal generator is connected to the input of the audio amplifier and an output indicator is used across the dummy load resistance. Using a test frequency of either 400 or 1000 cycles, some reasonable output indication is obtained. Then, the audio signal generator is varied in steps over the range of audio frequencies, and output readings are made for each setting. The amplifier input must be adjusted (by the signal generator output controls) at each

test frequency so that it is the same as it was at 400 cycles. Unless the audio signal generator has an output indicator, a vacuum tube voltmeter is needed to check this.

We can again take the ratio between the output at other frequencies to that at the standard frequency, and plot another curve similar to that for the over-all response. The difference, of course, is that the response characteristic is that of the audio amplifier alone, and hence may vary widely from the over-all response, particularly at the higher frequencies.

▶ As an alternative method, a power output meter calibrated in decibels

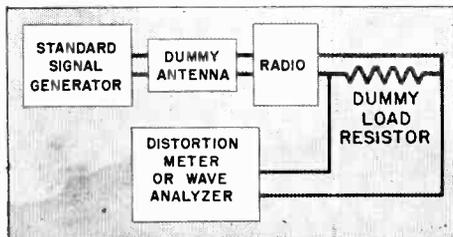


FIG. 12. Connections for determining distortion.

can be connected across the dummy load resistor and the output in db. can be read directly on the meter. If the output at, say, 1200 cycles is 10 db. and the output at 400 cycles is 8 db., we say the output at 1200 cycles is up 2 db. The curve prepared from such readings may be similar to that shown in Fig. 14A or 14B.

▶ If we get a flat curve like that shown in Fig. 14A, the amplifier is definitely a high fidelity type. This curve shows excellent response over the entire useful audio spectrum. However, it is quite likely that the amplifier response will be more like that shown in Fig. 14B, where the low frequencies drop off rapidly and there is some peak response around 4000 or 5000 cycles.

▶ Theoretically, the ideal amplifier



FIG. 13. How to get the response of an audio amplifier.

is one with an absolutely flat response. However, it may be necessary to "doctor" the response of the audio amplifier to compensate for deficiencies in the remainder of the receiver. A rising response characteristic or even a peak in the response may be desirable at the high frequencies to compensate for the side-band cutting which occurs in the r.f. stages. Thus, by over-emphasizing the high frequencies, we can make up for some of the loss in the r.f. amplifier and can improve the *over-all* response.

Similarly, a rise in response at the low frequency end of the band may be desired to compensate for a drop-off caused by the speaker or baffle characteristics.

▶ The relatively smooth curve normally obtained when checking an amplifier response may be utterly different if speaker responses are included. A typical curve in which speaker response is included is shown

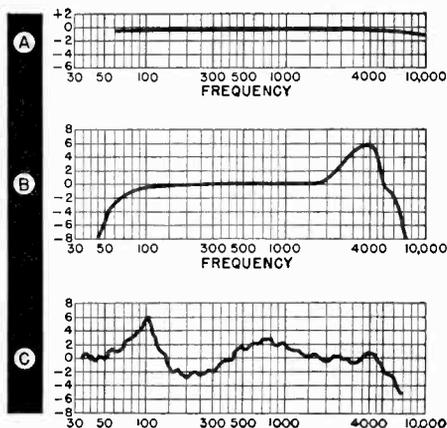
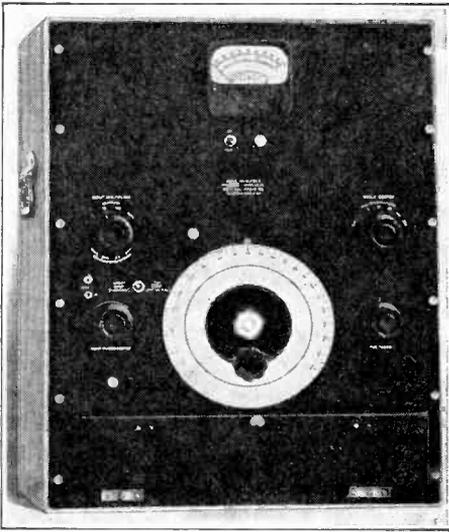


FIG. 14. Typical fidelity curves.



Courtesy General Radio Co.

**A wave analyzer.** A beat frequency is formed between the complex input wave and a built-in oscillator. This changes each audio frequency to a different i.f., as each frequency will form a different beat with the fixed oscillator. Thus the complex wave is spread over an r.f. band, and tuning circuits can be used to select each frequency component. An r.f. type v.t.v.m. then indicates the output at each frequency contained in the audio wave. Hence, the number and strength of each harmonic can be found.

in Fig. 14C. To make measurements for this kind of curve, a microphone and an amplifier are used, which have a combined response that is essentially flat. The microphone is mounted in front of the loudspeaker in a room with special acoustic properties.

The resonant characteristics of the loudspeaker cone-spider-voice coil assembly will cause numerous peaks and valleys in the characteristic curve. Hence, checking the over-all receiver response and the amplifier response merely gives us something with which we can make comparisons between similar amplifiers or receivers—it does not show the *actual out-*

*put* of the radio. Only an acoustical output response curve like that in Fig. 14C will give the actual sound output characteristics. Such a curve can be obtained only in laboratories equipped with the proper acoustical rooms and proper measuring equipment.

## FIDELITY MEASUREMENTS WITH SERVICE EQUIPMENT

The measurements which can be made in the service shop will depend greatly on the equipment available. If the shop has a standard signal generator and variable audio oscillator, over-all response curves can be made in the manner just described. Similarly, with a variable audio oscillator, it is possible to obtain the response curve of an audio amplifier.

► As always, the serviceman is usually looking for some particular defect and will use short cuts. Rather than plot the audio amplifier response, it may be possible to just vary the audio signal generator over the band and watch for any sharp peaks or sudden dips in the signal voltage measured across the dummy load resistor.

► If the serviceman has a “musical ear”, he can make a test by playing a record of known characteristics, and listening to the output of the audio amplifier. However, great care is necessary here: few people hear exactly alike, so the customer may object to a response which sounds good to the serviceman.

► As far as distortion is concerned, the methods of checking for distortion by using a c.r.o., given elsewhere in the Course, can be followed.

# Miscellaneous Measurements

It is of course possible to measure any receiver characteristic one might imagine. There are a few of these which are of some interest, although servicemen rarely measure them. Let's run through some of these.

## HUM MEASUREMENTS

In the laboratory, the residual hum level is measured by setting the radio volume control at maximum and short-circuiting the r.f. input so that no signals are picked up. The output voltage across the dummy load resistor is the fundamental hum frequency,

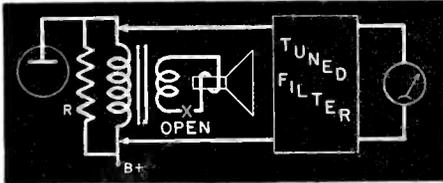


FIG. 15. A filter is necessary when measuring hum to eliminate noise components and to determine the amount of hum at each frequency. In practice, a serviceman would depend upon a listening test.

plus any harmonics of this frequency, plus any noise voltages which may be present. To eliminate noise, and also to make it possible to measure the frequency of the hum, a tuned filter may be used as shown in Fig. 15. This circuit is first tuned to 60 cycles and the amount of 60-cycle hum measured. Then the hum level is checked at 120, 180, and 240 cycles.

