B. FOZARD RADIO SERVICING—Vol. 3 NORMAN PRICE

RADIO SERVICING

Vol. 3—Final Radio Theory

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This volume has been enlarged to cover the new syllabus of the examination in RADIO AND TELEVISION SERVICING held by the City and Guilds and the Radio Trades Examination Board.

The volume covers the requirements of the final examinations but it certainly should be read by those taking the intermediate examinations although only small sections may be directly applicable. It will be of help in both written and practical examinations as the student will be concerned with a complete receiver.

Service engineers and all those interested in the theory and practice of radio receivers will find this book extremely useful.

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CHAPTER 1

POWER SUPPLIES

METHODS of supplying high tension and heater current to radio receivers operating from the public supply mains or from dry batteries have been dealt with in Volume 2, Chapter 8 (third edition 1960).

We must now consider the problem of supplying power to equipment which must be used where no supply mains are available but where, on account of large size and power consumption, it would be far too costly to use dry batteries. Typical examples of such equipment are radio receivers (and transmitters); recording apparatus; public address amplifiers; cinema equipment; and television equipment which are installed in all kinds of vehicles, boats, country houses and the like. In most cases a low-voltage (usually 12V but 6V and 24V are also common) d.c. electrical system based on an accumulator battery will be installed. In other cases (*e.g.* in a country house or small ship) a supply may be available at medium voltage—60V or 120V. Here again the system is usually based upon an accumulator battery and so must be direct current.

Another problem arises. In this country most people have an a.c. supply, and so most radio sets and other equipment are made for use on a.c. only. Difficulties may arise when an a.c. only set has to be used on d.c. (*e.g.* when the owner removes to another district or when equipment has to be installed temporarily for public address or similar work).

In the above cases some form of **converter** must be used between the supply and the equipment. The converter may be built in (as in a car radio) or may be a separate unit (as in d.c. to a.c. conversion).

Three main types of converter are in common use:

1. **The vibratory converter.**
2. **The rotary transformer.**
3. **The rotary converter.**

All three contain moving parts and so they generate a fair amount of acoustic noise. They must be carefully boxed and flexibly mounted so that the noise is not objectionable and so that mechanical vibrations are not transmitted to valves and other components.

1. THE VIBRATORY CONVERTER

The complete converter in its best-known form consists of four main sections: the vibrator; a step-up transformer; a rectifier; and suitable filter and smoothing circuits.

The principle of operation may be explained with reference to figure 1.1. *AB* is a reed of spring steel clamped at the end *A* and carrying contacts at *B*. By means of a driving coil (not shown) the reed is caused to vibrate on the electric bell principle. As it vibrates it makes contact alternately with the fixed contacts *C* and *D*. The primary of the step-up transformer is centre-tapped to provide two sections *E* and *F*. When the reed makes contact at *C* current flows "downwards" through section *E* and when the reed swings over to *D* current flows "upwards" through section *F*. Thus, an alternating m.m.f. of approximately square waveform is set up by the currents in the primary winding. This produces an alternating flux in the transformer core and so an alternating e.m.f. is induced in the secondary winding. This e.m.f. is rectified and smoothed in the usual way, it is filtered to remove interference and then used as high-tension supply. Sparking occurs at the contacts and causes contact wear and interference. The sparking may be very nearly

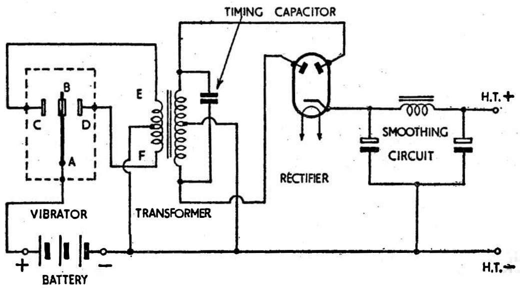


FIG. 1.1. BASIC CIRCUIT FOR VIBRATORY CONVERTER

eliminated by the connection of a **timing capacitor** across the transformer secondary. Converters of the type described are usually built in and supply h.t. current only, the valve heaters being connected directly (or in a series-parallel arrangement) across the battery. Recent developments have resulted in the production of vibrators capable of handling much bigger currents at the contacts and so giving greater output power. Such vibrators, in conjunction with a transformer, can give outputs of 100W or more at 230V and so enable the use of a.c. only equipment with low-voltage batteries.

Reed driving arrangement.—If the reed is to be kept going it must receive a driving impulse once during each vibration. Such impulses are produced by an electromagnet according to one of two methods (see figure 1.2). At (a) the reed carries an auxiliary contact which is used to energize a coil once per cycle when the reed bends to the left. To make sure of correct starting the reed may have a bias to the left so that the coil is in circuit at switching on. A simpler method is shown at (b) where the coil is connected across one of the pairs of main contacts. The reed has no bias but the driving coil is mounted to pull the reed to the left. On switching on, the reed is pulled to the left and the coil is short circuited by the closing of the main contacts. The reed is then released and moves over to the right by spring action. It then receives another impulse to the left and so on. The frequency of vibration of the reed (which is equal to the frequency of the alternating output voltage) is settled by its mechanical design (length, mass, etc.). Most vibrators operate at a comparatively high frequency (120c/s). This means that the reed can be made quite short ($\frac{3}{4}''$) and that the transformer and smoothing choke can be smaller and lighter than their 50c/s counterparts (see Volume 1, Chapter 13).

The driving coil is sometimes coupled to a short circuited coil wound on

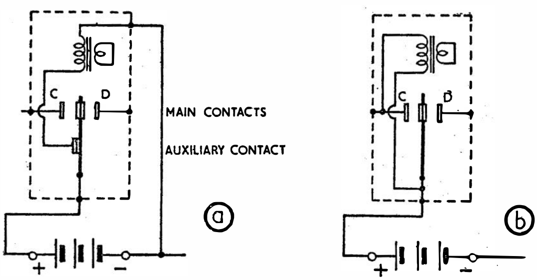


FIG. 1.2. VIBRATOR DRIVING ARRANGEMENTS

the same former. The function of the second coil is to absorb some of the energy in the magnetic field when the contacts open and so cut down sparking.

The Synchronous Vibrator.—By fitting an extra pair of contacts the vibrator may be used as a mechanical rectifier so obviating the need for a separate valve. Vibrators of this type are called **synchronous.**

A typical circuit for a power unit using a synchronous vibrator is shown in figure 1.3. Note that since the reed is common to both primary and second-

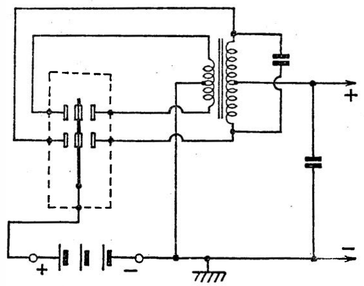


FIG. 1.3.
SYNCHRONOUS VIBRATOR
(Reed drive not shown)

ary circuits it must be at the negative end of the h.t. supply. Accordingly, the transformer secondary centre-tap forms the h.t. positive terminal. This is opposite to normal practice with mains-driven rectifiers and must always be borne in mind.

This difficulty can be overcome by using two reeds with a mechanical link between them but electrically insulated from one another. One reed is used in the primary circuit and the other in the secondary. This is the **split-reed** vibrator.

2. THE ROTARY TRANSFORMER

The rotary transformer, not to be confused with the static transformer which can be operated on a.c. supplies only, is a device for producing a high-voltage d.c. output from a low-voltage d.c. supply. Essentially it consists of a low-voltage motor driving a generator designed to give the required (high) output voltage. No rectifier is needed as the generator output is d.c. but filter and smoothing circuits must be fitted although component values may be unconventional from a 50c/s point of view.

Size, weight, and economy of battery power are essential features in any portable equipment. For these reasons the two machines comprising the rotary transformer are combined to the greatest possible extent. Thus, there is only one field system which is usually excited by permanent magnets or sometimes by a shunt winding connected to the low-voltage supply. The armature carries windings for both motor and generator in its slots, each winding being connected to a separate commutator, one at each end. The combination of two machines in one means that two bearings only are required, compared with four (or three) in a conventional motor-generator set. Losses in bearing friction are therefore reduced and the efficiency of conversion increased.

3. THE ROTARY CONVERTER

As is well known all rotating electrical machines carry alternating currents in their armature conductors, d.c. machines being fitted with a commutator which acts as a rectifier to produce a unidirectional output. If the armature winding of a d.c. motor is tapped at suitable points and the tappings are taken to slip-rings then an a.c. output will be available at the slip-rings whenever the motor is running. This is the principle of the **rotary converter** (strictly the name should be "inverter"). The general appearance is similar to that of the rotary transformer, there being a two- or four-pole field system and a single armature, but now the armature carries a commutator at one end and a pair of slip-rings at the other. As the armature winding is common to both input and output circuits these two circuits are not electrically independent. This may lead to difficulties with earthing in some cases. The frequency of the a.c. output depends upon the speed at which the converter runs. The frequency is usually arranged to be about 50c/s but it is not absolutely constant and may vary with the d.c. input voltage and with the load on the converter. Small changes in frequency are not usually important but may cause trouble with certain types of recording and playback equipment and with television receivers. The output voltage is usually 240V or thereabouts on full load. Here again the value depends on the d.c. voltage and on the load.

SUPPRESSION OF INTERFERENCE

All the converters mentioned depend for their action on the rapid reversal of currents by some form of mechanically-driven switch. This leads to the production, not only of acoustic noise, but also of high-frequency currents. These currents will cause severe background noise in the equipment associated with the converter (and in neighbouring radio receivers) unless they are suppressed at the source. Suppression capacitors are required between each brush and frame on rotary equipment and the frame must be efficiently earthed. For all converters adequate h.f. filter circuits must be incorporated in both input and output circuits to prevent mains-borne interference and the whole converter should be enclosed in an earthed metal box to prevent direct radiation. Further notes on interference suppression will be found in Chapter 9.

HEATER SUPPLIES

In recent years the practice of connecting the valve heaters all in series has become increasingly common. The idea was first used in a.c./d.c. sets but is nowadays quite common in sets for a.c. only. Two main factors have led to the widespread use of series heaters: it saves cost, size and weight by eliminating the mains transformer; modern valves are designed to work at h.t. voltages of 200 or less, and so can be fed from a half-wave rectifier connected directly to the (240V) mains. The simplest arrangement is that shown in

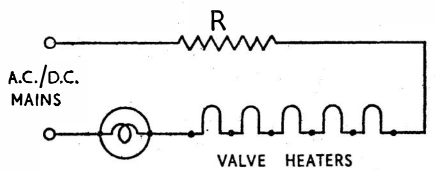


FIG. 1.4. VALVE HEATER SUPPLY

figure 1.4, in which the heaters and the dial lamps are joined in series with each other, and a suitable resistor R is used to "drop" the excess voltage (see Volume 2, Chapter 8). In general, the voltage across the resistor R is high—about 150V when the set is used on 240V mains—so that the power dissipated in the resistor is also quite high. In an older set using valves with 0·3 ampere heaters the power in R would be about 45W; in a modern set the heater current might be 150mA giving a power dissipation of 22·5W or so. In either case R must run at a high temperature in order to get rid of the heat. It has to be kept well away from other components (especially capacitors) and must be properly ventilated. The presence of a dropping resistor is one good reason why manufacturers recommend that a space of some 4 or 5 inches should be left clear behind a receiver. At one time it was a popular practice to incorporate the dropping resistor in the lead from the mains to the set. Special flexible cable was used known as **line cord** which, in addition to one or more insulated conductors of the usual stranded copper, contained a high-resistance lead in the form of a single strand of fine gauge resistance wire wound on an asbestos string, the whole being laid up and braided in the usual way. Such line cord, which is still obtainable although rarely used in modern sets, had a resistance rating (*e.g.* 120 ohms per foot) and a current rating (usually 0·2 or 0·3 ampere) depending on the gauge of resistance wire used. As the power dissipated in the line cord was spread over a length of six feet or so, the temperature of the cord was quite low—no more than warm to the touch. Line cord was widely used in a large number of sets of American origin which appeared in this country during and after the war. These sets were designed for operation on 110V mains and were easily converted for British use by fitting a suitable length of line cord in place of the original flex. These sets used 2-core line cord. The use of 3-core line cord is illustrated in figure 1.5 where the centre lead is used to supply the full

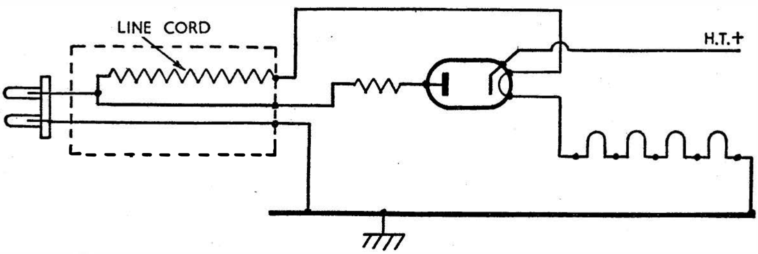


FIG. 1.5. USE OF 3-CORE LINE CORD

mains voltage to the rectifier. Line cord is more easily damaged than most flexible cables by abuse and is often a nuisance on account of excessive length. (Many people are sorry they shortened the "long flex" between set and socket.) The heating difficulties associated with a dropping resistor can be got rid of by using a choke or a capacitor in its place. This can only be done on a.c., of course. In these components the power loss is very small and so they do not get hot. However, they suffer from other disadvantages: choke is bulky and costly; capacitor must be suitable for a.c. working and can cause costly damage if its insulation fails. They are seldom found in commercial sets. A properly designed resistor, well ventilated and with no loose contacts to cause arcing and burning, makes for a cheap and reliable arrangement.

Use of Current Stabilizers.—Valve heaters for series operation are intended to be worked at the correct current value. If the current departs from the proper value by more than $\pm$ 7% for any length of time then, although the working of the set may not be affected to any noticeable extent, the life of the valves may be seriously shortened. Now the supply authorities are under a legal obligation to keep a consumer's supply voltage within $\pm$ 6% of the declared value and this they do except in times of emergency. It would seem that so long as a set is working at the right voltage tapping then variations in mains voltage would not affect the valves. In a set where the valve heaters are connected in parallel and fed *via* a transformer this is true, *i.e.* a $\pm$ 6% change in mains voltage produces a $\pm$ 6% change in heater voltage which is within the limits allowed. But in a series heater chain conditions are more complicated. Valve heaters are made from tungsten which has a relatively high temperature coefficient (0·0045 ohm/ohm/C°). Dropping resistors however, are wound with a nickel-chromium alloy which has a much smaller temperature coefficient (say 0·0001 ohm/ohm/C°). Suppose the mains voltage rises. The current in the heater chain rises and both valve heaters and dropping resistor get hotter. But the resistance of the dropper stays the same (nearly) while the resistance of the heaters goes up somewhat. The total increase in resistance is therefore less than it would have been if the circuit had consisted of heaters only. Therefore, the amount of current increase for a given change of supply voltage is greater in a series chain than in the parallel connection. The heater current can be kept constant for wide variations of mains voltage by replacing the series resistor, or a part of it, by a device called a **barretter.** This looks like a cross between a valve and an electric lamp; inside the bulb is an iron filament suspended in hydrogen. The barretter has the peculiar property that, over a certain range of voltage, the current which it will pass is almost independent of the voltage across it. A typical characteristic curve is shown in figure 1.6—after the initial rise the curve is flat. As the range of

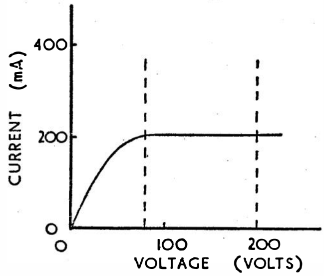


FIG. 1.6. CHARACTERISTIC OF BARRETTER

voltage over which the current is constant is large (typically 80-200 volts) the barretter is capable of dealing with large variations in mains voltage. If the total drop across the heaters is 70V the input voltage may vary over the range 150-270V without affecting the heater current. A set using a barretter does not need a voltage adjuster. Note that if the voltage across the barretter exceeds the rated maximum the current rises very rapidly and the filament burns out. This is unlikely to occur under working conditions but may happen at the moment of switching on.

Switching Surges.—Because of its high temperature coefficient the resistance of a valve heater when cold is likely to be about one-seventh of its normal working resistance when hot. The current flowing when the heater circuit is first switched on is thus much greater than the normal current. This happens whether or not a barretter is used as the following figures show. In the example above the normal voltage across the heaters is 70V. When the heaters are cold the voltage across them (with normal current) is $\frac{1}{7} \times 70 = 10$V. Thus, if the set is connected to mains of voltage greater than $200 + 10 = 210$V the barretter will be overloaded and will fail prematurely.

Where a simple series resistor is used it might be thought that the effect of the switching surge would not be serious. Something like two-thirds of the total voltage appears across the dropping resistor which has a nearly constant value and so the effect of the switching surge is less than in the parallel case. This is only true, however, if the valves all heat up together, and this does not occur in practice. The output valve and rectifier have much larger cathodes than the i.f. amplifier and double-diode-triode and take considerably longer to reach the operating temperature, and the dial lamps heat up fastest of all. Thus, during the heating-up period the lamps and the smaller valves have to pass an excessive current because the total resistance of the circuit is too low until all the valves have reached their working temperature.

As the heating-up period is quite long the effect on the valves is serious and many manufacturers take steps to reduce the overloading. The method usually adopted is to put in series with the heater chain a **current surge resistor** having a high negative temperature coefficient. Resistors of this type are made under the name *Thermistor* or *Brimistor* by Standard Telephones & Cables and associated companies. A typical thermistor for a 0·3 ampere heater circuit has a cold resistance of 3,000 ohms which drops to 150 ohms at the operating temperature. The graphs in figure 1.7 are drawn from measure-

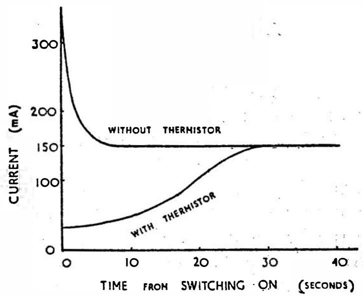


FIG. 1.7.
SWITCHING SURGE IN VALVE HEATERS

ments taken on an actual set with and without a thermistor in circuit. It will be noticed that the heating-up time is rather longer when the thermistor is used, which may be a nuisance but is more than compensated by the increase in valve life.

Thermistors are more often found in television heater circuits than in radio sets. The reason for this is that a television set has many more valves than a radio and the total drop across the heaters may be only 50V to 100V less than the mains voltage. In such a case the switching surge will be much more severe and the need for a thermistor more pronounced.

Example

In a radio receiver the total p.d. across the valve heaters is 50*V. If the heaters require* 0·2 *ampere what must be the value of the dropping resistor in order that the set may be used on* 240-*V mains? What will be the current flowing at the instant of switching on?*

Find the corresponding figures for a television receiver in which the total heater drop is 170*V at* 0·2 *ampere.*

Assume cold resistance is one-seventh of hot value.

RADIO

$$\text{Drop across resistor} = 240 - 50 = 190\text{V}$$

$$\therefore \text{resistance} = \frac{\text{V}}{\text{I}} = \frac{190}{0{\cdot}2} = 950 \textit{ ohms}$$

$$\text{Hot resistance of valve heaters} = \frac{50}{0{\cdot}2} = 250 \text{ ohms}$$

$$\therefore \text{cold resistance of valve heaters} = \frac{1}{7} \times 250 = 36 \text{ ohms}$$

$$\therefore \text{total resistance at switching on} = 36 + 950 = 986 \text{ ohms}$$

$$\therefore \text{current at switching on} = \frac{\text{V}}{\text{R}} = \frac{240}{986} = 0{\cdot}243 \textit{ ampere}$$

$$\text{Drop across resistor} = 240 - 170 = 70\text{V}$$

$$\therefore \text{ resistance} = \frac{V}{I} = \frac{70}{0\cdot 2} = 350 \textit{ ohms}$$

$$\text{Hot resistance of valve heaters} = \frac{170}{0\cdot 2} = 850 \text{ ohms}$$

$$\therefore \text{ cold resistance of valve heaters} = \frac{1}{7} \times 850 = 121 \text{ ohms}$$

$$\therefore \text{ total resistance at switching on} = 121 + 350 = 471 \text{ ohms}$$

$$\therefore \text{ current at switching on} = \frac{240}{471} = 0\cdot 51 \textit{ ampere}$$

Dial Lamps.—At one time the size and complexity of the dial and its lighting arrangements were important factors in the sales appeal of a set. A receiver might have upwards of a dozen lamps switched on or off according to the settings of various controls and push-buttons. In a.c./d.c. sets arrangements of this kind led to much complication—each lamp or group of lamps had to be shunted so that when it was switched off it did not cause an open circuit in the heater chain. Careful insulation from the chassis of the lampholders, wiring, and switches was needed. Shunting of the dial lamps is always desirable in a series heater chain. The lamps tend to fail before the heaters, especially where no thermistor is used, because of their small thermal capacity, and it is undesirable that the failure of a lamp should stop the set working. Some sets in current production do not use low-voltage dial lamps at all—a single 15W pygmy lamp running directly from the mains is used instead. Experience will show whether the high-voltage lamp is more reliable. Sometimes a lamp fails because it is situated too close to a loudspeaker or some other component which produces a stray magnetic field. Particularly if the set is a.c. operated, the lamp filament may vibrate strongly in the magnetic field and break at one of the supports after a short time. High-voltage lamps, which have long filaments, may suffer from this trouble unless they are carefully positioned.

CHAPTER 2

OUTPUT STAGES

In the larger domestic receivers and in amplifiers for public address and high-fidelity work power outputs in excess of those conveniently obtainable from a single valve are often required. Large valves are available which would give sufficient power but it is usually better in all ways to use two (or more) valves. Two methods of connection can be used.

1. VALVES IN PARALLEL

Figure 2.1 shows two output tetrodes connected in parallel. For anode, screen, and control-grid circuits, corresponding electrodes in the two valves are connected together. Note, however, that stopper resistors are usually necessary to prevent parasitic oscillations. The circuit gives twice the output power of one valve for the same drive but consumes twice as much h.t. current.

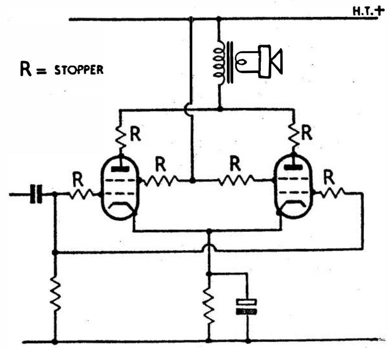


FIG. 2.1. VALVES IN PARALLEL

Valves in parallel are not commonly used in receiving and audio equipment. Four valves may be sometimes found in parallel push-pull.

2. VALVES IN PUSH-PULL

The basic circuit for two valves connected in push-pull is shown in figure 2.2. The two control grids are fed by signal voltages which are equal in magnitude but opposite in phase. Thus, as the anode current in one valve is increasing

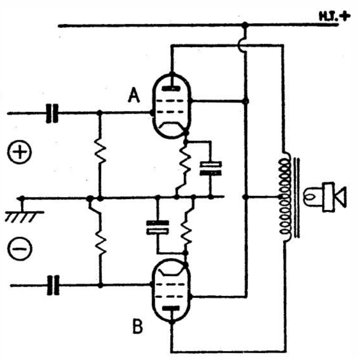


FIG. 2.2. VALVES IN PUSH-PULL

(push) the current in the other is decreasing (pull). This type of out-of-phase current variation is just the same as occurs in the two halves of a centre-tapped transformer winding. So the two anodes are connected to an output transformer having a centre-tapped primary winding, and the alternating anode currents add together to produce the signal flux in the core and the output voltage in the secondary. As can be seen, the circuit gives about twice the output power of one valve but requires twice the (peak-to-peak) drive while consuming twice as much h.t. current. On this statement the push-pull circuit seems to be not so good as the parallel circuit. As push-pull operation is very common it must have advantages in other ways, and here are some of the most important ones:

(*i*) **Effect on Output Transformer.**—The d.c. components of the anode currents produce m.m.f.s. in the output transformer which (theoretically) cancel each other out since they act in opposition. There is thus very little d.c. (static) magnetization of the core of the output transformer. The permeability of the iron and the primary inductance are therefore high, and the core can be made smaller for a given power handling capacity and distortion. In particular the performance of the output stage at low frequencies is improved.

(*ii*) **Effect on h.t. Line.**—It has been stated that the anode current of one valve rises as the anode current of the other valve falls. If the stage is balanced the rise and fall of current will be equal and so the total h.t. current taken will be constant. This means that there is no signal component due to the output valves in the power supply and so the chances of unwanted feedback *via* the h.t. line to other stages is greatly reduced. For the same reason any hum voltage on the h.t. line affects the anode circuits of the two valves in push-pull equally and in opposite phase—no hum is present in the output. It is therefore possible to supply h.t. to the output stage directly from the reservoir capacitor (even in a hi-fi amplifier if the capacitor is large, say 50μF) and so dispense with costly and bulky smoothing chokes.

(*iii*) **Distortion due to even harmonics is greatly reduced.**—If a pure sine-wave voltage is fed to the grid of a valve the voltage appearing at the anode will not be sinusoidal—it will be distorted because of curvature of the valve characteristic. Now the distorted wave can be looked upon as consisting of a mixture of sine-wave components of different magnitudes, frequencies, and phases. The frequencies are all integral (whole number) multiples of the frequency of the original signal. The component which has the same frequency as the original is called the **fundamental**; the components which have higher frequencies are called **harmonics.** The second harmonic has a frequency twice that of the fundamental; the third harmonic three times and so on. Just as 2, 4, 6, 8, etc., are called even numbers so the 2nd, 4th, 6th, 8th, etc., harmonics are called **even harmonics** while the 3rd, 5th, 7th, etc. (there is no first harmonic) are called **odd harmonics.**

Now consider figure 2.3. The full lines indicate one cycle of fundamental and two cycles of a second harmonic such as might be produced in the anode

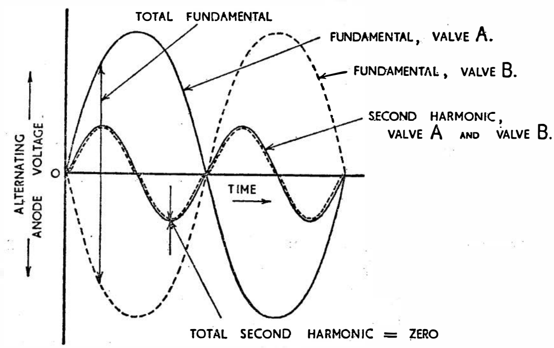


FIG. 2.3. ELIMINATION OF EVEN HARMONICS

circuit of a triode valve. The dotted lines indicate similar voltages generated at the anode of another valve connected in push-pull with the first. The total voltage appearing across the output transformer primary at any instant is given by the total distance from one curve to the other. For the fundamental voltages the total output is twice that of either valve but for the 2nd harmonics the curves fall on top of each other and so the total 2nd harmonic voltage is zero. Similar curves can be drawn for the other even harmonics and give the same result—harmonic output zero (or very small). This result applies only to the even harmonics. If figure 2.4 is studied it will be seen that in this case

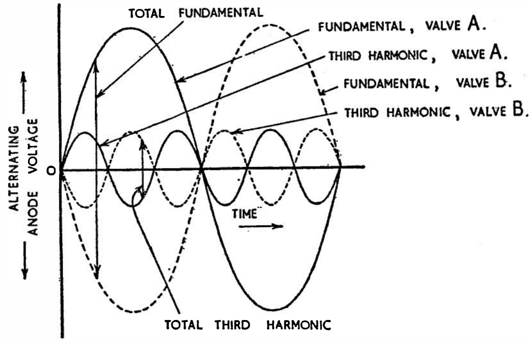


FIG. 2.4. EFFECT ON ODD HARMONICS

the harmonics add together and are reproduced as distortion in the output. Figure 2.4 is drawn for the third harmonic but the result applies to all the odd harmonics.

In general, the input is not sinusoidal (a musical instrument which produced a sinusoidal note would be very dull to listen to) and so itself contains harmonics. We must be careful to distinguish between those harmonics which are present in the original sound, and which must be maintained in their correct proportions, and those which are introduced in valve stages and produce distortion.

