# advanced service technique

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*Prepared*
*By the Editors of*

**RADIO CRAFT**

# Advanced Service Technique

# preface

A FLOOD OF information is available to the service-man starting out on his career. Books are published on elementary circuit theory, servicing by comparison, by means of signal tracers, or by merely referring to reams of service notes based on case histories. For simple servicing he finds plenty of written material.

It is when he wishes to know a little more than the simplest facts about the apparatus he services—when he wishes to do radio servicing of a more advanced nature —that he finds a gap in the literature. Should he wish to improve the quality of an amplifier's output, instead of merely locating and replacing a leaky blocking con-denser—should he wish to install a complex audio sys-tem—he can find very little information indeed. There is a dearth of material to cover the middle ground be-tween the "simple servicing" books and engineering literature. Texts for his group are rare.

RADIO-CRAFT has long recognized this fact, and has attempted to bridge the gap. Articles on advanced servic-ing have been printed whenever they were available, which has not been often enough. We present here a collection of the best of those articles. They have been systematized and augmented or abridged where necessary to integrate them into this book, which is intended to give the radioman a better insight into the equipment on which he works, thereby increasing his ability to install and maintain radio and audio equipment.

# VISUAL ALIGNMENT

VISUAL ALIGNMENT has been used little in the past. Many servicemen believe it to be radically different from the output meter method. This is not the case.

Equipment necessary for visual alignment consists of the usual alignment tools (wands, wrenches, etc.), a cathode-ray oscilloscope, and a frequency-modulated signal generator. The first two items need little or no explanation. The last one, though by no means a recent innovation,
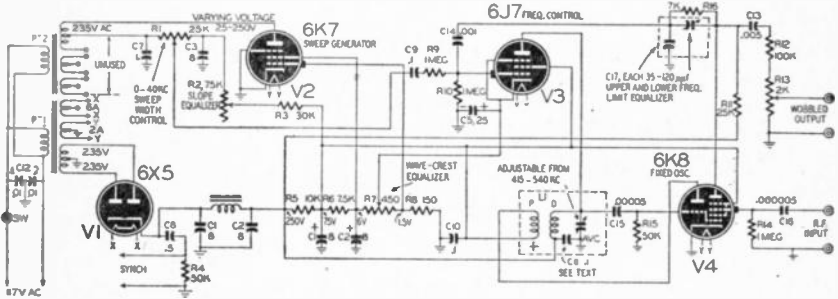


Fig. 1—Schematic diagram of a fully-electronic frequency-modulated signal generator.

has not been widely used in the servicing field and will bear some explanation.

A frequency-modulated signal generator is one whose output varies continuously in frequency above and below a given center frequency at a predetermined rate. For example, if the center frequency were 175 kc and the frequency deviation were ±5 kc, the frequency of the output signal generator would vary continuously from 170 to 180 kc and back again from 180 to 170 kc.

There are two general types of FM signal generators, or "wobbulators" as they are sometimes called. One type employs a motor-driven rotating condenser and the other develops the signal electronically. The rotating-condenser variety has become almost obsolete, so little need be said about it. There are a number of ways in which an FM signal can be electronically produced. The most popular method is to apply a low frequency control signal (usually 60 cycles) to the grid of a reactance

tube, causing the tube to act as a varying inductance across the oscillator tube, producing the tuning pattern of Fig. 2.

A schematic diagram of a typical instrument of this type which is capable of operating at mean frequencies from 415 to 540 kc and band widths of 0-40 kc is shown in Fig. 1. In this circuit the 6J7 is the reactance tube and the 6K8 is both a mixer and FM oscillator, the triode section performing the latter function. The band width may be regulated by raising or lowering the amplitude of the 60-cycle input to the 6J7 grid by means of R1. Once this adjustment is made, the mean frequency is swept back and forth through this range at a definite linear rate of 60 cycles per second by the 6K7 sweep generator. The band width is kept constant by using a second external variable frequency oscillator whose output is not modulated, and combining the two signals in the pentode section of the 6K8 tube as in a standard superheterodyne. The result of combining a frequency-modulated signal with an unmodulated r.f. signal is a third signal which is frequency modulated and whose mean frequency is equal to the difference between the two original frequencies. A few examples will make this clear.

We have a receiver with an i.f. of 175 kc and a pass band of $\pm 10$ kc. Assume that the mean frequency and frequency deviation of OSC. No. 1, which is the FM oscillator, are set at 460 kc and $\pm 10$ kc respectively. By adjusting the output frequency of OSC. No. 2, the external unmodulated r.f. oscillator, to 635 kc we would obtain the desired signal as follows:

| | | | |
|---|---|---|---|
| OSC. No. 2 | 635 kc | 635 kc | 635 kc |
| OSC. No. 1 | 450 kc | 460 kc | 470 kc |
| Output Difference Freq. | 185 kc | 175 kc | 165 kc |

If the i.f. frequency were 262 kc OSC. No. 2 would be set at 722 kc.

| | | | |
|---|---|---|---|
| OSC. No. 2 | 722 kc | 722 kc | 722 kc |
| OSC. No. 1 | 450 kc | 460 kc | 470 kc |
| Output Difference Freq. | 272 kc | 262 kc | 252 kc |

Note that in both cases, although the mean frequency is different for the output frequency, the band width remains the same. Of course, if the i.f. of the receiver lies in the range covered by the FM signal generator (415 to 540 kc) as is the case with the majority of receivers, it is not necessary to use the external signal generator because the wobbulator's signal is satisfactory in itself.
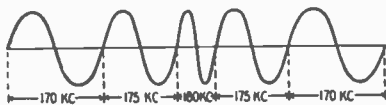


Fig. 2—The signal wobbles across the i.f. The amount of swing is fixed by the deviation control.

To discuss some of the units used to align FM receivers here is unnecessary, as their only essential difference from the unit in Fig. 1 is that they operate at higher mean frequencies and band widths in the neighborhood of 300 to 400 kc.

Amplitude of the frequency-modulated signal must remain constant throughout the entire frequency swing. The necessity for this will be seen in Fig. 3, which illustrates the shape of a typical resonance curve for a single- or double-tuned i.f. transformer which is peaked at 175 kc and has a band width of 10 kc. This curve tells us that the i.f. transformer for which it was drawn is designed to pass a signal frequency of 175 kc with minimum attenuation. Further, as the frequency is increased
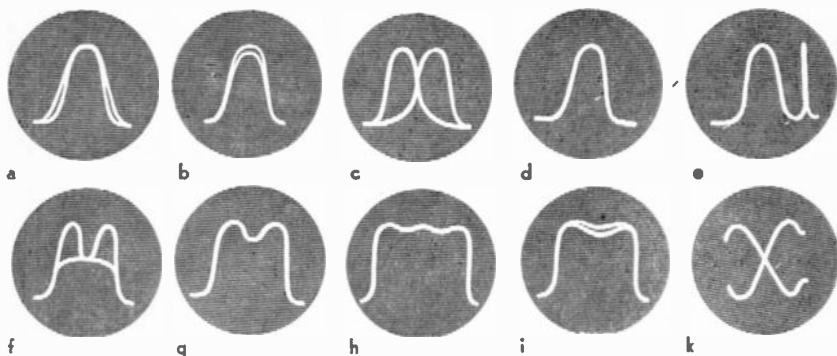


Fig. 5—These oscilloscope screen patterns indicate a variety of conditions in a set, which are explained in the text.

or decreased, the amplitude response drops off sharply until it falls below the level of audibility for all frequencies below 165 kc and above 185 kc. Therefore, the transformer, or strictly speaking, the resonant circuit, is said to reject or suppress all frequencies except those lying within the range from 165 to 185 kc. Note that the calibrations extend trom 160 to 190 kc to provide a base for the curve. The same must be done with the frequency-modulated signal to obtain a base for the trace on the screen of the oscilloscope.

The response curve in Fig. 3 could have been obtained by connecting a meter across the diode load resistor, recording the drop across it for a number of frequencies introduced by an ordinary signal generator, and then plotting the curve from the data. It is apparent that the amplitude of the input signal must be kept constant or results will be meaningless.

Because the cathode-ray oscilloscope serves as a voltmeter, it is possible to view the curve instantaneously in the form of a trace on the screen of the instrument. But to do this, the sweep frequency of the oscilloscope must be twice the rate at which the output of the signal generator sweeps across the band width and back again (from 170 to 180 kc and back to 170 kc). The trace representing the i.f. response curve will not stand still on the screen unless this requirement is fulfilled.

The sweep frequency circuit of the oscilloscope is a low-frequency oscillator usually combined with suitable amplifiers to produce the bright horizontal line across the scope face. The sweep-producing circuit is a *sawtooth oscillator*, producing a special type of curve. The voltage rises

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slowly, causing the spot of light to move across the oscilloscope screen, tracing out a line. On reaching its peak, the voltage falls suddenly to zero, causing the spot to fall back to the left side of the screen instan-taneously. Since the sweep frequency is set to twice that of the wob-bulator, the spot sweeps once across the screen as the signal generator is being swept from 165 to 185 kc, and a second time as it is being swept from 185 to 165 kc. In reality there are two curves instead of one, but the second curve is not visible because it is exactly on top of the first one. The reason for this will become apparent when it is noted that the response curve has the same shape whether you go from 165 to 185 kc or from 185 to 165 kc. Therefore, when the frequency of the FM signal reaches 185 kc the sawtooth voltage drops to zero, returning the trace to the left-hand side of the screen, and as the sawtooth voltage rises again, the frequency from the signal generator sweeps from 185 to 165 kc. As the response going from 185 to 165 kc varies exactly as it does in going from 165 to 185 kc, the second curve falls directly on top of the first, giving the appear-ance of a single curve.

In addition to double curves, single curves are also used. This is accomplished by keeping the saw-tooth voltage at zero during the period when the frequency of the generator passes from 185 to 165 kc. The resultant curve is shown at Fig. 4. The sharply rising line to the right of the curve is the second curve, but it is collapsed into a line due to the absence of the sawtooth deflecting voltage.
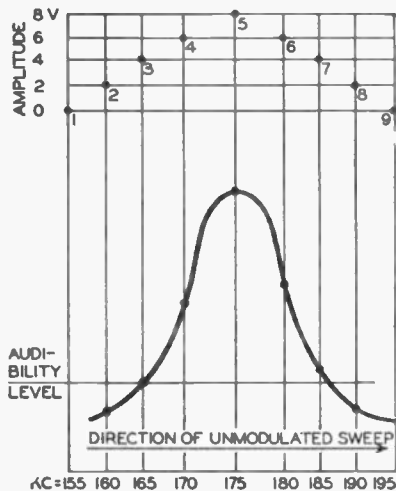


Fig. 3—Graph showing output curve, with constant signal input. The amplitude of the frequency-modulated signal must remain constant throughout the entire frequency swing.

The method of connecting the signal generator and the oscilloscope varies with the type of circuit to be aligned. In a receiver which uses a diode detector, the vertical plates of the oscilloscope should be connected across the diode load resistor. When the receiver employs a triode-biased detector, a satisfactory procedure is to open the bias resistor and by-pass condenser circuit, and then to connect the vertical plates of the oscil-loscope from the cathode to ground across a 0.5 megohm resistor which has been by-passed with a 250 $\mu\mu$f condenser. It is also advisable to short out the plate load; although this will invert the trace it does not affect the alignment procedure.

The FM signal generator is connected to the input of each particular stage, just as any other signal generator would be. The order of pro-cedure for visual alignment is identical to that which is followed when

an output meter is employed. Alignment starts at the back of the receiver and works toward the front, feeding the signal into the last i.f. amplifier input, the next-to-last, and so on, until the converter is reached. The object is to adjust the trimmers of the stage into which the signal is being fed, until a symmetrical curve is obtained. This applies equally as well to both the single- and double-curve methods. Fig. 5 shows some scope patterns commonly used.

The curves at a, b, and c of Fig. 5 indicate various conditions of mis-



Fig. 4—The "single-curve" alignment pattern as observed on the oscilloscope screen.
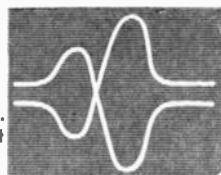
Fig.6—Alignment pattern for a.f.c. circuit. The trace shown is for correct alignment of such a circuit.

alignment. The curve at d indicates correct alignment. Adjustment of the trimmers will not always result in a symmetrical resonance curve. Slight departures from exact symmetry are permissible; however, if the departure is great, such as it would be when the degree of coupling is excessive and regeneration is present, the circuit itself should be suspected and examined.

Fig. 5-e is the same curve as that in 5-d except that the single-trace method was employed. Figs. 5-f and 5-g illustrate misalignment and alignment for a typical triple-tuned i.f. transformer such as might be encountered in a high-fidelity receiver. Fig. 5-h illustrates correct alignment for a 'transformer of similar design; the single-curve method was used. Note that there are two peaks in the curve at 5-g, and three in the curve at 5-h. The possibility of aligning a high-fidelity set as accurately as this with an output meter is extremely remote—the result is a loss in fidelity.

The same principles of alignment are applicable to FM receivers. Of course, the frequency range and the band width of the signal generator must be high enough to cover the FM band, as pointed out previously. The i.f. stages may be aligned by connecting the discriminator as a diode detector, so that the drop across the load will provide the vertical deflection voltage for the oscilloscope. Fig. 5-i shows the pattern which indicates correct alignment. After this the discriminator transformer is aligned so as to obtain a trace such as that shown at Fig. 5-k. This curve is obtained by first adjusting the primary until the linear portions of the curves are greatest; a secondary check is that peaks will have their maximum amplitude when correct alignment is obtained. The secondary is then adjusted until the intersection of the two curves lies halfway between the peaks.

Some higher-priced sets incorporate a.f.c., which must be checked for correct alignment. A representative trace which illustrates correct alignment is shown in Fig. 6.

# TRACKING PROBLEMS

EVERY SERVICEMAN has at some time been called on to align
a super which would respond to none of the usual alignment procedures.
In some cases an incorrect i.f. transformer replacement has been the
cause; in others a partially shorted coil or an unsuspected open in a
trimmer circuit. By far the greater number, though, are the result of
reconversion or unskillful adventure into set design engineering. Numerous
abandoned sets are from time to time modernized and put into action,
often by "cannibalizing" parts from other radios; t.r.f.'s are turned into
supers; and no few constructors "roll their own," in some cases winding
their own coils, in others taking them from old receivers.

Too many of these amateur engineers get unexpected results from their
completed jobs. Some of their receivers bring in stations at one end of
the dial only, others tune correctly on the high frequencies, while stations
are far from their correct markings on the lower ones. The opposite
trouble may be found, or all stations may be faint and crowded together
in one small section of the dial. The constructors are often sorely puzzled.

The reason for their troubles is that a superheterodyne includes two
distinct circuits tuned to different frequencies. These frequencies must
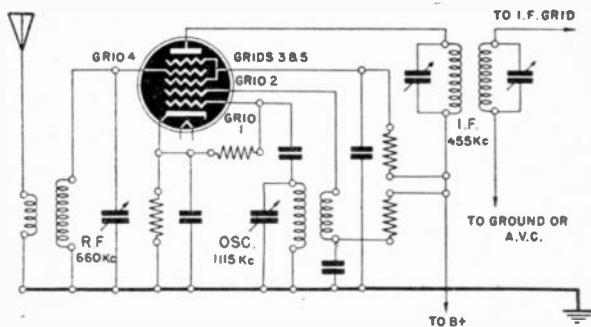


Fig. 1—The mixer stage of a simple superheterodyne, where
incoming signal and local oscillator output are combined.

be a definite distance apart at all points on the dial. Unless a set is
carefully constructed and adjusted, this distance is not maintained—the
set does not *track*—and such stations as are tuned in are the result of
accident, when the orbits of the two circuits cross each other or come
close enough to permit reception.

In its simplest form, a super starts out with a *mixer* tube, which is really two tubes in one envelope. See Fig. 1. One section of this tube (cathode, grid 4, and plate) acts like an ordinary r.f. amplifier. The coil
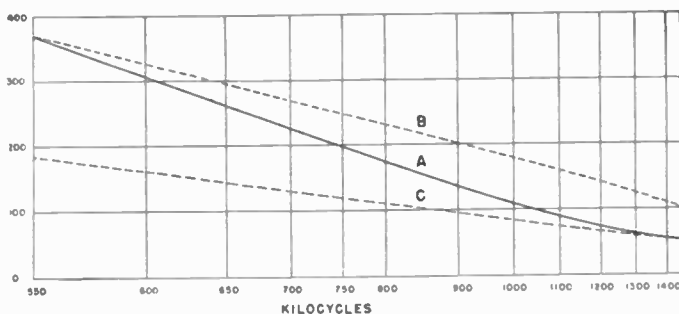


Fig. 2—These graphs show the result of the first attack on the problem of making superhet condensers "track".

and condenser connected to it are tuned to the frequency of the station received. The other section (grids 1 and 2) acts like a triode, and is connected in an oscillatory circuit tuned to a frequency usually higher by a defiinite number of kilocycles than the station being received. The screen-grid and plate circuits are shared by both sections of the tube. Consequently two r.f. currents flow in the plate circuit. One of these is at the frequency of the station being received, the other at the frequency of the "local" oscillator. These two are truly mixed in the plate circuit, The main result of the mixing is the appearance of a third frequency, which is equal to the arithmetical difference of the other two, and changes in strength with any variation in either of them. The signal from the oscillator section is fairly constant—that from grid 4 is modulated by the broadcast station, so the *difference frequency* (or beat frequency) is similarly modulated. An *i.f. transformer* in the plate circuit is tuned to the frequency of this modulated difference-frequency signal, and rejects or shorts the others.

If the difference frequency is 455 kc (as on many modern supers) and the i.f. transformers are tuned to that frequency, it is apparent that the oscillator must be tuned to 455 kc above the frequency of the station to be received. (It could be 455 below, as well, at the cost of making tracking problems worse.) To receive a station at 600 kc the oscillator must be tuned to 1055 to produce the correct "beat" frequency for the sharply-tuned i.f. transformers. If the r.f. section is tuned to 650 kc very little of the 600-kc signal will get through to the control grid of the mixer, and weak or no reception will result. The r.f. must keep in step with the oscillator, and 455 kc below it, all across the dial, for satisfactory reception.

These two circuits are usually tuned by one "gang condenser," so it is necessary that they be designed to track closely together. This is not easy. In a t.r.f. set, all stages are tuned to the same frequency at

the same time, and the only problem is to make all coils the same size. The two circuits of the super must be tuned to two *different* frequencies, and the difference between them must remain the same over the whole dial.