► If the hum is modulation hum, an unmodulated signal generator is connected to the input of the receiver and the hum output resulting is measured in a similar manner. Usually a high-sensitivity voltmeter must be used, as the hum voltage may be quite small even though it produces an objectionable amount of hum sound output.

► As a general rule, the serviceman just listens to the output of the receiver. If the hum is excessively loud, the serviceman is usually led right to the trouble by the frequency of the hum, which he can determine most easily by means of a c.r.o. or by having learned hum frequencies from listening to 60- and 120-cycle hum voltages. Modulation hum can be run down by moving the signal generator back through the r.f.-i.f. amplifier in the manner described in another lesson of your Course.

## POWER CONSUMPTION

The manufacturer usually checks the power consumed by a radio receiver, since he generally gives this figure on the receiver nameplate. He probably will use a wattmeter in the manner shown in Fig. 16A.

If the serviceman has a wattmeter, he should make similar connections.

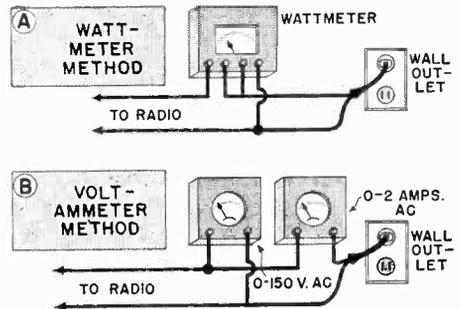


FIG. 16. Two methods of measuring the power consumed by a radio. The wattmeter is the more accurate.

If not, it is possible to use a voltmeter and ammeter by making the connections shown in Fig. 16B. The voltmeter-ammeter method does not indicate true power—multiplying their product by .8 will give a close approximation for the average a.c. radio using a power transformer.

Before using either method, it is a good idea to make sure the receiver is not in such a defective condition that it will damage the wattmeter or ammeter.

► Of course, any defects in the radio which cause a higher than normal current flow will be indicated by an increased power consumption. Thus, a leaky filter condenser or a short-circuited bypass condenser would result in an increased wattage consumption. However, the wattage test only indicates that trouble exists, without pointing out its location.

### FREQUENCY SHIFT TESTS

Once in a while the oscillator frequency of a superheterodyne receiver will shift progressively as some component in the oscillator circuit is affected by the receiver heat. This will be indicated by the program becoming more and more distorted, with the distortion clearing up if the receiver is retuned. The oscillator drift causes the production of an incorrect i.f. frequency, so the wave is distorted because of side-band cutting. Usually the drift will be in a single direction, so that the receiver must be continually tuned to higher and higher frequencies, or to lower and lower frequencies, depending on the particular part causing the trouble and its temperature characteristics.

Of course, many receiver oscillators drift *slightly* during their warm-up period, but they settle down within a few minutes. This is the reason the receiver should be allowed to warm up for half an hour or so before it is aligned. For the same reason, the signal generator should be allowed to warm up if it is a.c. operated.

The fact that retuning is necessary from time to time to obtain maximum response or to clear up distortion indicates clearly that there is an

abnormal frequency shift. If you want to measure this, you can connect a signal generator to the input of the receiver, tune them to resonance with each other, and allow both to operate for a period of time. Then, retune the signal generator for a maximum output indication. The difference in frequency between the original setting and the new setting indicates the amount of drift in the receiver for that particular period of time, provided the signal generator is itself free from drift.

### DEAD SPOTS

The laboratory and service tests for dead spots are identical. The receiver dial and signal generator dial are rotated in step over the entire frequency range of the receiver. For example, if we wish to check the broadcast band, both the generator and the receiver dial should first be set to 550 kilocycles, then to 560 kilocycles, and so on at 10 or 20 kilocycle intervals up to 1500 kilocycles. At each setting we should listen to make sure the output is normal. A skillful operator can turn the generator dial with his left hand and the receiver dial with his right, keeping the two in step, and get a continuous check throughout the band.

Naturally, a dead spot would be indicated by a lack of reception or by a sharp drop in output over some portion of the tuning range of the receiver.

### IMAGE REJECTION

Suppose a receiver dial is set to receive a 1500-kc. signal and that the i.f. of the set is 460 kc. This means the receiver oscillator will be working at a frequency of 1500 plus 460 or 1960 kc. Now, if a signal from a station at 2420 kilocycles is strong enough to get through the preselector, it also will produce the right i.f. value,

since 2420 *minus* 1960 is 460 kc. Thus, it is possible for the proper i.f. value to be produced by signals either above or below the oscillator frequency. The interfering signal (from the station the receiver is not tuned to) is called an image, and is twice the i.f. value above the desired signal frequency.

The ability of the receiver to reject image interference is determined by the following procedure. First, the signal generator is set to the frequency to which the receiver is tuned, and the input necessary to give standard output is determined. Then, with the receiver dial left at this point, the signal generator is tuned to the image frequency. The output from the signal generator is adjusted again to produce standard output. Dividing the signal input at the image setting by the signal input at the receiver dial setting gives the image rejection ratio. A ratio of 100 to 1 or greater is desired. A ratio below this value indicates poor receiver design or a receiver badly out of alignment. However, it is possible for image interference to exist even with a satisfactory image ratio if a very powerful station happens to be at the image frequency of a desired station. In this case, a change in the i.f. value of the receiver, or the use of a wave trap tuned to the interfering station would be an effective cure.

## TESTING FOR NOISE

In the laboratory, the receiver is tested for noise output by a method similar to that used for the hum voltage check. The receiver is placed in a shielded room and the power lines

leading into the room are thoroughly filtered, so that whatever noise is heard must come from the radio itself.

To distinguish between noise and hum, tuned filters may be used between the receiver and output indicator, tuned to *reject* the hum frequencies of 60, 120, and 240 cycles. Any remaining output from the receiver must then consist of noise components.

In the service shop, a shielded cage to prevent direct noise pick-up by the receiver is seldom available. Usually, the serviceman depends on a power line filter to remove any noise that may be coming in this way, and then compares the noise level heard on a suspected radio with that normally heard in the shop on other similar receivers. It is well to be cautious about judging the noise when no signals are tuned in, however, as the more sensitive the receiver, the greater the amount of noise pick-up by the circuit wiring, and also the greater the tube noise level. In fact, you probably will have to explain to many customers why their large sensitive receiver is so much more noisy than some inexpensive midget set they may have or may have heard.

Actually, of course, the amount of noise heard between the stations is no criterion of the performance of the receiver when it is tuned to a station. It is quite possible that the reduction in sensitivity brought about by the normal action of the a.v.c. circuit may cut out all the background noise. The important factor is the amount of noise heard when tuned to a station giving normal reception in your locality.