If the push-pull circuit eliminates the even harmonics only is its effect upon reducing distortion important? Yes, because for triodes at any rate most of the distortion produced is due to even harmonics. Many old (pre-war) output stages using triodes in push-pull are still giving good service. But most modern amplifiers use pentode or beam tetrode output valves for very good reasons (Volume 2, Chapter 7) and these types of valve tend to produce more third harmonic distortion than second. The designer can, however, by reducing the anode load, increase the second harmonic distortion and reduce the third and so take advantage of the push-pull connection. In modern circuits using pentodes and beam tetrodes in push-pull negative feedback is commonly used in an effort to further reduce distortion.

(iv) **Output valves may be operated with greater efficiency.**—It is shown mathematically in books on valve theory that by increasing the bias and the signal input to a power valve it is possible to operate the valve much more efficiently. That is to say a larger proportion of the power taken from the h.t. line appears as useful output in the loudspeaker or other load and less is wasted in heating up the anode. Great use is made of this result in r.f. power amplifiers for transmitters, etc. In such amplifiers the anode circuit is invariably tuned in some way and so the alternating anode voltage is bound to be sinusoidal (high-Q circuit) whatever the anode current waveform looks like. And the anode current waveform may be far from sinusoidal: see figure 2.5(a) which shows conditions in a valve working in what is known as class-B, *i.e.* biased nearly to the cut-off point. Compare the anode current waveform with that obtained when the valve is operated in class-A, figure 2.5(b), *i.e.* biased to a point roughly half way between zero bias and the cut-off point. In class-A the anode current is as near sinusoidal as possible; in class-B the anode current waveform looks like the output from an unsmoothed half-wave rectifier. But notice that the *average* anode current in class-B is less than in class-A and so the h.t. power required is less, thus giving the greater efficiency mentioned

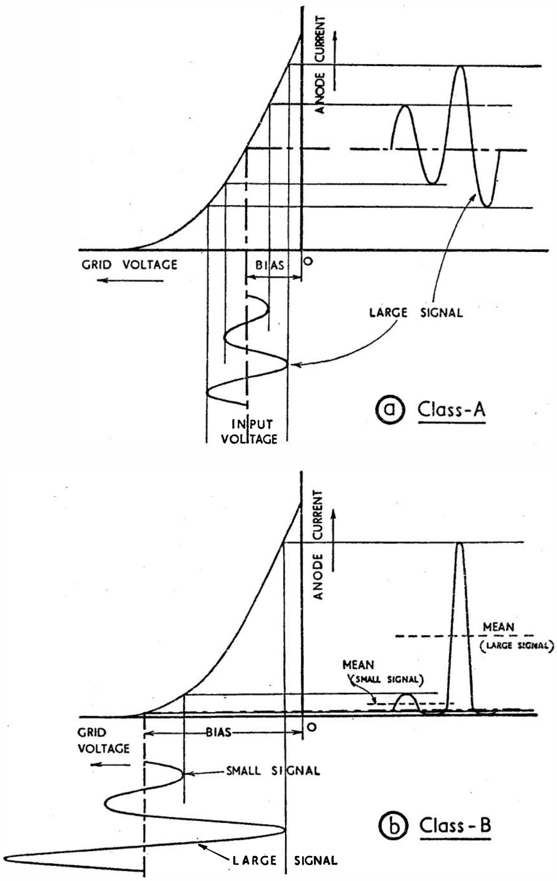


FIG. 2.5. OPERATION OF AMPLIFIER

above. For a.f. amplifiers there is a much more important result than the straightforward increase in efficiency. In any a.f. programme (speech or music) the time during which full power is required from the output stage is a small fraction of the total duration of the programme. For most of the time the signal to be handled is comparatively small. In a class-A amplifier the steady anode current is constant; the valve draws its 35mA (or whatever it may be) whether it is amplifying a very small signal or a very large one. In a class-B amplifier (see figure 2.5) the steady (average) anode current varies with the signal—on weak signals it is small and on strong signals large. Now, the true test of amplifier efficiency is how often the h.t. battery must be renewed or how many units are being taken from the mains per week. Thus, efficiency depends on electricity consumption which is proportional to current multiplied by time. Clearly, since a class-B amplifier operates with small anode currents for most of the time, it is going to be much more efficient than a class-A type. Again, a class-B amplifier for speech and music can be operated with a smaller power pack than an equivalent class-A amplifier. This means a considerable saving in initial cost and a greater expectation of valve life since neither output valves nor rectifier need be run at full rating; it also saves space and weight which are always at a premium especially in transportable equipment.

But the distortion introduced by working in class-B is intolerable. That is true for a single-ended stage but does it apply to the push-pull arrangement? If the valve characteristics were straight lines it would be possible to use class-B in an a.f. amplifier, for one valve would operate when the other was cut off and *vice versa.* Practical valves, however, have characteristics which are further from straight lines than many people appreciate and so a certain amount of distortion does occur, certainly more than could be tolerated in an amplifier which was supposed to give "faithful" reproduction. A compromise between efficiency and distortion can be effected by using push-pull and operating the valves with a bias between the class-A and class-B values. Such operation is known as class-AB. By careful design and the intelligent application of negative feedback it is possible to produce an a.f. amplifier capable of producing large output powers efficiently and with very little distortion. Such an amplifier will, in general, introduce less distortion than the loudspeaker (and pick-up) with which it is used. Class-AB operation has an advantage over class-B in that cathode bias can be used whereas a class-B amplifier requires fixed bias. The cathode by-pass capacitors must be large (100μF) in order to keep the bias voltage at its correct value on large signals.

PHASE SPLITTERS

The grids of a push-pull output stage must be fed with two voltages equal in magnitude and 180° out of phase with each other. The circuit used to produce such voltages is commonly called a **phase-splitting circuit.** There are two basic ways of arranging the circuit: one using a transformer; the other a valve or valves.

The transformer circuit is shown in figure 2.6. The primary of the transformer is shunt fed from the preamplifier anode in accordance with good

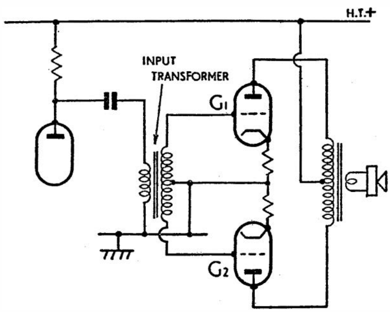


FIG. 2.6. TRANSFORMER PHASE SPLITTER

practice (Volume 2, Chapter 5, page 50) and the secondary winding is centre-tapped. The centre tap is connected to chassis and the two ends to the two grids. This arrangement is simple and gives excellent balance between the two halves of the output. Transformer coupling is not used nowadays because there are alternative circuits which give results as good as can be obtained from the best transformer at a fraction of the cost. The principal disadvantages of transformers are large size, weight, high cost or bad distortion, liability to pick up hum. Transformers are used for phase splitting in large amplifiers where the output valves may be driven into grid current. In such applications a path of low d.c. resistance from grid to cathode is essential. This can readily be provided by a suitable transformer, but is not so easily done in valve circuits.

A widely-used valve circuit is shown in figure 2.7. Sometimes called the **"concertina" circuit** it consists basically of a valve with equal load resistors in anode (R_a) and cathode (R_k) circuits. When a signal is applied to the grid of the valve it causes the anode current to vary. The same varying anode current flows through both resistors and so equal voltages are developed across them. The voltage across R_a is in antiphase with the grid voltage while the voltage across R_k is in phase with the grid voltage and so is in antiphase with the voltage across R_a. The two voltages are fed to the grids of the push-pull valves *via* the equal coupling capacitors C and grid resistors R_g. Coupling capacitors are required in both circuits because both anode and cathode are at a fair d.c. potential to chassis. Notice that there will be, in general, a d.c. voltage (about 100V) between heater and cathode of the phase-reversing valve. This should not cause difficulty with heater-cathode insulation breakdown in modern valves (this was a drawback in earlier days) but it may lead to hum, especially if the push-pull stages have a large voltage gain. In modern circuits it is usually only the output valves which are connected in push-pull. In such cases the small hum voltage introduced at the phase splitter is negligible. An important feature of this circuit is that the valve gives no gain—in fact the output voltage to each grid is a little (10 per cent. or so) less than the input voltage. This is because of the cathode load R_k which applies heavy negative feedback to the stage. The negative feedback has the effect also of increasing the apparent value of the grid resistor, so that the values of C_{in} and R_{in} given in circuit diagrams may appear rather small at first sight. The valve should have a high amplification factor. By-passing of the cathode bias resistor R_b is then unnecessary. Stray capacitances occur in this circuit in parallel with both R_a and R_k. That across R_k is usually the larger since it includes the cathode-heater capacitance of the valve. The output voltages may therefore be unequal at high audio frequencies but the effect is not serious.

Another circuit (**paraphase**) which can be used for phase-splitting (but is not so good as that of 2.7) is drawn in figure 2.8. G_1 is fed directly from the input and a fraction of the voltage applied to G_1 is taken from the tapped grid resistor P and fed to G_2 *via* a simple R.C.C. amplifier valve V. The gain of the amplifier must be adjusted so that the overall gain from G_1 to G_2 is unity—*i.e.* the gain of V is just equal to the attenuation of P. V thus provides no useful gain in the amplifier; its sole function is to reverse the phase of the grid voltage. Some people feel that the valve V is wasted and circuits have been devised in which a separate phase-reversing valve is not used. Two such circuits are given in figure 2.9. At (a) the potential divider P is placed across one half of the output transformer and the phase reversal is done in the output valve itself. At (b) the screen and grid of the first output valve are used as an internal triode and a suitable voltage, developed across the screen resistor, is fed to G_2. In this case the phase reversal is effected in the internal triode. The three circuits described are satisfactory and the latter two do a good job at low cost. In each case, however, there is no self-balancing action; any drift of valve characteristics would cause unbalance between the two output stages which would affect the reproduction.

Two further circuits will be described. They are used chiefly in high-quality amplifiers where cost is not quite so important as in a mass-produced domestic radio set. Both circuits give excellent results and there is little to choose between them. The first is in figure 2.10 and is a development of the paraphase circuit described above in which feedback is used to give self-balancing action. V_2 is the phase-reversing valve, its grid being fed partly from the anode of V_1 (so as to produce the correct phase) and partly from its

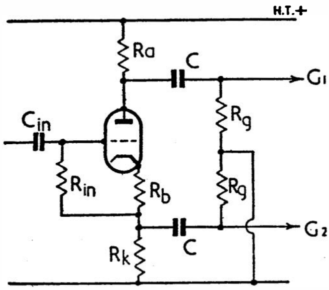


FIG. 2.7. "CONCERTINA" PHASE SPLITTER

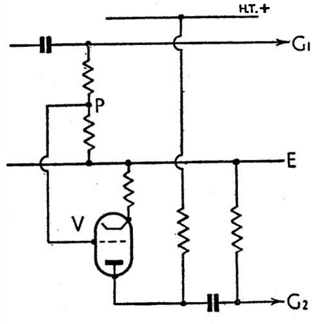


FIG. 2.8. PARAPHASE CIRCUIT

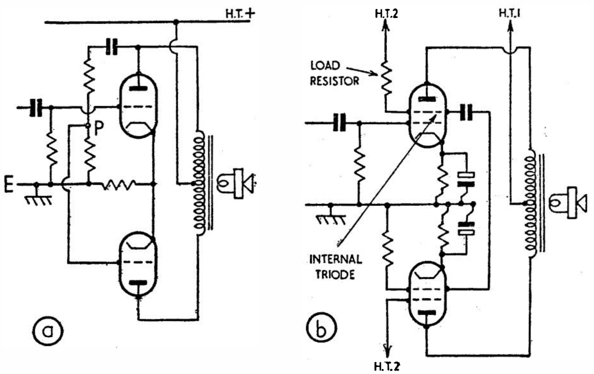


FIG. 2.9. "ECONOMY" PHASE SPLITTERS

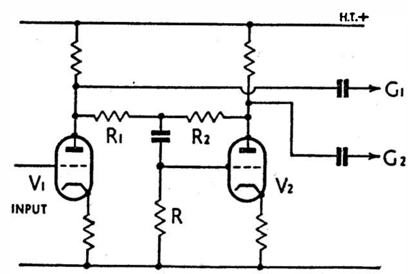


FIG. 2.10.
SELF-BALANCING PARAPHASE CIRCUIT

own anode (so as to produce the correct magnitude). If, for any reason, the output from V_2 departs from the correct value a compensating signal is fed back *via* the feedback resistor R_2 and balance is restored. V_1 and V_2 may be combined as a double triode, or pentodes may be used where high gain is required.

The other circuit is shown in figure 2.11. This circuit uses two valves with a common cathode resistor (long-tailed pair). The grid of the second valve

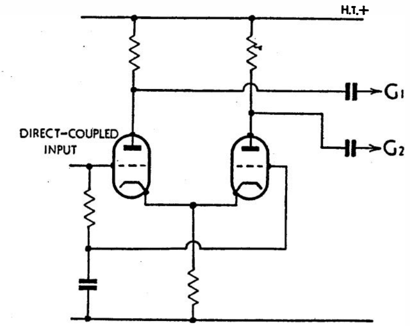


FIG. 2.11. LONG-TAILED PAIR

is earthed, the signal for this valve being injected at the cathode so giving the required phase reversal. High-gain valves are commonly used in this circuit in which case the anode load resistances may be equal without causing unbalance.

NEGATIVE FEEDBACK

Feedback is the term used in connection with a circuit when a part of the output is returned (fed back) to the input. Feedback may be intentional or not. Unintentional feedback often causes instability and may require circuits to be decoupled (see Volume 2, Chapter 11) in order to prevent oscillation. The principal effect of feedback is upon gain $\left(\text{ratio} \frac{\text{output}}{\text{input}}\right)$. If the gain of the circuit is increased by feedback, the feedback is called positive—too much positive feedback may cause instability or oscillation. If the gain is reduced the feedback is called negative. In a.f. amplifiers negative feedback is often used, but not, of course, primarily to reduce the gain. It so happens that by applying feedback to an amplifier in which the distortion is small, it is possible to reduce the distortion in the same ratio as the gain is reduced. Thus, a reduction of overall gain to one-fifth reduces the distortion to one-fifth, sufficient to make a worthwhile difference in the quality of reproduction. Where more gain can be sacrificed (*e.g.* 20 times or 26dB), as in a hi-fi amplifier, corresponding reductions can be made in distortion. In an average domestic radio receiver having one stage of i.f. amplification and a diode detector the a.f. amplifier usually consists of a high-gain triode or pentode voltage amplifier followed by a tetrode or pentode power valve. With modern valves the gain available between detector and speaker is greater than the amount required to fully load the output valve and speaker. It is therefore possible to reduce the gain by the use of feedback and so reduce distortion. In general, the voltage gain may be reduced by a factor of three to ten times (say 10 to 20dB). Negative feedback can be used to give other important effects, but as these effects are not usually required in radio receivers they will not be considered in detail here. Some of the principal effects are: (*i*) the gain of an amplifier may be made virtually independent of the constants of the individual valves. This is of importance in high-gain amplifiers for cathode ray oscilloscopes and valve voltmeters where constant gain is essential if the calibration of the instrument is to be relied on. (*ii*) The input impedance and the output impedance of an amplifier can be modified by the use of feedback. Thus, a cathode-follower stage which employs 100 per cent. negative feedback can be designed to give a low output impedance and can be used for direct matching between a high-frequency amplifier and a transmission line. Also, an a.f. amplifier can be so arranged that the output impedance, being low, acts as a damping resistance across the loudspeaker and output transformer and may cut down resonance effects.

Some of the effects of feedback depend upon whether the fed back signal is proportional to the output voltage (voltage feedback) of the amplifier or to the output current (current feedback). In so far as reductions of gain and distortion are concerned it does not matter which arrangement is used. Both types of circuit are thus found in domestic receivers, and sometimes a combination of the two is used.

Some typical feedback circuits in common use are illustrated in figure 2.12. At (a) the otherwise conventional output stage has no by-pass capacitor connected across the cathode bias resistor R_k. The signal voltage developed across R_k appears between grid and cathode in series with and in antiphase with the input voltage. There is thus negative (current) feedback. The gain of the stage is reduced to about one-half by this simple expedient. This is current feedback however, and so the output impedance of the stage is increased. This arrangement is used in cheaper sets.

Figure 2.12(b) shows a scheme for applying negative voltage feedback to an output valve, the anode voltage being applied to the grid in parallel with the input. Capacitor C acts as a d.c. block and should have low reactance compared with the resistors.

Figure 2.12(c) gives another circuit for feeding back from anode to grid.

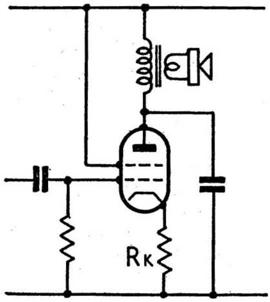
A common arrangement is shown in figure 2.12(d). Here the feedback is taken from the secondary of the output transformer and applied in the cathode circuit of the first valve. This scheme may be used where there are more than two stages and can be used for both single-ended and push-pull stages. In some cases a special *tertiary* (third) winding is provided on the output transformer to give the feedback voltage. Unless the terminals of the output transformer are carefully marked it is not possible to tell which way round the feedback circuit should be connected. A process of trial and error must be used. If the connection is wrong then positive feedback results and the amplifier may oscillate violently. It is always advisable to use an old speaker when reconnecting feedback lines (*e.g.* after replacing an output transformer).

Figure 2.12(e) is included to show how positive feedback may be used in a receiver. The circuit is that of a battery portable and the switch S is arranged to close when the lid of the set is closed. This connects the anode of the output valve to the *screen* of the preamplifier (compare with figure 2.12(d)). Positive feedback is then applied and the a.f. stages of the set oscillate to produce a warning note in the speaker.

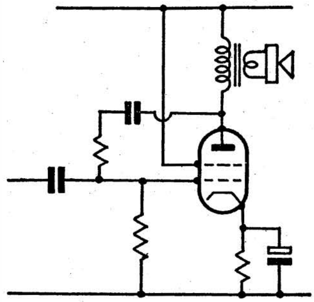
In the circuits described above the amount of feedback has been fixed by simple resistance potential dividers—any capacitors shown have been used as d.c. blocks and have had relatively high value (0·1 to 0·5μF) so as to present negligible reactance. Over the a.f. range the ratio of a resistance potential divider is very nearly constant, thus the effect of feedback in the circuits described above is independent of frequency. In the next section it will be seen that tone control can be conveniently arranged by putting reactive elements (*e.g.* capacitors) in feedback circuits.

TONE CONTROL

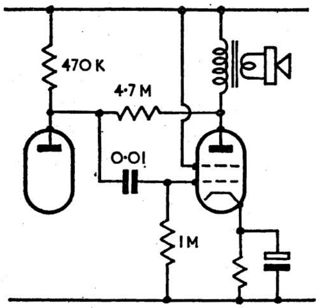
In order to give a perfectly faithful reproduction of the original sound entering the microphone, every item in the chain between microphone and loudspeaker (modulator, transmitter, receiver r.f., i.f., and a.f. amplifiers) should work in such a way that each frequency present in the original sound is reproduced in just the right amount. The original sound may consist of a mixture of all the frequencies which can be heard by the human ear and so the a.f. amplifier in the receiver should be "flat" *i.e.* its response should be



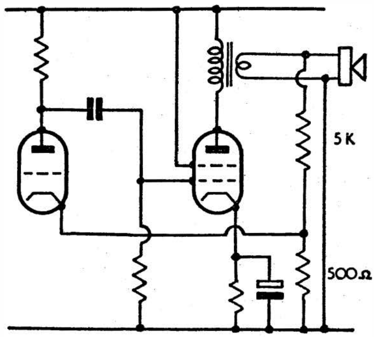
(a) Current Feedback



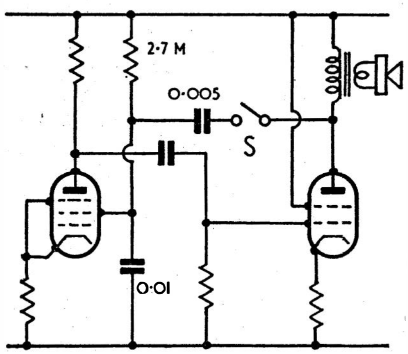
(b) Voltage Feedback over one stage



(c) Voltage Feedback to preceding Anode



(d) Feedback from Secondary of Output Transformer



(e) Positive Feedback

FIG. 2.12. NEGATIVE FEEDBACK

independent of frequency, from about 30c/s to about 15kc/s. This is not quite true when frequency modulation is used, nor, in general, is it true for gramophone amplifiers—the differences will be pointed out later. The attainment of such a flat frequency response is not possible in the average superheterodyne. The highest frequency is limited to about 5kc/s by the need for selectivity on medium waves in the i.f. circuits and so the reproduction is bound to lack brilliance. The worst effects occur, however, in the a.f. amplifier, in the output transformer, and in the loudspeaker and its cabinet. All these components are likely to cause both frequency distortion and amplitude distortion. The total effect is called the **tone of the set.** The designer, knowing that he cannot give perfection, must do the best he can to please the public at a reasonable cost. It seems clear that most people will not tolerate harsh sounds particularly at high frequencies, and since amplitude distortion has the effect of producing high harmonics the simplest and cheapest way of improving the tone of an amplifier is to cut out the offending high frequencies. Circuits used for doing this are called **tone compensation, tone correction,** or **tone control circuits.** The last term is commonly used when the listener is able to vary the effect.

In the main, therefore, tone control is used in the average set to reduce the high-frequency response. It has the additional advantage that it also reduces background noise, whistles, and/or needle hiss. But, of course, the quality of reproduced music suffers and speech is often well-nigh unintelligible. The circuits required to effect tone control in such cases are simple and well known and are described in Volume 2, Chapter 7.

Present-day equipment is capable of much better results. The top frequency has been extended far beyond 5kc/s by the use of v.h.f. broadcasting technique (f.m. for radio—a.m. for television) and microgroove recording. Distortion in the a.f. amplifier and output transformer has been reduced by the use of negative feedback which may also be used to damp objectionable resonances in the speaker. Thus, more advanced tone controls may be used. Their function is to allow for the difference between listening conditions at the microphone and at the speaker; or for individual taste, or to correct for defects in the equipment, *i.e.* for loss of sound output at the extreme top and bass frequencies.

Some of the components in tone control circuits must obviously be reactive, *i.e.* their impedance must vary with frequency. Thus, chokes or capacitors may be used. Chokes for a.f. work tend to be costly and heavy and may pick up hum: they are little used. The circuits to be described use capacitors as the reactive elements. A circuit embodying a choke may be understood if one remembers (*i*) that the impedance of a choke increases with frequency; and (*ii*) that it is possible to obtain resonance.

Where capacitors and resistors only are used for tone control it is not possible to obtain true **boost** either of top or bass. A so-called **bass-boost circuit** makes the full gain of the amplifier available at low frequencies and reduces the gain in the middle and upper registers. Such a circuit is shown in figure 2.13. At high frequencies the reactance of C is so small that it may be neglected and so the circuit acts as a simple potential divider and $\frac{Vo}{Vi} = R_2/R_1 + R_2$. At low frequencies, however, the reactance of C becomes large and the ratio $\frac{V_o}{V_i} = \frac{\sqrt{R_2^2 + X_c^2}}{\sqrt{(R_1 + R_2)^2 + X_c^2}}$ which may approach unity. The response curve of the circuit is shown in figure 2.13(b). The range of frequencies over which boosting occurs and the amount of boost are settled by the component values.

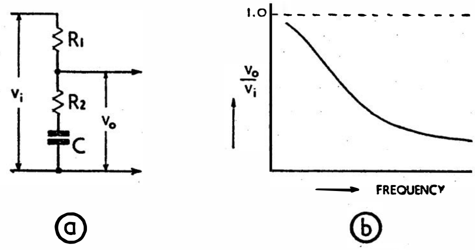


FIG. 2.13.
BASS-BOOSTING CIRCUIT

In a similar way the top frequencies may be accentuated (but only at the expense of the middle frequencies) by a top boost circuit of the type shown in figure 2.14. C is a small capacitance which may be assumed to have no effect

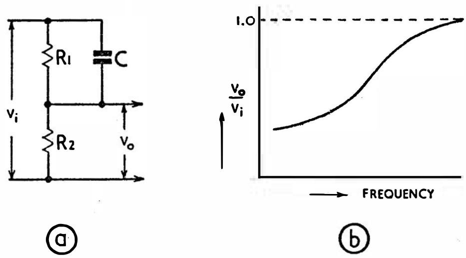


FIG. 2.14.
TOP-BOOSTING CIRCUIT

at middle frequencies. The ratio $\frac{V_o}{V_i}$ is thus $R_2/R_1 + R_2$ at middle frequencies as before. At the high frequencies, C provides a low impedance shunt path across R_1 and so the output rises and $\frac{V_o}{V_i}$ approaches unity.

It will be seen that there is a loss of gain in the amplifier at middle frequencies which is equal to the maximum amount of boost required. As this may amount to 20dB (10 times) then the provision of an **advanced tone control** very often requires the use of an extra valve.

Figure 2.15 shows a tone control stage in which both top and bass frequencies may be boosted or cut as desired.

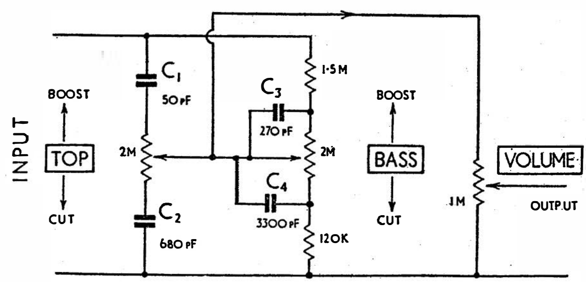


FIG. 2.15. TONE CONTROL CIRCUIT FOR HIGH-FIDELITY AMPLIFIER

A tone control circuit using negative feedback is shown in figure 2.16. When the slider S of the control is at the bottom there is no feedback and no

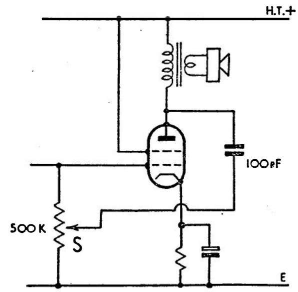


FIG. 2.16.
TONE CONTROL BY NEGATIVE FEEDBACK

tone control. As S is moved upwards some of the output voltage is fed back to the input and so the gain of the stage is reduced. Because of the varying reactance of C the highest feedback occurs at high frequencies. The circuit thus gives varying amounts of top cut. Circuits of this kind are very common.

COMPENSATED VOLUME CONTROL

Even if the whole chain of equipment between artiste and listener has perfect frequency characteristics, the reproduction would only sound natural if the volume level at the receiver were the same as that in the studio. Usually, the reproduced sound level is low and because of the way in which the human ear responds there is a pronounced fall in the apparent bass and a less noticeable fall in the apparent treble. Thus, if the reproduction is to sound natural at reduced volume there must be some degree of bass and treble boosting. This can be effected by the use of a circuit such as that shown in figure 2.17. Here a special tapped volume control is used and the lower portion is shunted in order to give the required boost.

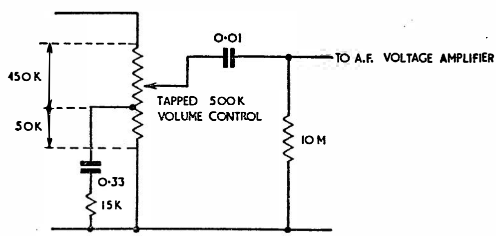


FIG. 2.17.
COMPENSATED VOLUME CONTROL

CHAPTER 3

THE SUPERHETERODYNE RECEIVER

THE intermediate frequency (i.f.) used in modern superheterodyne receivers for a.m. reception is usually 465kc/s or some very similar value. Occasionally, a frequency of about 110kc/s is used, which was at one time common but is not so much used nowadays. If one understands fully the reasons for the choice of a particular frequency for the i.f. then one will be able to

TABLE I

i.f. (kc/s)	(1) Percentage off tune at aerial (%)	(2) Frequency of interfering station in i.f. amplifier (kc/s)	(3) Percentage off tune in i.f. amplifier (%)	(4) Second channel frequency (Mc/s)	(5) Percentage off tune of second channel frequency in preselector (%)	(6) Oscillator frequency (Mc/s)	(7) Percentage difference of oscillator and signal frequencies (%)
465	$\frac{10{\cdot}01 - 10}{10} \times 100 = 0{\cdot}1$	455	$\frac{465 - 455}{465} \times 100 = 2{\cdot}15$	$10 + (2 \times 0{\cdot}465) = 10{\cdot}93$	$\frac{10{\cdot}93 - 10}{10} \times 100 = 9{\cdot}3$	$10 + 0{\cdot}465 = 10{\cdot}465$	$\frac{10{\cdot}465 - 10}{10} \times 100 = 4{\cdot}65$
110	0·1 as above	100	$\frac{110 - 100}{110} \times 100 = 9{\cdot}1$	$10 + (2 \times 0{\cdot}11) = 10{\cdot}22$	$\frac{10{\cdot}22 - 10}{10} \times 100 = 2{\cdot}2$	$10 + 0{\cdot}11 = 10{\cdot}11$	$\frac{10{\cdot}11 - 10}{10} \times 100 = 1{\cdot}1$

CARRIER FREQUENCY = 10 Mc/s

understand better why the superhet is so widely used. Some of the main factors involved are:

(*i*) The i.f. must be high enough to "carry" the modulation without introducing distortion. Now, a good rule is that, in general, the carrier frequency should be at least ten times the highest modulating frequency. Thus, if the highest a.f. is taken to be 10kc/s the i.f. must be at least 100kc/s.