The difficulty is illustrated in Fig 2. Curve A is made with a 365 (maximum) micromicrofarad variable condenser and a 230 microhenry coil. The frequency is 550 kc with the condenser at 365 $\mu\mu$f and 1450 at 50 $\mu\mu$f. The oscillator circuit of the set must be so designed that, at any given setting, the oscillator frequency is 455 kc higher than the corresponding resonant frequency of the r.f. circuit.

A capacity-inductance table shows that to tune to 550 plus 455 kc with a 365 $\mu\mu$f condenser requires a coil of 70 $\mu$h. (Values are approximate, having been taken with a table and a slide-rule, but are accurate enough for our purposes.) When the condenser is turned down to 50 $\mu\mu$f, the resonant frequency of this combination is 2500 kc, not the 1905 we would like to have. (See curve B, made by subtracting 455 kc from the curve of the 70 $\mu$h-365 $\mu\mu$f combination, to show how close it comes to perfect tracking.) Only one or two stations close to 550 kc could be received with such a combination.

The attack might be made from the other end—the high-frequency one. To tune to 1905 kc with 50 $\mu\mu$f capacity requires a 144 $\mu$h coil. Curve C—made the same as curve B—shows how that would work out. Constructors who received only high-frequency stations on their radios will see what caused their troubles.

The trick is to find some means of making the oscillator tuning curve lie exactly 455 kc above that of the r.f. coil-condenser combination. Experience with superheterodynes has already taught us that this *can* be accomplished by means of semi-variable condensers. An ordinary trimmer would be of little value to us, as can be seen from cu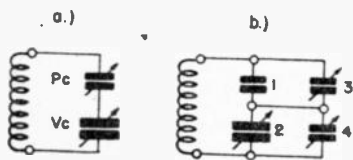rve C. To make the oscillator track at 1450 kc would require almost exactly 50 $\mu\mu$f trimmer capacity. Should we add that capacity by screwing down the



Fig. 3—Padders are used in most super-hets—two styles are shown above.

trimmer on the oscillator section of the condenser gang (supposing we had such a big trimmer) curve B would merely be lowered by 50 $\mu\mu$f right across the chart. Tuning would be out by 50 $\mu\mu$f at 550 instead of 1450 kc.

There is another adjustable condenser on most superheterodynes—the padder. This is *in series* with the oscillator variable condenser. Fig. 3-a shows the arrangement. It does not always look so simple. The padder on the broadcast band is usually made up of a fixed mica condenser with a trimmer shunting it, and schematics sometimes look like Fig. 3-b. The padder is 1; the oscillator section of the gang, 2; the large trimmer

across the padder, 3; and the trimmer on the gang, 4. The circuit is only that of Fig. 3-a with a trimmer across padder and tuner.

If two condensers are connected in series their joint capacity is smaller than that of the smaller one. This capacity cannot be arrived at by simple addition, but is expressed by the formula:

$$\frac{1}{C_1} + \frac{1}{C_2} = \frac{1}{C \text{ resultant}}$$

With this formula we can select a padder that will make the 144 $\mu$h coil and 365 $\mu\mu$f condenser track at the low-frequency end of the band.
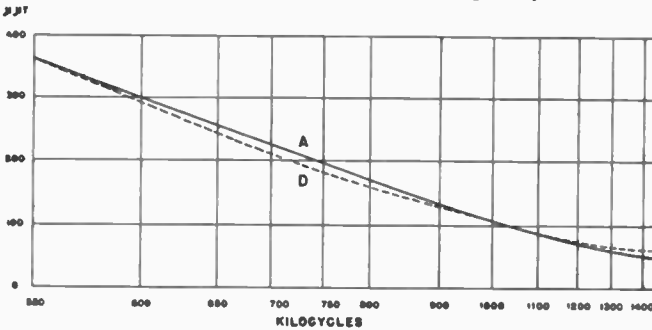


Fig.4—A is the original r.f. section tuning curve; D the result of adding a padder to the oscillator section.

According to the coil table, 176 $\mu\mu$f is needed. Subtracting 1/365 from 1/176

$$( \frac{1}{C_r} - \frac{1}{C_2} = \frac{1}{C_1} )$$    gives 1/350 approximately as the reciprocal of

the padder size. Using a 350 $\mu\mu$f padder, we get curve D (Fig. 4). A is our original r.f. tuning curve. Note that the padder throws tuning off only slightly at the high-frequency end, where it is much larger than the tuning capacity.

This is a great improvement. A set so lined up would work, though signals would be weak in the 650-850 kc region. It might be more effective to make the coil a little smaller, so that the two curves would coincide near the middle of the band. The padding condenser could then be made a little bigger to bring the curves together at some point near 600 kc and the trimmer could be adjusted to bring them together near 1400.

By varying the size of the coil, the padder and trimmer, it is possible to have the frequencies of the two coils in exactly the right relation at three points—near the top, middle, and bottom of the band, and to stray very little at any intermediate point. Fig. 5 is made with a 130 $\mu$h coil and a 390 $\mu\mu$f padding condenser. The tracking is almost perfect from

600 to 900. From there the two curves spread slowly apart. The trimmer can be adjusted at 1400 to bring them exactly together.

Because of the padder, a change in the trimmer capacity will not throw the tuning out as much at low-frequency points as it would in circuits without a padder. It requires about 9 $\mu\mu$f to bring the two curves together at 1400 kc. At 800, this extra capacity makes a difference of a little less than 5 $\mu\mu$f, and its influence rapidly disappears as the condenser is turned still further "in."

In actual practice, with two adjustments of trimmer and padder, the two curves can be kept close together practically from one end of the dial to the other.

## Be Sure I. F. Is Correct

The foregoing discussion is of little help to the person who has a set which fails to track, except as it acquaints him with the reasons for his difficulties. Fortunately, it is even easier to trim up a set of coils than to read about why they need such trimming. All that is necessary is to *free from the calibrated tuning dial all circuits but the one you are interested in at the moment.* Each circuit is thus adjusted independently of the others.

The first step is to be sure that the i.f. is correct. *This requires alignment with a good signal generator.* The signal generator is also useful in adjusting the coils, but not absolutely necessary, if you can identify a number of stations in different parts of the band. All that is needed is a variable condenser of capacity approximating that of one of the sections of the gang in the set.

Mount the new condenser firmly on some part of the chassis, or if impractical, make a good connection from its frame to the chassis and put it as close to the set as convenient. Then move the coil and grid connections from the stator of the r.f. section of the gang to the new "independent variable." Now the oscillator *and the oscillator alone* is tuned by the calibrated dial on the set.

Set the dial at 1400 or thereabouts, and rotate the independent con-denser. If the oscillator is tuned to a station it will be picked up at some position of the r.f. tuning condenser. If not, move the dial a de-gree or two and try again till one is tuned in. Identify the station and check its dial marking. If only slightly high or low, the oscillator coil may be the right size. If so, bring the dial to the correct point with the trimmer, then turn to a station near 600 kc and adjust padder, as in ordinary set alignment.

## Result of Too Large an Oscillator Coil

If the oscillator coil is too large, stations will tune in at much too high a figure on the dial—if too small, at too low a figure. Should the coil appear too large or small, it is a good idea to screw the padder

down pretty well, tune in a station between 800 and 1000 kc, and add or take off turns till the dial reads correctly. Then tune in a station near 1400 and adjust the trimmer till the dial is correct, afterward tuning in a station near 600 kc and adjusting the padder till the dial is also correct on it. Tune over the dial again, retrim and repad, possibly adding or taking off another turn or two for uniform tracking. Check again to see that all stations come in on the right dial setting.

To adjust the r.f. section, move the leads from the stator of the oscillator section of the gang to the independent condenser and put the r.f. leads back in their place. You will see now why the curves do not have
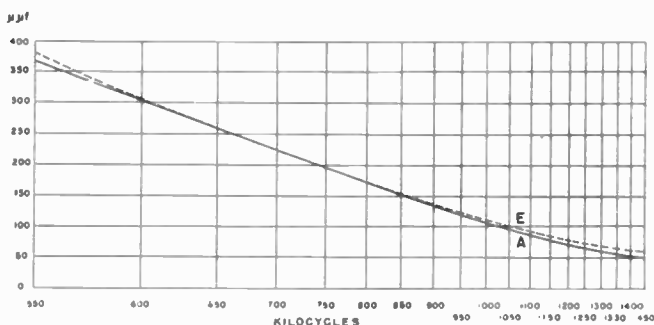


Fig. 5—With well-chosen coils, padders and trimmers, practically perfect tuning is possible, as explained in the text.

to lie exactly on top of each other. The r.f. section tunes rather broadly. Turning it a degree or so off maximum signal makes little difference in the strength.

Bring in the station near 1400 with the independent condenser and note whether the r.f. circuit is nearly correct or not. If reasonably close, adjust to the exact point with the trimmer and move up to the station near 600. If it is not loudest at its proper dial setting, add or take off turns till it also tunes in at exactly the right place. Correcting the r.f. section is easy compared to getting the oscillator lined up.

Bad tracking may occur because of wrong-size padding condensers rather than coils. If the padder is too large, stations will be crowded and displaced toward the high-frequency end of the dial, only those near that end being tuned in near their correct dial markings. If too small, the displacement is in the other direction, and stations are spread apart. A larger or smaller padding condenser is the remedy, of course.

Once the reasons for its action are understood a rebuilt super is not hard to adjust. If you follow the above method you can even wind your own coils with a fair chance of success.

# Superhet Coil Design

$\mathbf{M}$ANY EXPERIMENTERS like to design their own receivers and, if possible, design and construct their own coils. They are often stopped by the complexity of the design formulas concerned with finding the proper values of tuning inductance and capacity required for proper tracking of r.f. and oscillator circuits in a superheterodyne receiver. If they do happen to find long, involved, and exact formulas for the necessary calculations they then discover that exact values so arrived at will not work without further trial and error methods,

The simplified method outlined in this article is based on the assumption that for all practical purposes approximate values of inductance and capacity will enable the experimenter to go ahead and be able to select and construct his components intelligently. Necessary adjustments will always have to be made, regardless of the accuracy of the calculations. It is sufficient to find an approximate range of values which will serve as a goal to aim at when the experimenter begins actual construction.

## Design for 1 to 3 Megacycles

The first step consists of tabulating the characteristics of the projected receiver, its frequency range, its intermediate frequency, and so on. Let us follow a simple example through to completion as an illustration. For the sake of clarity round figures will be used wherever possible. Suppose you are designing a superheterodyne receiver to receive signals in the range of 1 to 3 megacycles and that you have available a two-gang variable tuning condenser with a capacity range of 20 to 180 micromicrofarads. The intermediate frequency is to be 500 kilocycles, or .5 megacycle. You must now know the value of inductance that will be necessary to permit your r.f. circuit to tune over the range of 1 to 3 megacycles with a 20- to 180-micromicrofarad variable condenser. This is found at either end of the band by the formula:

$$L = \frac{10^6}{4\pi^2 f^2 C} \text{ microhenries} = \frac{10^6}{39.44 \times 10^{12} \times 180 \times 10^{-12}} = 139 \text{ (approx.)}$$

Since the formula is in basic units—cycles and farads—the 180-$\mu\mu$f capacity is expressed as $180 \times 10^{-12}$. The other figures: 39.44 is $4\pi^2$ and $10^{12}$ is $f^2$.

The r.f. coil must have an inductance of 139 microhenries.

Next you will want to construct your oscillator tuning coil. Start at the high end of the first band first. The effective tuning capacity of the oscillator tank circuit at the high end of the band will be very nearly equal to the gang condenser setting at minimum capacity—in this case 20 micromicrofarads. We must therefore find the value of inductance which will tune with 20 micromicrofarads of capacity to produce a resonant frequency of 3.5 megacycles (since our intermediate frequency is .5 megacycle). Using the same formula again we now have:

$$L = \frac{10^6}{4\pi^2 f^2 C} \text{ microhenries} = \frac{10^6}{39.44 \times 3.5 \times 3.5 \times 10^{12} \times 20 \times 10^{-12}} = 100 \text{ (approx.)}$$

The oscillator coil must therefore have an inductance of 100 microhenries.

## Calculating Condenser Capacity

Now the inductance of the oscillator tank coil remains fixed under all conditions of tuning of the receiver. The oscillator circuit must be so designed that at the low end of the band it will produce a frequency of 1.5 megacycles with the same components. Starting with a coil of 100 microhenries' inductance, we now have to find that value of capacity which will make the combination resonant at 1.5 megacycles. This is given by the formula:

$$C = \frac{10^{12}}{4\pi^2 f^2 L} \mu\mu f = \frac{10^{12}}{39.44 \times 1.5 \times 1.5 \times 10^{12} \times 100 \times 10^{-6}} = 111 \text{ (approx)}$$

We must have a tank circuit capacity of 111 micromicrofarads. But at the low end of the band the capacity of our tuning condenser is 180 micromicrofarads. We must therefore connect a padding condenser in series with our oscillator tuning condenser of such a value that the series combination will have a capacity of 111 micromicrofarads. The correct value of padding capacity to do the job is found from the following:

$$\frac{1}{180} + \frac{1}{X} = \frac{1}{111}$$

$$X = \frac{180 \times 111}{180 - 111} = 290 \ \mu\mu f \text{ (approx)}$$

The padder will therefore have a capacity of 290 micromicrofarads. At this point our calculations have been completed and it is possible to proceed with our receiver.

Knowing the required inductance, it is possible to calculate the coil sizes to fit wire and forms at hand by the use of other standard formulas, or better, use one of the sets of charts by means of which number of turns may be found for almost any common size of wire and diameter of winding tube.

# NEGATIVE FEEDBACK

T HIS ARTICLE deals with the uses and types of negative feedback and the effects of ordinary types of feedback on power output.

Feedback as applied to most audio-frequency amplifiers is of the voltage type (i.e., no *current* or power is fed back). This feedback is classified as "voltage" or "current" according as the voltage fed back is proportional to the output voltage or output current. "Voltage" feedback is shown in Fig. 1-a, "current" feedback in Fig. 1-b. The voltage fed back is usually applied to the control grid (input grid or first grid) of the output tube(s) or to some preceding portion of the signal circuit, its phase being such as to reduce the gain. Why is this feedback used at all? It may be applied for one or more of four reasons:

(a) To prevent oscillation or frequency distortion caused by positive feedback (regeneration).

(b) To reduce wave-form distortion (amplitude or harmonic distortion).

(c) To provide frequency response variation as a tone control.

(d) To reduce the effects of an output load that varies with frequency.

An example of (c) is shown in Fig. 2.

Now it doesn't just mean that you apply some negative feedback and

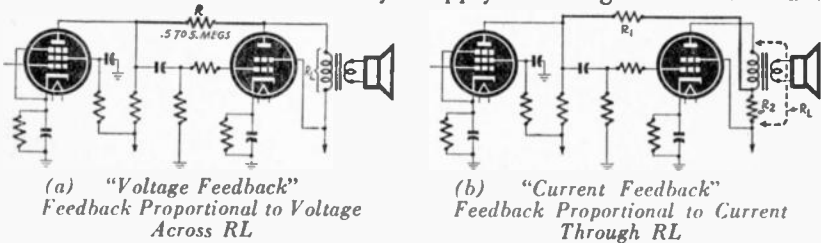| (a) "Voltage Feedback" | (b) "Current Feedback" |
| Feedback Proportional to Voltage | Feedback Proportional to Current |
| Across RL | Through RL |

Fig. 1—The two principal types of circuits for voltage and current feedback. The four general reasons for using feedback systems are explained in the text.

all the above happens straightaway! "Current" feedback, for instance, usually causes increased frequency distortion, especially where the distortion is due to a varying output load.

A résumé of the well-known mathematics of voltage feedback and the meanings of symbols employed will help the reader to understand.

$\alpha$=fraction of the output voltage that is fed back.

M=gain between point to which fed back voltage is applied and the output (before feedback is applied).

m=reduced gain after application of feedback.

$\beta$=gain reduction factor or $\dfrac{m}{M}$

Providing there are no phase changes then $\beta=\dfrac{1}{1+M\alpha}$ and this fraction $\beta$ gives the reduction in gain.

The same fraction, $\beta$, is also approximately equal to the ratio between



C GIVES FIXED BASS BOOST   R GIVES VARIABLE TREBLE CONTROL                         APPROX: WILL VARY WITH TUBES
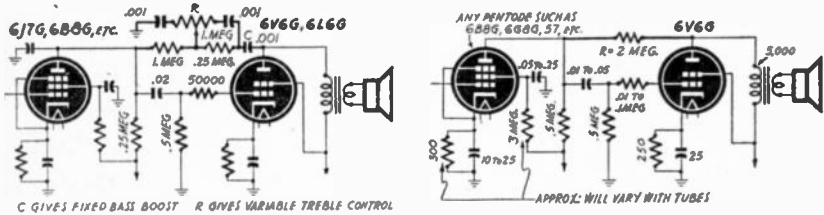
Fig. 2—left—Combined feedback and tone control. It provides both bass and treble tone control. Fig. 3—right—This diagram includes constants or circuit values for a given feedback ratio. The method of computing these values is explained by the author.

the amounts of frequency and harmonic distortion after and before application of negative feedback. Note the "approximately"!

Let us take an actual case: Suppose a 6V6G output tube operating under the usual conditions has a power output of 4.0 watts with a load of 5,000 ohms and a harmonic distortion of 8%. The peak output voltage is given by

$P = E^2/2R$, where $E$ is the peak voltage and P the power in watts.

$\therefore E = \sqrt{2\ PR} = \sqrt{4 \times 5000 \times 2} = 200$ volts.

As the input signal required for this output is 12 volts (peak), the gain M is given by $\dfrac{200}{12}$ or 16-2/3.

Suppose the harmonic distortion is to be reduced from 8% to 2%. Then a gain reduction factor of $\frac{1}{4}$ is required.

$$\beta = \frac{1}{1 + M\alpha} \qquad \therefore = \frac{1}{1 + 16\text{-}2/3\ \alpha}$$

$\therefore$ 16-2/3$\alpha$ = 3 and $\alpha$ = .18 or 18%.

That means that at least 18% of the output in voltage is to be applied to the control grid—say 20% or one-fifth to be on the safe side. One way of doing this is to connect the plate of the 6V6G to the plate of the preceding tube with a resistor which has a value equal to four (one *less* than five) times the resultant resistance of the preceding tube, its plate load resistor and the grid resistor in parallel. If the preceding tube is

a pentode with an a.c. resistance of approximately 2 megohms, the plate resistor and grid resistors each ½ megohm, then the resultant resistance is 2/5 megohm and the feedback resistor should have a value of 2/5 x 5 or 2 megohms. See Fig. 3.