# Receiver and Part Revitalization

Although revitalization is concerned with any of the receiver characteristics which may be below normal, most ordinary radio troubles can be located and cured by the usual servicing methods. Then, replacing any defective tubes, cleaning the chassis, and perhaps realigning the receiver will complete the service job.

However, there will be cases where the receiver has "lost its pep," so it has below-normal sensitivity, or has hum, noise or distortion levels somewhat above normal, but not high enough to present a real defect. These conditions may or may not exist together. The causes and cures of most of these have been given elsewhere in your Course, so now let's concentrate on those troubles causing below-normal sensitivity.

► As you know, radio parts do wear out through normal use. Tubes age, lose emission, and therefore reduce the stage gain. Paper condensers develop leakage because of dielectric fatigue and manufacturing imperfections. Electrolytic condensers dry out, develop high power factors, or become leaky. Speaker cones dry out, voice coil forms warp, and speakers using permanent magnets lose their magnetism.

In addition, misuse or accidents will cause trouble. For example, a receiver may be left near an open window during a rain-storm and the r.f. coils may absorb moisture, resulting in a decrease in the Q factor.

► Receivers designed for use in the tropics are thoroughly moisture-proofed. However, the average set is not so treated, and if it is used at the seashore or is left in a damp basement or damp recreation room, moisture is likely to get into the coils, transformers, and wire insulation,

lowering the over-all sensitivity. In addition, acid fumes from coal burning furnaces or from industrial plants can set up corrosion and cause leakage paths between wires. A heavy coating of grease (which is sometimes conductive) will be found on exposed parts and chassis of receivers used in kitchens. And, of course, a receiver which has been through a flood or has been drenched by a fire hose will be well water-soaked.

## DISASTER DAMAGE

Let us first see what to do to a receiver which has been damaged by a fire or a flood, as certain steps are necessary even to restore such a receiver to the point where ordinary procedures will be effective.

When a receiver is brought to you and obviously shows signs of disaster damage, the first thing to do is to remove the chassis and speaker from the cabinet. Remove the tubes and clean off the accumulation of mud or other debris. Some servicemen figure that since the receiver has already been water saturated, a little more water won't hurt, so they use a stream of warm water from a hose to clean the chassis. However, if possible, clean the chassis by using a dry cloth. If there is oil or grease on the chassis, carbon tetrachloride or Varsol, both good solvents and non-inflammable, may be used for cleaning. A rag or brush dipped in the solvent can be used to remove grease and other chassis dirt. (This work is best done outside, or in a well ventilated room, since the fumes from the cleaning solvent make some people ill.)

When the chassis has been cleaned, you must find a way of removing the accumulated moisture. A damp chassis put in a warm, dry place will not

become completely dry. Excess water will evaporate, but the moisture-laden air will be trapped in parts and under shield cans. To remove moisture from the chassis completely, a stream of dry, heated air should flow over the chassis and around and through moisture-laden parts. The moisture will be carried away by this stream of air.

For occasional jobs, a small electric fan and an electric heater can be directed against the chassis as shown in Fig. 17. The heater vaporizes the moisture and the fan drives the moisture-laden air away from the chassis. It is necessary to change the chassis position several times so that all parts will be dried equally.

► In larger shops, where work of this sort may be done more often, an outfit like the one shown in Fig. 18 may be used. The asbestos-lined box may be constructed from wood or

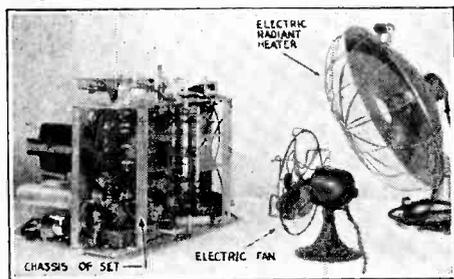


FIG. 17. How to use a fan and an electric heater to dry out a radio chassis.

sheet metal, or might be a portable cooking oven. After placing a chassis on the grid shelf, close the box tightly, and start the heater. After a few minutes, turn on the fan so it will drive the moisture-laden air up the stove-pipe exhaust away from the radio. Watch the temperature with a thermometer which has its bulb in the oven and its reading stem exposed to the outside. Keep the temperature at about 130° F. This tempera-

ture is sufficient to vaporize moisture but will not damage the receiver parts or cause undue melting of the wax or pitch used in sealed parts. Two or three hours in the oven should be long enough to dry out the average chassis.

► Once the chassis is perfectly dry, blow out all dirt and dust with a small hand bellows, a bicycle pump,

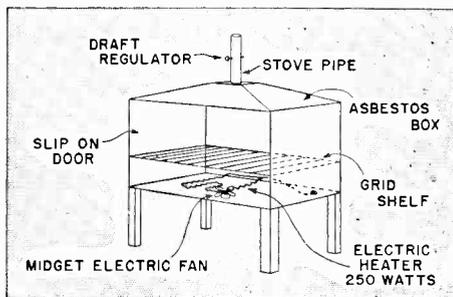


FIG. 18. A drying oven.

or a vacuum cleaner blower attachment. Clean all surfaces with a dry cloth. Use pipe cleaners (available at any tobacco store) to remove all dirt and dust from between the plates of the variable condensers.

**Operating Precautions.** Before trying the receiver out, first check for leakage within the power supply, by measuring across the B supply terminals with an ohmmeter. Place the ohmmeter test probes across either the input or output filter condenser leads—whichever are more accessible. The diagram will show if a bleeder resistor is used. If there is no bleeder, the leakage resistance should be that of the filter condensers, provided you observe proper ohmmeter polarity. If the resistance is abnormally low, disconnect the condensers and check them individually. Make replacements if you find the condensers are at fault; otherwise, run the trouble down to the defective part.

If the B supply resistance is normal, replace all the tubes *except the*

*rectifier*, and turn the set on. The tube filaments will place a partial load on the power transformer. (You cannot make this check on a.c.-d.c. receivers, since removing the rectifier tube breaks the filament circuit. However, there is no power transformer to worry about in such sets, so you can plug in the receiver directly.)

If the transformer shows no signs of overheating after half an hour, put the rectifier tube in its socket. This will supply tube electrode voltages throughout the chassis. You can now treat the receiver as if it were in for an ordinary repair job. Of course, the speaker cone will have been ruined, and will have to be replaced, and you will probably find other parts similarly damaged.

► After the receiver is restored to operating condition, very likely you will find it desirable to improve its performance with the regular revitalization procedures we will now give.

## ORDINARY REVITALIZATION

The following procedures will apply equally to receivers which have been damaged (by flood or fire) and to those receivers which have lost sensitivity with age. Normally, even if nothing else happens to a receiver, its performance will drop off gradually over a period of years. The distortion and hum levels will increase, the sensitivity will decrease, and the set will not separate stations as well as it did when new. Many a set owner, when he finally becomes aware that his radio has reached such a condition, thinks nothing can be done for it and buys another. Yet it is often possible to restore such a receiver nearly to its original condition. Let us see, first, what can be done to improve its sensitivity.