(*ii*) The i.f. should be as low as possible in order that stable high-gain amplification may be achieved. This, remember, is one of the two basic ideas upon which the operation of the superhet rests. Most of the gain between aerial and speaker occurs in the i.f. amplifier so that the i.f. must be low enough to enable much higher gain to be used than in an amplifier working at the signal frequency.

(*iii*) The i.f. must not coincide with any powerful transmitting station otherwise there will be i.f. breakthrough (Volume 2, Chapter 12) which will cause severe interference. Now, 465kc/s corresponds to 645 metres while 110kc/s corresponds to 2,730 metres: both these frequencies fall outside the medium and long wave broadcast bands. Notice that this condition may be harder to fulfil in a television receiver where the i.f. must be 10Mc/s or more.

Looking at these three reasons it would appear that of the two frequencies (110kc/s and 465kc/s) the former should be the better. However, we must also take into account the question of selectivity. We have already seen that the superhet is more sensitive than the t.r.f. receiver because of its stable, high-gain "low" frequency amplifier. It is also much more selective because after the frequency changer the *percentage* frequency difference between two stations is increased. Consider the effect on medium waves with the required station on 1000kc/s (300m) and an interfering station on 1010kc/s (297m).

The percentage difference in the aerial circuit is $\left[{}^{(1010-1000)}\big/_{1000}\right] \times 100$ $= 1$ per cent. If the i.f. is 465kc/s and the oscillator is set to tune in the required station then the unwanted station will produce a frequency at the mixer anode of $1000 + 465 - 1010 = 455$kc/s. Thus, the percentage off tune in the i.f. circuit is $\left[{}^{(465 - 455)}\big/_{465}\right] \times 100 = 2{\cdot}15$ per cent. If the i.f. is 110kc/s, however, the percentage off tune in the i.f. amplifier is $\left[{}^{(110 - 100)}\big/_{110}\right] \times 100 = 9{\cdot}1$ per cent. It therefore appears that while the superhet is more selective than the t.r.f. receiver, the low i.f. gives better results than the high one for an interfering station on an *adjacent channel.* The *second channel* frequency (see Volume 2) for the above example will be $1000 + (2 \times 465) = 1930$kc/s for the 465kc/s i.f. and $1000 + (2 \times 110) = 1220$kc/s for the 110kc/s i.f. Now it would be a pretty poor preselector stage which was incapable of separating a signal on 1,000kc/s from one on 1,220kc/s, so obviously second channel interference is not a serious problem—not on the long and medium wavebands. But, supposing we equip our set with a short wave band, as manufacturers first began to do around the year 1935. The figures in Table 1 are worked out for two stations separated by a frequency of 10kc/s as before, but the carrier frequency of the wanted station is now 10Mc/s (30 metres).

Column (3) gives the same results for adjacent channel selectivity as on medium waves.

Column (5), however, shows that there is only a 2 per cent. frequency difference at the preselector between the two stations at the lower i.f. If the preselector stage consists of a single tuned circuit it will not be able to separate the wanted signal from the unwanted one—both programmes will be heard at the loudspeaker. The higher i.f., however, gives a percentage difference of over 9 per cent. and so the risk of second channel interference is reduced and may be serious only if the interfering station produces a very strong signal at the receiver aerial.

Another important result appears in column (7) where the percentage difference between the signal and oscillator frequencies is given. This difference is only about 1 per cent. for the 110kc/s i.f. and would certainly cause "pulling" of the oscillator. That is to say that, because of slight capacitive or magnetic coupling between signal and oscillator circuits, there would be a great tendency for the oscillator to become pulled into step, or synchronized, with the signal. In such a case, of course, there would be no i.f. and the set would stop working. For the higher i.f. the percentage difference is much higher and the chances of pulling considerably less, provided the circuit is well laid out to avoid stray couplings and a suitable valve is used as frequency changer.

Clearly, a balance must be struck between adjacent channel interference and stability (which are better at low i.f.) on the one hand and second channel interference and oscillator pulling (which are less serious with high i.f.) on the other. If the set is to operate on short waves with the minimum of complexity in the preselector then the higher i.f. is essential. Notice that a 465kc/s i.f. gives a percentage off tune in the i.f. amplifier in the example considered of only 2 per cent. or so. This difference would cause difficulties in separation if a single tuned circuit were used. In practice, of course, there are at least four tuned circuits in the i.f. stages and so the figure of 2 per cent. does not usually present a great problem. It can also be shown that a set with a higher i.f. is less likely to suffer from whistling than one with a low i.f. A detailed discussion of this problem is outside the scope of this book.

CHAPTER 4

FREQUENCY CHANGING IN THE SUPERHETERODYNE RECEIVER

THE object of the frequency changer (or converter) stage in a superheterodyne receiver is to combine the signals from the aerial and from the local oscillator in such a way that the intermediate frequency signal is produced, ready for feeding to the i.f. amplifier. The frequency changer stage usually includes the local oscillator in addition to the actual frequency-changing (or mixing) circuit and may be arranged in several ways:

(*i*) Two separate valves may be used: one as oscillator and the other as mixer. This arrangement, common in the early days, is now found quite often in v.h.f. (f.m.) sets.

(*ii*) The same basic idea as in (*i*) may be adopted with the two valves enclosed in a common envelope. This scheme has been very common indeed for many years.

(*iii*) A special valve may be used in which both oscillator and mixer are truly combined in one valve, *i.e.* the same electrons pass through the two sets of electrodes in cascade. This type of frequency changer is not so suitable for short wave work as those above. Its main field of use nowadays is in battery portable receivers.

Although the valves used for frequency changing by the methods outlined above differ in principle and in construction, the circuits associated with them have many points of similarity as will be seen later. Thus, we can draw and describe a circuit in association with one type of valve and with very little modification use it with another type of valve.

FREQUENCY CHANGERS

As mentioned above the term "mixing" is often used for the frequency changing process. This is not a very good term since mixing of the two signals is not enough. If the two signals were fed, in series or parallel, to the grid of an amplifier working under class-A conditions the waveform of the anode voltage would appear as shown in figure 4.1. This shows a true mixing—a true addition—of the two signals such as is used in a.f. studio equipment for

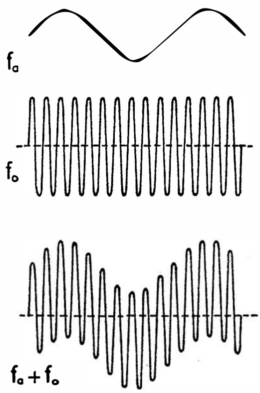


FIG. 4.1.
SIMPLE ADDITION OF TWO SIGNALS

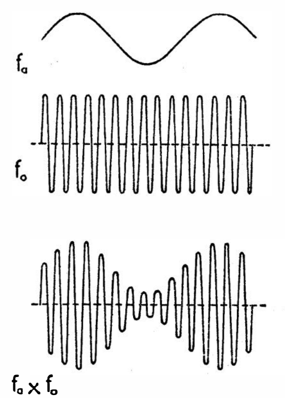


FIG. 4.2.
MULTIPLICATION OF TWO SIGNALS

(NO MODULATION OF THE AERIAL SIGNAL (f_a) IS SHOWN)

mixing or combining the output of several microphones or other items of gear. Such a mixture does not, however, yield the sum and difference frequencies which are essential to the operation of a superhet. In order to obtain these frequencies the two signals must be combined in a special way (in a process which is rather similar to modulation in a transmitter). The type of output waveform obtained in this case is shown in figure 4.2. It can be shown mathematically that a wave of this kind contains the required sum and difference frequencies. There are two principal methods of obtaining this form of mixing:

1. **Additive mixing.**—Here the signals from aerial and oscillator are added together as mentioned above and are then fed either to a rectifier or to an amplifier in which they are deliberately distorted. Additive mixing can be accomplished in ordinary valves (in triodes if required) and although rarely used in commercial receivers for a.m. reception it has become quite common in v.h.f./f.m. sets. Notice the reference above to the use of a rectifier (detector) in an additive mixer stage—for this reason the frequency changer is sometimes referred to as the "first detector". The second detector, of course, is the one used after the i.f. amplifier to extract the modulation from the i.f. carrier.

2. **Multiplicative mixing.**—In additive mixing the two signals are added together and then distorted. In multiplicative mixing the same result is

achieved by multiplying the two signals together electronically, *i.e.* the anode current of the mixer valve varies according to $f_o \times f_a$ where f_o and f_a are the instantaneous valves of oscillator and aerial voltages respectively. This operation requires a special valve with two signal grids constructed in such a way that the mutual conductance between the second signal grid and the anode is proportional to the voltage applied to the first signal grid. This idea is not difficult to understand if one understands clearly what is meant by mutual conductance (Volume 2, Chapter 3, page 33). Multiplicative mixing can be done in an r.f. pentode (using signal and suppressor grids) or even in a tetrode but it is usual to employ a valve specially designed for the purpose so that we may say that the simplest practical form of multiplicative mixer

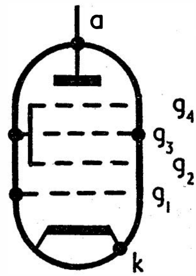


FIG. 4.3. HEXODE VALVE

is the hexode (six electrodes) shown symbolically in figure 4.3. There are actually four grids, numbered outwards from the cathode, thus:

g_1 and g_3 are the two signal grids; the oscillator voltage is fed to one (usually g_3) and the aerial voltage to the other. The mutual conductance between g_3 and the anode is proportional to the voltage applied to g_1. g_2 and g_4, as shown in the diagram, are usually connected together internally. Their function is to completely screen g_3 from g_1 and from the anode. g_2 and g_4 can thus be compared with the screen grid in a tetrode valve—but in the hexode screening is required on both sides of g_3. The screen grids g_2 and g_4 must be supplied at a suitable d.c. potential and must be decoupled to earth. A development of the hexode is often found. This is the heptode (seven electrode) valve in which an additional electrode in the form of a suppressor grid is inserted between g_4 and the anode—g_5 in figure 4.4. The

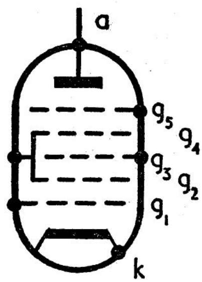


FIG. 4.4. HEPTODE MIXER

additional grid, which is connected to cathode or earth, tends to improve the valve characteristics in the same way as the suppressor grid in a pentode valve. In particular, the a.c. resistance is increased—this reduces the damping on the i.f. transformer and makes practicable the use of high-Q coils.

In both hexode and heptode it is possible to give the valve variable-mu characteristics so that it can be used for automatic gain control. A word of explanation is required here. The mutual conductance which is varied in order to control the gain is that between g_1 and the anode. A circuit of this kind is shown in figure 7.1.

A special term is required to describe the gain of a frequency changer stage since the output voltage at the anode is at a different frequency from the input voltage at the grid (usually g_1). The ratio of these two voltages is called the **conversion gain.** For similar reasons the effective mutual conductance of the valve (looked upon as an amplifier) *i.e.* the ratio

$$\frac{\text{i.f. alternating anode current}}{\text{signal grid voltage}}$$

is referred to as the **conversion conductance.** It is reasonable to expect that both conversion gain and conversion conductance will depend upon the voltage applied at the oscillator grid (g_3) and this is in fact true. For best results the oscillator voltage should be quite close to a particular value known as the **optimum heterodyne voltage.** If the oscillator operates too strongly or too weakly the sensitivity of the set is adversely affected.

The hexode and the heptode are both widely used, commonly combined with a small triode (for use as oscillator) in a single "bottle" and forming the well-known triode-hexode and triode-heptode frequency changers. The symbol for a triode-hexode is shown in figure 4.5. Note the internal connection between the grid of the triode and the injection grid g_3 of the hexode. Note also that because of internal connections between various electrodes it is possible to accommodate these valves on standard 8-pin or 9-pin bases.

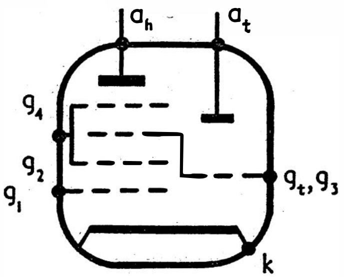


FIG. 4.5.
TRIODE-HEXODE FREQUENCY CHANGER

SINGLE-VALVE FREQUENCY CHANGERS

These valves, as mentioned in (*iii*) on page 30 combine oscillator and mixer in one electrode system. The symbol for such a valve is shown in figure 4.6—

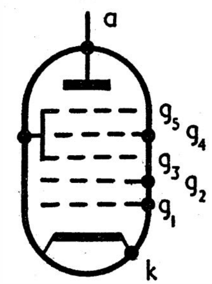


FIG. 4.6. EARLY TYPE OF HEPTODE FREQUENCY CHANGER

it has five grids (pentagrid) and so is called a **heptode.** It is important to realize, however, that the electrode arrangement and operation of this valve are quite different from those of the mixing heptode described above. The grids are numbered outwards from the cathode and perform the following functions:

g_1 and g_2 act as the control grid and anode respectively of a triode oscillator. Thus, the electron stream emerging from g_2 is modulated by the oscillator signal.

g_3 and g_5, which are connected together, act as screens, totally enclosing g_4.

g_4 is the signal grid, connected to the aerial input circuit.

The valve can be used for automatic gain control since the conversion gain can be varied by altering the bias on g_4.

The main differences between this valve and the mixing heptode are (*i*) electron coupling is used between oscillator and mixer, and (*ii*) the aerial signal is introduced at the grid nearer to the anode. Basically these differences should have no effect on the operation. In practice this is not the case, especially at high frequencies. Let us look briefly at the problems involved.

Electrons pass through the triode part of the valve and come into the zone of influence of the screen grid g_3. Most of the electrons pass through g_3 (although some are trapped, of course) and enter the region between g_3 (positive) and g_4 (negative). The electric field between g_3 and g_4 retards the electrons and a space charge builds up similar to that surrounding the cathode in any valve but with an important difference; the space charge between g_3 and g_4 in the heptode is varying in intensity at oscillator frequency. This causes a voltage at oscillator frequency to appear at the signal grid g_4 and unless an r.f. amplifier is used in the preselector stage the oscillator signal may be radiated from the aerial and cause interference with neighbouring sets. This effect is more apparent at high frequencies (even at the h.f. end of the medium waveband) since the percentage difference between oscillator and signal frequencies is then less. Again, since the oscillator frequency is always higher than the signal frequency in the kind of receiver under consideration, the aerial tuned circuit will exhibit capacitive reactance to the oscillator voltage induced on g_4. Thus, the two voltages on g_4 are out of phase and tend to cancel each other out, causing a reduction in conversion conductance and a loss of sensitivity. This effect may be more serious than radiation from the aerial. Both effects may be reduced by neutralizing, for the space-charge coupling is roughly equivalent to a small negative capacitance which may be neutralized by connecting a very small capacitance (*e.g.* $2\mu\mu$F) or a suitable C-R circuit between g_1 and g_4. The components of the neutralizing circuit may be located within the valve envelope.

From the heptode described above was developed the octode frequency changer by the addition of a suppressor grid between g_5 and the anode. The modern frequency changer of this type has five grids and is called a heptode, but is, in fact, an octode with one of the screen grids removed. Its symbol is shown in figure 4.7.

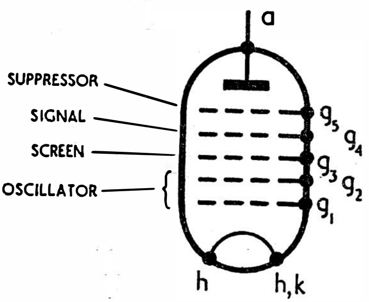


FIG. 4.7.
MODERN TYPE OF HEPTODE FREQUENCY CHANGER

There are therefore three different forms of the heptode valve: (*i*) the mixer valve of figure 4.4 (DK91, X17, etc.); (*ii*) the early frequency changer of figure 4.6 (DK32, etc.), and (*iii*) the modern frequency changer of figure 4.7 (DK92, X18, etc.; EK90, 6BE6, etc.; DK96, etc.). Notice that these are nearly all battery valves; the heptode, with its single electrode system, can be made to operate with less filament power and less anode current than a double valve of the triode-hexode type.

Also, most battery receivers do not have a short-wave band and so the difficulties associated with heptodes at high frequencies do not arise. For mains-driven sets the triode-hexode (ECH35, etc.) and triode-heptode (ECH81, etc.) valves are almost always used.

FREQUENCY CHANGER CIRCUITS

Typical basic circuits for various types of frequency changer are given in figure 4.8, triode-hexode; figure 4.9, modern heptode. Figures 4.10 (a) and (b) show the use of a heptode mixer (*e.g.* DK91 or 6BE6) as an electron-coupled frequency changer by using either the cathode or the screen grids as "live" electrodes.

Since additive mixers are used mainly in v.h.f. reception further consideration of these has been left until Chapter 11.

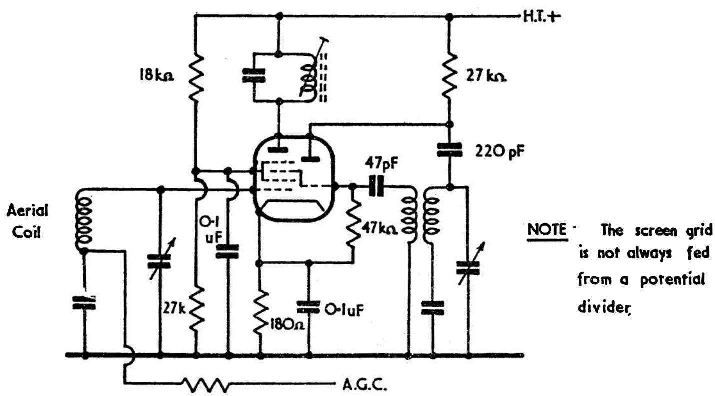


FIG. 4.8. BASIC TRIODE-HEXODE FREQUENCY CHANGER CIRCUIT

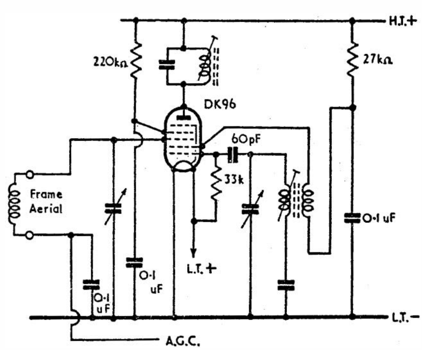


FIG. 4.9. SIMPLIFIED CIRCUIT FOR HEPTODE FREQUENCY CHANGER

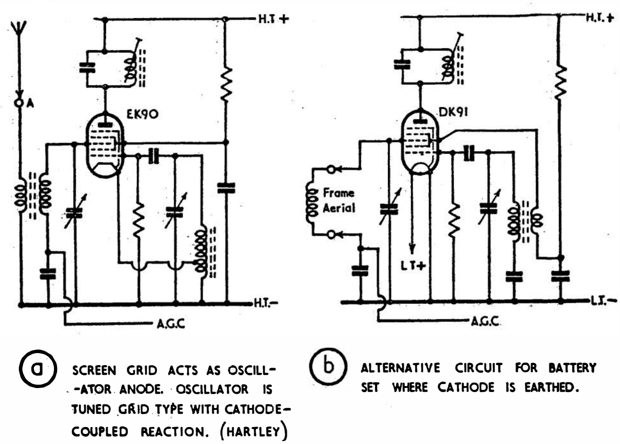


FIG. 4.10. USE OF HEPTODE MIXER AS ELECTRON-COUPLED FREQUENCY CHANGER (a) MAINS CIRCUIT; (b) BATTERY CIRCUIT

CHAPTER 5

OSCILLATORS

BASIC oscillator circuits have been considered in Volume 2, Chapter 11, where it is shown that the tuned-grid and tuned-anode circuits are commonly used in receivers for reception of frequencies up to about 20Mc/s (15 metres). These circuits can be made to work well at such frequencies if suitable precautions are taken and have the advantage that they can be arranged for band switching without great difficulty. At higher frequencies these circuits become awkward to manage and other arrangements must be used. Two important circuits for high-frequency oscillators are shown in figures 5.1 and 5.2. These are the Hartley and Colpitt's circuits respectively, which, with their variations, are widely used but are not always easily recognized in a circuit diagram. Bias for these oscillators is obtained in the usual way by means of a capacitor and leak in the grid circuit. The required 180° phase-shift between grid and anode is obtained in the Hartley circuit by the use of a tapped coil and in the Colpitt's circuit by means of a centre-tapped tuning capacitor.

Returning to the simpler circuits the following points should be borne in mind:

(*i*) **Tuned anode.**—In this case the parallel-fed circuit is always used in order to allow earthing of the tuning capacitor, but in some cases the d.c.

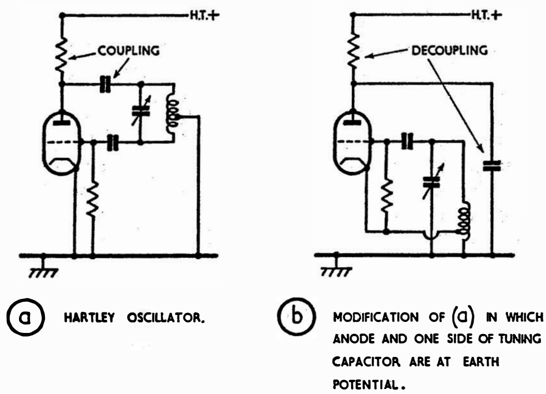


FIG. 5.1. HARTLEY OSCILLATOR CIRCUITS

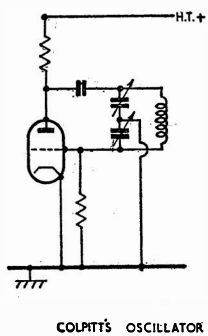


FIG. 5.2. COLPITT'S OSCILLATOR CIRCUIT

blocking capacitor may be omitted. This means that there is h.t. on one side of the tuning capacitor.

(*ii*) **Tuned grid.**—In this circuit either the series or parallel-fed arrangement may be used. In the parallel-fed circuit the reaction coil in the anode lead may be omitted, coupling between anode and grid being provided by the common impedance of the tracking capacitor. Some circuits employ a combination of reaction coil and common impedance coupling. A circuit of this type is shown in figure 5.3.

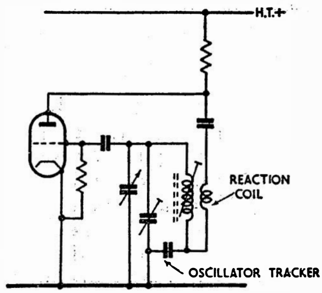


FIG. 5.3.
TUNED-GRID OSCILLATOR WITH REACTION COIL AND COMMON IMPEDANCE COUPLING

OSCILLATOR STABILIZING

As has already been pointed out the voltage to be applied at the oscillator grid of the mixer valve should have a certain optimum value. Departure from this optimum by a volt or two is not serious but larger variations may lead to lack of sensitivity, generation of whistles and/or increase in background noise in the mixer valve. Now the Q of the oscillator tuned circuit varies with frequency and so the oscillator operates much more strongly at the high frequency end of a band. If the valve is arranged to give the optimum heterodyne voltage at the high frequency end then at the low frequency end the oscillator voltage may be so low as to be ineffective in producing i.f. If, on the other hand, the oscillator is adjusted to give the optimum heterodyne voltage at the low frequency end the mixer may be over modulated at the high frequency end of the band. Some means of stabilizing the oscillator output voltage is obviously desirable One of the features of the grid leak bias arrangement always used with oscillators is that an increase in oscillator voltage automatically produces a bigger negative bias and so tends to reduce the output and *vice versa.* In many cases the stabilizing effect of this bias circuit is considered to be sufficient. Very often, however, additional stabilizing is employed, effected by the connection of suitable resistance in series or in parallel with the oscillator tuned circuit or the reaction coil. Such resistances vary in value according to the design of the circuit from a few ohms to several kilohms. In some cases the required resistance can be most easily obtained by winding a coil with resistance wire. Typical circuits are shown in figure 5.4.

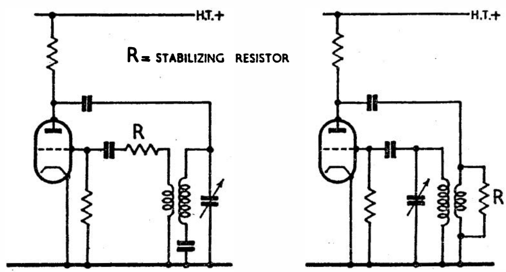


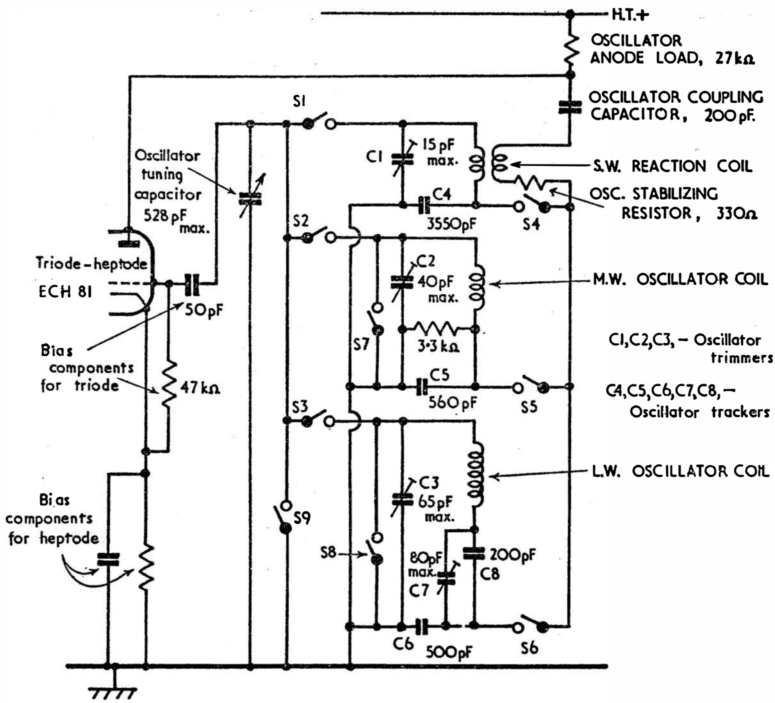
FIG. 5.4.
OSCILLATOR STABILIZING CIRCUITS

MULTI-BAND RECEIVERS

Most sets for a.m. reception have at least two wave bands—many have three or more. As the frequencies dealt with on the various bands are widely separated from each other it is usual to switch the whole of the oscillator circuit when changing from one band to another. The only components common to all bands are the tuning capacitor and the biasing capacitor and leak. In order to prevent any effect due to coupling between coils of the various bands those not in use are short-circuited by extra contacts on the wave-change switch. A three-band oscillator circuit embodying these features is shown in figure 5.5.

OSCILLATOR FREQUENCY DRIFT

The i.f. produced in a superhet is the difference between the signal frequency and the frequency generated by the local oscillator. Supposing a set is switched on and after warming up is tuned to a particular station, *e.g.* a short-wave station on 10Mc/s. The oscillator frequency must be 10·465Mc/s assuming a 465kc/s i.f. Now, for some time after the set is first switched on its internal temperature continues to rise and a period of half an hour to two hours or more (depending on the size and layout of the set) may elapse before the final



S1, S4, – closed on S.W. only S7 – closed on S.W.
S2, S5, – closed on M.W. only S8 – closed on S.W. and M.W.
S3, S6, – closed on L.W. only S9 – closed on GRAM.