## Effect of Feedback Often Falsely Judged

How will this affect the output power? Technicians generally answer this question pretty rapidly in one or two ways. Either the answer is, "It won't affect the power" or, "As the effect of feedback is to reduce the distortion of 4 watts from 8% to 2% then the power at 8% distortion must be more than 4 watts, so feedback increases the usable power output." *Both are wrong.*

Supplying feedback reduces harmonic distortion, but part of the output consists of this distortion, so the output is reduced also! Note: If the harmonic were out of phase with the fundamental, its removal would increase the effective power! More of this later. In practice it is found that (providing the tube is not run beyond cut-off or into grid current) the application of either voltage or current negative feedback to the control grid has a negligible effect on the power output. The closer the operating conditions are to optimum, the more negligible the effect. With pentode tubes there is generally a very slight decrease, with triodes hardly a change.

## Speaker Impedance Varies with Frequency

So far, all we've considered is a constant or "resistive" load. A loud-speaker has an impedance that varies with frequency. There is a fairly sharp rise of impedance around the bass resonant frequency and also a general rise in the "highs" due to the leakage reactance of the output transformer and inductance of the voice coil.

This rise of impedance causes a loss of power and a reduction in distortion in the case of a triode output tube but an increase in power and a big increase in distortion in the case of a pentode. In both cases there is an increase in gain, so if the feedback fraction $\alpha$ remains constant, there is an increase in the amount of feedback. This increased feedback has two results: First, there is a slight reduction in distortion due to the feedback itself. This effect is usually entirely masked by the second effect: The power output is decreased, not because there is less available (there *is* less in the case of the triode), but because the tube is no longer driven to its full extent.

At very low frequencies, the output load (speaker impedance) drops off rapidly, causing a drop in the available power and a slight increase in distortion. What about the feedback? This is reduced because the gain M is reduced, resulting in an increase of distortion and an over-driving of the output tube due to the excessive signal it gets, resulting in far worse distortion.

The results of applying voltage feedback, *assuming that the tube is driven just to its limit at midfrequencies,* are summarized in Table I.

One result is that triodes with inverse feedback are very unsatisfactory ‑‑they lack "highs" and the bass seems weak.

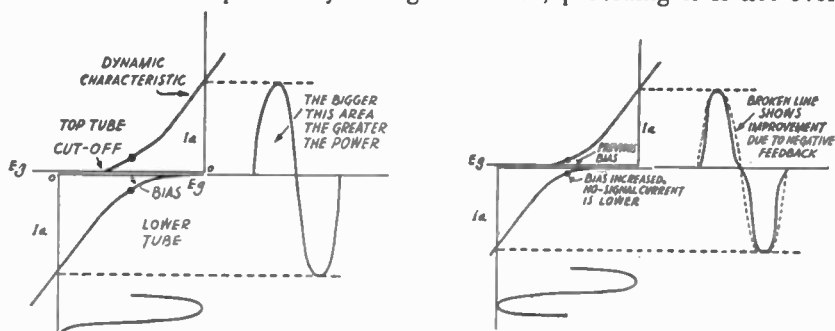Pentodes are improved by voltage feedback, providing it is not over-



Fig. 4-a—left—This diagram shows class B operation, with correct bias. Fig. 4-b—right—The effect of excess bias on class B operation is illustrated.

done. In both cases the frequency response curve is flattened by the application of feedback.

The table applies to tubes used singly or in class A1 push-pull while most of the remarks on power variation apply to single tubes.

When the output tubes are overbiased in an effort to obtain high efficiency, the application of negative feedback has a very beneficial effect in that it not only improves the wave form, but thereby increases the power output (for a sine wave). This is shown in Fig. 4-b where the excess bias causes the inverted grid characteristics to shift to the left.

One point not considered so far is the effect of the feedback on the stages *before* which it is applied. As the feedback reduces the gain, greater voltage outputs are demanded of the earlier stages, with a consequent rise of distortion. This is particularly noticeable when triodes are employed in the output stage because triodes require high driving voltages—often over 100 volts peak. In such cases the feedback should not be excessive and should be taken as far back as possible, over at least two stages and preferably three.

## TABLE I—Effects of Voltage Feedback

| FREQUENCY | TRIODE | PENTODE |
|---|---|---|
| Very Low Frequencies. | Distortion increases. Large reduction in power. | Slight increase of distortion compared to the decrease in middle register. Reduction in power. |
| Bass Resonant Frequency | Large Reduction in power. Decrease in distortion. | Slight increase in power. Increase in distortion due to load. Decrease in distortion due to feedback. |
| Midfrequencies. | Decrease in distortion. | Decrease of distortion. |
| Highs. | Large Decrease of power. Decrease of distortion | Slight increase of power sensitivity over that at midfrequencies. Rise of distortion due to load. Decrease of distortion due to feedback. |

# Tone Control Circuits

<span style="font-variant:small-caps">B</span>ECAUSE OF the expense, physical size, and sharper resonant peaks, inductance-capacitance types of tone control circuits are often impractical or otherwise undesirable. Here are five inexpensive, physically small, and broadly resonant resistance-capacitance tone control circuits.

Circuit 1 provides fixed bass reinforcement through the series parallel arrangement R1, R2, and C1. Condenser C1 offers very low reactance to the higher audio frequencies, thus attenuating the treble notes, and offers high reactance to the lower audio frequencies, permitting the bass notes to pass through to the grid of the bass amplifier tube, V1. C2 and R3 form a series circuit which offers higher reactance to the bass frequencies than it does to the treble, thus permitting the high notes to pass through to the grid of the treble amplifier tube, V2. R4 is an ordinary gain control, and by varying the signal input to the treble amplifier and keeping the bass amplifier at a constant level, the effect is that of a true tone control, giving full bass response at all times. There is no audible interaction between the two tubes, V1 and V2. (In the experimental circuit a 6N7 dual-triode tube was used. V1 is the upper and V2 the lower half of the tube in Fig. 1.)
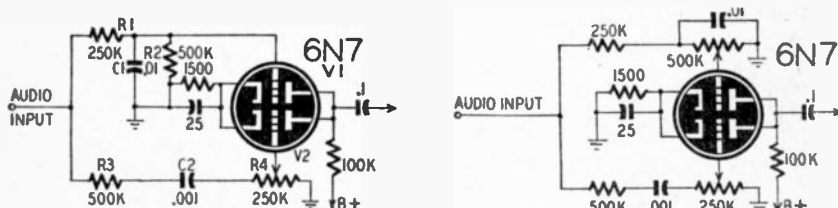


Fig. 1—left—A simple resistance-capacity circuit which provides fixed bass reinforcement. Fig. 2—right—In this tone-control circuit the treble and bass controls are variable.

Circuit 2 is similar to Circuit 1, with the addition of a gain control in the grid of the bass amplifier. This makes for a more flexible equalization of tone. It is particularly recommended for use in recording amplifiers where varied equalization problems are encountered. With this circuit, the fidelity response peculiarities of cutting heads, play-back pickups, amplifiers, and loudspeakers can be compensated for. Using this arrangement in a recording amplifier with a cheaper cutting head, the results

as judged by "good ears" were all that one could desire. Again, a 6N7 and the part values shown were used.

Circuit 3 is another arrangement of dual tone controls (bass-treble) using a single triode instead of a dual triode and mixing the bass and the treble in the same grid. The gain in this circuit is appreciably less
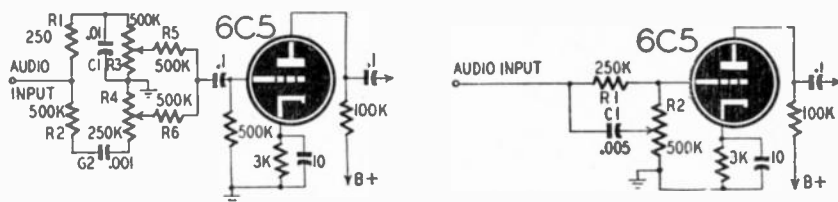


Fig. 3—left—Bass and treble controls are provided in this single triode circuit. Fig. 4—right—Simplest type of duo-control circuit; the tone response is varied as the potentiometer R2 is rotated.

than that of the previous tone controls, due to the 500,000-ohm series resistors (R5, R6) in the arms of the potentiometers (R3, R4). These resistors decouple the two potentiometers in order to limit their interaction to an inaudible degree. A single 6C5 triode and the parts values given were used in the finished model.

Circuit 4 combines the features of the three previous circuits. The condenser C1 goes to the variable arm of the potentiometer R2 and, as the potentiometer is rotated from the grid side to the ground side, the tone response is varied from treble to bass. With the condenser at the grid side, the bass notes are attenuated due to high reactance of C1 and resistance of R1. With the condenser at the ground side of R2, high notes are by-passed due to the low reactance of C1 at higher frequencies. Intermediate positions of R2 give varying response and fidelity characteristics to suit the ear of the listener. A 6C5 triode and the part values shown were used in the model.

## Familiar Type of Tone Control

Circuit 5 is the familiar tone control used in most home receivers and many public address systems. As R1 is varied, bringing C1 closer to ground, greater attenuation of the higher frequencies occurs, making the response more bass, or less treble. The disadvantage is that no bass accentuation, reinforcement, or compensation is provided for. Its simplicity of installation and comparatively low cost make it quite popular.

All of these circuits cause some loss in gain. This is caused by the resistors necessary to decouple the separate bass and treble mixers. However, the results achieved by these simple arrangements for controlling tone more than compensate for the lost gain.

The reason for the losses in gain become more apparent when considering R1 and R2 and their arrangement in Circuit 1. Resistors R1 and R2 in series are in effect a voltage divider, and when a signal voltage

appears across the input, a voltage drop appears across R1 and R2. This voltage drop can be computed by this simple formula and a little elementary algebra.

$$E_{grid} = \frac{R_g \times E_{input}}{R_{total}}$$

For example, Fig. 6. If we consider a phonograph pickup having a voltage output of .85 volts and connect it to points (a) and (c), what voltage will appear at the grid of the tube, points (b) and (c)?

$$(1) \quad E_{grid} = \frac{R_g \times E_{input}}{R_{total}}$$

$$(2) \quad E_{grid} = \frac{500,000 \times .85}{750,000} = 0.566 \text{ volt}$$

Note that in some of these circuits tube grids are connected directly to the audio input, without a series condenser. In these circuits, no d.c. may be applied from the preceding stage. Where the control connects
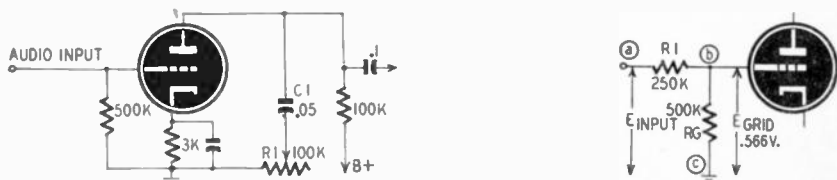


Fig. 5—left—The old treble-cutting tone control used in many small receivers and PA systems. No bass reinforcement is provided. Fig. 6—right—Theoretical tone control circuit, showing that a drop in potential occurs due to the resistance in series with the tube grid.

to the secondary of a transformer, as in certain audio and magnetic pickup circuits, no precautions need be taken. When connected to a crystal pickup, a resistor of at least one-half megohm should be bridged between input and ground. If operated after a resistance-coupled plate circuit, a *large* blocking condenser (0.1 $\mu$f) should be inserted following the plate resistor, to keep plate voltage off the grids of the tone control tubes. No grid leak resistors are needed, as they are provided for in all these circuits.
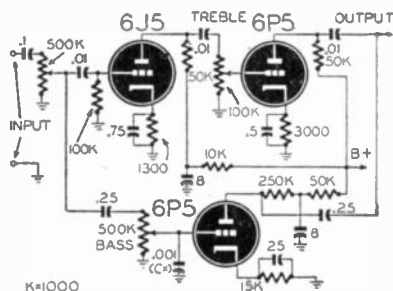
# An Improved Compensator

$S$EARCHING FOR a really effective tone compensator, we built and tried out every compensator found diagrammed in books and magazines—eight or ten in all. Unfortunately every one of them left something to be desired.

Going back to simple fundamentals, we contrived the tone compensator diagrammed here. It is thoroughly conventional, very simple, and one of the most effective compensators we have ever found. It uses only standard resistors and capacitors—no chokes to pick up inductive hum—and its separate controls for bass and treble have absolutely no interaction. It produces a concave response curve, the depth of which can be varied to any desired degree, which is exactly the correct effect for music played at low volume or moderate volume. It will also boost or attenuate either highs or lows alone.

The incoming signal is split between two channels. The upper channel handles the treble. The use of small coupling condensers, small cathode by-pass condensers, and resistors of low value in both the plate and the grid circuits, all tend to attenuate the lows. The lower channel handles the bass and is a straight stage of resistance-coupled amplification except for the condenser Cx shunted between grid and ground—an old-style "tone control," this one being fixed.

This compensator can be easily adapted to any given audio situation. If a stronger bass response is desired, the high-cut-off condenser Cx may be made larger, the exact value to be found by experimentation; or a second stage may precede the one shown. If a lighter bass response is desired, make condenser Cx smaller, the cathode by-pass condensers smaller, the coupling condensers smaller, the plate and grid resistors smaller—one or all of these. In the same way, if less



In this tone compensator the incoming signal is split between two channels; the upper one handles the treble, the lower one the bass.

emphasis is desired on the treble, the coupling condensers may be made larger, the cathode by-pass condensers larger, the plate and grid resistors larger, or only one stage may be needed. An understanding of the simple working of this compensator will enable you to adapt it to any need.

# The Bridged-T Circuit

$T$HE BRIDGED-T circuit has been placed upon a shelf and al·most entirely cast aside by the home experimenter, ham, serviceman, and design engineer, because of its seeming complexity. In reality, this circuit is very easy to handle, and may be applied with little difficulty by any-one with a little radio knowledge.

The Bridged-T is actually an R-C filter network, and when employed as shown in Fig. 1, will attenuate or suppress only one frequency. In other words, if we knew that several widely separated frequencies were present in the plate circuit V1, Fig. 1, we could by choosing the proper values of the Bridged-T following V1, suppress any one of those frequencies. All other frequencies would ·appear at the output of the Bridged-T (grid circuit of V2, Fig. 1).

The circuit as shown in Fig. 1 is applicable as a scratch filter in high-or low-fidelity phonographs, and as a "whistle eliminator" in high-fidelity,
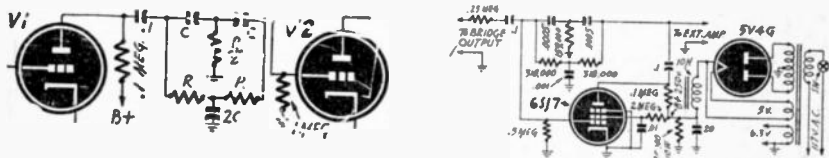


Fig. 1—left—A standard Bridged-T filter circuit, suitable as a scratch filter and as a "whistle eliminator" in high-fidelity, wide-band BC receivers. Right—Schematic diagram of the self-contained filter.

wide-band broadcast receivers. The whistle or heterodyne is caused by two stations in adjacent channels beating with each other and giving rise to a high frequency audio note. The scratch noises and whistles are subdued by choosing the components of the Bridged-T to suppress 10,000 cycles. Frequencies much above 10,000 cycles are generally not reproduced by even good phonographs or broadcast receivers, and therefore noise in this portion of the spectrum is not our concern.

The values of R and C for some very common frequencies are shown in the table; also the mechanics of computation for a 2000-cycle filter. For frequencies other than indicated, component values may be chosen and substituted in the formula for computation of the 2000-cycle filter.

The great advantage in using an R-C network of this type is that no bulky and expensive iron-core inductances are necessary as in many of the more common frequency-discriminating circuits and filters.

When the Bridged-T is applied as shown in Fig. 2 an interesting phenomenon takes place. The tube $V_2$ will amplify only one frequency. In other words the circuit becomes a selective amplifier and rejects all but that one frequency. The constants and frequencies shown in the table below may be substituted for R and C in Fig. 2 and the desired amplifier characteristics obtained.

Thus, if a selective amplifier is required in a c.w. receiver for code reception to cut down static, noise, and interfering signals, a Bridged-T



Photo shows Bridged-T filter with self-contained power-supply.

may be employed. Most c.w. signals sound best to the ear when heterodyned to give a 1000-cycle beat-note. A selective amplifier operating at 1000 cycles will permit only one signal to pass through and suppress any other signal producing heterodyne notes of a higher or lower frequency than 1000 cycles. Naturally, static and noise are also considerably reduced, as this type of interference is generally high-frequency audio and much above 1000 cycles.

The foregoing represents only a few of the many uses of the Bridged-T. The author constructed the Bridged-T selective amplifier shown in the photograph.

Excellent results were obtained when it was used in conjunction with a capacity and inductance bridge, to suppress the harmonics of the internal 1000-cycle mechanical vibrator.

## Other Applications

Other applications would call for changes in the R and C constants of the circuit to meet the special conditions of the given job. Used for phonograph scratch filter, as suggested earlier in the article, these constants might be adjusted to 5000 cycles or slightly higher. At 10,000 cycles, as stated previously, the circuit might be used to cut out 10-kc beat notes between adjacent broadcast stations.

The writer has not attempted to work with a Bridged-T circuit having a variable resistor, though the idea immediately suggests itself in connection with many kinds of experimental work. A unit could also be constructed which could be switched from a "rejector" to an "acceptor" circuit, thus making it more versatile for use in the home laboratory or shop.