A loss of sensitivity is caused by loss of stage gain. The gain may be

reduced by improper alignment, tube defects, changes in the value of the load into which the tube works, open bypass condensers, or changes in operating voltage. In addition, various defects may have reduced the Q or step-up obtained in the resonant circuits; in fact, this last is the most common reason for loss of sensitivity.

The first step in revitalization is to clean the receiver thoroughly. **DON'T USE WATER!** Wipe the receiver carefully with a *dry* cloth, blowing out the dust with a hand bellows or bicycle pump (or use compressed air). Clean the tuning condenser gang carefully with a pipe cleaner. If there is a heavy coating of grease or oil on the receiver, remove it with a solvent such as carbon tetrachloride or Varsol (not water).

Then, try realigning the receiver. Notice the action of the trimmers, particularly if the receiver sensitivity is not restored to normal. Frequently, you will be led to a defective tuned circuit directly by a sluggish, broad-tuning trimmer action, or even a lack of a resonance peak.

**Tuned Circuit Overhaul.** The next step is to eliminate resistance from the tuned circuits to improve their Q. Apply a hot soldering iron to each soldered joint in each tuned circuit, heating the joints until the solder runs. Any corrosion which has formed, or any cold-soldered joints, will be eliminated and the resistance in the tuned circuits lowered. Use a solvent such as carbon tetrachloride to clean the spring wipers which connect the rotors of the condenser gang to the frame. Also, bend the wipers to give a better wiping contact.

► If the stators of a condenser gang section are held in place by bolts on each end (they are soldered in place in recent receivers), the connection to each stator is made through these bolts. Sometimes corrosion on the

bolts increases the resistance of the circuit. Simply loosening and tightening these bolts, one at a time, will remove the corrosive deposit and reduce the connection resistance to its normal level.

► Besides the series resistance, leakage across the coil forms or across the tuning condenser gang also will lower the tuned circuit Q. Carefully clean the dust from between the plates of the tuning condenser gang. In addition, clean all of the insulating strips used in the condenser assembly.

Lowered coil Q is often caused by moisture absorbed in the coil form. You should wipe the coil form carefully to remove any surface moisture, then bake out the receiver by one of the methods shown in Figs. 17 and 18 to drive off absorbed moisture. (Incidentally, the speaker cone should not be baked out, for excessive drying of the cone will make it brittle. If anything is the matter with the speaker cone, it is best to replace it.)

**Circuit Troubles.** After you have improved operation of the tuned circuits, check over the operating conditions. The load into which an r.f. tube works is very frequently governed by a tuned circuit Q, so clearing up tuned circuit defects may cause normal operation. Of course, you should realign the radio after having worked over the tuned circuits. Should the sensitivity still be below normal, check the operating voltages and correct any defects so that proper voltages will be obtained.

A loss of speaker magnetism will lower the output of the receiver considerably, so if p.m. or magnetic speakers are used, it may be desirable to have them overhauled by the factory and remagnetized, or else replace the speaker.

► After you have followed all these procedures, you may be able to get a

little more sensitivity by "selecting" the tubes. Many servicemen overlook the fact that tubes of the same type do not all have the same gain. This is caused by small differences in the mutual conductance of the tubes, which will not show up on most tube testers. To select the tubes which will produce the maximum performance from a particular receiver, feed a signal from a signal generator into the set and connect an output meter so that it will measure the set output. Tune the set to resonance, and adjust the input to a value which produces some easily remembered output meter reading. Now try several different tubes of the proper type in each socket, and make a note of the output reading each tube produces. (Don't change the tuning or the volume control settings.) Leave in the tubes giving the highest output.

**Further Overhaul Steps.** If the receiver is being completely overhauled, check all fixed condensers for leakage and for capacity with a condenser analyzer. In addition, check the resistors with an ohmmeter and replace any that are more than 20% off from their rated values.

► Check all controls. If the dial cord is frayed, install a new one. Work powdered rosin into the cord if it slips. Should the receiver be noisy when you pull or push on the tuning knob, apply Grafoline (a mixture of vaseline and graphite) to all metal parts in the dial tuning mechanism. (Don't get any on the dial cord.) Clean the wave band switch contacts, and replace the volume control if it has a tendency to be noisy.

## MOISTURE-PROOFING RECEIVERS

When a receiver from a seashore cottage or pleasure boat has been re-

vitalized, the improvement will only be temporary unless steps are taken to prevent recurrence of the trouble. The procedure necessary for moisture proofing takes some time, so the process is rather costly. Be sure to explain this to the customer and get his O.K. before considering these steps. Here is what can be done about the parts most frequently affected.

**R.F. Coils.** The r.f. and i.f. transformers and coils in many receivers are untreated. Such coils and their coil forms are bound to collect moisture sooner or later, and their exposed terminal lugs are subject to corrosion.

It is frequently possible to buy a treated set of coils for the receiver. These coils will have been wax-dipped by the manufacturer in such a manner that they are less likely to absorb moisture.

If it is impossible to obtain treated replacement coils, the original coils can be thoroughly baked out, then treated by dipping them in melted "ceresin" wax, or by "painting" the windings, terminal leads, and connections with a thin coat of a moisture-proof insulating coil "dope." Preferably use a dope compound having a Polystyrene base, such as Amphenol 912 or Carron HQ-711. The coils must be thoroughly dried before being given such a treatment; if necessary, they can be removed and baked individually. Individual baking is best if the coil is covered by a shield can, since all the moisture may not be driven out by heat treating the entire chassis.

The coil dope just mentioned is a clear, transparent liquid which dries rapidly in air to form a hard, permanent surface. Ordinary insulating varnishes are not suitable for treating r.f. coils because the varnishes themselves have losses.

► Very frequently, the r.f. leads between the tuning coils and condensers

are ordinary cotton-covered wire. Replacing these leads with solid bare wire where possible, or with wire covered by varnished insulation or rubber, will frequently help to reduce trouble caused by moisture collecting in the tuned circuit.

Similarly, any shielded r.f. leads quite likely will have developed leakage between the wire and the shield. Replacing defective leads will pep up a receiver a great amount.

Many of these steps, individually, will not seem to make any difference in the operation of the receiver. Collectively, however, they will prove effective.

Most servicemen stop after carrying out the steps just given, since treating the tuned circuit coils and leads will be sufficient in most cases. However, several additional steps may be taken if the set requires them. Some suggestions follow.

**Variable Condensers.** Trouble with the wiping contacts between the shaft and frame is liable to occur again. The only way of eliminating this trouble permanently is to use pigtail leads. These leads are flexible wire, one end of which is soldered to the rotor shaft while the other end is soldered to the condenser frame or chassis. There should be a separate pigtail lead at each rotor element of the condenser; that is, a three-section condenser should have three pigtails.

It is sometimes impossible to solder pigtails to the rotor shaft. In such cases, the shaft may be drilled, a screw run through it, and the lead connected to the shaft under the screw head. Be sure the pigtail lead is long enough for the condenser to be rotated throughout its range without breaking the lead.

