INITIATIVE

The man who does only the routine tasks, the ordinary jobs in his profession, always waiting for the other fellow to take the lead, can expect only moderate returns for his labors. He who is continually on the alert for new ideas and new uses for his talents—who is alert to grasp each new opportunity—gets the greatest profits. The immediate financial returns from work in a new and specialized branch of your profession may not be great, but the reputation which you gain for progressiveness will soon result in more profitable routine jobs. It all boils down to these simple facts—you must do out-of-the-ordinary things, stand above the crowd in some way, to attract favorable attention. People remember you first for the unusual, then for your ability to do ordinary work well.

Radio interference-elimination, the subject of this book, can prove a very profitable radio side-line and reputation-builder for the man who has initiative backed by knowledge and "horse sense"; it is for you to decide just how much attention you want to devote to this particular phase of radio.

J. E. Smith.

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A LESSON TEXT OF THE N. R. I. COURSE WHICH TRAINS YOU TO BECOME A RADIOTRICIAN & TELETRICIAN
(REGISTERED U. S. PATENT OFFICE) (REGISTERED U. S. PATENT OFFICE)

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How To Eliminate Man-Made Interference

A GROWING PROBLEM

The radio public is today being supplied with receivers of greater sensitivity than ever before; short-wave reception of foreign as well as local programs is an accepted feature of the modern home receiver, and listeners are gradually becoming conscious of the superior performance of high fidelity receivers. These three important factors make the problem of man-made interference more and more important as new receivers reach the hands of the public.

Radio receiver manufacturers are now capable of building receivers which create only a negligible amount of interference within themselves; older receivers which develop internal noise can readily be repaired by the Radio-Trician, but still program-spoiling interference increases.

Oil burners, electric power-generating systems, refrigerators, motor-driven appliances, medical equipment, electric signs and scores of other new electrical appliances are man's contributions to radio receiver interference. Thus man creates more interference at the same time that he builds radio receivers which are more sensitive to interference; profitable work for the serviceman trained in interference elimination is the result. Remember that no radio installation is complete and satisfactory until it is as free from interference as is humanly possible. The man who can render this interference elimination service efficiently and intelligently will "cash in" on an opportunity for profit and prestige which grows bigger every day.

NOISE NOT DUE TO RECEIVER DEFECTS

We know that when noise is heard in a receiver, the first step is to eliminate receiver defects as possible causes of the trouble. A line filter is inserted in the power line of the receiver, the antenna and ground leads are disconnected from the receiver, and antenna and ground binding posts are shorted together; if, when this is done, the noise disappears or is reduced an appreciable amount, the trouble is definitely not a receiver defect. It is, therefore, an external disturbance which can or cannot be eliminated, depending upon its nature.

External noise disturbances which cannot be eliminated may be
divided into two groups: (1), those due to local electrical storms or lightning; (2), those due to the accumulated effects of distant electrical storms, sun disturbances and disturbances created by distant industrial or electro-medical equipment.

The new frequency modulation system of broadcasting almost completely eliminates atmospheric interference, but both broadcasting systems (f.m. and a.m.) have serious man-made interference problems.

The accumulated noise disturbance is often referred to as background noise; * this has a definite level (microvolts per meter) which will vary with the antenna location. Industrial towns and cities will usually have a high noise level, this being exceptionally high near factories and shopping centers. The only remedy in such cases is to cut down the sensitivity of the receiver or confine tuning to broadcasts whose intensities are much greater than the noise level. In localities of high noise level the customer should be taught to listen only to local or high-powered stations.

When receivers having automatic volume control are tuned off a broadcast signal, the AVC acts to boost the gain, and background noise becomes disturbingly prominent. This has led to the development and use of inter-carrier noise suppressors, found on a number of receivers.

Man-made static, usually of local origin and having an intensity comparable with that of the normal received signal, is often so annoying that the usefulness of a receiver is destroyed. It is the purpose of this text to show the origin of such disturbances and suggest ways and means of eliminating or at least greatly reducing such interference. The "cure" is generally applied in two steps: first, by seeking to keep the noise signal out of the receiver; and second, by "killing" the interference at its source.

ORIGIN AND NATURE OF NOISE SIGNALS

Wherever there is an electric spark or arc, there you will find a source of possible noise interference. The spark need not be large or even visible to create a disturbing effect. Contrary to general belief, the spark itself does not radiate interference, nor is it generally true that the spark creates a broadcast band radiation.