It is hoped that this will direct the attention of the technician to the many possibilities of the Bridged-T circuit, as well as to the ease of designing and constructing it. Design formula is given on the next page.
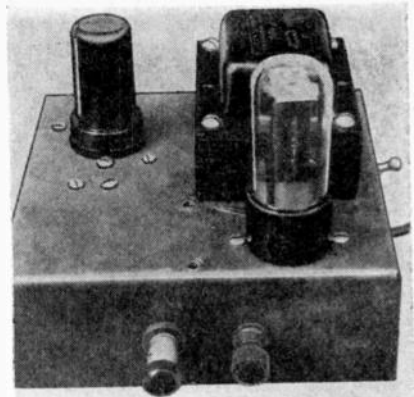
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| R AND C FOR COMMON FREQUENCIES | | | | |
|---|---|---|---|---|
| Cycles | C—$\mu$f | R—ohms | 2C—$\mu$f | R/2—ohms |
| 60 | .005 | 530,000 | .01 | 265,000 |
| 120 | .005 | 265,000 | .01 | 132,500 |
| 400 | .001 | 398,000 | .002 | 199,000 |
| 1,000 | .0005 | 318,000 | .001 | 159,000 |
| 5,000 | .00025 | 127,000 | .0005 | 63,500 |
| 10,000 | .0001 | 159,000 | .0002 | 79,500 |
| All tolerances should be held within 5%. | | | | |
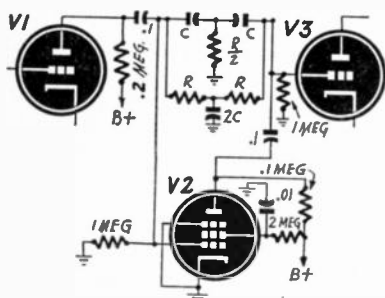
The basic formula is:

$$(1) \quad f = \tfrac{1}{2}\pi \ RC;$$

or solving for R:

$$(2) \quad R = \tfrac{1}{2}\pi \ FC;$$

F=frequency in cycles

R=resistance in ohms

C=capacity in farads

$2\pi = 6.28$

For audio frequencies other than those shown in the table and falling between 60 and 10,000 cycles, C is chosen as the value corresponding to the next highest frequency. This value of C is substituted in formula (2) and R obtained.

Let us suppose that it is necessary to have the filter operate at 2000 cycles. The next highest frequency above 2000 is 5000 cycles. The value of C corresponding to 5000 cycles is .00025 $\mu$f, and is substituted as shown below:

$$R = \frac{1}{(6.28) \ (2000) \ (.00025[10]^{-6})}$$

Therefore

R=318k ohms approx

C=.00025 $\mu$f

R/2=159k ohms

2C=.0005 $\mu$f

# Adding Microphones to Sets

$T$HERE IS a great deal of interest in adding "mikes" to radios. How the job can be done in a number of representative cases is our theme here. First, of all the types of mikes available, probably the easiest to get working is the old carbon-button variety. The single-button carbon mike has the highest output voltage, but the quality will not be as good as that obtained with a double-button unit or one of the more modern types such as the crystal or dynamic.

A simple hookup for a single-button carbon microphone is shown in Fig. 1. The value of R may be 250,000 ohms. In the average circuit, with the mike connected to the radio receiver, the load impedance will be about 100,000 ohms. The transformer may have a turns ratio of about 30 to 1, step-up, for a 100-ohm mike. About .13 volt may be obtained

Fig. 1.—right—The carbon microphone requires only a dry cell and a transformer. The output can be plugged into the set's phono jack.
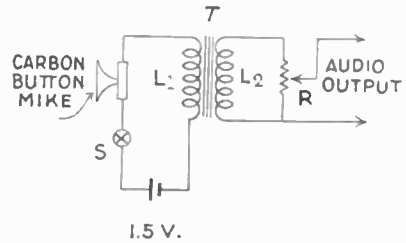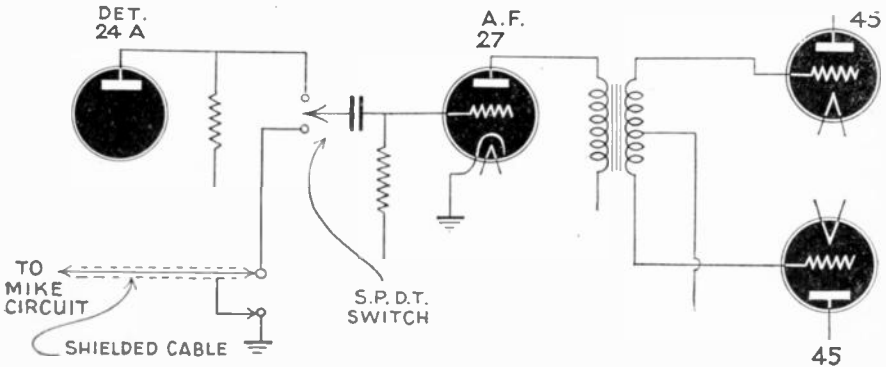
Fig. 2—How a microphone can easily be connected to a typical receiver, a Philco in this case.

across the primary and 4 volts across the secondary. (Steady whistle at 1 ft. distance into mike.) The output level of such a mike will be about —15 db (reference to .006 watt). If the bias on the first audio amplifier

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tube is 5 or 6 volts the tube will be swung to full output on peaks. (4 × 1.4 equals 5.6).

Voltage on carbon microphones should be kept as low as feasible to get good signal volume. They are commonly run underloaded, with one
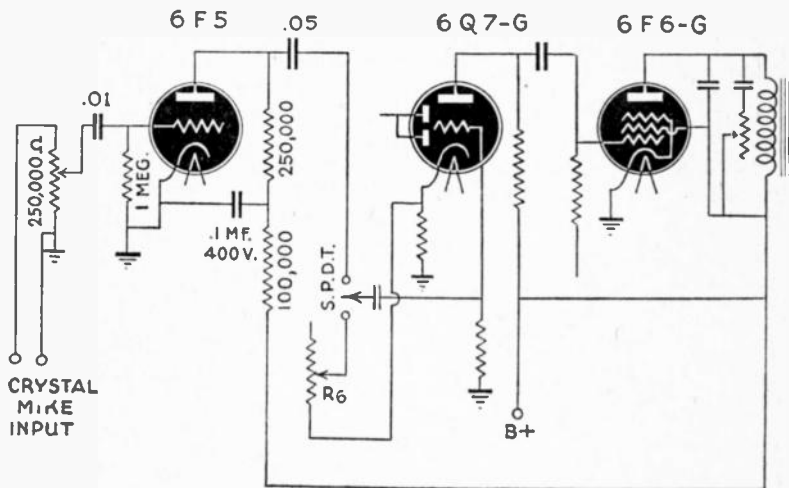


Fig. 3—The crystal mike usually needs one stage of pre-amplification; it is shown adapted to a Zenith 5516.

dry cell (flashlight battery) on this job, but if the sound is not loud enough, a second cell can be used. Consult the manufacturer's specifications in such cases, and use a rheostat (about 500 ohms).
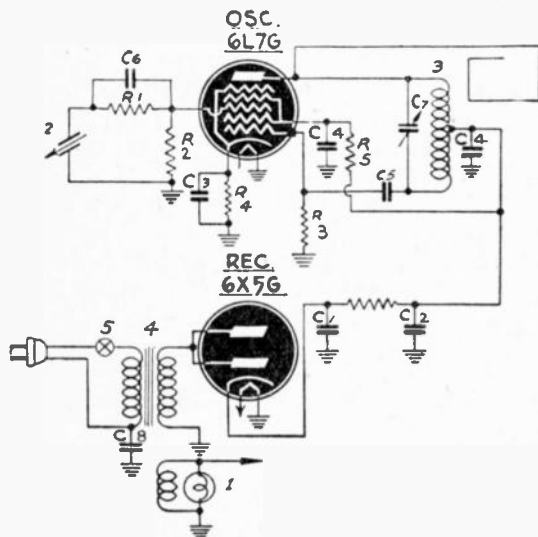


Fig. 4—The wireless phono oscillator has its own power supply and requires no connection to the radio set.

A hookup for the Philco 76 is shown in Fig. 2. (This adaption can be used for a record player as well as a mike.) The mike cable should be shielded and the mike itself must be turned on its axis so that a minimum of sound from the radio loudspeaker reaches it. It is usually best to use the mike in one room and the radio in another to prevent howling. Acoustic feedback occurs when the loudspeaker sound gets into the mike and is reamplified.

The same principle can be used in connecting any type of microphone to almost any standard radio set. In sets using a diode-triode or diode-pentode tube as detector and first audio amplifier, the switching circuit is of course placed in the lead to its grid. In some cases, the volume control in the set can be used in place of R in Fig. 1. The switch would then have its center arm connected to the "top" end of the volume control, and would switch it to either the return lead of the diode circuit (normal radio position) or to the microphone input.

The crystal mike has a high impedance, so it is impractical to use a step-up transformer for boosting the gain. Instead, an additional stage of amplification can be added. The power for operation of the stage is derived from the radio itself. Fig. 3 shows a hookup of a crystal mike to a modern receiver. The output of a crystal mike varies according to
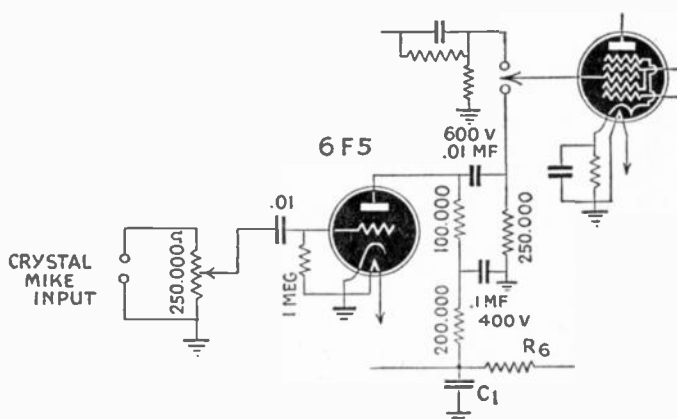


Fig. 5—How crystal mike may be connected to a Zenith S-6622 wireless record player.

the type and manufacture, ranging from —55 to —90 db, with —60 db being an average figure (.006 watt reference level).

The extra stage of amplification will give plenty of gain, so that you don't need to bellow into the mike to get a good output signal. Quality of reproduction will be better than that obtained with the carbon-button mike.

The tube can be mounted on a small chassis of steel or aluminum and the chassis can be screwed to the inside of the cabinet. A shielded wire can be run from the 6F5 plate to the single-pole double-throw switch and similarly shielded wires run to the volume control and 6Q7 grid. Any high-gain triode can be used, such as the 6K5, 6Q7, or 6F5-GT.

In many cases a wireless record player is available and the owner wants to have a mike added to it in place of the phono arm or as an extra gadget. If a crystal mike is used, additional gain will be required. Fig. 4 shows a typical record player of good quality. The circuit may be modified slightly as in Fig. 5 for the addition of a mike. If a single-button

carbon mike is used the circuit of Fig. 1 can be used to work directly into the record player without any need for an extra pre-amplifier stage. The change is illustrated in Fig. 6.

Where the wireless record player is an a.c.-d.c. type it is not convenient to add an extra tube for amplification of the signal derived from the microphone circuit, and the single-button carbon mike is used because of the ease of hooking it up. In Fig. 7 a typical record player of the a.c.-d.c. type is shown. Fig. 8 shows the circuit used, when a carbon mike is added. Fig. 9 shows a double-button carbon mike installation.



Fig. 6—The addition of a carbon mike to a Zenith S-6622 record player is shown above. A crystal mike will give better quality.

The double-button type has better fidelity but considerably lower output. The coupling transformer should have a turns ratio such that the voltage step-up is about 30 to 1 for 100-ohm buttons. In general it is better to use the carbon mike and a coupling transformer with a.c.-d.c. circuits to keep the 115-volt line voltage away from the untrained user of the mike (the average set owner) which will avoid danger of electrical shock.
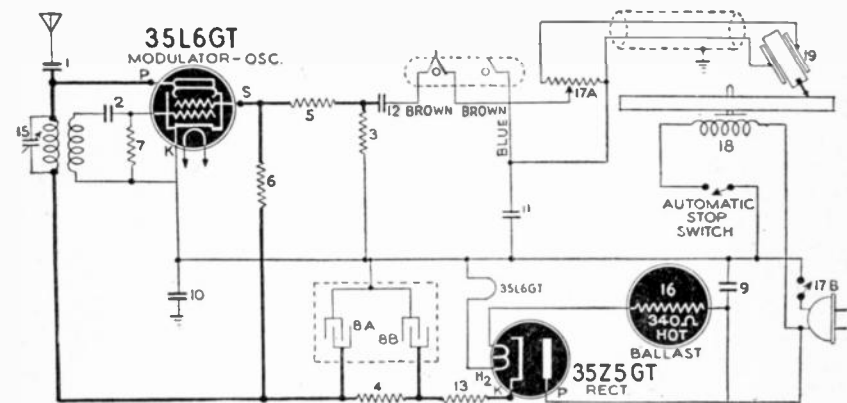


Fig. 7—The Stewart-Warner wireless record player is outlined in the above diagram.

Because the crystal type is high impedance and connects directly to the circuit it is unsuitable for a.c.-d.c. jobs. Such a mike could be coupled through a 1:1 ratio transformer, however.

As the carbon mike has high output, it eliminates the need for additional pre-amplification. If a pre-amplifier tube is added it will be

necessary to supply the filament through a separate voltage dropping resistance, since it is not possible to hook directly into the set without disturbing the receiver circuit. The filament circuit of an a.c.-d.c. set
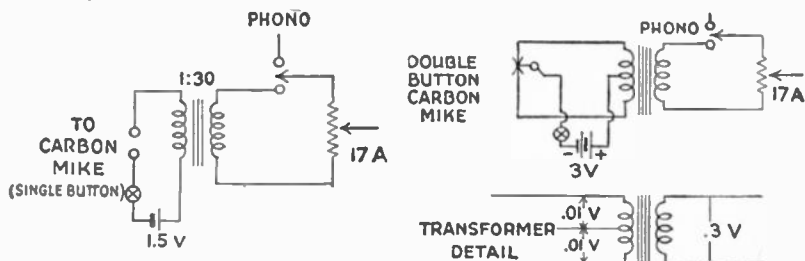


Fig. 8—left—How a carbon mike may be added to the Stewart-Warner record player. Right—Fig. 9—Connection for a double-button mike.

would be upset by the addition of an extra filament. This trouble may be avoided using the carbon mike. In adding a mike it is realized that the gadget is purely for entertainment purposes and not capable of high fidelity, such as would be required in making recordings or in public address jobs of high grade.
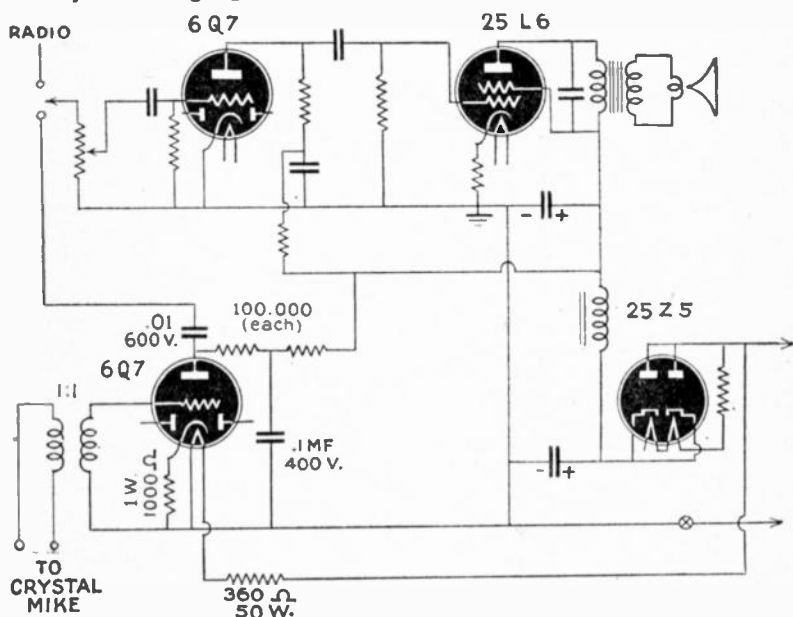


Fig. 10—Circuit for adding crystal mike to an a.c.-d.c. set.

How a crystal mike may be added to an a.c.-d.c. set is shown in Fig. 10. The 1:1 ratio transformer isolates the mike from the power line.

**33**

# Matching Loudspeakers

T HE TECHNIQUE of speaker matching is well understood by every radioman—up to a certain point. When two speakers of unequal impedance are to be attached to the same amplifier, this understanding is not so general. And if the speakers are of unequal wattage rating as well as voice-coil impedances—each one to receive its correct proportion of the total power—few radio servicemen indeed can toss off an answer to the problem. More than one compromise installation is the result of their inability to do so.

As a simple example: We need to connect a pair of 5-watt speakers with 16-ohm voice coils and one 20-watt speaker with an 8-ohm voice coil to a 30-watt amplifier. How are we going to hook up the speakers so that the power will be properly distributed? Remember that the speakers have to be connected across the secondary of a universal output transformer, and the speaker load must be properly matched to that required by the tubes.

The problem is not too difficult. One of the reasons so many radiomen are stumped by it is that they have learned too much about matching. They cannot imagine attaching an 8-ohm voice coil to anything but an 8-ohm tap. If it becomes necessary to hook a monitoring speaker across the 500-ohm line, they do it with dark forebodings as to what might happen at the other end. It is necessary to forget all that! If we are going to connect several speakers to the same winding, obviously we cannot proceed as if we had only one, and that means we can't "match" tap and speaker ohm for ohm.

## Purpose of Output Transformer

The chief reason for an output transformer is to match the impedance of the output tubes' plate circuit (usually between 2,000 and 10,000 ohms) to that of the speakers' voice coils (commonly between 2 and 16 ohms.) If the voice-coil impedance is 6 ohms and that required by the output plate circuit—for best results—is 6,000 ohms, the impedance ratio is 1,000 to 1. The voltage ratio is the square root of the impedance ratio, or in mathematical terms:

$$\sqrt{Z_1/Z_2} = E_1/E_2.$$

Our specimen transformer then has a voltage step-down of $\sqrt{1000}$, or

about 31.5. It is likely to have about 2,000 to 3,000 turns of wire on its primary and from 60 to 90 turns on its secondary.

The impedances of a universal output transformer are usually marked, but the turns or voltage ratios are not. Our only interest in these voltage ratios is that they help us to understand some of the electrical actions of the transformer, and thereby to figure out which of the secondary terminals we have to attach our speakers to.

To get the underlying principles straight let us try the simplest possible hookup (Fig. 1). The whole 30 watts is being fed into one 8-ohm speaker. Voltage across the 8-ohm voice-coil winding is 15.5 roughly ($W = \dfrac{E^2}{R}$, or $30 = \dfrac{E^2}{8}$). Amperage is 1.94. The impedance ratio, $Z_p/Z_s$, is 6,000/8 and the voltage ratio is the square root of that, about 27.4.
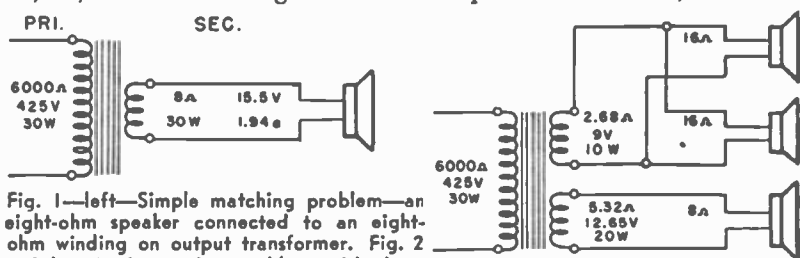


Fig. 1—left—Simple matching problem—an eight-ohm speaker connected to an eight-ohm winding on output transformer. Fig. 2 —right—An interesting problem, with three loudspeakers to be matched to two secondaries having different impedances.