FIG. 5.5. OSCILLATOR CIRCUIT FOR THREE-WAVEBAND RECEIVER

steady temperature is reached. During this period small changes are continuously taking place in the dimensions and properties of the components. Changes in the components of the oscillator tuned circuit are most serious for they cause a continuous change in oscillator frequency which is known as **oscillator drift.** If in the case mentioned the frequency drifts by 1 part in 1,000 then the oscillator frequency at the end of the period has changed by $\frac{1}{1000} \times 10{\cdot}465\text{Mc/s} = 10{\cdot}465\text{kc/s}$. The change in the i.f. generated is clearly equal to the actual change in oscillator frequency so that the i.f. also has changed by 10·465kc/s. This is sufficient (with a 9kc/s i.f. pass-band) to detune completely the required station. On medium and long waves the effect is not so severe but may cause sufficient detuning to affect the quality of reproduction **(side-band cutting).**

Several other factors affect the oscillator frequency and cause drift. Among the most important are:

(*i*) Effects on the components of atmospheric humidity and pressure which may cause warping and other dimensional changes. These effects are most important in aircraft equipment, etc.

(*ii*) Changes in the supply voltages to oscillator valve and mixer. These may be due to mains voltage variations or to h.t. voltage changes caused by operation of the a.g.c.

(*iii*) Changes in the input capacitance of the oscillator injection grid brought about by variations in a.g.c. bias.

The most serious effects are those due to temperature and to changes in valve input capacitance. Note that drift due to these causes is most noticeable at the high frequency end of a band where the amount of tuning capacitance in circuit is a minimum. The components worst affected by changes in temperature are capacitors and special types must be used. Now, the capacitance of a parallel-plate capacitor is given by

$$C = \frac{\epsilon\epsilon_0 A}{d}$$

where A is the plate area, d is the dielectric thickness, ϵ the permittivity of the dielectric, and ϵ_0 the permittivity of free space. Changes in temperature affect not only A and d causing the capacitance to rise when the temperature goes up but also the permittivity. Silvered mica and silvered ceramic capacitors are mainly used in oscillator circuits. Some ceramics have permittivities which increase with temperature, others have permittivities which decrease with temperature, *i.e.* the capacitors have negative temperature coefficients. By careful choice of suitable values and temperature coefficients it may be possible to neutralize an increase in capacitance in one part of a circuit by means of reduction in capacitance in another part. Circuits of this type are used to some extent and care must obviously be taken to make sure that any new component has the same characteristics as the one which it replaces.

Some of the capacitors used in oscillator circuits are variable or preset types. It is not easy to construct these to have the same degree of stability as fixed capacitors. The main variable tuning capacitor invariably uses air as dielectric and is stoutly built to give maximum dimensional stability. Preset capacitors used for padding and trimming, however, commonly have solid dielectrics and may be a source of frequency drift. To reduce dr ft from this cause it is common practice to reduce the size of preset capacitor s to the smallest practicable value. This can be done by using a fixed capacitor in parallel. Such an arrangement is shown in figure 5.6 where most of the

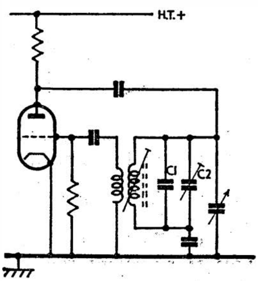


FIG. 5.6. OSCILLATOR TRIMMING

capacitance required across the coil is provided by C_1, any adjustment being made by the small preset trimming capacitor C_2.

Variations in valve input capacitance caused by changes in a.g.c. bias may affect the operation of the circuit in two ways as it is possible for both the aerial and oscillator circuits to suffer detuning. The net result is that the alignment of the set is upset and its sensitivity and selectivity are reduced. In many cases, therefore, a.g.c. is not used on the frequency changer stage, especially on short wave bands where the percentage difference between the oscillator and signal frequencies is small. Omission of a.g.c. control from the frequency changer causes considerable reduction in the effectiveness of the a.g.c. circuit, of course, unless an r.f. amplifier is used in the preselector stage. The designer must decide which effect is the worse and act accordingly.

CHAPTER 6

INTERMEDIATE FREQUENCY AMPLIFIERS

IT is upon the i.f. amplifier that the high sensitivity and selectivity of the modern superhet mainly depend. As this amplifier operates at fixed frequency a series-fed tuned anode circuit may be used as the coil and tuning capacitor can be easily insulated to withstand the h.t. voltage. In practice a tuned transformer is used in order to give a band-pass effect as explained in Volume 2, Chapter 12. The two requirements of high gain and wide bandwidth conflict with one another in amplifier design. In a set for a.m. operation on medium waves (and other bands) there is no point in making the overall bandwidth greater than 9kc/s since this is the maximum separation between transmitting stations on the medium waveband. This low bandwidth enables high gain to be used and so it is unusual to find more than one stage of i.f. amplification in the average a.m. receiver. A bandwidth of 9kc/s, however, severely limits the quality of reproduction obtainable from the set and before the advent of f.m. transmissions attempts were made to remedy this defect by using **variable selectivity** circuits in the i.f. amplifier. This enabled the bandwidth of the i.f. amplifier to be varied by means of a control on the panel. Increased bandwidth was used for the "high quality" reception of local stations where the loss of sensitivity occasioned by the wider pass-band did not matter, and where the possibility of adjacent channel interference was small because of the high signal strength of the local station. In receivers used for serious communication work on short waves very high selectivity may be required—for telegraph (Morse) reception the pass-band may be less than 2kc/s. Such a degree of selectivity cannot be obtained by the use of tuned circuits and so (quartz) crystal filters must be used in the i.f. amplifier. A detailed study of such filters is outside the scope of this book.

It is not an easy matter to vary the bandwidth of a tuned transformer and yet maintain the correct centre frequency and a flat-topped response curve. The most satisfactory method is to vary the mutual inductance between the primary and secondary circuits. This may be done by (*i*) actually sliding the coils relative to one another; (*ii*) using auxiliary coupling coils which may be rotated on their axes in order to vary the mutual inductance; and (*iii*) switching in auxiliary coupling coils having different numbers of turns. Note that methods (*i*) and (*ii*) require some mechanical arrangement (*e.g.* rods or wires) for their operation. In some cases a kind of variable selectivity effect has been obtained by connecting resistors (fixed or variable) across one or more of the tuned circuits. Such a scheme has the effect of increasing the damping of the tuned circuit concerned and producing a response curve of greater width but with a rounded top. Thus, the amplification given to the high frequency side-bands is less than that given to the low frequency side-bands and frequency distortion is produced.

In i.f. amplifiers for f.m. receivers and for television sound channels the i.f. is of necessity much higher (*e.g.* 10Mc/s) than in the ordinary a.m. set. For this reason the gain per stage is less and two or three stages of amplification in cascade may be required. At the same time the bandwidth may be increased (to 20kc/s or more) and so the high frequency response of the set is good. The required bandwidth for such high fidelity reception is often obtained by the use of damping resistors across the coils, but notice that the high mutual conductance valves required for good amplification of these frequencies themselves exert a considerable damping effect on the tuned circuits. The rounding of the response curve top is in this case much less than it would be at an i.f. of 465kc/s.

The valve most commonly used for i.f. amplification is the variable-mu r.f. pentode which can be used for automatic gain control by variation of the control grid bias. In some cases the hexode (or heptode) section of a triode-hexode may be used. A common example of this occurs in combined a.m./f.m. sets where, for f.m. reception, the mixer section of the a.m. frequency changer valve is used as an i.f. amplifier. Another example of the use of a hexode as an i.f. amplifier occurs in sets in which, in order to reduce the number of valve types used and the number of valveholders, two triode-hexodes are used. The first triode-hexode acts as (a.m.) frequency changer in the usual way; the hexode section of the second triode-hexode is used as i.f. amplifier; while the triode portion of the second valve is used as a.f. amplifier. If the third valve in the set is a double-diode-output pentode it can be seen that a complete four valve (plus rectifier) receiver can be made using only three valves of two different types.

Where, as in a combined a.m./f.m. set, the same valve is used for amplification of two different i.f.s no switching of the anode and grid circuits is necessary because of the large difference between the a.m. i.f. of 465kc/s and the f.m. i.f. of 10·7Mc/s (typical figures). Two i.f. transformers are simply connected in series in the anode and grid circuits. When the amplifier is used on one frequency the impedance of the tuned circuits for the other frequency is very small and can be neglected. (A parallel resonant circuit has its maximum impedance at the resonant frequency and very small impedance at frequencies widely removed from the resonant frequency.)

REFLEX AMPLIFIERS

A reflex amplifier is one in which two different frequencies or bands of frequencies are amplified simultaneously by the same valve. Reflex amplifiers are not common in commercial sets but are found occasionally. In such cases the same valve is usually employed to amplify the i.f. and a.f. signals. A typical circuit is shown in figure 6.1. The valve used here is a double-diode-r.f. pentode which performs all the functions of i.f. amplifier, detector, delayed a.g.c. valve, and a.f. amplifier. Each part of the circuit is fairly straightforward—the two signals a.f. and i.f. are fed in series to the grid and the outputs are developed across the two sections of the anode load, the i.f. across the primary of the i.f. transformer and the a.f. across the 10kΩ resistor.

Circuits of this kind, while attractive on paper, are often difficult to manage in practice—distortion and cross modulation are likely to occur as a result of the curvature of the valve characteristic.

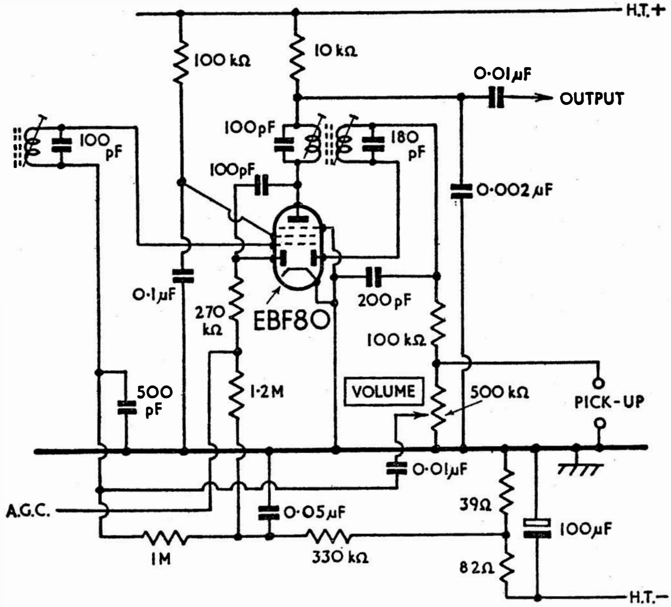


FIG. 6.1. REFLEX AMPLIFIER (Ferguson)

CHAPTER 7

AERIALS AND INPUT CIRCUITS

SUCH is the sensitivity of the modern domestic superheterodyne receiver that it will often appear to work satisfactorily with the bare minimum of aerial or even with no external aerial at all. To operate a set under these conditions, however, is not good practice and except for sets designed specifically for operation with an internal aerial (and the number of these is rapidly increasing) a good outdoor aerial should be used whenever possible. The use of such an aerial gives a much higher signal strength at the set and greatly improves the signal-to-noise ratio. This applies to noise picked up by the aerial and to internal valve noise generated in the set itself. Again, although a set may work satisfactorily without an aerial when first installed, when the valves are new and the circuits in perfect alignment, its performance will gradually deteriorate with age and the a.g.c. circuit will not be able to compensate for the fall in sensitivity since the set would be working at maximum gain when new. In areas where the signal from the local station is weak and/or where interference is very bad special types of aerial must be used; these are dealt with in Chapter 9.

INTERNAL AERIALS

It cannot be denied that outdoor aerials are something of a nuisance to install and maintain. In addition they are likely to be somewhat costly and it is difficult to persuade a customer to have one fitted when his set appears to work quite well without one. In addition, outdoor aerials may be unsightly and many people (in flats, etc) cannot easily erect one if they wish. For these reasons manufacturers have concentrated on making sets self-contained so far as the aerial is concerned by fitting internal aerials. Internal aerials are of two main types: the frame aerial consisting of a coil (or coils) of large dimensions often wound on the inside of the back cover of the set; and the ferrite rod aerial. The frame aerial has been widely used particularly in battery portables for many years; the ferrite rod aerial is a recent development made possible by the introduction of modern magnetic materials. Ferrites are non-metallic magnetic materials; being non-metallic they make possible the construction of magnetic circuits in which the loss of energy is very small indeed, even at high frequencies.

Both types of internal aerial have directional properties and early sets had to be placed in the best position for reception—some models incorporated built-in turntables for this purpose. The ferrite aerials are relatively small and in some sets are pivoted and connected by linkages to a panel control by means of which they may be rotated into the best position, the actual set remaining stationary.

RESONANT AERIALS

As has been stated in Volume 2, Chapter 6, an aerial possesses inductance and capacitance (and resistance) and so forms a resonant circuit. The resonant frequency depends upon the length and shape of the aerial. Now, the ordinary domestic receiver with three wavebands has to cover an overall tuning range of something like 16 metres to 2,000 metres—a ratio of 125: 1. Obviously a single resonant aerial cannot cover this large range. In addition, the length of a resonant aerial must be of the same order of magnitude as the wavelength to be received. A long wave aerial 1,000 metres (about three-quarters of a mile) long is undesirable to say the least. Aerials for the above sets are therefore non-resonant: they may be looked upon as simple conductors in which e.m.f.s are induced by the radio waves from the transmitting stations.

For reception of a small range of short wave stations, *e.g.* a particular short wave amateur band, the v.h.f./f.m. transmissions of the BBC, or a television sound and picture channel, resonant aerials may conveniently be used. The particular advantages are that they provide gain and have directional properties which enable them to be positioned so as to give the best signal-to-noise ratio. Also, being resonant, they are not likely to pick up interference from stations on different wavelengths. Resonant aerials usually have a low output impedance and must be connected to the set *via* a properly matched feeder or transmission line. There are many types of resonant aerial, the most common being the half-wave dipole, so called because the overall length of the two equal elements is approximately equal to one half of the received wavelength. The two elements are insulated from each other at the centre where the transmission line is connected. The output impedance of these aerials is about 75 ohms and so the feeder should have a characteristic impedance of the same value. This feeder should strictly be of twin-wire construction in order to maintain balance to earth. The aerial input coil is then centre tapped to earth. Twin-wire feeders, however, have a characteristic impedance of about 300 ohms and so difficulties with matching may occur. In practice a coaxial cable is commonly used as this has a characteristic impedance of 80 ohms or so. In such a case one side of the input coil is earthed, the outer conductor of the coaxial cable being connected to the earthy end of the coil. The directional and anti-interference properties as well as the gain of a half-wave dipole are often improved by the use of a **reflector** (H aerial). This consists of a metal rod mounted parallel with the dipole elements at a distance of approximately quarter of the wavelength. This aerial is most sensitive along the line joining the elements and the reflector on the element side, and least sensitive on the same line on the reflector side. Where interference is troublesome better results may often be achieved by pointing the reflector at the source of interference (so as to get minimum interference) rather than by pointing the elements in the direction of the transmitter (so as to get maximum signal). Additional elements known as **directors** may be fitted on the opposite side to the reflector in order to increase the aerial sensitivity still further. Aerials of this kind are only required in areas of very low signal strength. As the aim of the BBC and ITA is to bring more and more people into their primary service areas by increasing the number of transmitters the need for high gain aerials should gradually disappear except in ocalities where reception conditions are particularly bad, *e.g.* behind hills, etcl.

Other types of resonant aerial are used, especially for indoor installation and for television reception. In general, the principles involved are similar to those outlined above—detailed descriptions may be found in manufacturers' literature.

POLARIZATION

This term is used by radio engineers to indicate the method of propagation used for a radio transmission. The current in a transmitting aerial produces both electrostatic and electromagnetic fields which are radiated at right angles to one another. The direction of polarization is the direction in which the electrostatic field is propagated. A transmission is said to be vertically polarized if the electrostatic field is propagated in a vertical plane and so on. The BBC v.h.f./f.m. transmissions are horizontally polarized. Aerials used for the reception of these transmissions should be mounted with the elements horizontal. The television services in this country use both vertical and horizontal polarization.

INPUT CIRCUITS

This section may conveniently be divided into three parts dealing with internal aerials, non-resonant external aerials, and resonant aerials respectively.

1. Internal aerials.—Generally speaking internal aerials are suitable for long and medium waves only. Where a frame aerial is used the coil itself is often tuned directly on the medium wave band while a "loading coil" (used to bring circuit inductance up to the required value) is connected in series with the frame for long waves. Sometimes loading coils are used on both wavebands. A circuit of this type is shown in figure 7.1.

When ferrite rod aerials are used it is usual to employ two coils, one for medium waves and the two in series for long waves. These coils are mounted at opposite ends of the same rod. (See figure 7.2.) Alternatively, separate

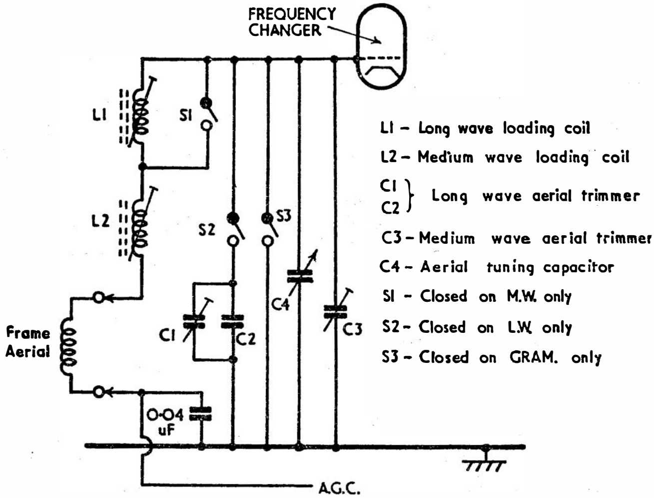


FIG. 7.1. FRAME AERIAL CIRCUIT FOR LONG AND MEDIUM WAVES

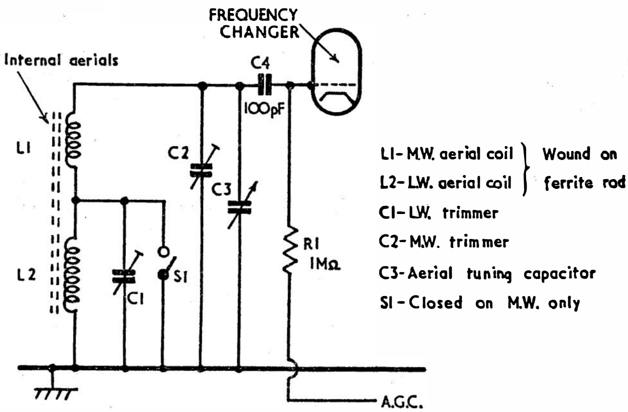


FIG. 7.2. FERRITE ROD AERIAL CIRCUIT

coils may be used, mounted on separate ferrite rods. These are switched into circuit as required and individually trimmed. A circuit of this kind having provision for an external aerial is given in figure 7.3. Note the use of common impedance coupling between the aerial and the tuned circuits.

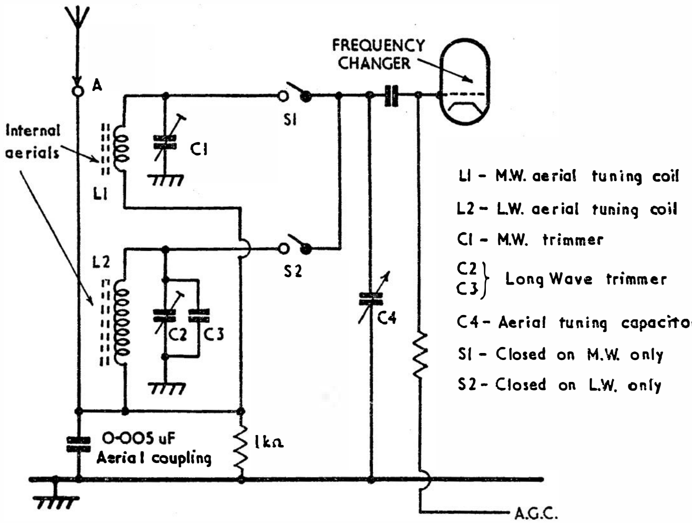


FIG. 7.3. EXTERNAL/INTERNAL FERRITE ROD AERIAL CIRCUIT

2. Non-resonant external aerials.—In this type of set the actual aerial may be a few inches of flex or a large outdoor aerial. So that the aerial does not seriously affect the alignment the coupling between the aerial and the first tuned circuit must be quite loose. Coupling may be achieved by use of a transformer or by common impedance as explained in Volume 2. In general (as is the case also with oscillator circuits) quite separate coils are used for the various wavebands, selected by the band switch as required. Figure 7.4

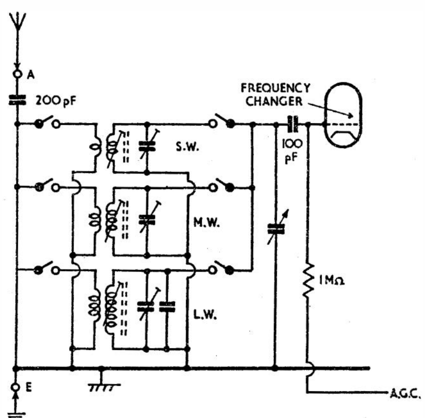


FIG. 7.4. THREE-BAND SET WITH AERIAL TRANSFORMER

shows connections for a three-band set with aerial transformers. The arrangement with common impedance coupling is similar to that of figure 7.3. Note, however, that transformer coupling is usually employed on short waves even in sets where common impedance coupling is used on the other wavebands.

In a.c./d.c. sets it is essential that no direct connection be made between the aerial and the chassis as the chassis may be connected to the live side of the mains. In such sets it is unwise to rely on the insulation of the aerial transformer, since these components are small and clearances between coils may be inadequate. High quality capacitors should be used to isolate the aerial from the chassis. This may lead to difficulty due to the accumulation of static charge on the aerial and so a d.c. path to (true) earth is commonly provided as shown by *R* in figure 7.5. The isolating capacitors in a.c./d.c. sets should always be checked with a high-voltage Megger or similar instrument when the set is serviced.

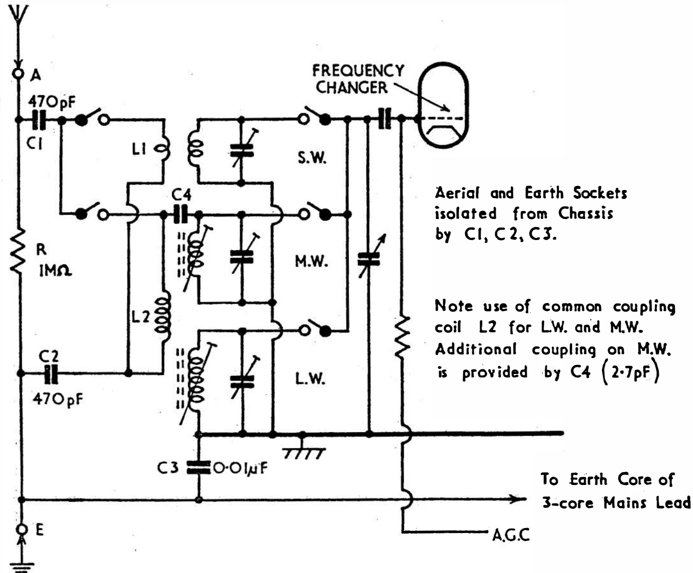


FIG. 7.5. AERIAL CIRCUIT FOR AC/DC RECEIVER WITH ANTI-STATIC SHUNT *R*

3. Resonant aerials.—The coupling between a resonant aerial and the first valve is usually by transformer. The transformer is used as an impedance changing device in order to match the feeder impedance to the valve. By tapping the primary winding it is possible to allow the use of either a 300 ohms twin feeder or an 80 ohms coaxial cable. (See figure 7.6.) The ratio of the aerial transformer depends on the arrangement of the first stage. For the f.m. transmissions the carrier frequency is about 100Mc/s (3 metres) and conventional r.f. amplifiers need careful handling at such frequency. Accordingly, an earthed grid r.f. amplifier is often used. This has a low input impedance but can readily be fed by a suitable transformer. Further work on v.h.f. circuits is given in Chapter 11.

BAND-PASS CIRCUITS

For reasons already given the i.f. bandwidth of domestic a.m. receivers is limited to about 9kc/s. It is, therefore, useless to provide a bigger bandwidth

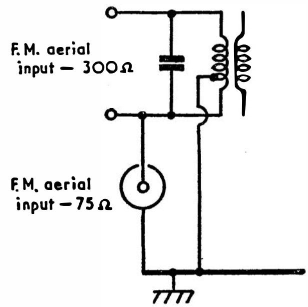


FIG. 7.6. INPUT CIRCUIT FOR RESONANT AERIAL

in the aerial tuned circuits. For this reason the band-pass circuits which were a feature of straight sets and some superhets several years ago are not now used. This statement is especially true since the introduction of the f.m. service. At the high carrier frequency of the f.m. transmissions adequate bandwidth is provided by single tuned circuits.

I.F. FILTERS

In some areas difficulty may be experienced due to the picking up on the aerial of stations broadcasting at or near the intermediate frequency of the set. Signals like these are amplified by the mixer valve and pass straight into the i.f. amplifier along with the wanted signal and so cause interference. Such signals may be reduced by the inclusion of an i.f. filter between aerial and mixer. This usually consists of a circuit sharply tuned to the i.f. and connected either as an **acceptor (series) circuit** across the aerial coil or as a **rejector (parallel) circuit** in series with it. Figure 7.7 shows these arrangements. I.F. interference of the kind described above can also be reduced by using an efficient preselector stage. For best results, however, this entails the use of an r.f. amplifier before the mixer which is undesirable from the point of view of cost. Direct pick-up of interference at intermediate frequency sometimes occurs, but is usually negligible because of the efficient screening used in the i.f. amplifier.

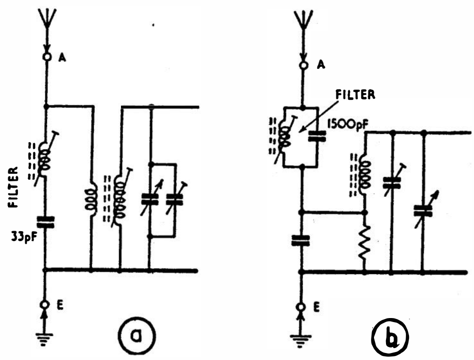


FIG. 7.7. I.F. FILTERS. (a) ACCEPTOR TYPE; (b) REJECTOR TYPE

CHAPTER 8

AUTOMATIC GAIN CONTROL

THE basic principles of automatic gain control (a.g.c.) have been outlined in Volume 2, Chapter 13, but a brief revision may not be out of place here. The function of the a.g.c. circuit is to compensate for fading and to prevent "blasting" when tuning from a weak to a strong station. At the same time distortion due to selective fading, *i.e.* the fading of particular sidebands, is increased and the set may be very noisy (depending on the level of interference) when tuning between stations. The required large variation in gain is made practicable by the use of variable-mu or "super-control" valves in the r.f., mixer and i.f. stages, while the negative control voltage for these valves is obtained from the demodulator *via* a suitable filter. The application of a.g.c. is usually confined to superheterodyne receivers because in the average straight set insufficient control voltage is available at the demodulator.

DELAYED A.G.C.

The simple a.g.c. system outlined above suffers from the disadvantage that the set can never operate at maximum sensitivity, because as soon as a station is received, however weakly, the gain of the set is reduced by the operation of the a.g.c. circuit. In order to overcome this difficulty delayed a.g.c. circuits are sometimes used in which the a.g.c. circuit does not become effective until a certain minimum voltage (the delay voltage) is produced at the demodulator. The delay is, of course, one of voltage and not of time (although in some cases a slight *time* delay in the operation of the a.g.c. can be detected. This is due to the overall time constant of the various decoupling resistors and capacitors used in the a.g.c. circuit.) The delay can most easily be arranged by biasing the diode anode negative with respect to cathode. As this bias must not be applied to the demodulator diode it becomes necessary to use two diodes in delayed a.g.c. systems. Typically, the two diodes have separate anodes and a common cathode (often being combined with a triode or pentode amplifier in a common envelope) but different arrangements may be employed when metal or crystal rectifiers are used.

Delayed a.g.c. systems are not at all common nowadays; there is, in fact, no double-diode-triode or double-diode-pentode 1·4V battery valve listed by valve manufacturers for use in current sets. Most battery receivers use the single-diode-pentodes DAF91 (ZD17, IFD9, 1S5, etc.) or DAF96 (1AH5, etc.) in a combined detector and simple a.g.c. circuit. The type of circuit used for delayed a.g.c. in earlier battery receivers is shown in figure 8.1.