Any trimmer or padder condensers associated with the tuning condenser gang must be carefully cleaned and should be checked for leakage.

Cracked mica means the trimmer must be replaced.

**Fixed Condensers.** Mica condensers molded in bakelite are usually moisture proof. However, the bakelite may crack, permitting some moisture trouble. If you find a cracked condenser, replace it with a new one, and treat the new condenser with coil dope to seal any cracks which may be in it.

Dry electrolytic condensers are preferable to the wet types where moisture trouble exists. The kinds sealed in metal containers should be used. (To prevent corrosion of the container itself, many high-grade dry electrolytic condensers are sealed in a metal container, which is itself encased in a wax-covered cardboard tube or box.)

Paper condensers should be of the type sealed in a moisture-proof wax cylinder.

**Fixed Resistors.** Resistance wire is very apt to be corroded by moist, salt air. Corrosion is particularly likely to develop where the resistance wire of a resistor joins the terminals. For this reason, resistors wound with bare resistance wire, or which have a portion of the winding left exposed for adjustment of resistance value by a slider, should never be used. It is best to use an adjustable resistor temporarily to find the right value, then replace it with a vitreous enameled resistor which has the correct value. If the original resistor was used as a voltage divider, then two resistors can be used to replace it.

**A.F. Transformers.** An a.f. transformer which is mounted in a metal case filled with sealing compound

will usually be relatively free from moisture trouble. However, any open transformer (or cased unit which does not contain a sealing compound) may easily be affected by moisture. Replace any defective units of this type (using a sealed transformer if possible). An open transformer can be coated with coil dope, but this is not always successful in preventing moisture troubles.

**Tube and Socket Contacts.** Tube sockets frequently fail, usually breaking down between the plate contacts and adjacent terminals. Such breakdowns are particularly common in wafer-type sockets, especially those used for rectifier or power output tubes. A defective socket should be replaced with one of a better type, such as a molded unit.

Corrosion is practically certain to cause poor contact between the tube prongs and the socket contacts from time to time. Cleaning the tube prongs with sandpaper and working the tube in and out of the socket a few times will usually clear up this trouble.

**Dry Batteries.** A modern dry battery is housed in a waxed container which usually does not cause much trouble as long as it can be kept relatively dry. However, sometimes leakage will develop through this case if it is placed where moisture can collect. Batteries should be given a thin coat of paraffin or beeswax, preferably the latter, to prevent this leakage. Don't try to separate batteries by pieces of ordinary cardboard or paper; such material absorbs moisture readily and will make matters even worse.

# Tube Testers

Tubes are responsible for a great many radio troubles. As a result, it is necessary to have some means of checking the condition of tubes.

Tube manufacturers have very elaborate testing apparatus. They can measure the mutual conductance, amplification factor, plate resistance, tube element capacity, and any other tube characteristic at will. Such testing apparatus might well fill a good sized room and cost thousands of dollars

Naturally, any such equipment is out of the question for servicemen. Therefore, simplified, portable testers were developed. These testers give an indication of the tube value or quality by checking one or two important characteristics of the tube, instead of all its characteristics.

Before we can determine whether or not a tube is good by measuring one of its characteristics, we must decide what variations in that characteristic are acceptable. As you already know, tube charts give average tube values. Manufacturers permit variations in many of these characteristics of 20 to 30% and frequently congratulate themselves upon getting even this close to the average. Thus, the fact that tubes have the same type number does not mean that they have exactly identical characteristics even when new.

Of course, radio circuits are designed for tubes with average characteristics. An exceptionally "peppy" tube will increase the sensitivity of a stage and may even cause oscillation if the circuit is relatively unstable. On the other hand, a tube with lower-than-normal characteristics may reduce the sensitivity.

▶ A tube may be considered unsatisfactory for any reason which causes

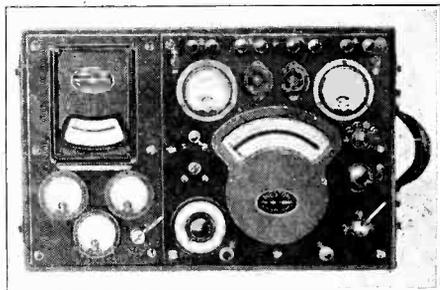
abnormal performance. We may list these reasons as follows:

1. Open element (usually the filament).
2. Incorrect emission.
3. Incorrect mutual conductance.
4. Gas.
5. Loose elements.
6. Shorted elements.
7. Leakage between elements.
8. Incorrect power output.

Of these, mutual conductance, power output and emission are the "quality" factors, so tube testers check one of these. In addition, most tube testers will check for shorts and leakages, and a few will check for other defects. Tube testers thus provide a means of quickly weeding out the tubes with major defects, leaving others to be found by their symptoms. Of course, it is always possible to try another tube in place of a suspected one, and this is often the only test which can be relied on completely.

## MUTUAL CONDUCTANCE TESTERS

The mutual conductance is recognized as a "figure of merit" of a tube. A measurement of the mutual conductance determines the ability of



*Courtesy Weston*

One of the early tube testers designed for checking for shorts, amplification factor, plate resistance, mutual conductance and gas. This tester was too elaborate for servicemen, but was a forerunner of the types used by tube manufacturers.

the grid to control the plate current of a tube and takes into consideration the amplification factor and plate resistance of that tube. It was natural, therefore, for the earlier tube testers to measure mutual conductance.

The circuit of a basic tester of this type is shown in Fig. 19. To use this tester, we first apply the proper filament voltage, set switch *S* to position 1 so that battery  $E_1$  furnishes the correct grid bias, apply the proper plate

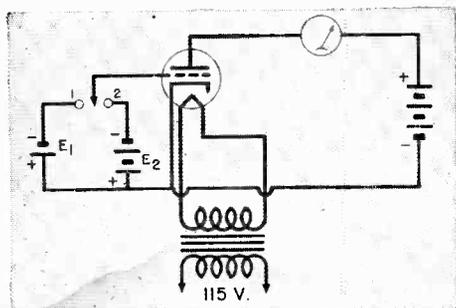
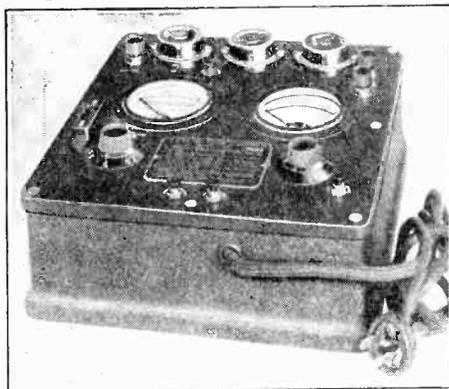


FIG. 19. The basic diagram of a grid shift or mutual conductance tester.

voltage, and read the resulting plate current.