The primary (a.c.) voltage is 15.5 x 27.4 = 425 approximately. This can be checked by calculating direct from the primary watts ($30 = \dfrac{E^2}{6,000}$).

Now we can try a hookup like that of Fig. 2, which has two secondaries. We connect the two 16-ohm speakers in parallel, making an 8-ohm load to place across one secondary. The 20-watt speaker, also an 8-ohm unit, goes across the other. What should be the impedance of the primaries?

First, 10 watts must be fed to the two-speaker unit. Using our formula, $10 = \dfrac{E^2}{8}$ or $80 = E^2$, the voltage across the voice coil works out to slightly less than 9. To get the voltage ratio, we divide 425 by 9, which is 47.2. Since $Z_p/Z_s = (E_p/E_s)^2$, we square 47.2, giving us 2,228. The impedance is 6,000/2,228, or roughly 2.68, which should be the rated impedance of a winding to supply the two small speakers.

The single speaker is to draw 20 watts. The same calculation makes the voltage about 12.65. ($20 = \dfrac{E^2}{8}$ or $160 = E^2$).

Dividing that into 425, the turns (voltage) ratio is near 33.6. Squaring this, we get 1,128. The impedance of the secondary coil is 6,000/1,128, or roughly 5.32 ohms.

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(All the foregoing figures are obtained from the slide-rule, and are approximations, but are more than accurate enough for this work.)

Since the secondary impedances are effectively in parallel, two separate windings are unnecessary. It is easier to hook each speaker to the proper impedance tap on a universal transformer, as in Fig. 3. This is what is done in actual practice.

Now, are these impedances correct? On the surface, it would not seem so. One 8-ohm load is attached to a tap whose impedance is slightly less than 3 ohms—the other to one of a little over 5 ohms impedance. Let us see if anything like 6,000 ohms is reflected back into the primary. If so, the speakers are matched to the output tubes.

## What is reflected Impedance?

The impedance reflected into any primary winding is due to the resistance of the secondary load and the transformer ratio. An 8-ohm load across an 8-ohm tap reflects the rated impedance (6,000 ohms in the case of our transformer) back into the primary. Placing the same load across a 4-ohm tap would reflect 12,000 ohms or 4/2 the normal impedance.

One of our windings will therefore reflect $8/2.68 \times 6,000 = 17,900$ ohms and the other $8/5.32 \times 6,000 = 9,060$ ohms. Adding the two parallel impedances, we get $1/17,900 + 1/9,060 = 1/6,000$. The impedance reflected into the primary is 6,000 ohms. The speakers are effectively matched to the amplifier.

In many cases the required impedance taps are not found on the output transformer. Connect to the nearest tap, checking the effect on power distribution and so averaging the mismatch of individual taps that the total reflected impedance will be as nearly correct as possible. If it is necessary to err in either direction, connection should be made to taps of slightly lower than correct impedance. Thus the reflected impedance will be a little high. This will merely reduce the power slightly, whereas too low an impedance facing the output tubes would harm fidelity.

## Matching Transformer Primaries

Speakers may also be matched by their output transformer primaries. Thus two 12,000-ohm primaries could be connected in parallel across the 6,000-ohm impedance of the output tube(s). This method is useful where speakers are some distance from the amplifier.

A quicker method of calculating the correct taps can be worked out from the example just given. Each speaker in Fig. 3 was so mismatched to its transformer winding that it got its own share of the power, yet all speakers when paralleled reflected the correct impedance back into the primary.

This method of distributing power is simple enough. Many servicemen do it unconsciously. With two 8-ohm speakers to attach to an output transformer, few would have to be told they could both be connected in parallel across a 4-volt tap. Each speaker is faced with an impedance

half its own, and receives half the power. What could be more simple, if we wish to divide the power in parts of one-third and two-thirds, to tap one speaker across a tap 1/3, and the other across 2/3 its impedance? All we have to do is *multiply the voice-coil impedance by the fraction of the output* we want it to take.

But will matching to the output tubes be correct? Back to the 8-ohm speaker and 4-ohm tap again! A 4-ohm speaker would reflect the correct impedance back into the primary, and maximum power would be drawn from the amplifier. The 8-ohm speaker reflects 8/4 or twice the correct impedance back into the primary. If two of them are paralleled, each reflects its 12,000 ohms. The resultant of these two 12,000-ohm impedances in parallel is 6,000, the correct load. If our two 8-ohm speakers are connected to divide the load into 2/3 and 1/3, the reflected impedances will be 3/2 x 6,000 = 9,000 and 3/1 x 6,000 = 18,000. Adding these impedances in parallel, the resultant impedance is 6,000, which is what we want.

The method can be extended to several speakers, as in the example of
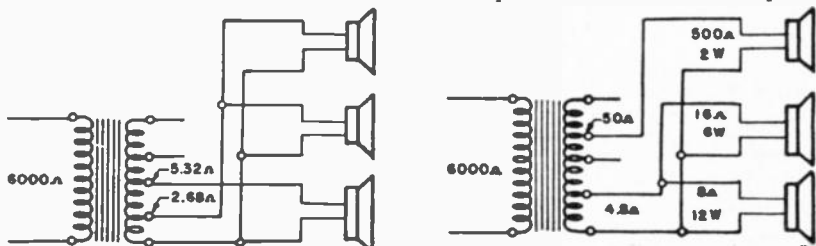


Fig. 3—left—Each speaker is matched so that it receives its share of the power; when paralleled as shown they reflect the correct impedance back into the primary. Fig. 4— right—Matching widely different speakers to one output transformer. The combined speaker impedances give a resultant impedance of 6,000 ohms reflected into the primary circuit, which is correct.

Fig. 4. Here we wish to supply 2 watts to a 500-ohm line, 6 watts to a 16-ohm and 12 watts to an 8-ohm speaker. This works out to 0.1, 0.3, and 0.6 of the total output (20 watts). Again assuming a primary of 6,000 ohms, we can get correct matching and power distribution by calculating output taps as follows:

For the 2-watt 500-ohm line, 500 x 0.1 = 50 ohms; for the 6-watt, 16-ohm speaker, 16 x 0.3 = 4.8 ohms; and for the 12-watt, 8-ohm tap, 8 x 0.6 = 4.8 ohms also. Reflected impedances are 6,000 x 10 = 60,000; 6,000 x 10/3 = 20,000 and 6,000 x 10/6 = 10,000. These paralleled impedances add up to 6,000 ohms.

By the above method it is possible to hook up the most complicated speaker combination. All that is necessary is to know the impedance of each speaker and the portion of the total amplifier power we want to put into each one. Another essential is an output transformer with a variety of taps. In conclusion, it might be well to mention that an output transformer has a large number of impedances not marked. For example, the impedance between the 2-ohm and 16-ohm tap is 6.6 ohms.

# REMOTE JUKE BOXES

**W**HILE THE use of telephone lines for distribution of music is not new, the following is written to clear up some of the mystery of that type of "Juke Box" where you are asked for the name of your selection and do not have to push a button.

In the first place, the building where the main equipment is located is known as the Central Office. The place where the "Juke Box" is located is called the Remote Station. As the reader proceeds the above two terms will be used to designate the two locations.

The speech equipment at the Central Office consists of: The metal rack about five feet high, divided into two sections. The entire rack is known as the Board. Each Board consists of ten complete units. Each unit controls a remote station location. Fig. 1 shows a front view of the Board. On top of the Board is a wire rack, into which approximately 1000 phonograph records can be inserted. At the bottom of the Board can be seen another wire rack which holds another 1000 phonograph records. This allows the operator to have any record at her finger tips the instant it is asked for.

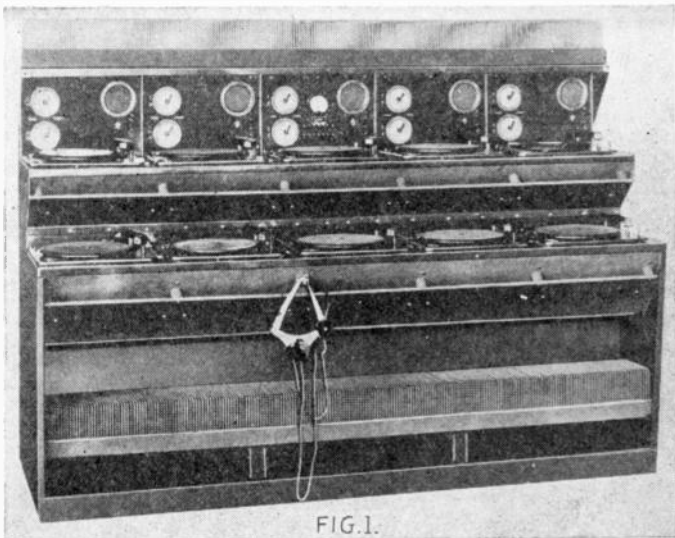Beginning at the left in Fig. 1 are seen two dials set one above the



Fig. 1—Front view of a typical Central Station panel.

other. Each dial is calibrated left to right from zero to twenty. These are the dials which light up and show the number of coins inserted at the Remote Station. A stepping relay is used to operate a pointer on this dial. Between the two dials and on either side can be seen the coin-counting relays that record the number of coins inserted continuously.
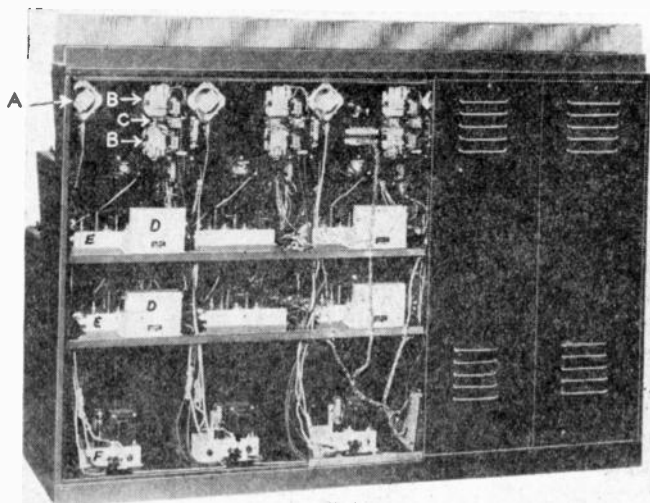


FIG. 2.

Fig. 2—Rear view of panel—text explains letters.

To the left of each dial can be seen pushbuttons which control a buzzer to notify the operator that a coin has been inserted in case the stepping relay fails to operate. To the right is the permanent magnet monitor loudspeaker which is used to check the quality of the program and also can be used to hear what the person at the Remote Location has to say in case the operator's headset fails. Directly underneath is a toggle switch which turns the power on and off for two complete units. To the right and just below the monitor speaker are two pushbuttons. The top pushbutton connects the monitor speaker to the output of the monitor amplifier for the top turntable and the bottom pushbutton connects it to the monitor amplifier for the bottom turntable. When not in use a dummy five-ohm load is cut in across each amplifier output.

There are two phonograph turntables with each section, each one being associated with a Remote Station. These turntables are powered by sturdy, variable-speed motors and are equipped with an electric stop. The operator merely puts the record on the turntable platter and moves the pickup arm to the right until a click is heard. By the time she has the pickup on the record the turntable is up to speed. These turntables require very little servicing. They are checked regularly once a week with a neon lamp and a stroboscopic disc for speed. They have no brushes and can be made to operate on 220 volts a.c. by a change in the connections of the motor windings.

## Talk-back System Provided

At the right and in the middle of each turntable is a triple-pole, double-throw switch. This switch is used to operate a talk-back system to the Remote Station after it has signalled the operator by means of the stepping relay or buzzer. When pulled forward, it connects a two-stage microphone pre-amplifier into the circuit. The amplifier consists of a 6SJ7 pentode, capacity-coupled to a 6J5 triode, with a volume control between the two tubes. The amplifier is transformer-coupled in and out. Equalization is inserted in this amplifier to make it peak in the voice range, and frequency response is from about 80 to 6000 cycles.

In the middle of the Board can be seen a volume level meter. This meter is used across the output of each program amplifier to indicate the correct operating level. Below the decibel meter is a row of push-buttons. Each pushbutton is associated with a program amplifier output.

At no time during the playing of a record does the volume rise to more than minus 2 db on the peaks. The program amplifier output is set by using a record which has an abundance of high and low frequency passages. I suppose it will be asked, "Why not set the maximum program level by means of a standard audio-frequency record?" This has been found by actual practice to be useless for a good many reasons, the main one being that all crystal pickup cartridges do not have the same voltage output for a given frequency, and also as they become weak through constant use this is a good check on their frequency response.

Hanging down from the front of the Board is seen the operator's breast-set. This consists of a pair of low impedance headphones, connected in parallel, and a dynamic microphone. Crystal microphones are little used in this type of work as they are not rugged enough for the abuse they get. Also the circuits are of low impedance to cut down noise and hum, and use of a high-impedance microphone would necessitate the use of an input transformer with its hum problems. The microphone can be raised or lowered to compensate somewhat for the different speaking voices of the operators.

Looking at the back of the Board as shown in the photograph, Fig. 2, is seen the following apparatus: At "A" in the upper left-hand corner is the back of the permanent magnet speaker. In the upper right-hand corner and labelled "B" are the two dial stepping relays.

## Phonograph Pick-up Amplifier

In the middle is seen the back of the turntable motor. On the first shelf and to the left labelled "E" is the program monitor amplifier chassis. The phonograph pickup amplifier consists of a dual 100,000-ohm potentiometer working into the grids of a 6N7 tube. The crystal pickup is not grounded on one side as is the usual practice. The 6N7 tube is transformer-coupled out to a 500-ohm line. The monitor amplifier consists of an input tranformer with a potentiometer across its secondary

into a 6J5 tube. This potentiometer not only controls the voltage on the grid of the next tube, which is a 6F6 tied triode, but also controls the volume level of the monitor speaker and the operator's headset. A second potentiometer in the grid of the 6F6 tube controls the volume level of the monitor speaker. On the above chassis is the copper-oxide rectifier
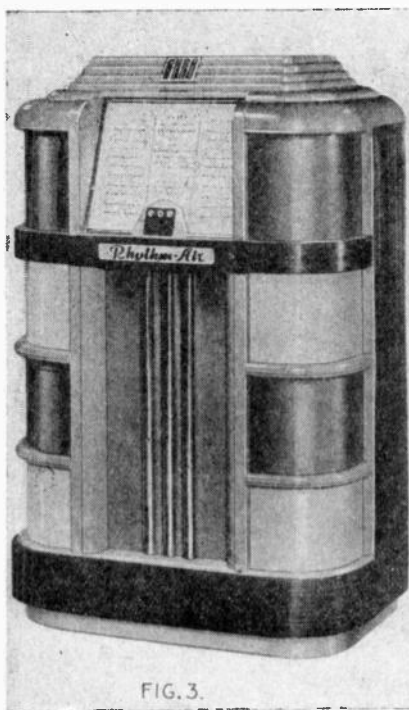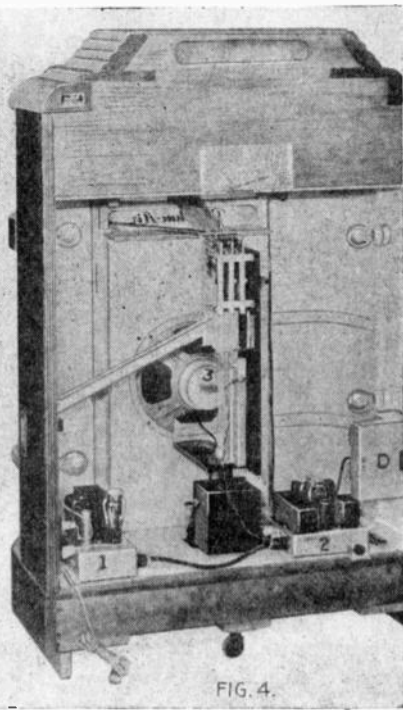


Fig. 3—Remote station resembles standard juke box.   Fig. 4—Rear view of cabinet shows special parts.

and relay that controls the stepping relay on the panel and the buzzer. On the same shelf as the above chassis and labelled "D" is the conversion unit chassis.

This unit consists of a resistance-capacity network and two 1:1 hybrid coils. A brief description of this unit, whch is very important to the frequency response and operation of the equipment, follows:

The amplifiers will operate over a maximum length of ten miles of telephone line. However, these conversion units plus the amplifiers are designed to work over a seven-mile class "C" telephone line or any unbalanced circuit of that length. Whether the line between the Central Station and the Remote Station be a full seven miles or any fraction thereof, the network in the conversion unit will make up the line difference so that to our equipment it still is a seven-mile line.

The capacitance of this seven-mile line was figured at 0.6 $\mu$f and its resistance as 1344 ohms. Thus each conversion unit (one being used

at each end of the line) is divided to have a capacity of 0.3 µf and a resistance of 672 ohms. Each of these units is divided into seven sub-units. There are three 1-mile line units, two ½-mile line units, one 1/3-mile line unit, and one ¼-mile line unit. Each of these units is arranged like an "H" pad, as shown in Fig. 5.

Each 1-mile line unit consists of four 56-ohm resistors, and a 0.1-µf condenser. Each ½-mile line unit consists of four 27-ohm resistors, and a 0.05-µf condenser. The 1/3-mile line unit consists of four 15-ohm resistors and a 0.02-µf condenser. The ¼-mile line unit is made up of four 12-ohm resistors and a 0.02-µf condenser. While 27-ohm and 15-ohm resistors do not figure exactly right, as stock resistors were used, they fall within the ten percent tolerance range and are all right for the purpose. There are two controls on the conversion unit, one for the high and one for the low frequencies. These controls peak the line at 100 cycles and 3,000 cycles respectively. To equalize the line requires an audio frequency oscillator and a calibrated volume level indicator with the necessary terminating equipment.

The second shelf is a duplicate of the first. On the bottom and labeled "F" may be seen the power supply which feeds the two program amplifier chassis above, including the lights for signalling, stepping relays, etc. The rectifier is a 5U4G used in a full wave circuit.