In mains operated receivers delayed a.g.c. is found in the more expensive sets—in others the second diode of the double-diode-triode is either shorted to earth or connected in parallel with the detector diode. A circuit for

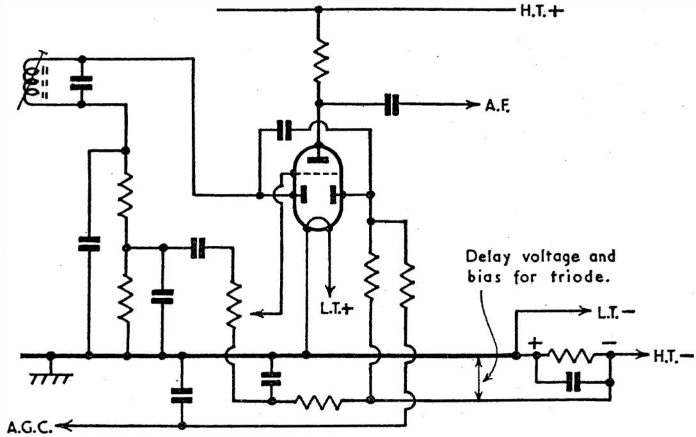


FIG. 8.1. DELAYED A.G.C. CIRCUIT FOR BATTERY SET

delayed a.g.c. in a mains set is shown in figure 8.2. The delay voltage is obtained by the voltage drop across R_1 which also provides bias for the triode section of the UBC41 used as a.f. amplifier. Notice that the a.g.c. diode load R_2 is returned to chassis while the detector diode load R_3 is returned to cathode. In this way bias is applied to one diode but not to the other. In the circuit of figure 8.2 the a.g.c. diode is fed from a tapping on the primary coil of the first i.f. transformer *via* the coupling capacitor C_2. A widely used alternative is to feed the a.g.c. diode from the anode of the i.f. amplifier. Again the a.g.c. diode may be fed from the i.f. transformer secondary in parallel with the detector diode. This last system is more likely to produce distortion than the others for the reasons given below. While the time constant of the a.g.c. circuit is such that its output (as fed to the controlled valves) does not vary.

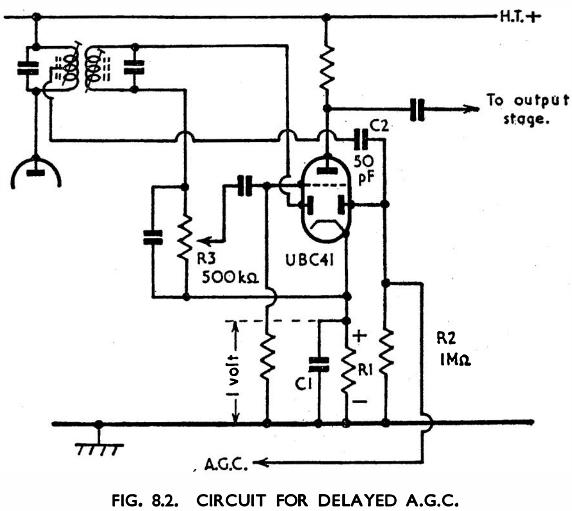


FIG. 8.2. CIRCUIT FOR DELAYED A.G.C.

with the depth of modulation, the a.g.c. diode itself (together with its load resistance and delay voltage) is connected across the i.f. tuned circuits. The resistance of a diode with negative bias on its anode is infinite until the anode voltage exceeds the bias when the resistance suddenly drops to a comparatively low value. When an amplitude modulated wave is fed to a biased diode it may happen that the diode conducts during part of the a.f. cycle only. The gain of the i.f. amplifier is less when the a.g.c. diode conducts because the low resistance of the diode is shunted across the i.f. tuned circuits. Therefore, the i.f. gain varies during the a.f. cycle and amplitude distortion of the a.f. is produced. The amount of distortion depends on component values but even with the best arrangement bad distortion may occur with certain signal strengths and depths of modulation.

Circuits have been devised in which the amount of distortion is reduced —one such is shown in figure 8.3.

In this circuit the a.g.c. bias is the negative p.d. developed across the signal diode load resistor R_1. Delay is effected because the a.g.c. line is held at chassis potential by the conduction of the "switching" diode d_2 which is connected to h.t. + *via* the high resistance R_2. When a high enough voltage is developed across R_1 to cause the anode of d_2 to go negative, d_2 is cut off and a.g.c. is applied to the controlled valves.

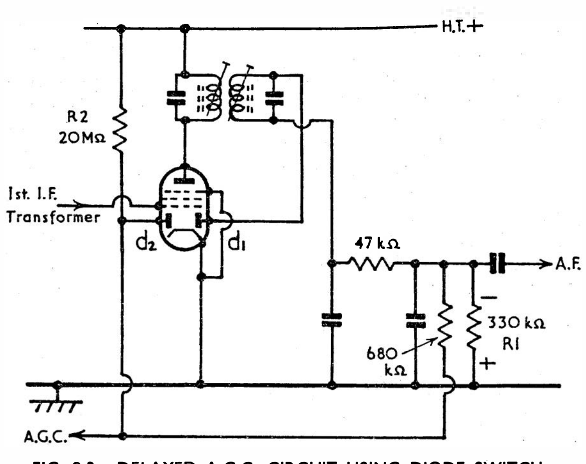


FIG. 8.3. DELAYED A.G.C. CIRCUIT USING DIODE SWITCH

APPLICATION OF A.G.C. VOLTAGE

The negative voltage produced by the a.g.c. diode must be fed to the grids of the controlled valves in such a way that it does not interfere with the signal circuits. Two methods are commonly used and are shown in figure 8.4.

At (a) the d.c. bias from the a.g.c. line is fed to the grid in series with the signal voltage from L_1. The bottom end of L_1 must be disconnected from the chassis to prevent shorting of the a.g.c., the a.c. impedance from the bottom end of L_1 to chassis being kept down to the required small value by C_1. C_1 is now in series with the tuning capacitor and unless C_1 is large the tuning range of the receiver is reduced. If C_1 is made too large, however, the time constant of the a.g.c. circuit becomes too great. Because of these difficulties this series circuit is little used in r.f. and mixer stages, its application being confined mainly to i.f. amplifiers where the tuning capacitance is fixed in value and need not be earthed. This is shown in figure 8.4(b).

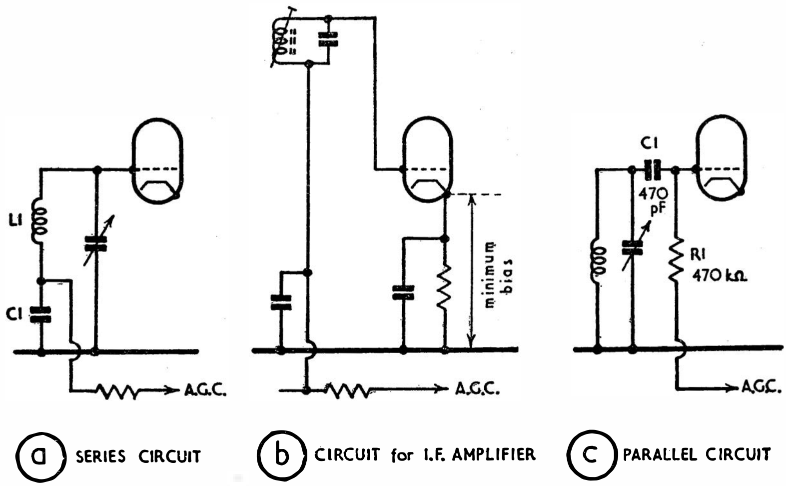


FIG. 8.4. METHODS OF APPLYING A.G.C. BIAS

Where variable tuned circuits are involved the circuit of figure 8.4(c) is commonly used. In this case the a.g.c. bias is fed to the valve grid in parallel with the signal, d.c. blocking being provided by C_1. The grid resistor R_1 is now shunted across the tuning capacitor but this is not serious as R_1 can have a high value (*e.g.* 0·5MΩ) and so its effect is negligible.

AMPLIFIED A.G.C.

For a.g.c. to be really effective greater control is required than can be provided by the simple and delayed circuits so far described. This extra control involves extra amplification which may be obtained from an amplified a.g.c. system. There are three possibilities:

(*i*) A separate amplifier channel may be used after the mixer parallel to the signal channel but having greater gain and feeding a separate a.g.c. diode.

(*ii*) An extra stage of i.f. amplification may be fitted and used to increase the signal applied to the a.g.c. diode only. The block diagram of figure 8.5

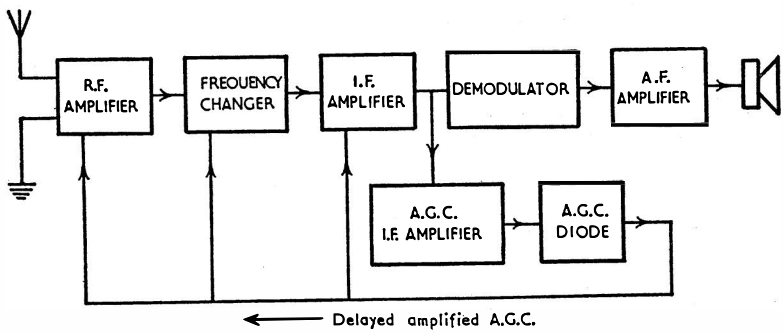


FIG. 8.5. BLOCK DIAGRAM OF A.G.C. AMPLIFIER

shows this scheme. The extra i.f. amplifier follows normal practice except that it operates at fixed gain so no detailed circuit is given.

In the above two arrangements the a.g.c. diode is isolated from the signal circuits by at least one stage of amplification and so delayed a.g.c. may be used without introducing distortion of the a.f.

(*iii*) Instead of using an a.c. amplifier before the a.g.c. diode, a d.c. amplifier may be used after the diode to amplify the d.c. control voltage before it is fed to the controlled valves. In general, d.c. amplifiers are rather awkward to manage as their performance depends much more upon the actual values of the components used than is the case in an a.c. amplifier. However, it is possible to obtain d.c. amplification for a.g.c. purposes without using an extra valve and an ingenious circuit of this kind is given in figure 8.6.

The triode section of the double-diode-triode is used both as a.f. amplifier and as d.c. amplifier for the a.g.c. voltage. The cathode of this valve is returned,

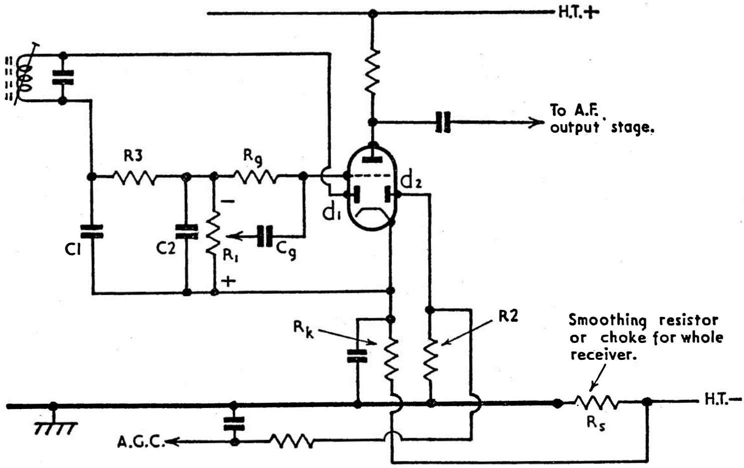


FIG. 8.6. AMPLIFIED A.G.C. CIRCUIT

not to chassis, but through the resistor R_k to h.t.—. The set is arranged with the smoothing choke or resistor R_s in the h.t. negative line so that the chassis is some 20 to 30 volts positive with respect to h.t.—, this voltage being constant. The p.d. between chassis and valve cathode is the difference between the constant voltage across R_s and the drop across R_k which varies with the triode anode current. When this anode current is large the drop across R_k is large and the cathode is positive with respect to chassis. Now, the a.g.c. diode d_2 is returned to chassis *via* its load resistor R_2 and so its anode is negative with respect to cathode. The diode is thus non-conducting and the a.g.c. line is at chassis potential. Suppose that the triode anode current is gradually reduced—as it decreases the drop across R_k decreases until this drop is equal to the p.d. across R_s. At this point the p.d. between cathode and chassis is zero and d_2 just begins to conduct. As the anode current is further reduced the cathode goes negative with respect to chassis. The a.g.c. diode now conducts and its anode, together with the a.g.c. line, also goes negative with respect to chassis (the diode anode is, of course, slightly positive with respect to *cathode*). In the extreme case where the triode anode current is zero (this should not occur in practice) the a.g.c. line has the full drop across

R_s applied to it. Some actual voltages are worked out in Table 2 for $R_k = 10\text{k}\Omega$ and a steady drop across R_s of 30V.

TABLE 2

Triode anode current I_a (mA)	p.d. across R_k $I_a R_k$ (V)	p.d. across R_s (V)	p.d. between cathode and chassis (V)	a.g.c. voltage (V)
	$R_k = 10\text{k}\Omega$	constant		
4	40	30	$40 - 30 = +10$	0
3	30	30	$30 - 30 = 0$	0
2	20	30	$20 - 30 = -10$	-10
1	10	30	$10 - 30 = -20$	-20

The required variation in anode current is effected at the grid of the triode by applying a negative bias which depends on the i.f. carrier amplitude. The diode d_1 is used as demodulator, R_1 being the load resistance (which also acts as volume control) while C_1, C_2, R_3 act as reservoir capacitor and filter components in the usual way. The a.f. voltage from R_1 is fed to the triode grid *via* C_g and the grid is returned *via* the leak R_g (1MΩ or so) to the upper end of R_1. Now, when a signal is being received a d.c. voltage is developed across R_1 proportional to the signal strength, the upper end of R_1 being negative. This voltage is applied as bias to the triode grid and gives the required control. With a high-gain triode a change of bias of a volt or two causes changes in anode current of the kind shown in the table so that a considerable degree of d.c. amplification may be achieved. At the same time, of course, the input to the triode grid must be limited or the valve may be cut off causing severe distortion.

If the circuit values are properly chosen this scheme provides delayed as well as amplified a.g.c. for d_2 is inoperative when the anode current is large (*i.e.* on weak signals). This is shown in the top line of the table.

QUIET A.G.C.

One of the disadvantages of a.g.c. is that the set works at maximum sensitivity in the absence of a carrier and so tends to be very noisy when tuning from one station to another. To cut down this noise and to cut out stations which are too weak to provide entertainment some form of **quiet a.g.c.** (q.a.g.c.) is required. The names **muting circuit** and **inter-station noise suppression circuit** are also used. Quiet a.g.c. is usually achieved by applying a large negative bias to either the demodulator or the a.f. amplifier when signals below a certain level are being received. This level can be controlled by the listener so that the set may be operated without q.a.g.c. when maximum sensitivity is required. A q.a.g.c. circuit is shown in figure 8.7. V_1 is a double-diode-triode used for detection, delayed a.g.c., and a.f. amplification in the usual way; V_2 is the muting valve. The grid of V_2 is biased by the negative voltage developed across the signal diode load resistance while the cathodes of V_1 and V_2 are strapped together and share the common resistor R_k. When large signals are being received V_2 is cut off and the bias on the triode grid of V_1 has its normal value. When the signal strength is low, however, some anode current flows in V_2 causing an increase in the drop across R_k. Under these conditions the drop across R_k is sufficient to cut off V_1 triode completely. The signal strength at which the muting action occurs can be selected by

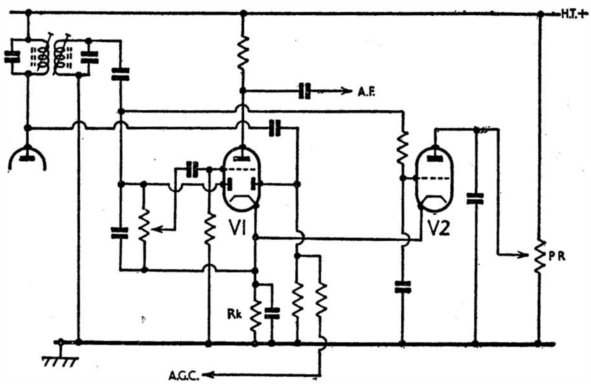


FIG. 8.7. QUIET A.G.C.

means of the control *PR* which alters the anode voltage on the muting valve.

Other circuits have been used for q.a.g.c. Most of them, like the one described above, require an extra valve and so add to the cost of the set. For this reason q.a.g.c. is very rarely found in post-war domestic receivers, much the same effect having been obtained quite cheaply by the use of flywheel tuning. This enables the listener to "spin" the tuning control from one station to another so quickly that inter-station noise suppression is unnecessary.

TUNING INDICATORS

The unskilled person tunes a radio receiver by adjusting the tuning control to give maximum loudness. When a set is fitted with a.g.c., however, it is possible to tune in a station of reasonable strength at equal loudness over a considerable range on the dial. (Notice that a set fitted with a.g.c. appears to be less selective than one not so fitted.) Accurate tuning is essential, however, if the best quality of reproduction is required with the minimum of background noise and this requires a little skill. Manufacturers have approached this problem in two ways: by the introduction of press button tuning which is dealt with in Chapter 10; and by the use of visual tuning indicators which enable the unskilled listener to *see* when his set is correctly tuned instead of having to rely on his ears.

Visual tuning indicators are operated from the a.g.c. line, maximum bias clearly indicating correct tuning. Early tuning indicators of the electromagnetic and neon types depended on the change in anode current or d.c. anode voltage of the i.f. amplifier and were not very sensitive. Modern tuning indicators are invariably of the cathode ray type described below.

In the cathode ray or electronic tuning indicator ("magic eye" of the publicity leaflets) a stream of electrons from a heated cathode is caused to fall on a fluorescent screen connected to the h.t. line. The area of fluorescence is controlled by a deflector electrode which is internally connected to the anode of a small triode. (See figure 8.8.) The grid of the triode is connected to the a.g.c. line. As the negative a.g.c. voltage increases on tuning in a station the anode current of the triode decreases and so the deflector electrode becomes more positive with respect to cathode. This causes a reduction in the angle of shadow thrown on the fluorescent screen by the deflector electrode. The

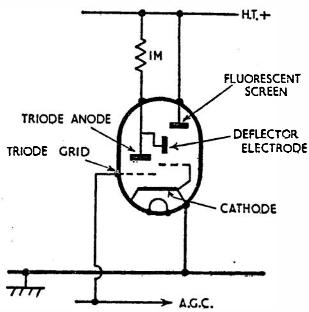


FIG. 8.8. CATHODE RAY TUNING INDICATOR

tuning point is, therefore, indicated by minimum area of shadow (or maximum area of fluorescence).

In order to deal satisfactorily with very weak and very strong stations dual-sensitivity indicators are sometimes employed. The same principle is used as described above but there are two internal triodes of different amplification factor but with a common grid. These are connected to two deflector electrodes which produce two shadow areas. The shadow angle associated with the high-gain triode is reduced to minimum with a grid voltage of about -3V, while that associated with the low-gain triode requires a grid voltage of about -12V for minimum. Where a dual-sensitivity tuning indicator is used in a set employing delayed a.g.c. it should be fed from the signal diode load and not from the a.g.c. line otherwise the delay bias prevents operation of the tuning indicator on weak signals and the advantage of the high-sensitivity section of the indicator is lost.

CHAPTER 9

INTERFERENCE SUPPRESSION

Interference with reception has been a problem since the earliest days of radio communication, but it has become an issue of major importance in recent years requiring, amongst other things, the passing of a new Act of Parliament and the provision of a new broadcasting service. There are three main types of interference:

1. **Atmospheric interference (static).**—This is mainly a problem in long distance reception except in the case of a local thunderstorm. The crackling due to electrical discharges in the atmosphere and fading due to changes in the ionized layers of the upper atmosphere are natural phenomena about which little can be done by the listener. In communication work different frequencies are used at different times of the day or year so as to keep interference to a minimum; also, complex receiving aerials may be erected.

2. **Interference from other stations.**—In this case the interference takes the form of a second programme or a continuous whistle superimposed on the programme from the wanted station. This is not always a simple matter of lack of selectivity as the interfering station may be a "pirate", working at the same or nearly the same frequency as the wanted station. Interference of this kind on medium and long waves was one of the more powerful reasons for the introduction of the v.h.f. transmissions of the BBC.

3. **Man-made interference.**—To the average listener, trying to hear the BBC or the more powerful continental stations on long and medium waves, this is the kind of interference which causes most annoyance. There has been a tremendous increase in the use of domestic electrical appliances since the war so that in the quietest suburban road there may be dozens of small electric motors, thermostats, switches, television receivers and so on pouring out a positive torrent of interference throughout the day. Possibly the worst sources of the "motor whine" kind of interference are hairdryers, electric sewing machines, and portable tools, since these are commonly used in the evening at times when many people are listening or viewing. It is this kind of interference (from industrial as well as domestic sources) which has been mainly responsible for the passing of legislation requiring all new apparatus to be satisfactorily suppressed by the manufacturers and giving the GPO the power to order the suppression of appliances in bad cases.

GENERATION OF INTERFERENCE

All electric circuits possess inductance and capacitance. In many cases the values are very small and have no effect at all on the steady-state operation of the circuit. Nevertheless energy is stored in the magnetic and electric fields and when this energy is liberated, *e.g.* when the circuit is suddenly switched off, it causes high frequency currents to oscillate in the resonant circuits formed by the inductance and capacitance. As these resonant circuits are complex and numerous the oscillations set up in them cover a wide range of frequencies, often sufficient to interfere with reception over a whole waveband. Thus, the operation of a switch in a simple d.c. lighting circuit may cause radio waves to be radiated from the associated wiring. Such waves may be picked up on the aerial of a neighbouring receiver where some will be amplified and detected producing a click in the loudspeaker. If the switch is replaced by a commutator motor then, as the brushes make contact with each segment in turn, a click is radiated. The rapid succession of clicks is heard in the loudspeaker as a whine which varies in pitch with the speed of the motor. Again, in a slow-break thermostat on an a.c. circuit arcing may occur during several cycles as the contacts part, producing a rough 100c/s "roar" at the receiver. In a similar way motor-car ignition interference may be reproduced as clicks or as a buzz depending upon the speed of the offending engine.

TRANSMISSION OF INTERFERENCE

The r.f. currents described above may reach the receiver in three main ways:

(*i*) By direct radiation, the appliance and its supply leads acting as a transmitting aerial;

(*ii*) By conduction through the mains—the street cables and house wiring acting as an r.f. feeder—into the set, to h.t. line and heaters;

(*iii*) By re-radiation to the receiving aerial from the supply mains in its vicinity.

ELIMINATION OF INTERFERENCE

By far the best way of reducing interference is to fit efficient suppressors to the offending appliance *i.e.* to suppress the interference at its source. This must now be done for all new equipment. Where such a course is impracticable it is necessary to reduce in some way the amount of interference getting into the set. This entails siting the aerial as far as possible from the interference and from any mains wiring which may cause radiation. An aerial in the form of a vertical rod (for long and medium waves) mounted on a chimney stack fulfils these conditions but care must be taken that interference is not picked up on the lead between aerial and set. In bad cases this lead must be screened. Now, a screened lead possesses considerable capacitance and acts as a low-impedance shunt on the aerial so that the signal at the set is very small. To overcome this difficulty matching transformers must be used: a step-down transformer at the aerial to match the aerial to the lead (which is now used as a transmission line), and a step-up transformer at the set to match the feeder to the input coil in the receiver. As these matching transformers have to operate over a wide frequency range they are not easy to design and this type of aerial system is rather costly. Efficient earthing arrangements are essential.

Mains-borne interference can be prevented from entering the set by the use of a suitable filter which may conveniently be installed by the side of the socket outlet used for supplying the set. A typical filter circuit for long and medium wave work is shown in figure 9.1. The chokes present a high

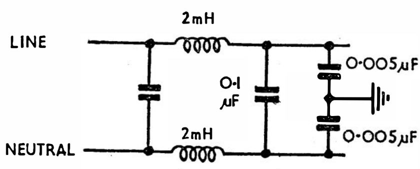


FIG. 9.1.
FILTER FOR MAINS-BORNE INTERFERENCE

impedance to r.f. currents while the capacitors provide low-impedance paths between lines and to earth. When a filter of this type is used care must be taken to ensure that heavy-current appliances such as fires are not used from the socket as the series chokes are wound with fine wire and are not usually suitable for currents in excess of one or two amperes. To save space the two chokes are sometimes wound on a common former in opposite directions. In this way the d.c. or 50c/s fields cancel each other but the chokes are still effective at radio frequencies.

INTERFERENCE SUPPRESSION

As stated above the best way of eliminating interference is to suppress it at its source. This can only bc done properly, of course, if the actual contacts at which the interference is generated are in good condition. A small amount of sparking is nearly always produced by a commutator motor, especially when operated on a.c.; this is quite normal and such a machine can be satisfactorily suppressed. A bright vicious arc, however, usually indicates that something is wrong (although the unfortunate fact must be faced that some motors always spark badly, even when new) and the machine should be given a complete overhaul before any attempt is made to fit suppressors.

Many suppressors are commercially available and manufacturers' lists should be consulted for details. It is important to realise that separate suppressors are required for broadcast band (*i.e.* 16 to 2,000 metres) and television frequencies. For suppression of interference on the broadcast bands the suppressors are often incorporated in a special mains plug or socket, a circuit similar to that of figure 9.1 being used. For Band I television the filter should be fairly close to the appliance and is often accommodated in a moulded housing on the flexible lead. For the best results, especially at the highest frequencies, the suppression components should be mounted on the appliance itself as close as possible to the contacts or brushgear. In many cases adequate suppression may be obtained by the use of capacitors only, a commonly used circuit employing three capacitors being shown in figure 9.2. The three

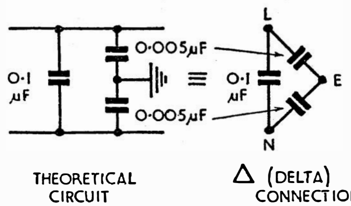


FIG. 9.2. FILTER CAPACITORS

capacitors are often delta-connected and enclosed in a common moulding with three leads for connection to the appliance. In this circuit, which is intended for broadcast work, a large capacitor (*e.g.* 0·1μF) is connected across the supply mains to act as a low impedance shunt for the r.f. interference current while a small capacitor (0·005μF) is connected from each side of the mains to the motor frame. The motor frame should be earthed *via* a 3-core cable and a properly installed 3-pin socket outlet. Unfortunately many people do not bother to take this simple safety precaution which means that the appliance frame (if of metal and not double-insulated) is connected to the live side of the mains *via* one of the 0·005μF capacitors. When the appliance is used on a.c. there is a small capacitance current (impedance of 0·005μF at 50c/s is about 0·5MΩ) which may cause a slight tingling sensation in anyone handling the appliance but cannot be regarded as dangerous. The value of 0·005μF must not, however, be exceeded and a properly designed suppressor capacitor from a reputable manufacturer should always be used. Failure of a suppression capacitor may have fatal results. No one should take risks with unearthed appliances. For Band I television the suppression capacitors are smaller *e.g.* 500pF, although tuned chokes (rejectors) may be required. These are also used on Band III.

CHAPTER 10

PUSH-BUTTON TUNING

MENTION has already been made of the difficulty experienced by many people in tuning in a superheterodyne receiver accurately. The tuning indicator is one answer to this problem; push-button tuning is another. Theoretically push-button tuning is to be preferred as it tends to simplify the set, making it easier to understand and operate, while the tuning indicator is an added complication which may not always be properly used. At one time many sets were available having both these tuning aids, a number (usually four or six) of favourite stations being selected by push-buttons while other stations could be logged by manual tuning supplemented by a tuning indicator.

Some modern sets have push-button tuning only—these cater for the people who never listen to anything other than the BBC and perhaps one of the Continental commercial transmissions.

Three main methods of push-button tuning have been used:

The first to be described is a purely mechanical system. (See figure 10.1.)

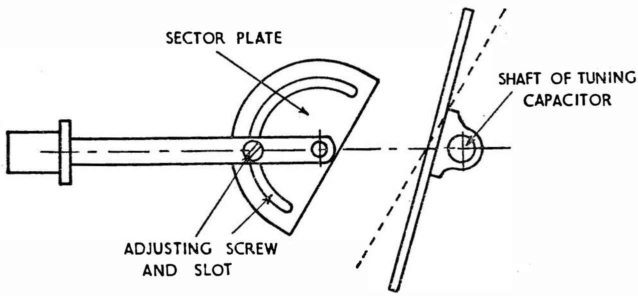


FIG. 10.1. PUSH-BUTTON TUNING: MECHANICAL SYSTEM

When the button is pushed (hard) a sector plate is brought to bear against a flat plate attached to the tuning capacitor shaft and causes the tuning capacitor to be rotated to the required position. The angle between the straight edge of the sector plate and the button axis can be changed by slackening a screw and so the station selected by a particular button can be changed if required.