We then throw switch *S* to position 2, which provides a different bias. This results in a new plate current. By dividing the difference in the plate current readings by the difference in grid bias voltage, we get a measure of mutual conductance. (In other words, for a constant plate voltage, dividing the change in plate current by the change in grid bias gives us the mutual conductance.) If the plate current change is in milliamperes, multiplying the result by 1000 will give the mutual conductance in micromhos.

This tester was called a grid shift tester because of the method of changing the bias. It was rather inconvenient to use, since two readings

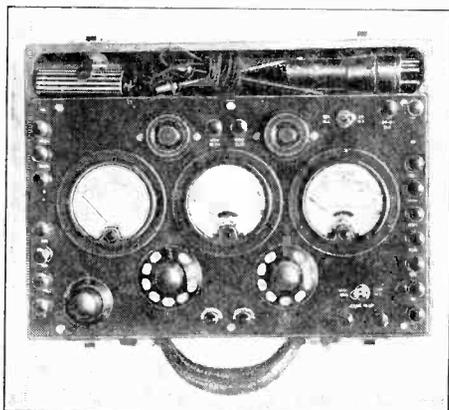


Courtesy Weston

One of the first grid shift or mutual conductance testers designed for servicemen.

and some figuring were necessary before the mutual conductance could be found. As a result, tube testers were soon developed which eliminated much of this inconvenience; they had elaborate circuits to balance out the first meter reading, and had the meter arranged so that it indicated the difference in current directly with its second reading. This arrangement allowed the meter dial to be calibrated in mutual conductance, so that no figuring was required.

► However, using a d.c. voltage



Courtesy Weston

Many of the early set analyzers had a built-in grid shift test. Readings were only relative, as the analyzer depended on the radio for voltages, but tubes could be compared this way.

change on the grid does not indicate the dynamic or operating characteristics of the tube. For this reason, as soon as a.c. power supplies became common, a dynamic mutual conductance tester was developed in which an a.c. voltage was applied to the grid as shown in Fig. 20.

This circuit has the advantage that, when the a.c. grid voltage is adjusted to exactly 1 volt, the mutual conductance (in micromhos) of the tube is equal to the resulting plate current reading in milliamperes, multiplied by 1000. Because the plate current wave is distorted, it is desirable to have a meter which will indicate the effective or r.m.s. value of an a.c. wave, so a dynamometer is used. In Fig. 20, one coil of the dynamometer is connected to the transformer. This makes the meter read *only* the a.c. plate current, ignoring the initial d.c. current altogether, as only a.c. will produce an adding field and give a deflection.

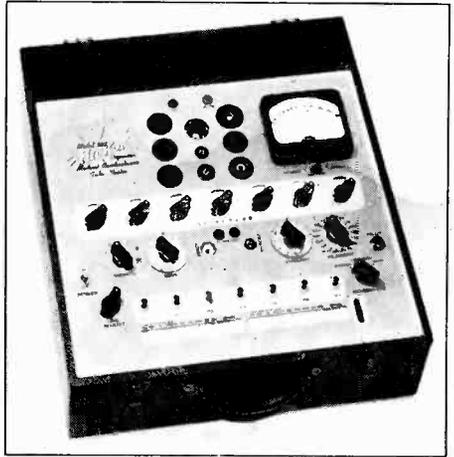
#### Advantages and Disadvantages.

Mutual conductance testers were fine in the early days of radio, when there were but a few tube types. The first models all required that normal operating voltages be applied to the tubes. As new tube types came out, an increasingly wide variety of filament, grid, and plate voltages had to be supplied. Then, screen grid, pentode, and other multi-element tubes came out, which hopelessly complicated the power supply situation.

► At this time, manufacturers decided that it was unnecessary to determine the *actual* mutual conductance, as long as some comparative reading could be obtained. To get a basis for such readings, arbitrarily determined voltages were applied to the grid and plate elements of tubes known to be in good condition, and the mutual conductance of each was measured under these conditions.

The resulting readings were compiled into a chart which then became a standard for tubes measured under the same conditions. In other words, all one had to do to determine whether any particular tube was satisfactory was to compare the readings with those on the chart for that particular tube tester. If the readings obtained came within normal tolerance limits of those given on the chart for that type of tube, then the tube being tested was good.

The chart values, of course, were not true mutual conductance values, but they were just as useful for tube testing. Further, this method of using



Courtesy Hickok Elec. Inst. Co.

A modern dynamic mutual conductance tube tester. Through the use of an elaborate switching arrangement, mutual conductance values may be read, or by switching, the results may be read on an "English" scale.

arbitrary voltages on tube elements allowed some elements to be connected together. In a screen grid tube, for example, the screen grid could be tied directly to the plate as long as the applied voltage was kept within safe limits. This vastly simplified the power pack requirements.

Finally, since the chart readings were not the true mutual conductance values, it was possible to go a step

further. The chart itself could be eliminated by proper selection of voltages, so that the so-called "English reading" dial scale could be used on the tube tester. This scale was divided into sections marked BAD—QUESTIONABLE—GOOD. When the controls were properly set, the tube plate current would cause an indication in one of these sections and so show the condition of the tube at once. Such a scale is far easier to read; it can be marked to take care of tolerance limits; and—most important of all—the customer can understand the readings. In fact, this scale has so many advantages that it is now standard on practically all tube testers.

The first testers of this improved type had several sockets, each of which was wired to make the correct connections to a particular group of tube types. However, the great number of tubes developed soon made this method unsatisfactory because too many sockets were required. Finally, a design was evolved which had one socket for each type of tube base. Connections between the tube elements and to the power supply were made by a rather elaborate switching arrangement. Even at best, this was a complicated type of tube tester.

## POWER OUTPUT TESTER

While more modern equipment has largely superseded the mutual conductance tester, some models of it are still being sold and used. The tester is entirely satisfactory for tubes intended for voltage amplification. It falls down somewhat on testing power tubes, however, and does not duplicate operating conditions for the ordinary amplifier tube, because it has no load in the plate circuit.

The tester shown in Fig. 21 is known as a power output tester. The

chief difference between it and the mutual conductance tester lies in the fact that the tube operates into its rated load, which is adjusted by varying  $R_L$ . The mutual conductance can be found under conditions more closely resembling actual operation, and the output power can be determined. (The power output equals

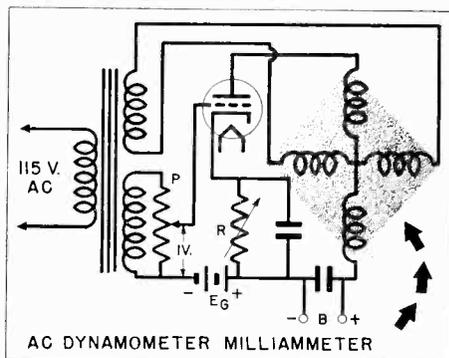


FIG. 20. A dynamic mutual conductance tester, using an a.c. voltage on the grid.

the load resistance value multiplied by the square of the a.c. plate current.) Instead of calculating these values, the voltages are arranged so the meter reads on an English scale.

The power output tester gives a more accurate test under reduced voltage conditions than does the mutual conductance tester. Several testers of this basic type are now being made.