The Remote Station: In Figs. 3 and 4 can be seen a front and back view of the "Juke Box" used at the remote location. Behind the metal grill at top center is the microphone which the customer uses to tell the operator the number of the phonograph record he or she wants to hear. Either a crystal mike or a small two-inch permanent magnet speaker with an input transformer is used as a microphone. Below this and just above the words "Rhythm-Air" are coin slots for the nickels, dimes, and quarters. Behind the three metal bars in front of the Box is the 12-inch, permanent magnet speaker.

A back view of the Box and its interior is shown in Fig. 4. To the left and on the bottom is the power supply with its 5U4G rectifier tube. This is labelled 1. Rear view of the speaker is 3 in the photograph. At top rear middle is the coin scavenger mechanism. This rejects any coin
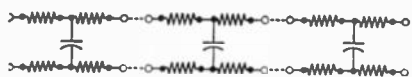


Fig. 5—Three sections of the artificial line. The converted juke box will work over a seven-mile line; shorter lines may be balanced by artificial line units like those shown.

that is of a ferrous nature and also any slugs that might be inserted. Below this and shown with a twin-pair conductor is the coin counting mechanism. This causes one pulse for a nickel, two pulses for a dime, and five pulses for a quarter to be sent over the telephone line to the Central Station and operate the stepping relay. The voltage used is anywhere from 30 volts a.c., 60 cycle, to 110 volts a.c., 60 cycle. It depends on the length of the line and other factors, determined by trial.

On the right-hand side and fastened to the wall, labelled D, is the

conversion unit. Just below this unit and marked 2 is the chassis containing the remote amplifier. The talk-back amplifier is on this chassis and depending on the type of microphone used, has either a resistance coupled input or transformer coupling to two 6N7 tubes in resistance-capacity coupling, push-pull. The output is transformer-coupled to a 500-ohm line. A volume control is used in the grid of the first 6N7 which is a phase inverter for the second 6N7.
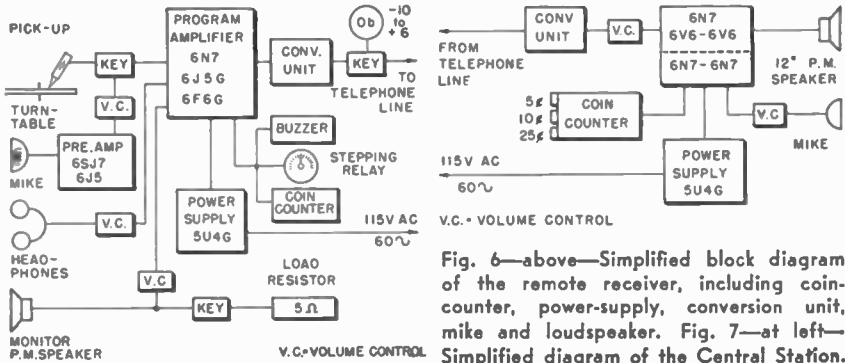


Fig. 6—above—Simplified block diagram of the remote receiver, including coin-counter, power-supply, conversion unit, mike and loudspeaker. Fig. 7—at left— Simplified diagram of the Central Station.

A brief description of just what takes place when a coin is inserted in the Remote Location juke box is as follows: A coin is dropped through the counting mechanism and—depending on its value—causes a contact to close. This operates the double-pole, single-throw relay on the Remote Amplifier chassis. When this relay closes it sends the voltage which has been chosen (30 to 110 volts a.c. 60 cycle) from an isolation transformer on the power supply, over the telephone line to the Central Station. When reaching the Central Station this voltage impulse is rectified by a full-wave copper-oxide rectifier and as a d.c. voltage operates the single-pole, single-throw relay on the program amplifier chassis. When this relay closes it operates either the stepping relay on the front panel or the buzzer.

When the operator sees the stepping relay operate or hears the buzzer, she throws the triple-pole, double-throw key between and at the right-hand side of the turntables. This connects her headset through the monitor amplifier to the incoming telephone line, and also connects the output of the microphone pre-amplifier to the same line. She then can not only hear what record the customer would like to hear played but also talk back to him. If the record is not available, she can ask him to request another. When the operator throws the key to talk to the Remote Station, it automatically drops the level of the phonograph record that might be playing at that time so the customer can hear her voice over the music.

A block diagram of the Central Station equipment and also the Remote Station equipment is shown, in Figs. 6 and 7. When the telephone lines are equalized and balanced their frequency response is within $\frac{1}{2}$ db, from 100 to 6,000 cycles.

# Servicing Record Changers

W HAT IS the most efficient method of attack on record changers in need of repair? The answer: simply the same logical approach used on electronic circuits. Putting it into general terms:

*Visualize the mechanical actions step by step in a manner similar to the way electrical reactions are considered stage by stage in a radio set.*

Many faults are so logical as to be perfectly obvious to anyone possessing sufficient curiosity to try a few simple adjustments.

The greatest variety of troubles occurs in newly installed machines.

## Improper Unpacking

The unpacking of any radio-phonograph consists of more than merely removing it from the shipping box. During shipment the radio chassis and phonograph unit are both secured by "packing bolts." These are usually painted red for easy identification. and when removed leave the radio and phono unit floating freely on rubber or spring mountings.

*Symptoms:*

Jamming of the mechanism in the midst of a repeat cycle or failure to operate at all may be due to binding of the mechanism against the cabinet, caused by packing bolts being tightened down.

If a rim-drive motor spins but the turntable does not revolve, the motor may have a separate packing bolt preventing it from floating on its pivoted mounting. This particular bolt is sometimes never removed but merely loosened enough to free the motor. In this case, it may not have been loosened enough to allow for wearing-in of the moving parts.

## Improper Packing

The above method of packing would be used by the factory or a service man in preparing a set for shipment, but the customer often ships the set himself. He may fail to secure loose parts safely (probably not even using a box if he intends to carry it in the family car). That physical damage can occur is obvious, but loss of rubber or spring mountings can make it impossible for the phono unit to rest on an even keel. This may interfere a great deal with the balance of the many small tension springs used to position the levers.

*Symptoms:*

Dropping records two at a time, jamming one side of a record and dropping the other; starting too far in or outside the edge of the record;

repeating before the selection is finished; all these faults are typical of a unit that is not level.

Don't forget to check the floor! Older buildings sometimes settle badly and can contribute to faulty operation unless the legs of the set are leveled by placing a block under one corner.

## Forcing of Mechanism

This is probably the most prolific source of serious trouble. Many people have become familiar with old-style hand-operated phonographs, and seemingly cannot overcome the habit of reaching for the pickup arm when it is time for another record to start playing. They inadvertently forget that something else is handling the pickup, the sensitive gears and cams of the repeat mechanism located below the motor board.

Levers can be bent and springs broken this way, but usually adjustment screws are forced out of position.

*Symptoms*:

Failure of the needle to come down at the proper place on the edge of the record and failure of the repeat cam to disengage, resulting in the repeat action occurring over and over without stopping to play a record, is one of the typical faults resulting from such maloperation. The list of troubles attributable to this cause may exceed the imagina-
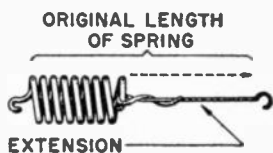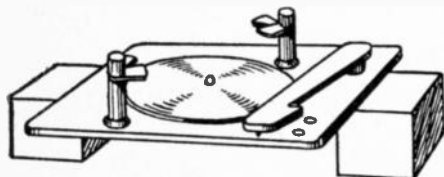


Fig. 1—left—This sketch shows a simple system for blocking up the record changer on the bench for repairs. Fig. 2—right—If a spring breaks, it often can be repaired temporarily by adding an extension wire as shown.

tion. Care used in placing the blame for the cause of breakdown may help to prevent its recurrence.

If the set owner has developed a habit of operating his set in an improper manner, he may not be aware of the bad effects since they appear gradually. As long as his way of doing things gets results he will persist in doing them that way. A few tactful remarks can result in the discovery of such a situation and, once the true cause of the trouble is known, a bit of advice to him will eliminate a possible call-back.

## Use of Old Records

An example of unintentionally improper operation is the use of phonograph records of ancient vintage. Many still in use date back to "tin-horn" days. Unfortunately, the man who made them had none of the qualifications of Nostradamus. Not being able to predict the future, he failed to design them to work on an automatic record changer.

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This is not difficult to explain if a new record is available for comparison. The *eccentric groove* around the center can be pointed out as the means of actuating the repeat cam. This will not be found on the old record. Also, old records can be seen to vary from the standard thickness of modern records.

(Never suggest the obvious cure of discarding the old records. Remember, they are "heirlooms," to be broken only by grandchildren.)

*Symptoms:*

Failure to repeat and jamming of the record selector mechanism.

## Bent Records

The worst treatment that can be accorded phonograph records is to forget to remove them from the selector blade supports and leave them there for several days. In this position, the stack of discs is not resting on a flat surface like the turntable, but is supported either from two points at the sides, or at one side and the center. In the first case the discs will sag in the center and in the second they will sag at the unsupported edge. When this sag becomes fixed the recordings may be so badly warped that they cannot rest flat on one another and the selector blades will not be able to slide between them to separate one record at a time from the stack.

*Symptoms:*

The "wow" effect of warped records on music is well known and many complaints that the motor is "dragging" or alternately speeding up and slowing down can be traced to this reason, especially if the unit refuses to misbehave when the repair man is present.

### Record Changer Trouble Chart

| ADJUSTMENT ON: | SYMPTOMS OR TROUBLE |
|---|---|
| 1. Friction Clutch on tripping pawl | Too loose—Repeat mechanism fails to trip. Too tight—Repeats grooves over and over. (using a good record) |
| 2. Height of Pick-up (Turntable) | Needle drags on top record when 10 or 12 records are stacked on turntable. |
| 3. Needle landing place for 10" records. (Set screws on pickup shaft) | Needle misses edge of record or starts too far in on 10" records. |
| 4. Needle landing place for 12" records. (Eccentric stud) | Same as above but for 12" records. This must be set after adjustment for 10" records. |
| 5. Distance between Selector Blades. (Screw and lock nut) | Record selector blades strike the edges of records instead of separating them and sliding in between. |
| 6. Distance of rotation of Selector Blades. (Set screw on Selector Blade shaft) | Records, when released, fail in a lopsided manner instead of both sides simultaneously. |
| 7. Reject Lever (Requires bending or reshaping) | Reject Lever inoperative because of too forceful handling by inexperienced operator. |
| 8. Leveling of player. (Spring retaining screws at each corner) | Needle may fail to slide into starting groove of record if turntable is not level. |
| 9. Oiling. (Use SAE No. 10 oil on motor and petrolatum on other parts) | Squeaks that can spoil listening pleasure and "dragging" of motor during operation. |

## General Repair Procedure

Before actual repair work can be accomplished a device must be provided by which the player can be supported in a right-side-up and level position, since this is the only position in which it can be tested for proper operation. Two small boxes taken from the stock shelf are quite practical for this purpose (see Fig. 1). Many service men prefer more elaborate supports installed in a permanent phono repair position. Naturally, some operations will require placing the player carefully upside down (preferably on a felt pad).

The most expedient method usually begins with an inspection of the
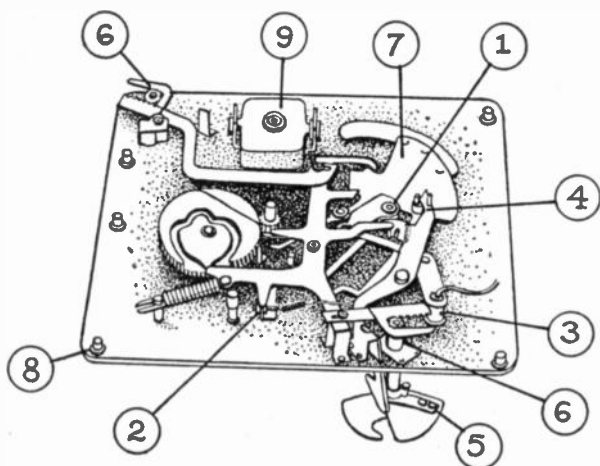


Fig. 3—Simplified drawing of the under-side of a well-known record changer. The reference numbers can be checked against the trouble charts appearing on page 46.

mechanism for broken and defective parts. These are either replaced or repaired.

In this connection an important caution should be observed in regard to replacing broken springs. (See Fig. 2.) If it is necessary to stretch a broken coil spring so that it can be put back in its original location, some arrangement must be used to avoid excessive tension. Remember that springs are used merely to bring the levers back to their original positions and minimum tension should be used to avert wear at points of friction.

After mechanical repairs have been effected, resetting of adjustments can be undertaken and will invariably be necessary due to the considerable effect one adjustment has upon another in this type of mechanism.

The underside of a widely used record changer is shown in Fig. 3 with its various components numbered to correspond with their adjustments as listed in the accompanying table.

# Hum Elimination Methods

ⅡUM, THAT bane of every engineer's existence, the spoiler of recorded and transcribed music, can be eliminated. It merely requires careful analysis and a large dose of common sense.

First, hum may be eliminated through proper design. One of the most violent sources of hum is lack of filtering. Proper filter components are arrived at by designing the power supply—for the unit under consideration—to have a certain percentage ripple when loaded with the load the unit will draw. This is not always enough. It is sometimes necessary to utilize a voltage-regulated power supply to reduce the ripple to the desired amount. This may take the form of the well-known RCA circuit which uses a sharp cut-off pentode to receive the variations in the d.c. output voltage and with a resistor in its plate circuit to influence the grids of several power tubes such as 6B4G's or 2A3's, connected in series with the power supply. Their resistance is increased or decreased, and, thereby, the voltage held constant and the ripple wiped out. For smaller loads, a gaseous regulator of the VR150 type may be employed.

## Several Causes of Ripple

In the former case, the regulator may function well with a resistive load, but when coupled to the circuit it is intended to power, ripple develops. This may be caused by several things. First and foremost is the shielding of the grid circuit of the pentode which controls the regulator tubes. If 60 cycles is introduced at this point, the tube will amplify the 60 cycles and swing the d.c. voltage at this frequency. Secondly, *this type of power supply must not be overloaded*. It must be designed to handle the maximum current to be delivered. The regulation and ripple increase very rapidly after its maximum output has been reached.

If a VR tube is used as a regulator, its current range cannot be exceeded. This tube must be supplied with a minimum of 5 milliamperes, else it will not remain in a conducting condition. It must be remembered that the line voltage may change, so a low line voltage condition should be used to figure the minimum current value and the highest line voltage for the maximum current value. This current should not exceed the current rating for the tube.

These tubes will oscillate under certain conditions. This condition may be cured in some cases by inserting a series resistance of about 20 ohms between the VR tube and ground. It is interesting to note that VR tubes have an effective resistance in the order of 40 ohms.

The power supply should be laid out with the fields generated by the power transformer and chokes properly oriented. The chokes and power transformers should be magnetically shielded and placed so their cores are at right angles to each other, as there is no point in building a good low-ripple power supply and then inducing a volt or so of 60 cycles from the power transformer into the chokes. This hum source will cause even more trouble if audio transformers are involved, as there is usually a considerable amount of amplification following these items and the induced hum is amplified accordingly.

Audio transformers should of course be kept away from power transformers and chokes and should be oriented properly as well. If these components are mounted on a common iron or steel chassis, their cores should be kept away from the chassis by mounting them on aluminum or brass bushings, as the magnetic chassis becomes a common lamination
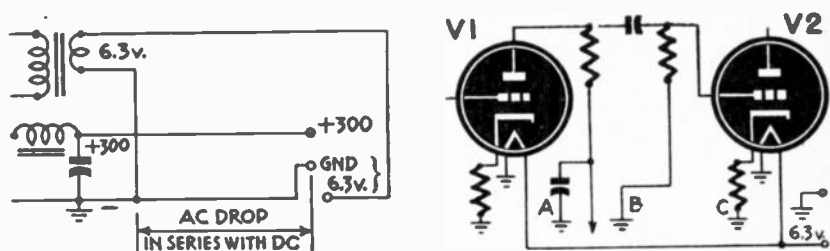


Fig. I—left—How an a.c. hum component is often unwittingly piped into a d.c. circuit. The hum current may be small but amplification can increase it to alarming proportions. Fig. 2—right—Receiving sets often manifest noticeable hum due to improper ground connections of the various units to the chassis. Troublesome voltage drops may occur as the diagram shows.

in the core of each transformer and acts as a convenient path for the transference of magnetic fields.

In general, the smaller an audio transformer is, the fewer external magnetic lines will cut it and, consequently, the less it will be bothered by external fields.

Remember that the field around a conductor is proportional to the current that conductor is carrying, so be very careful with leads such as supply lines for the power and filament transformers, and filament wiring which supplies a large number of tubes. Grounding one side of the heater winding on the filament transformers *in one place* is helpful, but keep the filament wiring to itself. This, of course, applies also to high voltage wiring to the rectifier tubes. Don't forget to shield all mercury-vapor rectifiers.

Where the power supply and the unit to be powered are separated, do *not* run the negative B voltage and the grounded side of the filament

through a common wire, because (as a look at Fig. 1 will show you) you will effectively connect an a.c. voltage in series with the d.c. supply which you consider hum-free. Although this voltage is small, being due to the filament current, it may be amplified many times by several tubes and assume much greater proportions in the output of your device.

It is somewhat risky to ground the filaments of high-gain amplifiers in several places or depend on the chassis to carry one side of the filament current. This is quite all right from the current standpoint, but if a previous plate is by-passed at Point A in Fig. 2 and a subsequent grid is returned to point B, then a hum voltage may be placed in series with the grid of the second tube. Even though this voltage is in the order of .01 volt or smaller, if that tube is a pentode with an amplification of several hundred followed by several more amplifier stages, as in a microphone amplifier, this hum voltage can become very appreciable.

## Cathode May Be Source of Hum

In high-gain stages, incidentally, hum may be caused by emission from the heater to the cathode, or emission from the cathode to the heater. This condition may be reduced or eliminated, in the first instance by applying a positive voltage on the heater, sufficient to make it as positive as the cathode. In the second instance, the application of a negative voltage on the heater sufficient to make the heater-cathode voltage difference small is the remedy.

Although a tube is supposedly cut off by the application of bias to the cathode, the cathode does not always cover the heater completely, and emission can occur from the heater to the plate. If this emission takes place from one exposed heater wire, it will not be from an equipotential surface and will vary as the heater supply frequency.

It is best, in audio amplifier design, not to rely on multiple grounds to the chassis. It is better to carry the ground circuit from point to point with a heavy copper wire, and this ground, insulated from the chassis, should be grounded in one spot only. This spot will have to be found by trial and error. This is particularly effective in high-gain amplifiers. A spot will be found at which the hum will drop appreciably and even reduce to the vanishing point.