* This background noise should not be confused with noise originating within the receiver due to thermal agitation of the electrons in the conductors and to the impact or shot effect of electrons as they hit the plates of vacuum tubes. It is this noise which is heard when antenna and ground terminals of a high quality receiver are shorted and the gain turned up.
A spark is accompanied by a sudden current change in the circuit where it originates, the change being transmitted to all parts of the circuit. This sharp current change gives rise to a fundamental audio frequency noise signal whose frequency depends upon the duration of a single disturbance, and a large number of audio and super-audio frequency harmonics of this fundamental. These noise signals may reach the receiver by conductive, magnetic or capacitive coupling, and may either affect the audio stages directly or, more likely, enter a resonant R.F. circuit. The latter is more troublesome, for through shock-excitation it results in the formation of an R.F. current which is modulated with the original noise signal. Because the original noise signal wave form is not destroyed or altered, the expert is usually able to judge, after listening to the noise emitted from the loudspeaker, what the probable source of interference may be.

When a spark occurs in an electric circuit, the current surge is transmitted through the connecting wires, away from the origin of the spark, in both directions and out of phase. In a power transmission circuit this means that a large area—several blocks—will be affected. This disturbance will continue to travel until it is dissipated by the system. If the circuit contains transformers or other circuit-changing components, part of the disturbance will be reflected back to the origin at the first of these points, be reflected again at the disturbance source, and continue to travel back and forth until the losses in the circuit wipe out the disturbance. The remainder of the surge passes through the first obstacle and out over the line to the next, where it in turn is partially reflected, partially transmitted.

Whenever the surge of current meets an electrical obstacle in the line, be it a transformer, a change in wiring construction, or even a noise-eliminating filter introduced into the line improperly by an untrained radio man, the surge moves back and forth between its origin and this point, creating a standing wave or ultra high frequency oscillation whose frequency is determined by the line length. This wave is radiated through space in much the same way that R.F. currents are radiated by a transmitter antenna.

Sparks in auto ignition systems are typical examples; because the ignition wires are short, the natural wavelength of the radiation is somewhere between 1 and 10 meters. This explains why 5 to 10 meter ultra short-wave reception is so greatly affected by auto ignition disturbance.

* If you were to drop a stone in a long trough filled with water, the disturbance would likewise travel away from the source to both ends of the trough, and then would be reflected back to the origin of the disturbance.
Bear in mind that a spark or arc produces a current surge or impulse which is fundamentally of an A.F. or super-audio frequency. Because the circuit in which this impulse is created has reflecting points, ultra high frequency radiations are produced. The original A.F. impulse currents, flowing along the transmission lines, also produce strong magnetic and electrostatic lines of force which may travel an appreciable distance through space. Magnetic and electrostatic interference fields of this nature get into the radio receiver through the aerial and ground, over the power supply lines or directly through the chassis. As these impulse fields induce strong impulse voltages in the R.F. or I.F. oscillatory circuits, forced oscillations modulated by the original noise currents are produced.

A study of Fig. 1, which shows a typical “man-made static” problem, will bring out many of the facts just discussed. An electric motor, located in a house, is sparking at $S$, one of the brushes. Impulse current, therefore, passes out of the feeder line to points marked $1$, where a part divides to flow to points $2$ and $4$, and the remainder is reflected back to the motor to produce a radiation whose wavelength is determined by the distance between $S$ and $1$. At point $2$ the impulse current will again divide, a part going to house $B$ before being reflected back. The radio antenna on house $B$ picks up noise radiation from all electric wires in the house and from the power line system, and the radio receiver itself receives the impulse current directly through the power line. A radio in house $B$, therefore, picks up more interference than a radio in house $C$, which is unwired and therefore receives noise signals only through space.

It would appear that because of the parallel power leads in this system, out-of-phase impulse currents in the two wires would produce canceling fields. This is not true, because spark $S$ is rarely produced in the electrical center of the disturbing device. In this example, where sparking is occurring at one brush, one impulse passes directly into the line while the other passes through the armature and the other brush first. The inductance of the armature thus reduces the strength of one of the impulse current signals and prevents cancelation of the currents. It is safe to say that any line which is connected electrically to a spark source will send out an interfering induction field.