The second method is electro-mechanical. The tuning capacitor is driven by a small a.c. motor working at low voltage from a tapping on the mains transformer. By the use of contacts mounted on the edge of a disc attached to the tuning capacitor shaft or by means of cam-operated switches brought into circuit by the push-buttons the driving motor is stopped at the required station, a clutch being used to give accurate positioning. Trip switches are fitted to effect automatic reversal of the drive at the end of a wave-band. A muting contact is fitted to cut out noise between stations. With this system the push-buttons for station selection can be cut out of circuit and the motor used for manual tuning under the control of stop and start buttons, there being no tuning knob of the orthodox kind. As it is difficult to accurately tune with a motorized control an automatic frequency control circuit may be incorporated which enables the set itself to accomplish the *fine* tuning of the required station. Receivers of this kind, while quite fascinating to operate, are very complex and rather costly. They are not, so far as is known, made nowadays.

The third method is electrical, push-buttons or a rotary switch being used to select circuits tuned to the required frequencies in the aerial and oscillator stages. Various detailed arrangements may be used *e.g.* a fixed coil with variable capacitors switched by the push-buttons; a fixed capacitor with switched adjustable coils; or individual tuned circuits switched as required (coil turrets). The second method using adjustable coils is commonly used for the oscillator, figure 10.2 showing the arrangement of the aerial and

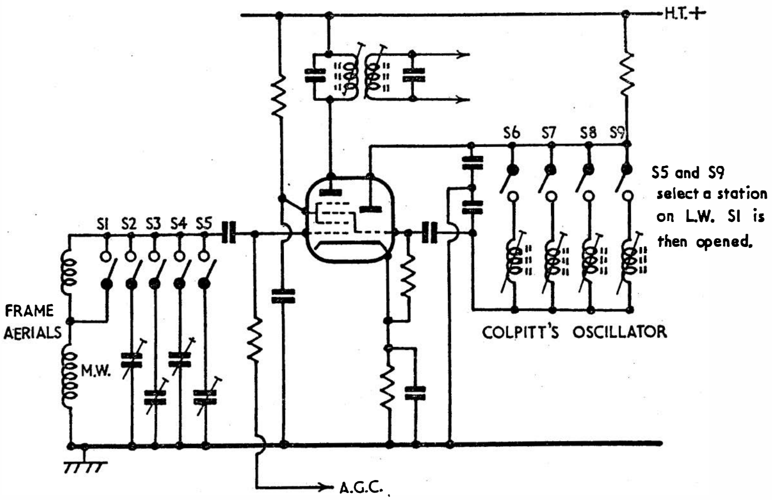


FIG. 10.2. R.F. CIRCUITS OF RECEIVER FOR PUSH-BUTTON TUNING ONLY

oscillator circuits in a set for push-button tuning only. As the coil inductances are individually adjusted to the required station there is no need for padding capacitors. This method is used because, as has been previously noted, preset capacitors are not very stable: it is better, therefore, to use a fixed tuning capacitor. Drift in the tuned circuits, particularly that of the oscillator, is much more serious in a push-button receiver as the listener has no means of compensating for it. If the drift is bad the set may become quite useless after a time—at best it loses sensitivity, requiring more a.f. gain and increasing the noise level. Automatic frequency control goes some way towards reducing the effect of drift. The control itself depends upon the frequency stability of a tuned circuit but as the frequency is relatively low drift in this circuit is not very serious. For example, when a set is tuned to a station on 1Mc/s the oscillator frequency, with an i.f. of 450kc/s, is 1,450kc/s. A drift of 0·2 per cent. in oscillator frequency produces a frequency change of 2·9kc/s while a similar drift in a frequency controlled circuit, operating at the intermediate frequency, causes a change of only 900c/s. Drift can be eliminated by connecting a small variable capacitor across the oscillator tuned circuit to act as a fine tuning control (a scheme widely used in multi-channel television receivers). Doing this reintroduces the difficulties associated with accurate tuning already referred to.

AUTOMATIC FREQUENCY CONTROL

Automatic frequency control (a.f.c.) circuits are most commonly used in motor-tuned receivers where the required station is tuned approximately by the motor and the a.f.c. circuit effects any fine tuning which may be required to bring the station "spot on".

A.F.C. circuits comprise two parts: a frequency-sensitive detector (discriminator) which determines whether or not a station is off tune by comparing the actual i.f. produced with the standard i.f. of the set; and a variable reactance circuit to which the discriminator feeds a control signal. The variable reactance circuit is usually connected across the oscillator tuned circuit and causes the oscillator frequency to be changed by the correct amount. Notice that the oscillator frequency can never be made exactly right, otherwise there would be no control voltage to the variable reactance circuit, but in a well designed set the i.f. produced in practice is so near to the correct value that no audible difference can be detected.

VARIABLE REACTANCE CIRCUIT

A commonly used variable reactance circuit is given in figure 10.3. The valve is a variable-mu pentode with an r.f. choke in the anode circuit and

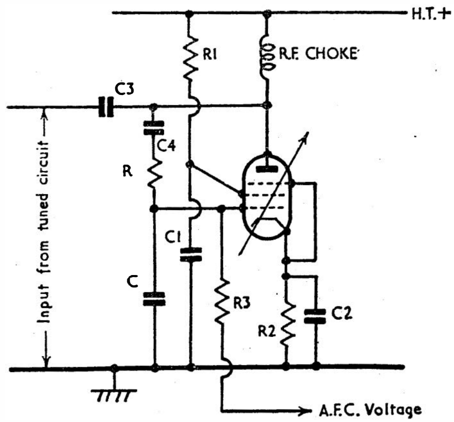


FIG. 10.3.
VARIABLE REACTANCE CIRCUIT

normal screen-feed components R_1 and C_1. R_2 and C_2 are used to provide cathode bias such that the valve operates, without control, at about the centre of its I_a–V_g characteristic. C_3 and C_4 are d.c. blocking capacitors in the leads to the tuned circuit and grid respectively. R_3 is the grid leak. The operation of the circuit depends upon the values of R and C. If the reactance of C at the operating frequency is very small compared with R then the current flowing in R and C in series is very nearly in phase with the anode voltage. Now, the voltage applied to the grid is that appearing across C which lags on the current, and therefore on the anode voltage, by 90°. The alternating anode current, of course, is in phase with the grid voltage and so also lags 90° on the anode voltage. Thus, an alternating signal voltage applied across the input produces a current in the valve lagging by 90° and so the circuit as a whole behaves as an inductance. The effective inductive reactance which is given by the ratio $\frac{\text{applied voltage}}{\text{current}}$ can be varied by controlling the d.c. bias applied to the grid. If the bias is reduced by making the a.f.c. line positive with respect to chassis then the mutual conductance of the valve is increased and the effective inductive reactance is reduced, whereas if the a.f.c. line is made negative the mutual conductance is reduced and the reactance goes up. Other circuits can be devised which behave as variable capacitors but these are not used for a.f.c. as it is important to keep the circuit capacitance down in order that a complete wave-band may be covered in one sweep of the tuning capacitor.

THE DISCRIMINATOR

A well-known discriminator circuit is the Foster-Seeley shown in figure 10.4(a). V_1 is the i.f. amplifier and L_1 and L_2 are the primary and secondary coils of a special i.f. transformer with centre-tapped secondary. L_1 and L_2 are both tuned to the intermediate frequency. d_1 and d_2 are diodes with load resistors R_1 and R_2 (and reservoir capacitor C_2) so connected that their output voltages are in opposition. The algebraic sum of these voltages constitutes the a.f.c. control voltage fed, through the filter R_3–C_3, to the reactance valve. By virtue of the capacitor C_1 the p.d. V_{d1} applied to the diode d_1 is the vector sum of the i.f. transformer primary voltage V_1 and the voltage across the top half of the i.f. transformer secondary V_{2a}. (See figure 10.4(b).) The dots under the letters indicate vector addition. Similarly, the p.d. V_{d2} applied to d_2 is the vector sum of V_1 and the voltage across the bottom half of the secondary V_{2b}. When the intermediate frequency is correct the primary voltage V_1 and the secondary current I_s are in phase and so the

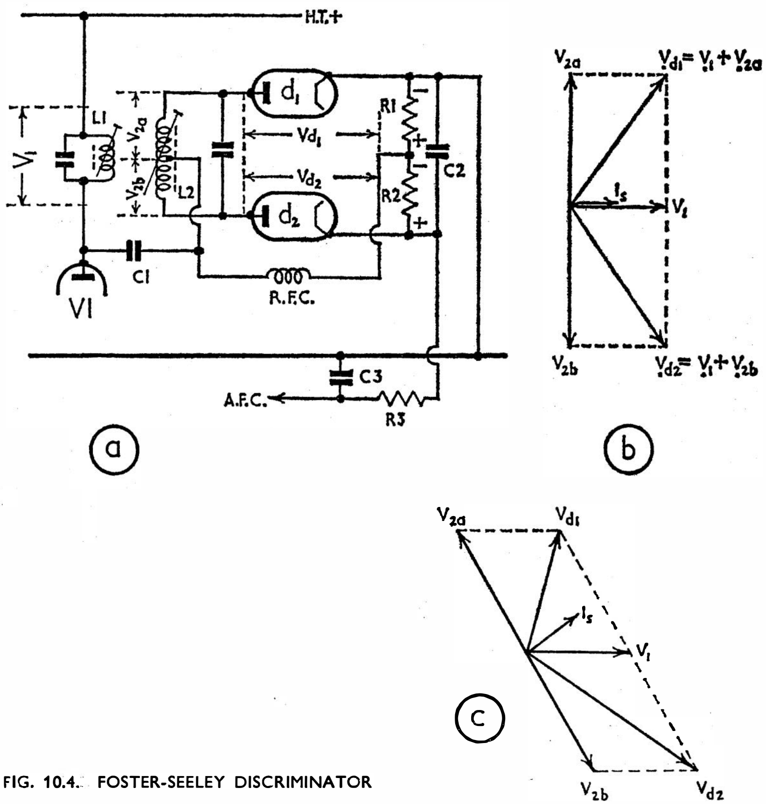


FIG. 10.4. FOSTER-SEELEY DISCRIMINATOR

p.d.s applied to the diodes are equal. The rectified voltages across R_1 and R_2 cancel and there is zero output to the a.f.c. line. When the intermediate frequency is too high the tuned circuits, no longer resonant, behave as reactive impedances and the secondary current leads the primary voltage. The two half-secondary voltages are still 180° out of phase with each other and 90° out of phase with the secondary current and their vectors swing round as shown in figure 10.4(c). The diode voltages are now no longer equal, V_{d2} being the greater. Hence, the p.d. across R_2 exceeds that across R_1 and the a.f.c. line goes positive with respect to chassis, producing a decrease in effective inductance in the oscillator tuned circuit and so increasing the oscillator frequency. If the intermediate frequency is too high the secondary voltage vectors swing the other way (clockwise), the a.f.c. line goes negative and the oscillator frequency goes down.

Note that an a.f. output can be obtained (on an amplitude modulated transmission) across either diode load, but not across the two in series. It is thus possible to use the a.f.c. discriminator as audio detector and a.g.c. valve, taking the output from R_2. This is not common practice as it gives rise to decreased selectivity. The a.f.c. discriminator is usually fed from a separate i.f. amplifier.

CHAPTER 11

FREQUENCY MODULATION

In this chapter it is intended to amplify what has already been said about frequency modulation in Volume 2 and to point out some of the principal points of difference between receivers for amplitude-modulated transmissions and those designed for reception of the BBC v.h.f./f.m. service.

PRINCIPLES OF FREQUENCY MODULATION

For faithful reproduction any system of modulation must be capable or transmitting, without distortion, the **intensity** (or volume), the **pitch** (of frequency) and the **tone** (or waveshape) of the sound at the microphone.

In a frequency modulated system these quantities are applied to the carrier as follows:

The **intensity** of the sound determines the amount by which the carrier frequency is varied. This is called the **deviation** or **frequency swing.** Thus, if the unmodulated carrier frequency is 90Mc/s a low-intensity sound which produces a swing of $\pm$ 5kc/s causes the carrier frequency to vary between 89·995Mc/s and 90·005Mc/s, while an intense sound may produce the maximum deviation of $\pm$ 75kc/s; the carrier varying between 89·925Mc/s and 90·075Mc/s.

The **pitch** of the sound determines the frequency at which the carrier frequency is varied. Thus, if the intense sound mentioned above has a pitch of 1,000c/s then the carrier goes through the cycle 90Mc/s to 90·075Mc/s to 90Mc/s to 89·925Mc/s, and back to 90Mc/s one thousand times per second.

The **tone** or **quality** (*i.e.* that which enables one musical instrument to be distinguished from another) determines the manner in which the above cycle is carried out.

ADVANTAGES AND DISADVANTAGES OF FREQUENCY MODULATION

A great deal has been written about the advantages of f.m. over a.m., and some of the claims made for f.m. have had to be modified in the light of experience. As far as the listener is concerned the principal benefits which he may expect from a f.m. receiver operating within the primary service area of the transmitter are: (*i*) a lower level of background noise and interference, and (*ii*) a much wider dynamic range. These are discussed in some detail below.

It is sometimes asserted that a f.m. transmission gives higher fidelity because of its wider bandwidth. This is true if one compares a f.m. transmission at v.h.f. with an a.f. transmission on medium waves. An a.f. transmission on v.h.f., however, can give wide bandwidth as is shown by the performance of those television receivers in which the sound channel has been designed to take advantage of the high quality of reproduction available.

A disadvantage of frequency modulation is that it tends to complicate the receiver. Much of this complication is due, however, to the use of a v.h.f. carrier and would have been needed anyway if it had been decided to use amplitude modulation for the v.h.f. sound service.

REDUCTION OF NOISE

Noise in a receiver may be generated either externally (*e.g.* atmospherics or ignition interference) or internally (being particularly important in the r.f. amplifier and the first tuned circuits). Loose connections are another matter. In either case the noise voltages are largely impulsive in character and cause amplitude and phase modulation of the carrier. A properly designed f.m. receiver does not respond to amplitude modulation of the r.f. or i.f. carriers and so the a.m. noise component is not reproduced. The phase modulated component does produce some noise in the output but this is small and the overall effect is that an f.m. receiver is some hundreds of times less noisy than its a.m. counterpart. Part of this improvement is due to the use of pre-emphasis at the transmitter and de-emphasis at the receiver. In pre-emphasis the amplitude of the high-frequency components of the audio signal is artificially increased (as in top boost) before modulation so giving a higher signal/noise ratio. Correct tonal balance is restored by a simple de-emphasis (top cut) filter after the demodulator.

DYNAMIC RANGE

This is the ratio of the loudest to the softest sound power occurring in a programme. A symphony orchestra has a dynamic range of the order of 10 million to 1 (70dB) while an amplitude modulated transmission has a range from about 5 per cent. modulation (limited by noise) to 90 per cent. modulation (limited by distortion) *i.e.* about $\left(\frac{90}{5}\right)^2$ which is 324 : 1 or 25dB (power is proportional to voltage squared). In order to accommodate an orchestral transmission in an a.m. system a large amount of volume compression must be used. Some of this is desirable as reproduction of the full orchestral range in the average living-room might lead to difficulties. The amount which must be used, however, is excessive and so the reproduced sound lacks depth and is unrealistic. Various volume expansion circuits have been used in the past—are still used for the high fidelity reproduction of recorded music, especially from cinema sound tracks—in an effort to overcome the difficulty. With f.m., because of the higher signal-to-noise ratio the dynamic range may be increased by something approaching 100 times (20dB) *i.e.* to 45dB or so.

BANDWIDTH

When a carrier is amplitude modulated side-bands are produced which differ from the carrier frequency by an amount equal to the modulating frequency. For a simple sinusoidal (no harmonics) modulating frequency f_1 applied to a carrier of frequency f_o the side-band frequencies are $f_o - f_1$ and $f_o + f_1$ and the bandwidth required to receive the transmission without distortion is $2f_1$. A high fidelity a.m. system can be made in which the bandwidth is not greater than 30kc/s (corresponding to a maximum modulating frequency of 15kc/s). With frequency modulation, however, a much greater bandwidth is needed for the same maximum audio frequency. Instead of there being two side-bands for each modulating frequency there is, with f.m., a large number of side-bands whose frequencies differ from the carrier frequency by multiples of the modulating frequency. The side-bands having frequencies very different from the carrier can be neglected but the others must be correctly received if high quality reproduction is required. For the BBC transmissions, in which the highest modulating frequency is 15kc/s and the maximum frequency deviation is 75kc/s, an overall bandwidth of 200kc/s is required. The use of such a high bandwidth for radio transmission necessitates a carrier frequency in the v.h.f. band, for while a 200kc/s bandwidth could be accommodated on a relatively low-frequency carrier, very bad distortion would be experienced on account of fading. Also, of course, the number of stations on a band would have to be very small. Band II, 87·5 to 100 Mc/s is used for the BBC transmissions. At these frequencies only the groundwave is effective, the skywave penetrating the ionized layers without reflection. Fading, such as is experienced on the medium and short wave bands, does not, therefore, occur within the service area. Because of the high carrier frequency the superheterodyne principle must be used in the receiver—it would be quite impossible to obtain adequate gain with a straight set. The intermediate frequency must, of course, be much higher than that used in an a.m. broadcast receiver and a value of 10·7Mc/s is commonly used. The gain obtainable at this frequency is limited and at least two stages of i.f. amplification are required. In practice three i.f. stages may be used, each having a voltage gain of about 30, in order to avoid the need for neutralizing.

THE F.M. RECEIVER

In figure 11.1 is shown a block diagram of a f.m. receiver. If this is compared with that for an a.m. receiver it can be seen that the main points of

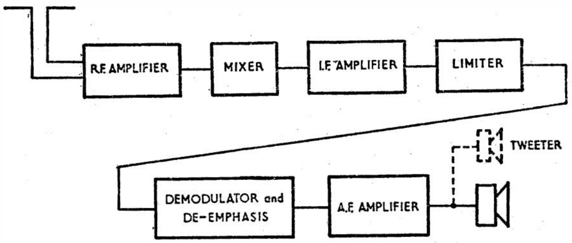


FIG. 11.1. BLOCK DIAGRAM OF F.M. RECEIVER

difference are the use of an r.f. amplifier in the preselector stage, and the limiter stage preceding the demodulator. These two features, together with the limiter and demodulator, are discussed in detail below. Several other points, not apparent in the block diagram, should be borne in mind:

Power Supply.—Because of the low noise level the slight hum often apparent in the output of a.m. sets cannot be tolerated. Extra smoothing may, therefore, be used.

A.F. Amplifier.—The f.m. transmissions are intended to provide a high-fidelity local station service. The output stages may, therefore, be built to do justice to the quality of the transmission. Many sets have push-pull amplifiers and/or are fitted with feedback and advanced tone control circuits. As the maximum modulating frequency is 15kc/s the sound output may be divided between two or more speakers, including a high-frequency "tweeter".

Combined Circuits.—As f.m. is suitable for short range work only any distant station must be received on a.m. and so sets capable of operation on both systems of modulation are required. Some of the stages may be common to both f.m. and a.m. and ingenious circuits are used in some of these sets in order to cut down the number of valves and other components to a minimum. A typical combined set is arranged as indicated in the block diagram of figure 11.2.

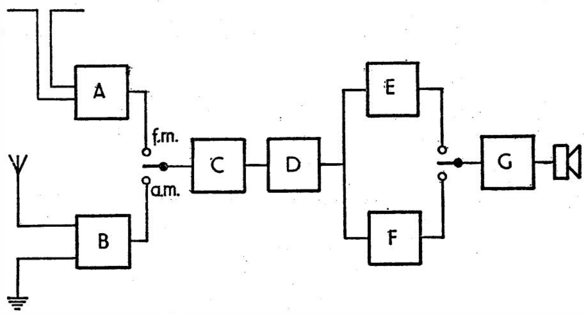


FIG. 11.2. BLOCK DIAGRAM OF COMBINED A.M./F.M. RECEIVER

A is the v.h.f. r.f. amplifier and frequency changer from which a 10·7Mc/s i.f. output is delivered.

B is the a.m. preselector circuit which does not usually include an amplifier.

C is the a.m. frequency changer *e.g.* a triode-hexode. When the set is switched to f.m. the oscillator in *C* is switched off and the hexode section acts as an i.f. amplifier for the f.m. signal.

D is the i.f. amplifier. This has two sets of transformers as mentioned elsewhere.

E is the discriminator (f.m. demodulator).

F is the a.m. demodulator. *E* and *F* may feed a common a.g.c. line.

G is the a.f. amplifier which is switched to *E* or *F* as required.

In battery sets the filaments of the valves in *A* (and possibly in *E*) are switched off when not required in order to economize in battery power.

R.F. AMPLIFIERS FOR V.H.F.

Most domestic f.m. receivers are fitted with an r.f. stage. There are a number of reasons for this:

(*i*) In the type of mixer circuit commonly used at v.h.f. there is considerable coupling between the oscillator and signal circuits. It is important that there should be no radiation from the oscillator *via* the receiver aerial as this would cause interference with other sets *e.g.* television receivers on Band III. The r.f. stage isolates the mixer from the aerial and so cuts down the amount of oscillator radiation.

(*ii*) The stage acts as a buffer between the aerial and mixer and so prevents mistuning of the oscillator by changes in the aerial and input circuits.

(*iii*) Mixer valves are inherently noisier than r.f. amplifiers. The use of an r.f. amplifier, therefore, improves the signal-to-noise ratio for a given overall gain.

(*iv*) The selectivity of the preselector circuits is improved by the use of two signal tuned circuits instead of one ensuring freedom from second channel interference.

(*v*) The gain of the amplifier gives a useful increase in sensitivity. Alternatively, for the same overall gain, the gain of the i.f. amplifiers may be reduced and these stages may be made more stable.

A frequency of 100Mc/s is just about the upper limit at which high-gain pentode r.f. amplifiers of conventional design may be operated without difficulty. At such a frequency it may be necessary to neutralize the stage in order to maintain stability. In such a case a neutralized high slope triode may just as well be used, although the gain is less. The triode has a considerable advantage in that it generates less noise than the pentode. Both types of valve are found in the r.f. stages of f.m. sets, the triode probably being the more common in this country—a double-triode is often used as r.f. amplifier and mixer.

Two circuits for use with triodes are described below.

The Neutralized Triode Amplifier.—The conventional triode amplifier circuit cannot be used at r.f. because of excessive feedback from output to input *via* the grid-anode capacitance. In figure 11.3 the grid-anode capacitance ${}_gC_a$

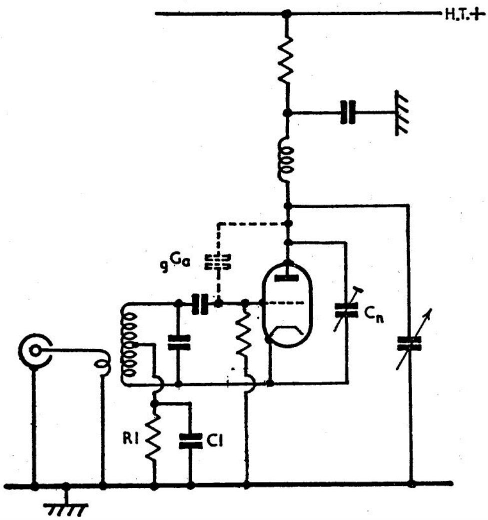


FIG. 11.3. NEUTRALIZED TRIODE AS V.H.F. AMPLIFIER

(shown dotted) is neutralized by means of the small trimmer C_n. Note that the input coil is centre-tapped, the centre-tap being connected to earth *via* the bias components R_1 and C_1. Thus, the voltages fed back from the anode *via* ${}_gC_a$ and C_n to the two ends of the coil cancel each other out. In this diagram no variable tuning arrangement is shown in the input circuit—the input circuit is heavily damped by the aerial and by the valve and so its response is fairly flat over the f.m. band. There may be some reduction in gain at the ends of the band but it is not generally considered worth while to tune the input circuit on account of the difficulty of ganging.

The Earthed-Grid Triode Amplifier.—The circuit of this amplifier, which is very popular with designers, is shown in figure 11.4(a). Feedback through the grid anode capacitance is eliminated by connecting the grid to earth (using a very short lead) and applying the input between cathode and earth. The input is now in series with the output in such a sense that negative feed-back occurs and makes the input impedance very low. This low impedance

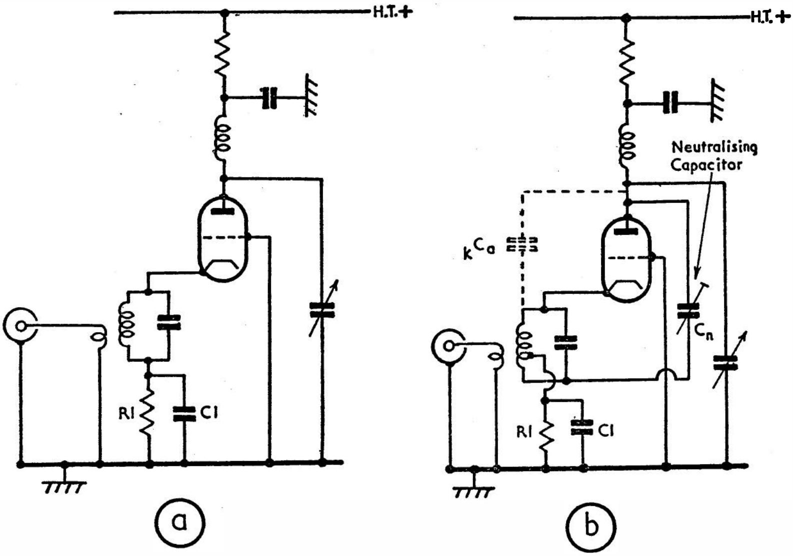


FIG. 11.4. EARTHED-GRID TRIODE AMPLIFIERS

can readily be matched to that of the aerial feeder by means of a suitable input transformer. R_1 and C_1 provide bias, of course; the input circuit is fixed tuned as before. The capacitance between anode and cathode $_kC_a$, through which feedback may occur, is very small (about 1·5pF in a typical valve). This can be neutralized if required by using a circuit such as that of figure 11.4(b).

FREQUENCY CHANGERS FOR V.H.F.

At v.h.f. the multi-grid multiplicative mixers described in Chapter 4 cannot be used, mainly because of valve noise. Most f.m. sets employ a triode as additive mixer and a simple circuit of this kind using a separate oscillator is given in figure 11.5. In this circuit the aerial and oscillator signals are fed together to the grid and steps must be taken to prevent the oscillator output from reaching the aerial. The bias required to ensure that rectification takes place (an essential part of additive mixing) is produced by the grid leak and capacitor. Because the i.f. is low compared with the signal frequency it is possible, with modern valves, to use a self-oscillating type of mixer. Circuits of this kind are very common; as already mentioned, the second half of the r.f. amplifier double triode is used. A typical circuit is given in figure 11.6. The basic circuit is that of a tuned grid oscillator. L_2 is the grid coil, oscillator tuning being effected by a cam-operated core in L_2; C_4 is the oscillator trimmer. Reaction coupling is provided by L_1; R_2 is a stabilizing resistor to prevent oscillation at the natural frequency of L_1. Bias is produced by C_5 and R_1.