## Addition of R C Filter Often Effective

It sometimes becomes impractical from an economic standpoint to use brute force methods, such as complicated filters and regulators, to effectively filter the entire power supply, when the addition of an RC network, in one or more circuits, will accomplish the desired result. RC filter networks are usually figured on the basis of $T = RC$ (see Fig. 3) where T is the time in seconds, R is the resistance in ohms, and C is the capacity in farads.

This formula states the time, T, it takes the voltage across the con-

denser, C, to charge to 2/3 of its full value or to discharge to 1/3 of its original value. In any event, when unequal wave forms are considered, it is more convenient to consider time constants rather than impedances and reactances.

Time can be saved and time-constant problems can be worked in the head, if you remember that microseconds equals ohms times microfarads or that microseconds equals megohms times micromicrofarads. One of these formulas puts the parameter you want in the terms that you want it and eliminates converting farads, for instance, to microfarads or micro-microfarads, or ohms to megohms. This will also eliminate the possibility of error in adding and subtracting negative and positive exponents.

Decoupling is necessary where several stages are to be cascaded, as oscillation can result. It is conventional, therefore, to isolate several or all the stages, not only in the interest of hum reduction, but to eliminate the unwanted regeneration introduced by common coupling in the power supply.

Hum which cannot be eliminated in any other manner can be reduced by the application of degeneration which will reduce the hum, within the loop, proportional to the amount of feedback. In severe cases, this feedback loop could be resonant at the hum frequency only, so that a maximum of degeneration is introduced at the hum frequency.

In cases where low-frequency response is not important, it is feasible
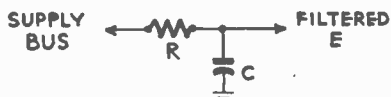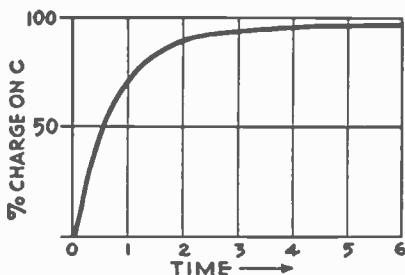


Fig. 3—The use of RC filter networks may eliminate hum in many cases. The charging time is set by the values of R and C. Fig. 4—The condenser in a filter network charges or discharges according to an exponential function, so that it takes an appreciable time for it to receive its full charge.

to reduce the coupling time constants by reducing the coupling condensers and grid resistors, reducing the gain at the hum frequencies. When all other means fail or when economy dictates, hum may be balanced out by applying to the circuit in question a hum voltage of proper amplitude and 180° out of phase. This voltage may be applied to any of the tube elements in a number of ways.

Remember that the field of a dynamic speaker, when used as choke in the power supply, has a very strong hum field surrounding it. An input transformer should not be located close to it. A velocity microphone will pick up a strong hum voltage when located as far as 8 or 9 feet from such a speaker field. Microphone orientation will reduce this pickup to a minimum.

When connecting a self-powered tuner to an amplifier, remember to reverse the line connections if undue hum results. If both units have

by-pass condensers from one side of the line to ground, it is possible to plug the units into the supply mains in such a way as to place some, or all, of the 110-v 60-cycle a.c. power source in series with the common ground between the units.

Consider, when using a cathode follower, that the cathode is swinging with the signal, and, at some parts of the cycle, it may well be a hundred volts above the filament, which may give rise to cathode-filament emission, or capacitive hum coupling.

Electrolytic condensers are a potential hum source. Their capacity reduces with age and their impedance increases, which can also give rise to oscillation. They should not be depended upon for r.f. by-passing and, in such instances, should be paralleled with a good paper or mica condenser.

Don't forget to by-pass to ground all the cathodes in which degeneration is not desired. The value of the condenser should be large so that its impedance at 60 cycles will be much lower than that of the cathode resistor, so that any hum developed between the heater and the cathode will be by-passed to ground. It is important that this condenser be grounded to the same place as the grid and cathode resistors, so that an a.c. circulating current in the chassis is not connected in series with the cathode by-pass condenser. (See Fig. 2.)

Electrolytics used for by-passing sometimes give rise to tunable hums in radio tuners. Tunable hums have also been known to be caused by unwanted 60 cycles introduced into the oscillator of a superheterodyne receiver, due either to faulty by-passing or a faulty tube.

## Hum Sources in Phonograph

Hum can be caused by induction from a phonograph motor to a magnetic pickup. This can be reduced by magnetically shielding the pickup and using an iron turntable. Of course, the leads from the pickup should be run through shielded wire and, in low-level pickups or microphones, the shield braid should not be relied upon for the ground return. Rather, use double wire shielded cable, connecting shield braid to the amplifier chassis and the ground wire to the amplifier ground. In this instance, do NOT connect the braid to the ground at the microphone or pickup but only to its frame. In connection with phonographs, it may be noted that what appears to be a hum only when records are played, may be turntable rumble. This may be reduced by replacing the rubber drive wheel on the rim drive models or by reducing the low frequency response of the amplifier.

Last, but not least, mechanical vibration can be responsible for hum. In such cases mechanical isolation is indicated. Mounting on rubber or removal of the source is the answer.

There are very few cases where the annoying hums cannot be eliminated or reduced to the point where they are no longer objectionable and the full capabilities of the unit under consideration can be realized.

# VOLTAGE REGULATORS

**G**OOD VOLTAGE regulation is important in many radio circuits. A radio transmitter's frequency may shift if the plate supply voltage swings. Changes in voltage may cause instability and oscillation in some receiving circuits. A few types of electronic instruments require supply voltages fixed as closely to the predetermined value as human ingenuity can achieve. This voltage regulation may be attained in three chief ways: by careful attention to apparatus used in construction of power supplies; by use of special voltage regulating and ballast tubes; and by the use of special power supplies with built-in circuits which act to oppose and neutralize any tendencies to change in the supply voltage.

A basic diagram of a source coupled to a load is shown in Fig. 1. If the source resistance ($R_1$) is kept very low, voltage regulation will be
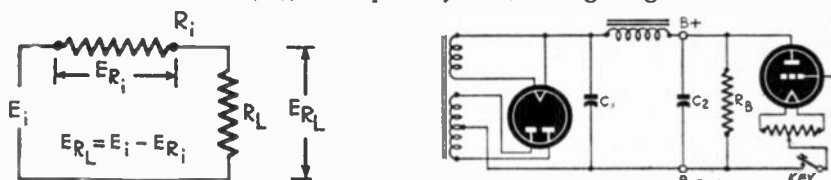


Fig. 1—left—Basic diagram of voltage supply and load. If source resistance is kept low the voltage regulation will be good. Fig. 2—right—Voltage regulation may be tied in with the storage of energy in a condenser and choke.

good. With an incorrectly chosen source resistance, varying the load resistance between two limits above and below the resting value will mean an abnormally large swing in the current through $R_1$ (internal resistance). The varying current flowing through $R_1$ will produce a variable voltage drop across the source resistance. Since the output voltage is equal to the internal voltage minus the drop in the internal resistance, it will fluctuate too much. If $R_1$ is very low in value, clearly the variation in potential across it will be low and the output voltage will be steady for reasonable changes in load resistance. *This points to the desirability of low-resistance chokes and transformer windings in power supplies.*

Voltage regulation may also be tied in with the storage of energy in a reactance. In a radio transmitter with the key open, the load resistance connected to the power supply in effect is made much higher in ohmic

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value and the load current is decreased. Referring to Fig. 2, the circuit current through the choke may be low in value with the key open. The output voltage across B plus and B minus is then high, and the filter condensers are charged up to high values of potential and store energy.

When the key is pressed, a load is put on the supply and the output voltage tends to drop. The choke's resistance causes power dissipation and reduction of output voltage. As the output voltage drops with increased load, it becomes lower than the potential of the output filter condenser. Energy from this condenser is then returned to the circuit. *Electrons flow from that condenser into the tube (the load resistance), tending to maintain the output voltage constant and thus serving as a voltage regulator.* Ordinarily, we don't think of a condenser as a voltage regulator; yet, in reality, it may act as one. By using large value of C and L, not only the a.c. ripple can be reduced in compliance with the radio law and FCC regulations, but also the voltage regulation can be improved, aiding in getting a "clean" note in radio code transmission.

The circuit of Fig. 2 may be simplified into the circuit of Fig. 3. If a steady current is bled through $R_1$ due to the presence of $R_b$ a steady voltage drop will be developed across $R_1$, tending to maintain the output voltage constant, since that output voltage is always equal to the internal source potential minus the voltage drop in the internal resistance $R_1$. Then, if the load resistance swings between reasonable limits above and below some normal resting value, the percentage change in current through and voltage across $R_1$ will be relatively smaller.

In Fig. 2, the bleeder resistance $R_b$ connected across the output filter condenser not only helps to stabilize the output voltage but also
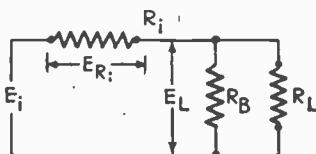


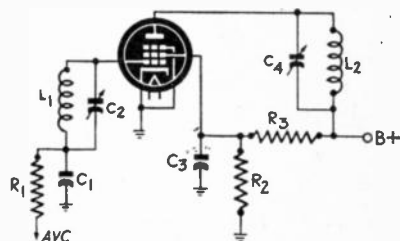Fig. 3—left—Simplified theoretical circuit of the fundamental voltage regulator.

Fig. 4—right—A typical i.f. stage.

allows charges to leak off the condensers when the transmitter is not in operation, an important safety feature in high-voltage equipment. The same circuit may be used with a voltage regulator tube. The tube acts to maintain a constant voltage across the load. The regulator resistance is $R_b$ and the load is $R_L$.

In some radio receivers, a shunt resistance is used for voltage stabilization of the screen grid potential of a tube. A typical i.f. stage is shown in Fig. 4. Again the basic circuit of Fig. 3 applies. We substitute $R_2$ for $R_b$ and $R_3$ for $R_1$. When the control grid of the tube (nearest the cathode) is made highly negative due to automatic volume control bias potential, the screen grid current decreases and there is a reduction in the voltage

drop across $R_3$. A reduced drop across $R_3$ would mean an increased screen grid potential at resonance. Rise in the screen potential is limited by the action of $R_2$. As the voltage across $R_2$ rises, the current through it tends to rise. This increased current flowing in $R_2$ and $R_3$ will tend *to produce an increased voltage drop in $R_s$, which works against any rise in voltage on the screen,* providing a stabilizing action.

The series resistance of a supply may be made *either* very high or very low to secure a stabilized output voltage. In both cases, what is sought is a stabilization of the voltage across the element in series with
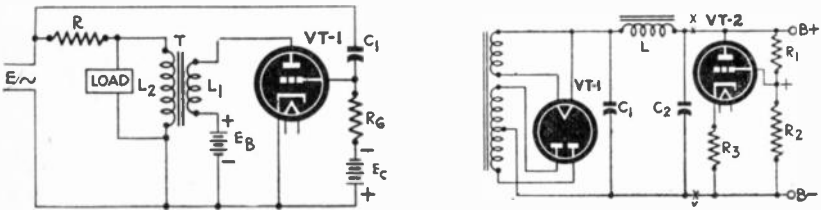


Fig. 5—left—Voltage regulator which employs a saturable reactor T and tube to balance the rise and fall of potentials in the circuit. Fig. 6—right—Voltage control circuit utilizing a vacuum tube as shunt resistor. The effect is that of connecting a variable bleeder across the output of the B supply.

the load circuit in order to obtain a stabilized output voltage, since the output voltage is the internal source voltage minus the drop in the internal resistance.

High-resistance supplies may be used with devices which draw very little current. By having more than 90% of the total circuit resistance in $R_1$ (Fig. 1) any change in the load impedance can only have a 10% effect on the circuit as a whole.

In some communities trouble may be experienced due to excessive line voltage at certain times of the day. Tubes may fail in the receiver prematurely or may even be burned out. By putting a resistance in series with the line, a measure of protection may be gained. As the line voltage rises and the circuit current tends to rise, the increased current flow through the series protective resistance develops an increased voltage drop which subtracts from the line voltage applied to give the receiver working voltage. A special form of this regulator is the "ballast re-sistor." This is constructed to work on a critical point of its temperature-resistance curve. If current through it increases slightly its resistance rises greatly, thus holding down the voltage supplied to the load.

A more refined device is that of Fig. 5, which may be a part of an industrial control. A saturable reactor T is used. A bias potential $E_c$ is applied to the grid of a control tube, and a certain amount of plate current, determined by the tube design, the bias, load impedance, plate potential, etc., flows through the winding $L_1$. The magnetic flux due to this d.c. circulates in the core of the saturable reactor transformer T. As long as this plate current does not become too large, saturation and decrease of the primary inductance $L_2$ does not take place. When the

line potential E rises, an increased signal voltage at the line frequency is applied through condenser $C_1$ to the grid of the control tube, which is so biased that any increase in signal voltage due to E will cause a linear rise in the current through the plate circuit and $L_1$. With increased d.c. in $L_1$, the core flux rises and the core approaches saturation, causing a decrease in the effective inductance of $L_2$ and the impedance of $L_2$. We have, in effect, a variable impedance connected across the load. Decreasing the value of the impedance reduces the load voltage and increasing it causes the load voltage to rise. By suitably choosing operating parameters the circuit conditions can be set.
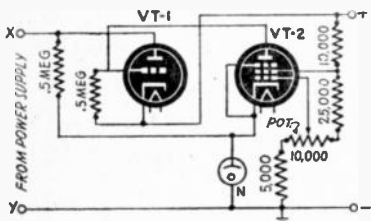


Fig. 7—Fully electronic voltage regulator; tube VT-1 acts as a variable series resistor. Tube VT-2 acts as variable shunt resistor. Interaction of 2 tubes stabilizes voltage effectively.

A different form of control action is used in Fig. 6. A d.c. potential is applied to the grid of the tube. If the output voltage of the supply rises, there will be a tendency for the grid to go more positive. When this happens, an increased amount of current tends to flow in the plate circuit of the tube and through the cathode resistance. Any increase in the voltage drop across $R_3$ will tend to reduce the plate current since, with increased drop across $R_3$, the grid is made more negative with respect to the cathode. Accordingly, a certain amount of degenerative action and stabilization may be expected. In effect we have a shunt resistance across the output of the B supply, between the B plus and B minus terminals, consisting of the tube's plate-cathode resistance in series with $R_3$. Somewhat better regulation may be obtained by substituting a special bias battery for $R_2$, thus maintaining constant voltage between grid and negative terminal.

A completely electronic voltage regulator is shown in Fig. 7. The voltage-regulator tube of Fig. 6 is used in conjunction with another, VT-1, which acts as a variable resistor. This circuit may be substituted for that of Fig. 6 at the points X and Y. An increase of output voltage increases the voltage on the grid of VT-2 thereby increasing its plate current. This current is drawn from the output voltage terminal through the half-megohm resistor between grid and cathode of VT-1, increasing the voltage drop across it and proportionately dropping the grid voltage of VT-1. This reduces current through that tube and increases the voltage drop across it, effectively reducing the output voltage. VT-1 is an electronically variable resistor.

A neon-tube, N, or a battery may be used to maintain the cathode voltage at a predetermined level above ground. The half-megohm resistor between it and the rectifier output causes the tube to remain "struck" at all times, thus avoiding oscillator action sometimes experienced otherwise.

# FM SERVICING POINTS

$S$ ERVICING AN FM receiver is much like servicing an AM set, but there are certain fundamental differences and peculiarities in design that make FM servicing a more difficult problem.

Like AM sets, FM radios have by-pass condensers that break down, resistors and coils which open-circuit, and tuned circuits that may require alignment. FM radios are more critical than AM sets where tuned circuits are concerned. This is true for two reasons. First, the FM set uses a detector arrangement similar to the discriminator circuit in an automatic frequency control system, which is very critical. Second, the r.f. and oscillator circuits operate on high frequencies in FM; much higher than those usually encountered in AM work.

The new sets tune from 88 to 108 megacycles, while the older FM sets tuned from about 40 to 50, and the broadcast receivers from .55 to 1.6 megacycles. The intermediate frequencies are also much higher. The FM set may have an intermediate frequency of 4.3 mc in the older sets (Fig. 1) or 10.7 mc in the postwar receivers designed in conformity with RMA standards. Other intermediate frequencies may be found, which do not agree with either the old or new standards. All these are almost equally high as compared with common broadcast i.f.'s, which are usually near 455 kc.

Fig. 1—The curve above shows typical FM i.f. response at frequencies near the i.f.

An FM radio has a limiter stage whereas an AM set does not. (Postwar FM sets using *ratio detectors* do not have limiters.) The tuned circuits in an FM radio are designed to pass a wide band of frequencies (approximately 75 kc either side of the center reference frequency). FM radios used in communications systems of police and fire departments and in amateur radio have a narrower pass band.

The Stromberg-Carlson Model 505 FM receiver is shown in Fig. 2. It uses a 6AC7 as an r.f. amplifier, coupling the antenna stage to the following mixer stage. The mixer works into a 6AB7 first i.f. amplifier followed by a second 6AC7 which also serves as an i.f. amplifier, further boost-
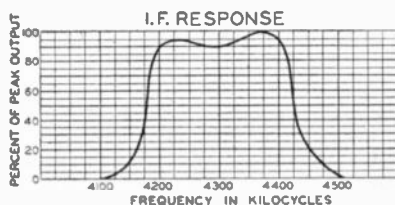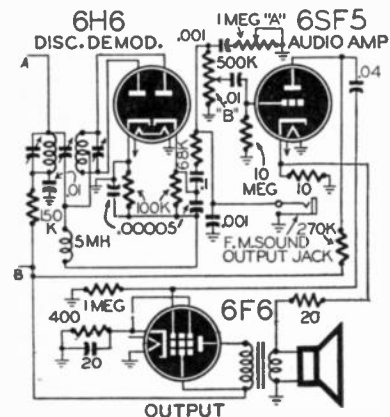
ing the signal strength. This amplified signal is then fed to the 6SJ7 limiter. The function of the limiter is to "iron out" any variations in the signal amplitude so that the FM detector will have a signal of varying frequency but fixed amplitude. (Some sets employ two limiters to further level out the variations in amplitude and to increase the noise reduction.)