## EMISSION TUBE TESTERS

Both the mutual conductance and power output testers run into trouble in testing diode tubes. Since a diode has no grid, it has no mutual conductance. Therefore, an emission test is the only one which these testers can make on a diode tube. This test is made by applying a chosen voltage to the tube and measuring the resulting plate current. This will give some idea of the ability of the cathode to

function under normal conditions and deliver normal emission.

► Since emission tests are all that could be made on diodes, it is natural to consider checking just the emission of all tubes, assuming that if the cathode emission is normal, the mutual conductance and other factors are probably acceptable. This led to the development of the emission type tester, in which all tubes are converted to diodes and the plate current is measured. The basic circuit for a tester of this type is shown in Fig. 22.

As you can see, the grid elements are all tied to the plate of the tube, and a fixed voltage is applied between the cathode and the elements which are tied together. The fixed low voltage reduces the danger of excessive current flow. However, since a power



Another modern emission tester. This is the NRI Professional model. The lid is detached when the instrument is used on the workbench. The chart containing operating instructions is carried in the lid when the instrument is used as a portable tube tester.



Courtesy Triplett Elec. Inst. Co.

A late type emission tester. Toggle switches are used to make the "shorts" tests, and to select the proper elements for the emission test. There is a built-in roll chart that is set to give a listing of the switch settings for the tube type. This model is also a multimeter.

tube will naturally pass far more current than a voltage amplifier, some means of changing the meter range must be used. In this particular model, resistor  $R_2$  acts as a shunt on the meter and resistor  $R_1$  acts as a

series resistance. These two resistor units are usually ganged together and operated by a single control. When the resistors are adjusted by the control to the proper value for the type of tube being tested, the plate current reading on the meter  $M$  will indicate the worth of the tube on the BAD—QUESTIONABLE—GOOD scale.

#### Advantages and Disadvantages.

Today, the emission tester is the most common of the tube tester types. It is the simplest to construct, the easiest to operate, and the lowest in cost. The circuit arrangement makes it easy to test new tube types as they are brought out.

Admittedly, it is not the best tube tester, for a tube may test good on it and then prove faulty when installed in a radio circuit. However, in general, any tube *rejected* by this tester is definitely bad.

Since only the emission is measured, not the ability of the tube grid to control the plate current, the mu-

tual conductance of the tube may be below normal without this fact being disclosed by the emission tester. However, this tester will pick out most of the defective tubes, and a trial in the receiver will quickly determine whether a tube which tests GOOD is actually in the best operating condition.

► While there are many different makes of emission testers, all of them have much the same features. Any one you buy will have a tapped filament winding, so that the proper filament voltage can be supplied to any tube type. It will have a series of toggle switches, push buttons, or a selector switch to connect all elements except the cathode (and filament) to the plate, and will have some means of varying the resistors  $R_1$  and  $R_2$  together. Incidentally, the latter control is commonly marked the "load" control.

### OTHER TUBE TESTS

Today, service-type tube testers fall into one of the foregoing basic

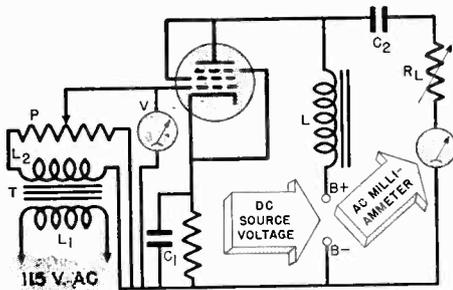


FIG. 21. The power output tester measures the power fed into the rated tube load.

types, testing quality by measuring the mutual conductance, the power output, or the emission. In addition to the foregoing "quality" tests, many of these tube testers can make additional tests. Let's see what some of these tests are.

**Shorts and Leakages.** Practically all modern tube testers can check a tube for short circuits. Most of them also test for leakage between the elements as well. In fact, it is necessary to test a tube for shorts and leakage before making a quality test; if such tests are not made, the tube tester may be damaged by excessive current flow.

A basic short checker is shown in Fig. 23. In the positions shown, the switches tie all of the elements to-

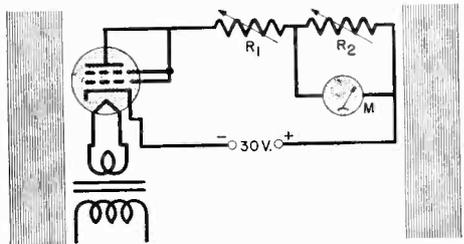


FIG. 22. The basic emission tester circuit.

gether. One side of the filament circuit is then connected to the power line, while the other side of the power line is connected through a neon lamp and resistor to additional switch contacts.

After the filament voltage is adjusted to the normal value, the switches are thrown one at a time. Each switch is returned to the position shown in Fig. 23 before the next is thrown. Each throw of a switch to the left connects one individual element to the resistor-neon lamp assembly, and thus forms a circuit in which the power line voltage is connected between this one element and all others through the lamp and resistor. If there is any short or leakage between this element and any of the others, the lamp will glow. The condenser  $C$  prevents rectified current flow, so the neon lamp will not be lighted by currents resulting from



Courtesy Weston

This modern emission tester checks batteries as well as tubes, the meter being used as a voltmeter for this purpose. You will find that many tube testers are "combination" testers like this.

emission and the rectifying action of the tube. The charging of the condenser may result in a momentary flash (but no steady glow) as the switches are closed.

The switches can be individual switches, thrown one at a time, or they can be steps on a selector switch that is rotated through the short-testing positions. This test always precedes the regular tube test.

The most important leakage to check is that between the cathode and filament, which is shown up by this test. It is important to operate the tube filament at its normal temperature, for some leakages will show up only when the filament is heated.

**Gas Tests.** The fact that gas causes a grid current flow makes it possible to check for gas by measuring this current. Some of the earliest tube testers used a micro-ammeter in the grid circuit. Such a meter, however, is rather expensive, and there is dan-

ger of burning it out. A somewhat simpler test device, like the one shown in Fig. 24, is now used on some testers.

To make the gas test, the push button switch  $S$  is first left in its normal closed position so the resistor  $R$  is shorted out of the circuit. The C bias and other voltages are adjusted to normal values, and the plate current, indicated by meter  $M$ , is noted. Then switch  $S$  is pressed; this opens the switch and places the resistor  $R$  in the grid circuit. Any grid current caused by gas now develops a bucking bias voltage across  $R$ , which, if it is appreciable, will cause a change in the plate current. This gives a very simple test for gas in a tube; if the tube has no gas content, depressing switch  $S$  will not cause a change in the plate current; if it has, pressing the switch will cause a large plate current change.