From the point of view of the grid circuit the valve may be replaced by a capacitance C_v, this being the combined grid-anode and grid-cathode capacitances together with the wiring capacitance. The grid circuit may now be

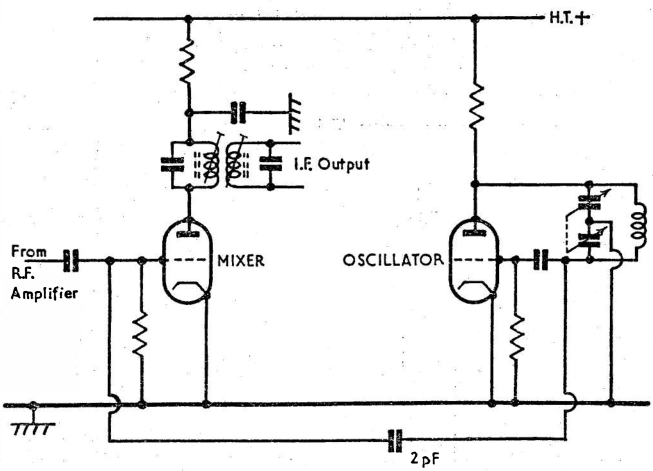


FIG. 11.5. TRIODE MIXER WITH SEPARATE OSCILLATOR

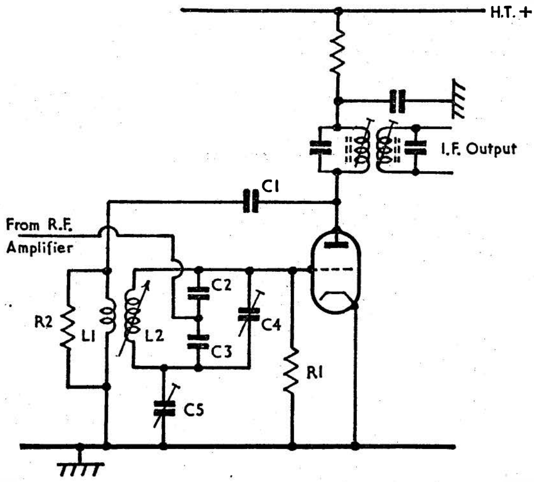


FIG. 11.6. SELF-OSCILLATING TRIODE MIXER

redrawn as in figure 11.7 from which it can be seen that C_2, C_3, C_5 and C_v form a capacitance bridge supplied from the oscillator coil L_2. By adjustment of the trimmer C_5 this bridge may be brought to balance (like a Wheatstone Bridge). When the bridge is balanced there is no oscillator voltage across the points P and Q and so the signal input from the r.f. amplifier may be coupled in between P and Q without any fear of oscillator radiation. A further advantage of this scheme is that "pulling" of the oscillator by the signal circuits is made negligible.

In some sets the oscillator is tuned by means of a variable capacitance in which case the r.f. input may be fed to a tapping on L_2, and C_2 and C_3 may be dispensed with.

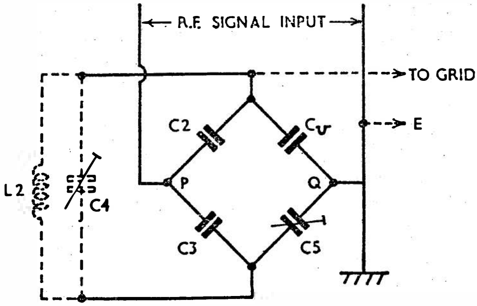


FIG. 11.7. BRIDGE NEUTRALIZING CIRCUIT

THE LIMITER

One of the chief advantages of the f.m. receiver is its freedom from noise However, as has been mentioned previously noise causes amplitude modulation of the carrier and this modulation must be prevented from reaching the demodulator. It is the function of the limiter to remove all amplitude modulation from the i.f. signal before it is passed to the demodulator. Now the ratio detector is very commonly used as a demodulator in domestic f.m. sets. This detector possesses quite a large amount of inherent limiting action and so a separate limiting stage is not often used. When other types of demodulator are employed, or in the more expensive sets, a limiter of the kind shown in figure 11.8 may be found. This consists of an r.f. pentode with screen and anode fed at low voltage from a potential divider. In this way the grid base of the valve is made very short, *e.g.* two to three volts. The grid leak and capacitor R_1, C_1 produce negative bias at the grid. Signals

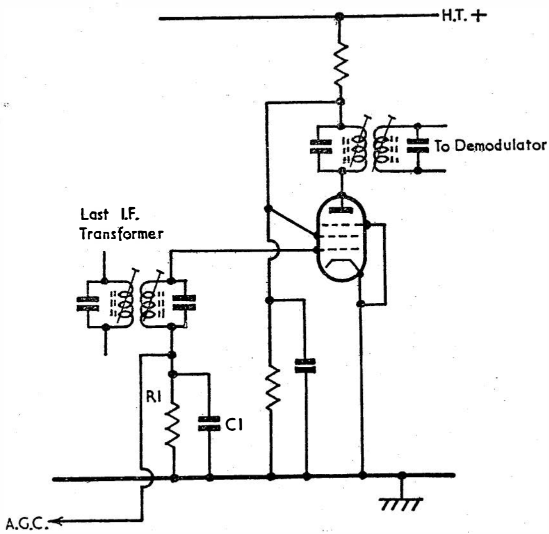


FIG. 11.8. CIRCUIT OF LIMITER

above a certain level are cut off on one side by the bottom bend and on the other side by grid current, producing a constant output at the anode irrespective of the input. The negative voltage produced across R_1 may be used as a.g.c. bias for the r.f. and i.f. amplifiers.

F.M. DEMODULATORS

An excellent demodulator for f.m. is the Foster-Seeley discriminator (or phase discriminator) described in Chapter 10 (figure 10.4, page 64). The action of this circuit for a.f.c. purposes is described in that chapter and it is shown that when the input frequency is low a d.c. voltage of one polarity is produced at the output, while when the input frequency is high a voltage of the opposite polarity is produced. If the circuit is fed with a f.m. signal it produces a voltage at the output which alternates at the modulation frequency and whose amplitude depends upon the change in carrier frequency. This is exactly what is required of a f.m. demodulator. The Foster-Seeley circuit is very linear and gives excellent results. It is less used than the ratio detector in domestic receivers because it requires a limiter stage while the ratio detector works quite well without one.

The Ratio Detector.—Figure 11.9 shows the arrangement of this well-known circuit. L_1 and L_2 are the primary and secondary windings of the last i.f.

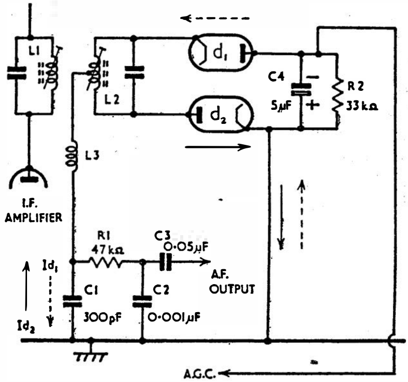


FIG. 11.9. RATIO DETECTOR

transformer, L_2 being centre-tapped and joined in series with the tertiary winding L_3. The two diodes d_1 and d_2 are connected in series aiding around the local circuit through R_2 and L_2 but produce, when they conduct, currents in opposite directions in the a.f. load capacitor C_1. These currents are shown in the diagram by I_{d1} and I_{d2}. The voltage applied to each diode is the vector sum of the e.m.f. induced in L_3 and one half of the voltage across L_2. When the two diode voltages are equal the diodes pass equal currents through C_1 and so produce zero output. This is the case when there is no modulation. When the modulation causes the intermediate frequency to change, the voltage across L_2 swings in phase relative to that across L_3 (rather as in figure 10.4(c) p. 64), the diode voltages are not equal, and the difference current produces an output across C_1.

The a.f. produced across C_1 is fed to the a.f. amplifier *via* the blocking capacitor C_3 and the de-emphasis circuit R_1, C_2 which also acts as an i.f. filter.

Now, so long as the amplitude of the i.f. carrier is constant the voltage acting in the circuit L_2, d_2, R_2, d_1 is also constant (being the voltage across

L_2 only) and so a steady p.d. is set up across R_2, the top end being negative. This voltage may be used for a.g.c. and tuning indicator if desired. If the amplitude of the carrier suddenly changes, as when amplitude modulated by noise, the voltage across R_2 is held constant by the action of the large reservoir capacitor C_4 and so the noise is not reproduced in the output. (Slow changes of amplitude, *e.g.* as when tuning, are accompanied by relatively slow changes in p.d. across C_4). This is the self-limiting feature of the ratio detector. The limiting action occurs with signals at any level so that weak signals can be received free of interference. Such is not the case with the separate limiter as this does not operate correctly for inputs below about two volts, which therefore usually requires an extra stage of i.f. amplification. In either case the limiting action is lost when there is no carrier and so the set tends to be very noisy between stations.

CHAPTER 12

SOUND REPRODUCTION

In many cases it is desired to reproduce sound which does not arise as the result of a broadcast transmission but is produced more or less near to the loudspeaker from which it is heard. Some of the more important sources of such sound are:

(*i*) Microphones, as used in public address work;
(*ii*) Gramophone records or discs—see Chapter 13;
(*iii*) Magnetic tapes—see Chapter 14;
(*iv*) Sound on film;
(*v*) Electronic musical instruments of various kinds.

Items (*iv*) and (*v*) are too specialized to come within the scope of this book.

REPRODUCTION OF RECORDED SOUND

The general principles involved in the reproduction of recorded sound are similar to those met in the audio frequency stages of a radio or television receiver but these principles have to be extended in many cases because of various special requirements. As these extensions are largely common to all forms of sound reproduction equipment they are considered in some detail below.

SIMPLE A.F. AMPLIFIER

Let us consider first an a.f. amplifier of the simplest type as used in a typical radio or television receiver of the less expensive kind. In this case the signal input is of the order of half a volt from the demodulator and the maximum power output is perhaps one or two watts from a single power valve or transistors feeding a single loudspeaker. The speaker is commonly housed in the same cabinet as the amplifier. The amplifier may consist of a single stage only, but it is more usual to employ two stages and to make use of some of the additional gain in the application of a degree of negative feedback. Such an amplifier may be used for the reproduction of recorded music, as in a radiogram or record-player, but it may be necessary to increase the gain if a modern high-quality pick-up is used with an older set. A typical circuit, as used in a portable record-player, is given in figure 12.1. Facilities for tone

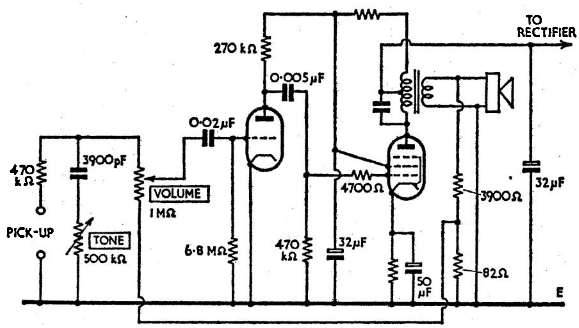


FIG. 12.1. CIRCUIT OF SIMPLE A.F. AMPLIFIER

control are quite simple, consisting of a straightforward top-cutting network. Circuits like this, when properly designed and used in conjunction with good-quality pick-ups and loudspeakers in good condition, are capable of giving a fidelity of reproduction which is far superior to what could be achieved before the war with equipment costing many times the price and have brought to many people a great deal of new enjoyment. At the same time it must be pointed out that equipment of the kind described above cannot reproduce the whole range of frequencies which is present in a radio (f.m.) transmission or on a record. Deficiencies may exist in the amplifier, in the loudspeaker, or in the pick-up (or whatever other source of signal is used). In most cases some distortion occurs in all three. To those who want "concert hall realism" in their reproduced music this distortion, or lack of response at the ends of the frequency spectrum, is intolerable and so it is necessary to have high-fidelity, or hi-fi, equipments in which the frequency range is extended to cover the whole of the audio band and in which all forms of distortion are virtually eliminated. At one time hi-fi was a highly specialized field in which enthusiasts largely built their own equipment with the aid of certain manufacturers who specialized mainly in pick-ups and loudspeakers. The advent of the microgroove record and v.h.f. broadcasting has made the pursuit of hi-fi much more rewarding and has vastly increased the interest and participation of the general public. Nowadays, therefore, many commercial hi-fi outfits are available. Some of these succeed better than others in their attempts at truly faithful reproduction. They may be judged by the way in which they fulfil the requirements set out below.

HIGH-FIDELITY SYSTEMS

The special requirements of any high-fidelity sound reproducing system must include the following points. It is assumed for the purposes of the argument that the signal source is a gramophone record—similar problems would be met with other sources.

(1) An extended frequency range. Ideally this should cover the whole audio spectrum; in practice it is limited to the range available on the disc. In an amplifier which may be used with different types of input, *e.g.* radio or gram, some way of altering the frequency response is essential in order to match the response of the amplifier to that of the signal and to give the desired overall response. This may be done by manual setting of separate top and bass controls to suit the type of input; or the amplifier response may be modified automatically by the use of preset tone control or equalizer circuits ganged to the input selector switch. Examples of this kind of arrangement are given in the chapters on recording.

(2) Freedom from distortion. If we assume that the groove on the disc represents a perfect signal then the points where distortion may occur are: (*i*) in the pick-up; (*ii*) in the amplifier, including the output transformer; and (*iii*) in the loudspeakers. The effects of room acoustics must also be considered.

(3) Volume expansion. It has been pointed out elsewhere in connection with radio transmission that it is not practicable to deal with the full dynamic range of many programmes, particularly orchestral music, and so **volume compression** has to be applied between microphone and transmitting aerial. Similar restrictions are required when recording on disc or tape. Thus for true high fidelity the signal must be expanded in the reproducing equipment to compensate for the original compression. This may be done in a variety of ways. First, it may be done manually, the listener following the musical score and adjusting the volume control of his equipment as required; this method is clearly for the enthusiast. Secondly, it may be done automatically. The scheme here is to use a sort of a.g.c. system in reverse. A variable-gain stage is included in the low-level section of the amplifier chain, so designed that its gain may be controlled by the variation of a d.c. voltage. (In an r.f. or i.f. amplifier a variable-mu valve is commonly used for this kind of circuit but this is not a good choice for a.f. because of the greater possibility of audible distortion). The d.c. control voltage is applied in such a sense, of course, that the gain is increased on large signals. The control voltage may be produced by rectification of the signal, but in this case the control circuit cannot respond fast enough to give undistorted reproduction of transients, and circuits using this idea are not now in favour. Alternatively, a suitable control signal may be recorded on an auxiliary channel. This is a better method, though costly, as the timing and amount of the volume expansion can be made to just compensate for the compression used during recording. Volume expansion is used in the cinema, where the control signal is recorded on a second sound track running parallel to the main track. It is not very much used, however, in domestic equipment. One reason for this is that the reproduction of an orchestra with full dynamic range in average domestic circumstances would be too overpowering for most people.

(4) Adequate power output. Whether volume expansion is used or not the dynamic range of reproduced sound is quite extensive, not to say impressive. The actual figure depends upon the type of system but is likely to be about 40dB in most cases. That is to say the ratio of peak power output during the loudest passages to the smallest output during the quietest passages is

10,000 to 1. This means that if the average level of reproduction is such that the quietest passages can be heard without strain then the peak putput may be of the order of 10 watts—for a domestic equipment. Clearly, the amplifier and loudspeaker must be capable of handling this power without distortion if the sound is to be reproduced with something approaching full depth and realism. Peak powers of this magnitude are not intolerable to the listener in a room of fair size but they have been known to cause trouble with the neighbours.

(5) Freedom from noise and hum. In the simple amplifier discussed earlier with its limited frequency response the requirements as to hum are not very stringent since it is unlikely that the speaker will be very sensitive at 50c/s or even at 100c/s. Furthermore, the gain employed in the a.f. amplifier is unlikely to be so high that valve noise or microphony would form a serious problem. Such is not the case with hi-fi equipment, and special care must be taken in design and maintenance to ensure that hum and noise do not become obtrusive.

AMPLIFIERS FOR HIGH-FIDELITY SYSTEMS

With modern circuits and components there is little difficulty in producing an amplifier which will meet the requirements listed above. Such amplifiers, which are nearly always of the push-pull type, have been dealt with in Chapter 2. It must be emphasized that a high-fidelity amplifier can only give its best performance (and anything less than the best is not hi-fi) if it is carefully maintained—component values are often of closer tolerance than is usual in general servicing practice; output valves may have been carefully matched; pre-amplifier valves may have been selected for low microphony; and the h.t. line voltage may be more critical than in other kinds of equipment. In addition great care has to be taken with earthing, particularly where several units are used together and the formation of earth loops, either within or between units, must be avoided. Failure to do this may result in the production of hum or instability by the installation as a whole while the individual units are apparently in perfect order.

LOUDSPEAKERS FOR HIGH-FIDELITY SYSTEMS

The basic principle of the loudspeaker has been described in Chapter 7 of Volume 2 and it is now necessary to give further thought to this. Let us consider a simple moving coil loudspeaker as shown in figure 12.2. When

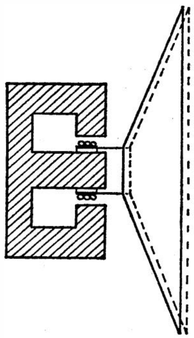


FIG. 12.2. SIMPLE LOUDSPEAKER

current is fed into the driving coil the cone is caused to move, the direction of motion depending on the direction of current. Consider conditions when the coil is carrying alternating current. The cone moves to the right during one half-cycle and as it does so it compresses the air in front of it. During the next half-cycle the cone moves to the **left** and as it does so the pressure of the air at the front of the cone is momentarily reduced—a rarefaction is produced. One complete cycle of the current thus produces one compression and one rarefaction in the air at the front of the cone. Such local variations in pressure are unstable, and so they travel outwards from the loudspeaker as sound waves. At the same time as compressions and rarefactions are being produced in the air at the front of the cone, similar compressions and rarefactions are being produced at the back of the cone but those at the back of the cone are 180° out of phase with those at the front. That is to say a compression at the front of the cone corresponds to a rarefaction at the back of the cone and *vice versa.* If no steps are taken to separate the sound waves produced at the front and rear of the cone they tend to merge together and in doing so, because of the phase difference, they cancel each other out. The extent to which this occurs depends upon the ease with which air can "spill" round the cone from front to back and this in turn depends upon the frequency of operation. At high frequencies the cone is moving at high speeds and the inertia of the air surrounding the rim of the cone is such that little spillage takes place. At low frequencies, however, the cone moves much more slowly and it is consequently much easier for the air to move round the edges of the cone from front to back. The efficiency of the cone as a source of sound is therefore much less at low frequencies than it is at medium and high frequencies. In order to increase the efficiency of the loudspeaker it is usual to fit it into some "enclosure" the prime function of which is to separate, or combine in an effective way, the sound radiation from the front and back of the cone.

Before going on to describe different types of enclosure it will be well to say a practical word of warning. As has already been seen the loudspeakers used in hi-fi systems may have to deliver peak powers of the order of 10 watts. In order to do this the speaker cone has to move through relatively large distances and the suspension must be designed to allow such movement. Now if the efficiency of the loudspeaker is low the power fed into the driving coil, instead of appearing as sound power in the air, goes into producing very large movements indeed in the loudspeaker suspension. These movements may be so large that the loudspeaker may be permanently damaged. It is therefore of great importance that a high-power loudspeaker should never be used without a proper enclosure.

Many different kinds of enclosure have been designed and developed and descriptions of these, together with their theory of operation, may be found in more specialized textbooks dealing with loudspeakers. From this large variety of loudspeaker enclosures there are three which are in very common use and these will be described in the succeeding paragraphs.

The simplest way of improving the efficiency of a loudspeaker at low frequencies is to mount it on a **baffle.** This may be a large flat sheet of wood or other material with a hole cut in or near the centre to accommodate the loudspeaker. If the baffle is very large it separates completely the front and back of the loudspeaker. In practice the size of the baffle has to be limited and this means that the baffle is only partially effective. Some people have mounted loudspeakers in the dividing wall between two rooms, or in a door, thus obtaining the effect of an infinitely large baffle. One disadvantage of this scheme is that all the sound radiated from the back of the loudspeaker is lost and so amplifier power is wasted. If an attempt is made to reduce the size of the baffle by bending it back to form a box or cabinet then difficulties arise on account of resonances in the enclosed air. The frequencies at which resonances occur depend upon the dimensions of the box, *i.e.* its volume and the ratios length/breadth/depth. In general the reproduction becomes worse as the depth of the enclosure is increased.

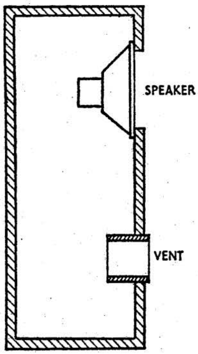


FIG. 12.3. BASS REFLEX CABINET

A type of enclosure which is very popular with hi-fi enthusiasts is the **vented chamber** or **bass reflex cabinet.** The arrangement of this is shown in figure 12.3. It consists of a stoutly constructed box of fair size (about 8 cubic feet in volume) which is totally enclosed apart from a vent which is seen in the sketch at the bottom and a port into which the loudspeaker fits tightly. The interior of the reflex cabinet may be lined with sound-absorbent material so as to reduce the effect of cabinet resonances on the radiation from the loudspeaker. The purpose of the vent is to provide a path of such length that bass radiation from the speaker cone traverses half a wavelength before being emitted from the vent. In this way the bass radiation from the back of the cone is emitted from the front of the chamber through the vent in phase with the bass radiation from the front of the cone and therefore the bass response of the loudspeaker, and its efficiency at low frequencies, are improved.

The most efficient way of producing sound with a loudspeaker is to use the loudspeaker in conjunction with **a horn.** The principle of the horn has been known for centuries and is widely used in wind instruments. There are three main shapes of horn and these are shown in the diagrams of figure 12.4. At (a) is shown the plain conical horn which flares out uniformly from the throat at one end to the mouth at the other. Diagram (b) shows one form, one shape, of parabolic horn. In this case the area of cross-section at any point in the horn is proportional to the distance measured from the throat. In the type of parabolic horn shown in the diagram two sides are parallel to each other; the other two are straight and are inclined at an angle to each other. Diagram (c) is of the very common exponential horn. In this case the logarithm of the cross-sectional area (to the base ϵ) is proportional, very nearly, to the distance measured from the throat, the rate of increase being referred to as the **flaring constant** or **flare.** The exponential horn is more efficient and has a more uniform response than either the conical or the parabolic and for these reasons is much more commonly used. In addition to its high efficiency the horn can also be arranged to have powerful directional properties and these are often made good use of—for example, in public address work out of doors.

It is of interest to note that the introduction of the exponential horn to the acoustic gramophone in 1925 greatly improved the performance of the latter and was, we may say, the first step in the very early stages of hi-fi. If it were possible to build an infinitely large exponential horn it would have the characteristic shown in figure 12.5. In this graph acoustical resistance is plotted to a base of frequency. The acoustical resistance of a loudspeaker is a measure of the useful power which can be converted into sound waves, just as the electrical resistance of an electrical circuit is a measure of the power

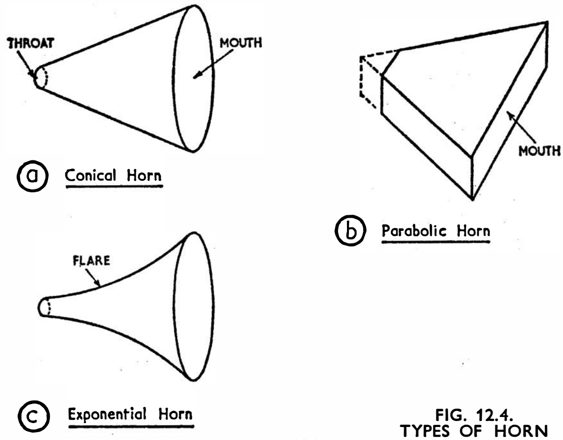


FIG. 12.4.
TYPES OF HORN

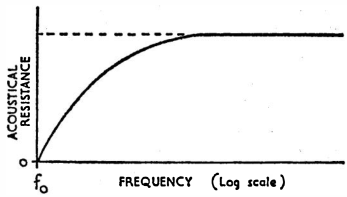


FIG. 12.5
CHARACTERISTIC OF INFINITELY LARGE EXPONENTIAL HORN

which can be converted into heat. Notice that the acoustical resistance of the horn loudspeaker is zero at some value of frequency f_0. This frequency is known as the **cut-off frequency** of the horn and its value depends upon the flare. The greater the flare the higher the cut-off frequency. The theoretical infinite exponential horn mentioned above would be incapable of reproducing sound below its cut-off frequency. In practice, of course, it is impossible to make an infinite horn and the characteristics of a practical horn are rather different from those shown in figure 12.5. The horn still has a cut-off frequency but its response is not zero for frequencies below the cut-off. The response is, however, greatly reduced at frequencies below the cut-off value. The characteristic of a horn loudspeaker is also affected by the size of the mouth. (In the theoretical infinite horn, the size of the mouth has no meaning). The horn can be looked upon as an acoustical matching device, matching the high impedance of the loudspeaker diaphragm to the low impedance of the air. If the mouth of the horn is very large the matching is good, but if the mouth of the horn is small, of the same size as the wavelength of the sound which it is required to reproduce, the matching is poor, reflections occur at the mouth, and the response becomes peaky. Thus for good results the diameter of the mouth should be at least equal to the wavelength of the lowest frequency with which the horn is required to deal. The relationship between wavelength and frequency for sound waves is

$$\lambda = \frac{c}{f}$$

where c is the velocity of sound in air. This may be taken as 1100ft/second. By substituting in this formula we find that for a frequency of 500c/s the wavelength is approximately 24 inches. Thus, if a horn is to be used down to low frequencies with good results the size of the mouth must be very large indeed. In an attempt to accommodate large horns within reasonable dimensions the technique of folding has been used. Folded horns are not particularly satisfactory for hi-fi work especially where more than one fold is used because the lengths of the paths of the sound waves are different for different paths.

MULTIPLE LOUDSPEAKER SYSTEMS

Because of the very wide frequency range required in high-fidelity systems (say 30c/s to 15,000c/s) and the corresponding wide range of sound wavelengths (37 feet to 0·9 inch) it is very difficult to design and manufacture a single speaker which has high efficiency and uniform response over the whole range. For this reason it is fairly common to divide the whole frequency range between two, or three, loudspeakers each designed specifically for a certain relatively narrow band of frequencies. Thus, in a typical three-loudspeaker system the low frequencies up to, say, 800c/s are handled by a large loudspeaker **(woofer)** in a suitable cabinet such as a vented acoustic chamber; the middle range, from 800c/s to 5000c/s are handled by a smaller, *e.g.* 8 or 10-inch speaker in a baffle; and the top range of frequencies, from 5000c/s up to the maximum is handled by a small high-frequency unit fitted possibly with a horn or a series of horns, and known as a **tweeter.** Some means must be provided for dividing the incoming sound into the three ranges required by the various loudspeakers. This is done by a set of filters or crossover networks. These may be arranged in the low level stages of the amplifier and followed by separate power amplifiers as shown in figure 12.6(a) or, more commonly, as shown at (b), a single power amplifier stage is used and the electrical filtering to separate the various bands is done by crossover networks connected across the terminals of the output transformer. One of the biggest advantages of using separate speakers is that the large excursions required to produce high powers in the bass do not modulate the middle and high frequencies and so an annoying form of distortion, known as **intermodulation distortion,** is eliminated.

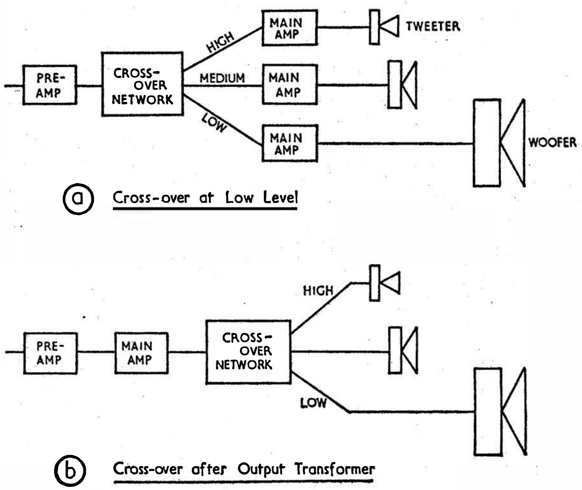


FIG. 12.6. CROSSOVER ARRANGEMENTS

Crossover Networks

The function of the crossover network is to divide the input among the various speakers according to frequency and in doing this to maintain the overall response as flat as possible. The networks are built up of components whose reactance varies with frequency, *i.e.* chokes and capacitors.