From the limiter stage, the signal is applied to the 6H6 FM detector. This tube converts the frequency variations into amplitude variations



Fig. 2—Schematic layout of an FM receiver, the Stromberg-Carlson No. 505. A microammeter may be inserted in the circuit by removing the strap at the ground end of the 22,000 ohm limiter grid-leak.

and supplies an audio signal which is fed to the 6SF5 first audio amplifier. The amplified signal is then applied to the 6F6 final output stage.

Principal differences between an AM set and an FM set are, therefore, that the operating frequencies for the mixer and i.f. stages are different and that limiter stages are employed. A "frequency-change detector" rather than an amplitude-change detector is used. And a wideband i.f. amplifier is used in the FM receiver. This means that a somewhat different alignment technique is required, but if the manufacturer's instructions are carefully followed, alignment will not be a difficult matter at all.

In aligning an FM set, it is best to allow the signal generator to warm up for at least half an hour so that drift will be minimized. The generator calibration may be checked by zero-beating against stations of known frequency on the FM band. While many generators are very precise, even the best of them will not be as good as the stations themselves as far as frequency stability is concerned.

The receiver should be allowed a sufficient warm-up period. Due to the very high operating frequencies, a slight change in the oscillator circuit constants of the receiver may result in a relatively wide change in frequency. If the set is aligned when cold, the performance under actual working conditions may be poor. The oscillator and second-detector circuits are the most critical in many FM radios.

Before beginning the alignment, the tubes should be checked in a tube tester and the operating voltages checked. Any routine faults, such as leaky by-pass condensers, changed values of resistors, faulty tubes, leaky coupling condensers or filters, etc., should be corrected before alignment is attempted. The alignment of an FM set calls for an accurate signal generator and a vacuum-tube voltmeter or high-sensitivity galvanometer, *at least*.

If a high-frequency signal generator is not available, it is possible to use an ordinary signal generator and employ harmonics of the lower fundamental frequencies. For example, the fourth harmonic of 20 mc would be 80 mc. This range could be used for alignment in some cases.



Right half of diagram shown on opposite page. Points AA and BB join together.

A frequency-modulated signal generator and an oscilloscope combined provide the most accurate and convenient method of FM alignment, but a satisfactory job can also be done with an ordinary signal generator covering a range in the vicinity of the i.f. The object of alignment is to adjust the trimmers so that the i.f. system has a band-pass from i.f. —75 to i.f. +75 kc, and then to adjust the discriminator to cover exactly the same band.

## Aligning FM Receivers

Alignment procedure varies with different sets. *Follow the manufacturers' recommendation.* Standard alignment methods for FM receivers are explained on the following pages.

# Adjusting FM Receivers

$\mathbf{T}$EST EQUIPMENT required to maintain FM receivers satisfactorily should not present any problem to the experienced serviceman. The essential instruments required are an accurately calibrated vacuum tube voltmeter, preferably of the zero center scale type; an accurately calibrated signal generator, which should cover the FM band; and the other common instruments used to service AM receivers.

In Fig. 1 is shown the schematic of a conventional FM receiver. We will assume that the i.f. of this receiver is 10.7 mc, and that it is push-button tuned. We will follow through all of the adjustment and maintenance steps in such a manner as to make them applicable to any other FM receiver. An itemized list of the parts used in the receiver is given in chart 1, on page 64.

## Adjusting the Discriminator

As in AM we work from the detector back to the converter in alignment. The only instruments required to align the discriminator are a signal generator and a vacuum tube voltmeter which reads down to at least .5 volt:

(a) Set tuning dial of signal generator to 10.7 mc.
(b) Connect "hot" lead of signal generator to grid of 2nd limiter V5 through a .01 μf paper condenser. (Point 1 on Fig. 1.)
(c) Connect "cold" lead of signal generator to chassis (gnd.) of receiver.
(d) Short out the oscillator section of V2 to prevent spurious beats in the mixer during alignment.
(e) Connect the diode plates of V6 together, so V6 will act like a simple half-wave rectifier.
(f) Connect the "hot" lead of the v.t. voltmeter to the negative side of R26 (point 2 on Fig. 1) and the "cold" lead to ground.
(g) Set the v.t. voltmeter range selector switch to the position that gives the lowest practicable reading.
(h) Feed just enough signal from the signal generator to the limiter grid to give a substantial reading. *Note:* The modulation switch of the signal generator should be in the *off* position during the entire alignment of the receiver.
(i) Adjust the primary trimmer of T4 at point 3 for *maximum* voltage indication on the v.t. voltmeter.
(j) Disconnect the lead across the diode plates of V6.
(k) Disconnect the "hot" lead of the v.t. voltmeter from point 2 and connect it to point 4 on Fig. 1. You will now be reading the total d.c. voltage across R25 and R26.
(l) Adjust the secondary trimmer (5) of T4 for *zero* voltage indication on the v.t. voltmeter.
(m) Swing the signal generator tuning dial to 10775 kc and note the voltage reading on the v.t. voltmeter; *it should be negative.*
(n) Swing the signal generator tuning dial to 10625 kc and note the voltage read-

**60**

ing on the v.t. voltmeter; it should be positive, and the same value as in (m).
*Note*: Checks (m) and (n) are made to indicate whether or not the discriminator is responding linearly to the plus and minus 75 kc deviation in frequency encountered. If the same voltage readings are not obtained in (m) and (n) all of the constants of the discriminator should be checked to determine the fault. This method is the simplest and most accurate for aligning the discriminator circuit if carried out systematically.

# I. f. Amplifier Alignment

In aligning the i.f. transformers it should be remembered that the bandpass of these transformers is about 200 kc to accommodate the frequency swing, and in adjusting these stages this factor must appear as part of that adjustment:

(a) Set the tuning dial of the signal generator back to 10.7 mc.
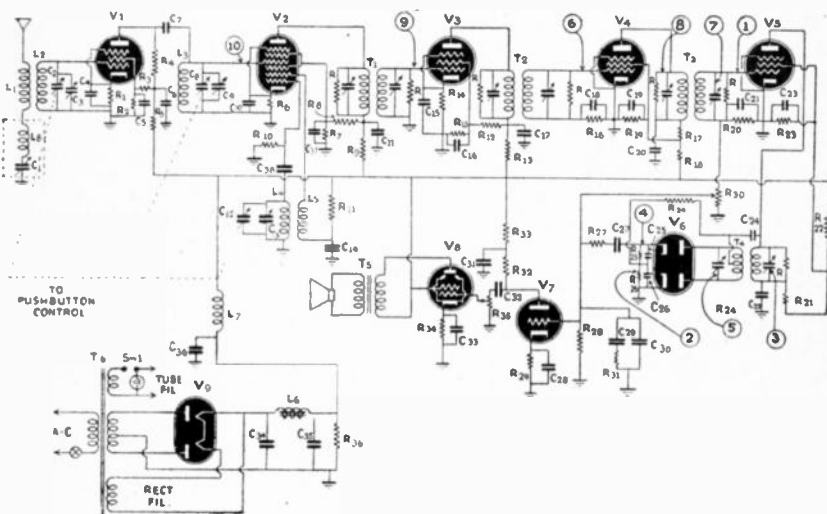


Fig. I—Schematic diagram of an FM receiver. The circled numbers are referred to in the text. The adjustment of Discriminator, Limiter, and Squelch circuits is discussed.

(b) Connect the "hot" lead of the signal generator to the grid of the 1st limiter V4 at point 6 in Fig. 1.

(c) Connect the "hot" lead of the v.t. voltmeter to the grid of the 2nd limiter V5 at point 1 on Fig. 1. *Note*: At this point a *negative* voltage will be indicated. This is the voltage drop across grid-leak R20, and the attenuator of the signal generator should be turned up until this negative voltage appears.

(d) Adjust the secondary trimmer (7) of T3 for maximum negative voltage reading on the v.t. voltmeter.

(e) Adjust the primary trimmer (8) of T3 for maximum negative voltage reading on the v.t. voltmeter.

(f) Swing the signal generator tuning dial to 10775 kc (maximum deviation frequency, plus) and note the voltage reading.

(g) Swing the signal generator tuning dial to 10625 kc (maximum deviation frequency, minus). The voltage reading should be the same as in (f); if not, repeat steps (d) and (e). An error of 1% in the readings is allowable.

(h) Leaving the v.t. voltmeter connected to the 2nd limiter grid connect the signal generator "hot" lead to the i.f. amplifier grid at point 9.

(i) Adjust the primary and secondary trimmers of T2 as described in (d), (e), (f) and (g).

(j) Connect the signal generator "hot" lead to the converter control grid at point 10.
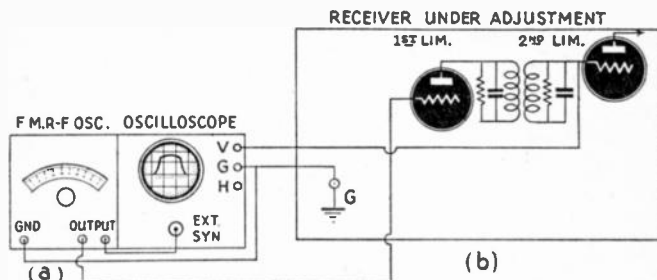


Fig. 2—A cathode-ray oscilloscope, with a frequency-modulated signal generator may be used advantageously in checking and adjusting FM receiver circuits. A good electronic voltmeter is also necessary.

(k) Adjust the primary and secondary trimmers of T1 as described in (d), (e), (f) and (g). This completes the alignment of the i.f. transformers.

## Visual Alignment

If the serviceman is fortunate enough to have at his disposal a good cathode-ray oscilloscope and a frequency-modulated r.f. oscillator that covers the desired 150 kc swing, visual alignment can be accomplished.

Fig. 2 shows method of connection for alignment. The output of the FM r.f. oscillator is connected to the external synchronizing terminal of the oscilloscope and to the grid of the stage ahead of the i.f. transformer to be aligned. The vertical amplifier of the oscilloscope is connected to the grid of the stage following the i.f. transformer to be aligned as shown at 2(b). With the deviation control on the FM r.f. oscillator set at 75 kc, and using a three-inch cathode-ray tube, an image width of 1.5 inches on the screen would indicate a bandwidth of 150 kc. This is shown in Fig. 3. By adjusting the primary and secondary trimmers of each i.f. transformer an image similar to Fig. 3 appears. Each transformer adjusted should produce the same bandpass pattern on the screen. This method will provide the most nearly perfect alignment.

Wave traps are used in FM receivers for the same reason as in AM receivers; to eliminate response to signals at the intermediate frequency.

(a) Connect the "hot" lead of the v.t. voltmeter to the grid of the 2nd limiter V5 at point 1.

(b) Remove the jumper shorting the oscillator section of the converter.

(c) Set the signal generator tuning dial to 10.7 mc.

(d) Connect the "hot" lead of the signal generator to the antenna terminal of the receiver.

(e) Adjust C1 for *zero* limiter grid voltage indication on the v.t. voltmeter. Any other wave trap that would be present in the receiver would be adjusted in the same manner.

## Push Button Adjustment

Regardless of the type of push button assembly employed, the following procedure can be carried out successfully:

(a) Leave the v.t. voltmeter "hot" lead at point 1.
(b) Set the signal generator tuning dial to the desired station channel.
(c) Connect the signal generator "hot" lead to the antenna terminal of the receiver.
(d) Loosen the push button tuning gang locking screw or screws.
(e) Depress the push button that corresponds to the desired station channel.
(f) Adjust the receiver tuning control gang to the desired channel and maximum voltage indication on the v.t. voltmeter. *Note*: If the dial pointer must be set off the correct channel number to obtain a maximum voltage, set the tuning dial exactly on some high frequency channel and correspondingly set the signal generator to this same frequency; then adjust the r.f. tank trimmers C3, C9, and C13, respectively, for maximum voltage.
(g) After the maximum has been found swing the signal generator tuning control to 10775 kc and 10625 kc, respectively, noting the voltage values; these values should not vary more than .1% for good accuracy and circuit tracking. If these values are off by any larger degree touch up the r.f. sections. Follow steps (a) through (g) for the additional push buttons on the receiver. If the receiver's r.f. sections are padded to spread the low end of the dial select some low frequency channel and adjust the padders for maximum limiter grid voltage. Thus alignment of the r.f. stages is completed.
(h) Tighten the push button locking screw.

## Squelch Circuit Adjustment

The squelch circuit is incorporated in some FM receivers to disable them during no-signal intervals. The following has been found to be the simplest method of adjustment:

(a) Set the manual tuning dial of the receiver to some point where no signal is received.
(b) Adjust the squelch potentiometer R30 (usually a screwdriver adjustment) to the point where the sensitivity hiss just disappears. *Note*: This should be done with the manual volume control advanced to *maximum*.
(c) Depress a push button, and the signal should open up the squelch and come through the receiver. If not, R30 will have to be adjusted to some optimum setting which will allow opening of the squelch, with suppression of as much of the sensitivity hiss as practicable.

If noise comes through the receiver with the signal, the trouble will usually be traced to the limiter stage or stages. The following should be checked: R16 and R20 for changes in value; C18 and C21 for open; R17, R18, R21 and R22 for changes in value. Regeneration sometimes occurs due to open plate and screen grid by-passes C19, C23, C20 and C32. Regeneration in the limiter can be detected by pulsating squeals in the audio output. These troubles are the most common ones.
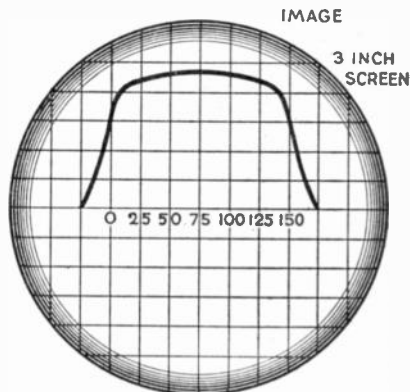


Fig. 3—Correctly adjusted FM discriminator curve as seen on the oscilloscope screen. Bandwidth is a full 150 kilocycles.

Very little trouble is ever found in the discriminator circuit if it is properly balanced. Sometimes distortion arises due to an open, or short, or an intermittent in C24. A fringe on the audio sometimes occurs due to open r.f. by-passes C25 and C26 across the load resistors. Outside of these few cases the discriminator usually behaves itself.

If precision parts are used for R30, R31, C30 and C29 there is nothing to worry about from the squelch circuit. If the squelch takes hold too fast the value of C29 should be increased (or R31). If the squelch lets go too quickly on noise peaks the value of C30 should be increased to allow a longer discharge period for C29. Improper adjustment is the biggest fault with this circuit and as the parts are few, it is comparatively simple to correct.

## CHART 1
### Function of Component Parts of Fig. 1

**VACUUM TUBES**
V1—R.f. amplifier
V2—Pentagrid converter
V3—I.f. amplifier
V4—1st overload limiter
V5—2nd overload limiter
V6—Discriminator
V7—1st a.f. amplifier
V8—Power a.f. amplifier
V9—Full-wave rectifier

**RESISTORS**
R1—R.f. amp cathode bias
R2—R.f. amp screen-grid bleeder
R3—R.f. screen-grid dropping
R4—R.f. amp plate load
R5—R.f. amp plate filter and decoupling
R6—Converter cathode bias
R7—Converter screen-grid bleeder
R8—Converter screen-grid dropping
R9—Converter plate filter and decoupling
R10—R.f. oscillator grid leak
R11—R.f. oscillator anode grid dropping
R12—I.f. amp screen-grid dropping
R13—I.f. amp plate filter and decoupling
R14—I.f. amp cathode bias
R15—I.f. amp screen-grid bleeder
R16—1st limiter grid leak
R17—1st limiter plate dropping
R18—1st limiter screen-grid dropping
R19—1st limiter screen-grid bleeder
R20—2nd limiter grid leak
R21—2nd limiter plate dropping
R22—2nd limiter screen-grid dropping
R23—2nd limiter screen-grid bleeder
R24—Discriminator cathode return r.f. filter
R25, R26—Discriminator load
R27—1st a.f. grid series resistor
R28—1st a.f. grid leak
R29—1st a.f. cathode bias
R30—Squelch sensitivity control
R31—Part of squelch time constant filter (with C29 and C30)
R32—1st a.f. amp plate load
R33—1st a.f. amp plate filter and decoupling
R34—Power a.f. amp cathode bias
R35—A.f. volume control (manual)
R36—Power supply bleeder
R—I.f. transformer damping

**TRANSFORMERS**
T1—1st i.f. converter to i.f. amplifier
T2—2nd i.f., i.f. amplifier to 1st limiter
T3—3rd i.f. 1st limiter to 2nd limiter
T4—Discriminator; 2nd limiter to discriminator

T5—A.f. output
T6—Power supply

**MISCELLANEOUS**
S1—Off-on switch
P—Pilot light

**CONDENSERS**
C1—Part of series wave trap (L8)
C2—R.f. amplifier tank tuning
C3—R.f. amplifier tank trimmer
C4—R.f. amplifier cathode by-pass
C5—R.f. amplifier screen-grid by-pass
C6—R.f. amplifier plate by-pass
C7—D.c. blocking r.f. coupling plate to converter control grid
C8—Mixer section tank tuning
C9—Mixer section tank trimmer
C10—Converter cathode by-pass
C11—Converter plate by-pass
C12—R.f. oscillator tank tuning
C13—R.f. oscillator tank trimmer
C14—R.f. oscillator plate by-pass
C15—I.f. amplifier cathode by-pass
C16—I.f. amplifier screen-grid by-pass
C17—I.f. amplifier plate by-pass
C18—1st limiter grid leak by-pass
C19—1st limiter screen-grid by-pass
C20—1st limiter plate by-pass
C21—2nd limiter grid leak by-pass
C22—2nd limiter plate by-pass
C23—2nd limiter screen-grid by-pass
C24—D.c. blocking
C25, C26—Discriminator load r.f. by-passes
C27—Part of frequency correcting network (with R27)
C28—1st a.f. amplifier cathode by-pass
C29—Part of squelch time constant network (with C30 and R31)
C30—Part of squelch time constant network (with C29 and R31)
C31—1st a.f. amplifier plate by-pass
C32—D.c. blocking a.f. coupling 1st a.f. plate to 2nd a.f. grid
C33—Power a.f. amplifier cathode by-pass
C34, C35—Power supply filters
C36—D.c. supply bus r.f. by-pass
C37—Converter screen-grid by-pass
C38—Converter grid leak condenser

**COILS**
L1—Primary antenna transformer
L2—R.f. amplifier tank (secondary of antenna transformer)
L3—Mixer section tank
L4—R.f. oscillator tank (primary)
L5—R.f. oscillator tank (secondary)
L6—Filter choke (power supply)
L7—D.c. bus r.f. choke
L8—Part of wave trap (with C1)