► Not all tube testers have this gas test feature. (Emission testers do not have this feature.) If yours does not, you can check a tube you suspect of

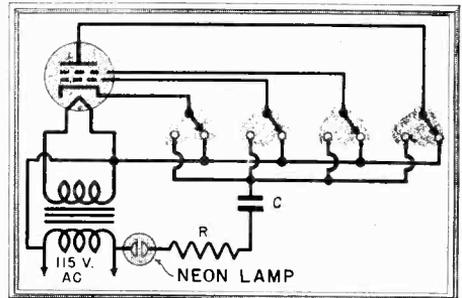


FIG. 23. One form of "shorts" tester.

being gassy by making measurements within the radio itself. As you will recall, a gassy tube causes trouble if there is a high resistance in the grid circuit, as in the ordinary resistance-capacitance coupled amplifier. Measuring for a voltage drop across the grid resistor when the set is turned

on (but with no signal tuned in) will quickly show whether the tube is gassy or not. If a voltage is across this resistor, then either the coupling condenser is leaky or the tube is gassy. Withdrawing the tube or disconnecting the coupling condenser will let you determine just which defect has caused the voltage drop.

**Miscellaneous Tests.** Tubes may also be tested for unusual defects by special procedures. You can very easily check to see whether vibration

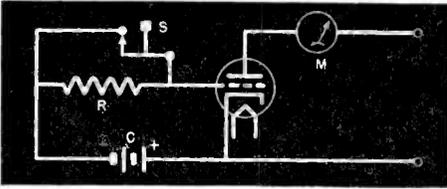


FIG. 24. A typical gas test circuit.

can cause tube elements to touch: while you are testing for leakage between the elements, just tap the tube and watch the neon lamp or short circuit indicator for signs of leakage. Or, you can tap on the tube while it is in the radio. If the radio makes a noise when you do so, then the tube is sensitive to vibration.

► If an earphone is connected in series with the plate of the tube, any variation in current caused by an intermittent short or loose element will result in noise in the headphone. Some tube testers have a phone jack to make this test possible.

In some cases, you will find that when you check a tube in a tester the meter pointer will swing up and then gradually swing down—or it may vary, first going one way and then the other. This indicates a varying emission which may cause fading reception. When you test a tube, leave your finger on the test button for a few seconds so that you can observe the steadiness of the meter reading.

If the tube is good, a relatively steady meter reading should be obtained. Don't worry about a small variation or movement of the meter pointer; this is probably the result of poor filtering in the power supply or fluctuations in line voltage. Consider only changes which cause relatively large variations in the meter reading.

## TUBE TESTING PROCEDURE

The exact procedure for operating a tube tester will vary with the type of tester. Similarly, the setting of the controls will depend on the tester type. Always be sure you follow the manufacturer's instructions exactly.

However, in general, the following procedure will be used with most types of tube testers:

- 1. Turn on the tube tester, then rotate the line voltage adjustment control knob until a pointer comes to rest behind an illuminated shadow-graph scale (or, in other types of testers, until the meter needle comes up to a mark on the scale). The voltages applied to the filament and tube elements depend on this procedure.
- 2. Look up the tube type on the tester instruction chart, and set the filament control for the proper filament voltage. Set the circuit selector switch to the "short test" position. Plug in the tube. After the tube has warmed up, check the line voltage adjustment and, if necessary, reset the control.
- 3. Now test for shorts or leakage. Depending on the tester, this may be done by rotating a selector switch through various positions; by moving toggle switches according to the tester instructions; or by depressing push-buttons one at a time. Watch the neon lamp for a glow, indicating leakage or a short circuit. As you go through each of the short-testing

positions, tap the tube lightly with a lead pencil, or thump it by flicking your finger against it, to see if vibration will cause shorts or leakage. If there are shorts, the tube is bad and no further tests should be made.

► 4. If the tube successfully passes the short test, check its quality. Set the load control to the position given in the manufacturer's instructions, then throw the circuit selector (or toggle or push-button) switch to the proper position. The tester may now automatically indicate the tube quality on the meter scale, or it may be necessary to depress a button to get the reading.

► 5. If the tube gives an indication in the GOOD region of the meter scale, make any special tests the tester provides (such as a check for gas or for noise).

## OBSOLESCENCE

Tube testers are subject to rapid obsolescence. When new tubes are brought out which have a different base arrangement, it is necessary to adapt the tube tester to accommodate them. The early tube testers were frequently out of date within just a few months, because of the complex nature of their switching arrangements and the rapid introduction of new tubes.

Sometimes adapters were made available to prolong the useful life of the tester. These adapters were plugged into one of the sockets on the tube tester, and the new tube was plugged into the adapter. The wiring in the adapter was arranged to make it possible to test the tube. However, soon so many adapters were necessary that it became impractical to continue this system.

► Today, tube testers have switching arrangements designed to minimize the possibility of new tube arrangements making the tester out of date. Of all the testers, the emission type is the easiest to arrange in this manner, which is one of the very important reasons why such testers are more popular than more elaborate ones.

Today, most tester manufacturers release data on each new tube to purchasers of their equipment, giving the control settings for testing the tube.

► When you buy a tube tester, make sure that it is the latest model available and that it is a type which will not go out of date quickly. Even so, you can expect to replace tube testers from time to time with later models. Bear this item of service expense in mind, and set aside funds from service earnings so you can replace such test equipment when necessary.

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# Lesson Questions

Be sure to number your Answer Sheet 45RH-1.

Place your Student number on *every* Answer Sheet.

*Send in your set of answers for this lesson immediately after you finish them, as instructed in the Study Schedule. This will give you the greatest possible benefit from our speedy personal grading service.*

1. What two revitalization steps are usually made during each service job?
2. What are the two standard output levels used for sensitivity measurements?
3. What advantage does a serviceman find in using a signal tracer for sensitivity measurements rather than the standard input-output measurements?
4. If the manufacturer lists gain values for an r.f. stage at 600 kc., would you expect measurements at 1400 kc. to give the same values?
5. Why must the a.v.c. be disconnected and a fixed bias be used when making gain measurements?
6. Suppose you make a signal voltage measurement across the primary of a double-tuned i.f. transformer. Would you expect the secondary voltage to be: *1, greater than; 2, the same as; or 3, less than* the primary voltage?
7. When drying out a chassis, what is the purpose of the fan?
8. What are the three basic types of tube testers?
9. What test must always be made in the tube-testing procedure before the quality test is made?
10. Why should tubes be jarred while making a test for "shorts"?



## WASTED TIME

A minute seems such a little thing—something most of us thoughtlessly throw away. But, just as pennies make dollars, so do minutes make hours. Few people realize that ten minutes wasted daily make over sixty hours—more than a work-week—in a year's time.

Study the habits of most successful men and you will find that they made use of odd moments, reading or writing, or *thinking*. Those precious minutes gave them the *extra* weeks, months, and years of time necessary to prepare and to advance themselves.

Now, time spent in healthful recreation is not being wasted. But, how much of your time is spent in idle amusements instead? How much time do you waste "stalling" before starting a task—doing unnecessary or useless things—or doing nothing at all?

Study your actions during the day. Make a list of the things you do. You'll be surprised at the number of five- or ten-minute intervals you can put to better use, in studying or planning for the future. Be ready for your opportunity when it comes!

*J. E. Smith*