Dealing first with two-loudspeaker systems, figure 12.7 shows simple high-pass and low-pass networks which may be readily understood by

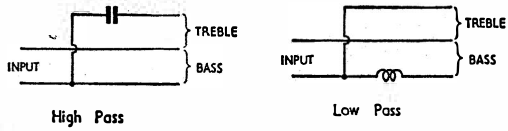


FIG. 12.7 SIMPLE HIGH-PASS AND LOW-PASS FILTER NETWORKS

remembering that the reactance of a choke rises with frequency while that of a capacitor falls as the frequency is raised. Simple circuits of this kind do not give a uniform overall response nor do they present a constant impedance at the input terminals. More advanced circuits, embodying properly designed filter sections are shown in figure 12.8 and figure 12.9 (three speakers). These

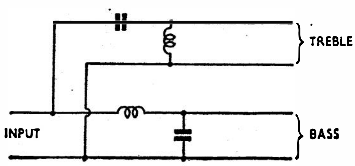


FIG. 12.8
CONSTANT RESISTANCE (HALF SECTION, PARALLEL) CROSSOVER NETWORK FOR TWO SPEAKERS

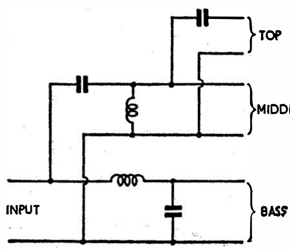


FIG. 12.9
CONSTANT RESISTANCE CROSSOVER NETWORK FOR THREE SPEAKERS

are examples of so-called **constant-resistance** circuits. Calculations for these are based on the assumption that the speaker impedance is purely resistive. This is not true in practice—the coil has complex impedance—but forms a good basis for design. Values are not given in these circuits as these depend upon the required frequency and on the impedance of the loudspeakers used. In general the chokes have inductances of a few millihenrys and the capacitors are of a few microfarads (say 4 to 20). It will be seen that suitable components for these networks are not difficult to find nor are they especially costly. Air-cored coils are best for this purpose as there is no chance of distortion due to saturation of the magnetic circuit. However, the working of the network is upset by excessive resistance in the chokes and it may thus be necessary to introduce iron cores in some cases in order to allow a reduction in the number of turns and in the resistance. Electrolytic capacitors are not suitable for use in crossover networks on account of their relatively low insulation resistance and the need for a polarizing voltage.

The crossover effect may be enhanced by using acoustic filters in addition to the electrical ones. For example, the baffle of a middle-frequency speaker may be of such size that the speaker response falls below a chosen frequency, or the cut-off of a horn may be employed.

The choice of crossover frequency depends upon the characteristics of the loudspeakers used and upon their enclosures; it is usually between 500 and 1000c/s, the actual choice being a matter for compromise. If the crossover frequency is made low the middle-frequency speaker has to be large to handle frequencies at the low end of its range, while if the crossover frequency is increased the risk of intermodulation distortion in the woofer is increased.

The directional effects in loudspeakers increase as the frequency is raised and are therefore most pronounced in the tweeter. Some means may be used to diffuse the sound over the listening area, a multiple (cellular) horn may be used, or the tweeter may be pointed upwards to face the ceiling.

STEREOPHONIC SOUND

The aim of a hi-fi system is to bring to the listener in his own room the sound exactly as he would hear it if he were sitting in the concert hall or studio where it originated. This the modern hi-fi system can do with near perfection save for one important exception. The listener in the concert hall receives his impressions of sound *via* two ears working together and receiving impressions by two different paths. One important aspect of this is that the listener has a full sense of direction in the music—he can tell automatically from which part of his field of hearing a particular sound comes. Whether this in itself adds to the enjoyment of music is a matter for doubt, but if one may compare the effect of listening with one ear to that of seeing with one eye then one may say that the music must be somewhat "flat" or lacking in roundness. Unfortunately there are no technical terms to describe the effect and no scale upon which to measure it. It is a thing about which people form their own ideas and about which therefore there will always be arguments. In stereophonic sound systems an attempt is made to get over this difficulty by providing more than one channel between performer and listener. For economic reasons the number of channels is usually limited to two (although more may be used—in the cinema up to six) and a schematic diagram of a two-channel stereophonic sound system is shown in figure 12.10. Two

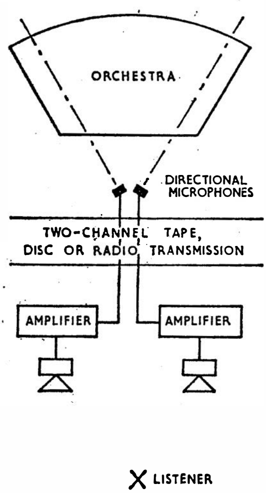


FIG. 12.10. STEREOPHONIC SOUND SYSTEM

microphones are used for which the directional characteristics and spacing as far as possible match those of a pair of human ears. These are coupled *via* two similar transmission (or recording) channels to a pair of loudspeakers or loudspeaker systems positioned so as to simulate, on the scale of the listening room, the centres of the two halves of the field of sound. For best results it would seem that the two channels should have exactly similar characteristics and that both should be up to full hi-fi specification. In practice it seems that good results can be obtained if only one channel has full frequency response provided that the power is balanced between the two channels and that neither produces audible distortion.

For the ordinary listener stereophonic sound is confined to gramophone records and tape although occasional experimental transmissions are broadcast by the BBC on Saturday mornings.

For the most part stereo equipment is similar to monaural equipment but there is twice as much of it. There are, however, some special problems which are peculiar to stereo equipment and the more important of these are discussed below.

CROSS-TALK BETWEEN CHANNELS

The stereophonic effect depends upon the small differences in the sound outputs from the two loudspeakers. Any interaction or cross-talk between the channels tends to average out the two signals thus causing the stereo effect to be diminished or lost. Cross-talk may occur in gramophone pick-ups, especially in magnetic types, and also in the pre-amplifiers and main amplifiers of disc or tape systems. It is unlikely to occur in the main amplifiers unless they are fed from a common power supply or are very badly laid out. Separate single-core screened cables must always be used for low-level signals.

HUM

Hum tends to present a greater problem in stereo equipment because (*i*) the recorded signal level may have to be reduced in order to avoid cross-talk in the recording, thus requiring greater overall amplification; and (*ii*) there is a greater chance of earth-loops being formed due to the duplication of equipment. Each channel should be earthed at the amplifier input and a direct connection to a good earth may be required in some circumstances.

BALANCE

If a sound has been recorded in the centre of the field it is reproduced in the centre only if the outputs from the two channels are properly balanced—with identical amplifiers and speakers the outputs should be equal. It may be necessary to make small adjustments to the gain of one of the channels in order to compensate for lack of balance in the recording, in the reproducing system, or in the acoustics of the listening room. While this can be done by using independent gain controls for the two channels it is much more convenient to have a main gain control which varies both amplifiers simultaneously and a separate balance control which alters the channel gains differentially, *i.e.* increasing one while decreasing the other. A balance control may consist of two ganged potentiometers so connected that one raises the signal fed to one output stage as the second reduces the signal fed to the other output stage.

PHASING OF LOUDSPEAKERS

The directional effect in the reproduction of stereophonic sound is largely due to small differences in phase between the signals fed to the two loudspeakers. The effect is lost if the two leads to one of the speakers are reversed for then the input to that speaker is changed in phase by 180°. If the speaker leads are colour coded or if non-reversible plugs and sockets are used care must be taken to see that phasing is not upset if the leads are lengthened or otherwise altered. Correct phasing can usually be checked by temporarily joining both loudspeakers in parallel and feeding them with a monaural input. If the phasing is right the sound appears to come from a point midway between the two sources: if wrong the apparent source of sound is indeterminate.

It is clearly desirable that the correct signal be fed to each loudspeaker, *i.e.* the sound picked up by the left-hand microphone must be reproduced from the left-hand loudspeaker.

For further details of loudspeakers and stereo see Bradley's *Records and Gramophone Equipment* (from the same publishers).

CHAPTER 13

DISC RECORDING

THE modern gramophone disc has a history which goes right back through the acoustic gramophone to the original phonographs of Edison and Bell. The early systems were wholly mechanical. The performer sang into the mouth of a large horn which concentrated the sound waves and caused a diaphragm located at the throat of the horn to vibrate. Vibrations of the diaphragm were transmitted through a system of levers to a cutting point which caused tracks corresponding to the sound waves to be impressed in a disc or cylinder of soft material such as wax or shellac. At the reproducing end the process was reversed—the sound being produced through the chain of reproducing stylus, lever system, diaphragm and horn. By manufacturing techniques which cannot be dealt with here it eventually became possible to make mass-produced copies of the original recording. The way in which the gramophone disc lends itself to mass production is possibly the greatest advantage which it has over other systems of recording for the commercial market.

The mechanical or acoustic gramophones suffered from several serious disadvantages. In order to give a loud enough result the modulation of the recording grooves had to be of large amplitude and it was therefore necessary to space the grooves relatively far apart—about 100 grooves to the inch. The bass response was poor because of the effects of the horns at recording and reproducing ends. The high frequency response was limited for reasons which a little arithmetic will make clear. Consider a groove on the inside of a record running at 78r.p.m. The radius of the groove is about 2 inches and so the speed at which the groove moves under the recording stylus is about 15i.p.s. (inches per second). A 5000-c/s note, having a periodic time of one five-thousandth of a second, thus has a wavelength in the recording groove of

$$\frac{1}{5000} \times 15 = 0{\cdot}003 \text{ inch.}$$

It is obviously impossible to record notes of this frequency with a heavy mechanical recording cutter. Conditions at the outside of the record are a bit easier. Here the radius is rather greater, almost 6 inches for a 12-inch record, but even so the high frequency response is strictly limited and the acoustic gramophone cannot go above about 2500c/s. Better results might be obtained by using either higher playing speeds or bigger disc radii but to do either of these would be to shorten the playing time of a record or to make the record unwieldy. In any case the materials used for record production in the early days produced a large amount of surface noise which masked any high frequency reproduction.

Another major disadvantage of the acoustic system was that the arrangements at both ends included a large number of acoustically responsive components, which produced many resonant peaks within the recording range. Electrical recording was introduced in 1926 and this led to immediate improvements in the quality of reproduction from commercial records. The following techniques used in the production of modern gramophone records should be compared with those mentioned above for the acoustic gramophone. New materials of non-granular form (*vinylite*) are now used for discs and as a result surface noise has been greatly reduced and is no longer a serious nuisance. These new materials also allow very fine indentations to be moulded with great accuracy and this, coupled with the full use of electronic amplification, has enabled the modern microgroove record to be developed. In this there may be up to 300 grooves to the inch—a common figure is 240. The finer grooves and smaller styli have enabled the high frequency response to be greatly improved and this may go up to 20,000c/s in experimental records. For commercial records in which the top limit is perhaps 15,000c/s it is possible to reduce the rotational speed as in the case of the 33⅓ and 45r.p.m. records now in common use. Records running at 16r.p.m. are used for speech where the frequency requirements are not so stringent as for music. Good bass response is obtained by the use of **de-emphasis.** This is a process whereby the amplitude of the bass frequencies is reduced before the signal is fed to the recording stylus by the use of frequency selective circuits in the recording amplifier. Thus the need for large excursions in the groove at low frequencies is done away with, but of course a suitable network must be included in the reproducing amplifier, or built into the pick-up characteristic, such that the bass response is boosted before the signal is fed on to the main amplifier and loudspeaker. A similar kind of thing is done at the high frequency end but in this case the object is to improve the signal-to-noise ratio and this is effected by **pre-emphasis.** In this process the top frequencies are boosted before being fed to the recording cutter and again a suitable equalizing circuit should be used in the reproducing amplifier. International standards were agreed for recording characteristics in 1954 and these are now in common use. These standards are shown in figure 13.1. Improvements in record materials and record techniques have had to be matched by improvements in pick-up design. Very lightweight pick-ups are now available with which it has been possible to remove all resonant peaks outside the recorded range.

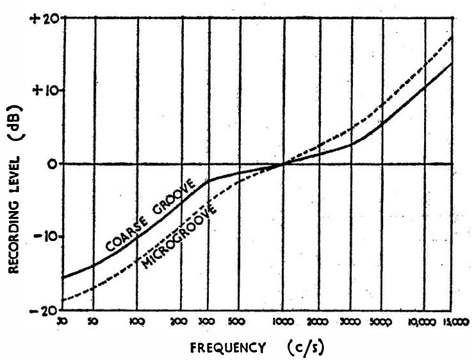


FIG. 13.1. INTERNATIONAL RECORDING CHARACTERISTICS

GRAMOPHONE PICK-UPS

Whatever type of pick-up is used some form of stylus is necessary to ride in the groove and transmit the recorded information from the groove to the pick-up itself. In order to cause minimum wear to stylus and record the stylus has to be very hard. Fabricated sapphires are commonly used for styli, but these are being replaced to some extent now by highly polished diamonds. These are dearer to buy than sapphires but may last fifty times as long. The stylus has a spherical tip of slightly greater diameter than the bottom of the groove, and it therefore rides on the walls of the groove somewhat less than half-way up. As the grooves for standard (78r.p.m.) and long-play records have different dimensions, different sizes of stylus are required for the two types. Other records, particularly old ones, may have other sizes of groove which require yet other stylus dimensions. The requirement to play different types of record has led to the adoption of pick-ups with interchangeable heads having different sizes of stylus. Where only two types of record are to be played, *e.g.* in the modern commercial radiogram, a turnover pick-up head is often used. This has two styli, either of which can be brought into action by rotating the pick-up head. (See figure 13.2). Interchangeable pick-ups and turnover heads are colour coded: GREEN for coarse groove; RED for micro-groove. Because of the minute dimensions and great precision which is needed in records and pick-ups it follows that these should always be treated with the greatest possible care.

THE CRYSTAL PICK-UP

This type of pick-up depends for its operation upon the phenomenon of piezo-electricity. The piezo-electric effect is exhibited by suitably prepared crystals of certain substances and results in the appearance of a potential difference between opposite faces of the crystal when this is strained mechanically. The magnitude of the potential difference is proportional to the strain applied. Modern crystal pick-ups use thin wafers of Rochelle-salt crystals or artificial crystals made from new ferro-electric materials such as barium titanate and lead zirconate. The latter is as sensitive as Rochelle-salt but it is not so fragile, its properties are not greatly affected by temperature, and it is not hygroscopic, *i.e.* it does not, as Rochelle-salt does unless carefully sealed, absorb atmospheric moisture. The crystal is supported in a gel whose characteristics are chosen to give optimum response over the audio range, and is driven by a simple cantilever. Two driving cantilevers with different sapphires can be accommodated, one at each side of the crystal, so forming a turnover head. A diagram showing the general arrangement of such a pick-up head is given in figure 13.2.

In modern crystal pick-ups the output voltage is lower than in early types but the frequency response characteristic is much smoother and the reaction

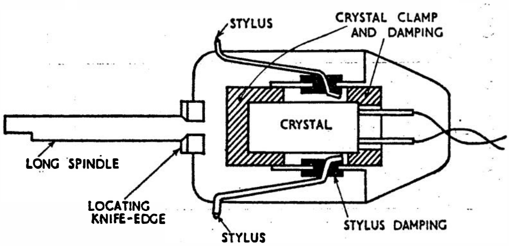


FIG. 13.2. TURNOVER PICK-UP HEAD

of the pick-up on the record is much less. The crystal pick-up gives a higher output voltage than either of the other types in common use and because it is surrounded by a viscous gel having carefully chosen properties which affect the response at high and low frequencies no equalizing networks are usually required in the reproducing amplifier—the required amounts of de-emphasis and pre-emphasis are inherent in the properties of the crystal cartridge as a whole. Crystal pick-ups act as high-impedance sources and should therefore be used with amplifiers having high input impedances, of the order of 0·5 megohm. The bass response of the pick-up is adversely affected if the input impedance of the amplifier is less than this figure.

THE MOVING IRON PICK-UP

In this type of pick-up e.m.f.s are induced in a fixed coil which is wound on a magnetic circuit energized by a fixed permanent magnet. The magnetic circuit includes an air gap in which is situated an armature of ferromagnetic material which is driven by the stylus. The reluctance of the magnetic circuit, and therefore the amount of flux from the permanent magnet linking the coil depends upon the position of the armature in the air gap and for small movements of the armature the change of flux is proportional to armature displacement. Modern forms of this pick-up are often referred to as **variable reluctance** pick-ups. A schematic diagram of a pick-up of this type is given in figure 13.3 where *A* is the armature; *B* is a flexible support for the armature made of rubber or a similar synthetic material; and *C* is the coil.

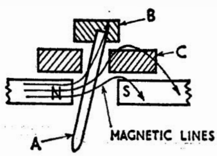


FIG. 13.3.
VARIABLE RELUCTANCE (MOVING IRON) PICK-UP

Moving iron pick-ups give output voltages of the order of one-tenth of a volt and so require a fair amount of amplification. As the output voltage depends upon pick-up velocity and as there is no gel or other medium to modify the characteristics of the pick-up equalizing networks are required and should be fitted in the amplifier. Moving iron pick-ups, by virtue of their construction, are always likely to pick up hum and care must be taken to ensure that this does not happen.

THE MOVING COIL PICK-UP

In this type of pick-up the output from the reproducing stylus is fed to a coil which moves in the field of a permanent magnet. As it does so e.m.f.s are induced in the coil. This type of pick-up is not common in commercial equipment but is capable of giving very good results. The number of turns on the coil is limited by the need to keep down the coil mass and so the output voltage is very small. It is usual to employ a step-up transformer between the pick-up coil and the input terminals of the amplifier. Both the pick-up coil itself and the transformer are liable to pick up hum from gramophone motors and mains transformers. The pick-up transformer especially may have to be carefully shielded and carefully sited in order to keep hum pick up to a satisfactory level. The moving coil pick-up, like the variable reluctance type, is a velocity-dependent device and so requires equalizing networks in the associated amplifier.

STEREO PICK-UPS

For monaural records there are two main ways in which the modulation can be put into the groove. Some early records used the "hill-and-dale" technique which as its name implies means that the stylus moves up and down in the groove in sympathy with the recorded sound. All modern monaural records use the technique of lateral recording in which the modulation causes the stylus to move from side to side. With the introduction of stereo it became necessary to record the output from two sets of microphones simultaneously yet independently in the same groove. One way of doing this would have been to use lateral recording for one channel and hill-and-dale recording for the other and this was used in experimental systems. For commercial use this system was discarded in favour of one in which the two channels are recorded each at 45° to the vertical, one on each side of the groove. The groove is traced by a single stylus which is connected by two systems of levers to two crystals or two magnetic heads. As the two sets of modulation in the groove, and the two sets of levers are at right angles to each other, then theoretically the modulation of either channel should go to its corresponding pick-up cartridge without producing any signal in the other cartridge. In practice there is a certain amount of cross-talk between channels but this can be kept down to a very low level in well designed equipment. The problem of cross-talk is much more difficult to solve with magnetic pick-ups because the two coils tend to be linked by stray flux. In this country crystal pick-ups are most common for stereo work. In this case the two pick-up elements are quite separate from each other electrically and so the production of cross-talk in the pick-up itself is minimized. The basic arrangement of a crystal pick-up for stereo work is shown in figure 13.4.

A smaller stylus (0·0005 inch, *i.e.* half the size of a monaural long-playing stylus) is required to trace the complicated groove of a stereo record and in order to minimize record wear due to the increased pressure on the stylus it is essential to cut down the playing weight. This demands greater precision in the tone arm.

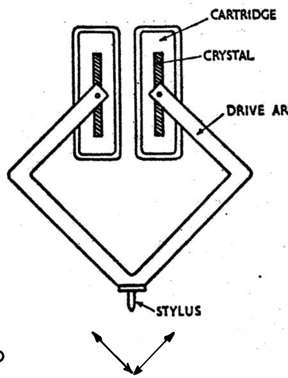


FIG. 13.4. CRYSTAL PICK-UP FOR STEREO

GRAMOPHONE TURNTABLES AND MOTORS

A gramophone turntable should run at dead constant speed and should not introduce any spurious signals into the pick-up circuit. Spurious signals may be produced either electrically or mechanically: electrically in the form of hum as already mentioned in connection with magnetic pick-ups; mechanically by vibrations, usually of low frequency (**rumble**) set up by unsatisfactory or worn bearings or in the transmission system between motor and turntable. Rumble may be particularly evident with stereo pick-ups because of their sensitivity to vertical motion of the stylus. With most commercial turntables constant speed is achieved by using a constant speed motor—the alternative, used in older equipment, is to use a variable speed motor in conjunction with a mechanical governor, but this arrangement is not satisfactory at low speeds, *e.g.* $33\frac{1}{3}$r.p.m. The most popular type of constant speed motor is the shaded pole induction motor; another type which has similar characteristics and which is widely used in tape recorders as well is the capacitor induction motor. On constant load the speed of either of these motors depends upon the frequency of the supply (they are suitable for use on a.c. supplies only) the applied voltage having relatively small effect. Changes of load, *e.g.* from loud to soft passages on a record, can cause changes of speed which show up as changes in pitch (**wow**) in the reproduction. Such speed changes can be reduced by various means—by the use of a more powerful motor or, more elegantly, by the use of a heavy turntable with a large inertia or flywheel effect.

As recording speeds are now standardized and as the frequency of the supply mains is kept constant within satisfactory limits—well within the $\pm 6\%$ required to cause a change in pitch of one semitone—there is little need for variable speed turntable drives.

Most modern gramophone motors are four-pole machines and run at approximately 1500r.p.m. on a 50c/s supply. The drive is transmitted to the turntable *via* a three-step or four-step pulley on the motor shaft and a rubber idler wheel which runs in contact with the inside face of the turntable rim. The three or four diameters of the motor pulley correspond to the different speeds at which the turntable is required to run. The mechanical qualities of the transmission must be of a high order to avoid speed fluctuations and rumble.

It should be noted that induction motors, like transformers, can be burnt out if connected to d.c. mains. Provided reasonable care is taken to keep the mechanism free by following the manufacturers' instructions regarding oiling and cleaning and provided the turntable is not held stationary while the motor is switched on there is little likelihood of electrical faults in the motor. In the shaded pole type of motor there is only one winding connected to the mains but in the capacitor motor there are two, one joined in series with the capacitor from which the machine gets its name. See figure 13.5. An open

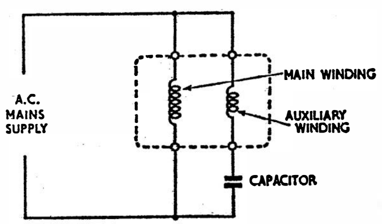


FIG. 13.5. CIRCUIT OF CAPACITOR MOTOR

circuit or short circuit in the capacitor will cause failure of the motor to start and rough running when taken up to speed by hand. A faulty motor capacitor should always be replaced by one suitable for operation on a.c. supply.

Great advances have been made in recent years in the design of highly efficient small d.c. motors for use with dry batteries. These are now available as driving units for portable record players. Direct current motors of this type use permanent magnet field systems and should not in any circumstances be dismantled or the magnet may be demagnetized.

AUTOMATIC RECORD CHANGERS

Modern automatic record changers are simple and robust and should not cause trouble unless they are abused or damaged provided that the simple maintenance required is carried out.

Adjustments are usually provided for stylus dropping position, pick-up arm height, and stylus pressure or playing weight. These adjustments must be carefully maintained if damage to stylus and record is to be avoided.

For further details of disc recording equipment see Bradley's *Records and Gramophone Equipment* (from the same publishers).

CHAPTER 14

TAPE RECORDING

THE idea of recording sound in the form of a pattern of magnetized particles on a wire or tape is a very old one but tape recording in its modern form dates from developments made in Germany during the second world war. Progress in the last few years has been rapid and good modern commercial recorders are capable of producing results at least equal to those from long-playing gramophone records. In addition it is relatively easy for an ordinary person to acquire the skill required to make good tape recordings in the home (disc recording is much more difficult and costly) and the making of recordings of music or speech is a very popular pastime. Tape recorders are also extensively used as dictating machines. This book, for example, has been recorded rather than written.

Tape recorders can also be used in competition with gramophone discs for the reproduction of commercial pre-recorded tapes. Future developments in this field, particularly with stereo tapes, may be awaited with interest. At the moment the number of pre-recorded tapes available is small and they are costly. A major difficulty is that of printing tapes in large numbers. Another difficulty has been the lack of standardization in tape recorder parameters both in the matter of tape and head dimensions and speeds, and of recording and replay characteristics. The latter problem has been solved in part by the issue of a standard by the International Consultative Committee for Radio Communication (C.C.I.R.) and this has also been adopted by the British Standards Institution. A tape recording made on a machine embodying C.C.I.R. standards can be played back on another such machine without the need of any modification.

THE RECORDING PROCESS

Space does not permit a detailed description of the theory of tape recording. For this the reader is referred to specialist books.

The basic idea used in tape recording may be understood by reference to figure 14.1. The signal to be recorded is fed to a coil wound on a magnetic circuit made up of high permeability steel laminations split into two halves. The two halves are separated by non-magnetic shims or gaps. The magnetic tape which consists of a coating of iron oxide particles in a suitable binding agent supported by a strong flexible plastic backing runs across the head at constant speed with the coating in contact with one of the gaps. As the magnetic particles pass the gap they are turned so that their axes lie along the direction of the magnetic field produced by the coil and so the tape leaves the head with a magnetic impression corresponding to the variations in current in the coil. In order to obtain a linear relationship between the electrical signal fed to the coil and the magnetic impression on the tape it is necessary to apply a high frequency bias (at 50 to 100kc/s) simultaneously with the signal.

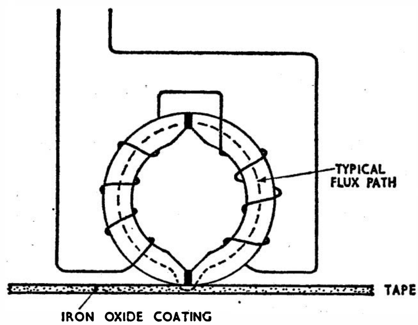


FIG. 14.1. BASIC ARRANGEMENT FOR TAPE RECORDING

The bias current must be of pure waveform and is obtained from a suitable oscillator. The output of the bias oscillator is also fed to an **erase head** whose function is to remove any previously recorded signal from the tape before it passes to the recording head.

On playback all bias is cut off and the tape passes over a gap in the playback head. This may be the same head as that used for recording or a separate unit designed specifically for its job. In either case the magnetized tape produces flux variations in the coil of the playback head and the resultant e.m.f.s are fed through equalizing circuits to the playback amplifier and loudspeaker.

On recorders equipped for superimposition and other trick effects means are provided to vary the current fed into the erase head. On others an interlock is usually provided to ensure that recordings are not erased inadvertently.

Most domestic recorders are two-track machines, *i.e.* only half of the tape width is used at a time and a second recording can be made in the other half, either by moving the heads through a distance equal to half the tape width and reversing the drive, or by interchanging the positions of the two tape spools. Four-track recorders are also available in which four separate tracks are recorded side by side on a standard one-quarter inch tape. On account of the narrowness of the track and the corresponding reduction in the strength of the recorded signal the signal-to-noise ratio in four-track machines may be inferior to that in two-track machines. Four-track recorders have two sets of heads and lend themselves well to superimposition effects and multiple recording.

TAPE SPEED AND DRIVE

The same basic principles relating noise, high-frequency response, speed and playing time apply equally to tape and disc methods of recording. Thus the faster the speed the better the high-frequency response for a given gap width (stylus size). Professional machines may thus run at 30 or 15i.p.s. while early domestic recorders had speeds of 15 or 7½i.p.s. Developments in head design and improvements in tape materials have enabled modern four-track recorders to be offered which run at 3¾i.p.s. with a frequency range of 30 to 16,000c/s or at 1⅞i.p.s. with a frequency range of 30 to 9000c/s. Recorders with these characteristics are not cheap: cheaper models running at the same speeds give results which are acceptable for the reproduction of popular music and speech.

Induction motors are used on recorders for a.c. mains operation while miniature d.c. or clockwork motors may be used for portable equipments.

In all cases the motors are similar to those discussed in connection with gramophones in Chapter 13.

The tape is driven at constant speed by an accurately machined pulley or capstan against which the tape is held by an idler wheel. In modern recorders the capstan rotates continuously on record or playback. When the start button is operated the idler wheel is moved smartly into position by an electromagnet which also carries the tape into contact with the heads. As only the mass of a short length of tape has to be accelerated very rapid starting and stopping can be obtained by energizing or de-energizing the electromagnet. This "quick-stop" feature is very useful in practice for editing. It also enables the recorder to be remotely controlled.

The mechanical driving arrangements of a tape recorder are complicated by the need to maintain uniform tension on the tape during recording and playback and by the need for fast wind and rewind facilities coupled with adequate braking arrangements. Some form of drive to the spool spindles is thus needed. This may be done mechanically, *e.g.* by friction drive, all the power coming from one motor; or separate motors may be used on the spool spindles.

STEREOPHONIC TAPE RECORDERS

The simultaneous recording and playback of two channels on a tape recorder presents little difficulty—it is an obvious extension of the two-track or four-track operation already mentioned. The amount of cross-talk between channels on the tape can be kept down to a very low level. The machine is complicated of course by the need for two-channels throughout and care is required in the arrangement of switches and wiring layout to avoid cross-talk from these components.

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