Reflection of the current impulses at points $1$, $2$, $3$ and $4$ produces standing waves on the line; radio waves modulated with noise signals are, therefore, radiated by the line to create troublesome interference in all-wave receivers.
REDUCING MAN-MADE INTERFERENCE

In tackling any interference-elimination job, the practical aspects of the problem must be carefully considered, and even human nature itself must not be overlooked. Broadly speaking, however, the interference-eliminating procedure may be divided as follows: 1, eliminate or reduce the sparking, if possible; 2, prevent the interfering current impulses from leaving the disturbing device; 3, prevent the various interfering signals from reaching and affecting the radio receiver. It is generally conceded that elimination of interference at its source is the best procedure, but in cases where this is impractical, filters and other devices which will keep the signal out of the radio receiver must be used.

Reducing the interference at its source is not always the simplest

![Diagram](image)

**FIG. 1.** Diagram illustrating how interference created by sparking brush 5 on motor in house A can reach radio sets in vicinity.

procedure, nor is it always permitted by the owner of the disturbing device. If a customer calls you on an interference job and you can directly trace the trouble to some device in the customer's home, the logical procedure is to kill the interference at its source. On the other hand, where your tests show that the interference is being created outside of the customer's home, you must decide whether to search for the location of the interfering device or prevent the interference from affecting your customer's receiver; remember that once a disturbing device is located you must convince its owner that there is a need for interference-eliminating work, and that this work will make his own receiver more free from interference.

After proving to yourself that the chassis of the receiver in question is not picking up noise directly (which of course includes trying a line filter to prove that the interference is not coming in over the

5
power line), your next important move is to install a noise-reducing antenna. You cannot, of course, guarantee that this will entirely eliminate noise interference troubles in the receiver, but you can be sure that it will improve radio reception as well as give a worth-while reduction in interference pick-up. Always make this perfectly clear to a customer who is ordering a noise-reducing antenna. Then, if the antenna fails in its primary purpose, you will not be blamed by the customer for something which is beyond your control, and you will be allowed to tackle the more difficult procedure of locating and eliminating the source of interference.

NOISE-REDUCING ANTENNAS

You are already sufficiently familiar with noise-reducing antennas, so they will not be discussed in detail in this lesson. The type of antenna which you select for a job depends upon the type of receiver encountered, the antenna location, and to some extent upon your personal preferences gained through experience with the products of different manufacturers. An all-wave receiver calls for an all-wave antenna, while a broadcast band antenna should be put up where only American broadcast band stations are to be received. The length of the antenna and lead-in wires will vary according to the space available.

The effectiveness of any noise-reducing antenna depends upon your ability to locate this antenna in a position where it will pick up a minimum of noise interference. You can determine the ideal position with a battery receiver, using a loop or pole antenna and moving the set about until you locate a zone where the least noise is heard, but these three general rules for locating noise-reducing antennas will often allow you to "spot" a good location at a glance: 1, Place the antenna as high as is reasonably possible, keeping all unshielded vertical wires short; 2, keep the horizontal or straightaway portion of the antenna at a maximum distance from known sources of interference; 3, place the horizontal portion at right angles to nearby trolley lines, main power lines or transmission lines. The antenna on house C in Fig. 1, for example, is at right angles to the main power line running from points 2 to 4; the antenna on house B is not at right angles to this line, and is, according to the general rule, incorrectly placed. This antenna may actually give better results than an antenna which is perpendicular to the power line, for oftentimes interference radiated from various points will cancel itself in certain regions. If an antenna erected according to general rules fails to reduce the noise sufficiently.
try it in various directions. An antenna located in a noise-free zone, with the shielded or twisted leads correctly balanced and grounded, may be expected to prevent pick-up of noise signals.

In a few instances it may be necessary to locate the exposed portion of the antenna at distances as great as 1,000 feet from the receiver, in order to get the antenna into a noise-free zone. Very little signal strength is lost by a long lead-in such as this, provided that both the antenna and the receiver are correctly impedance-matched to the lead-in, using shielded R.F. transformers for this purpose. Quite often, as in locations near railroad tracks along which run high tension power lines, or in locations near high power cross-country transmission lines, the placing of the antenna at a remote point is the only practical solution to the problem of interference elimination.

SUPPRESSING NOISE AT THE SOURCE

Assuming for the moment that the disturbing device has been located, you will invariably find it to be a spark, an arc or a rubbing condition. (All conductors such as pipes in homes acquire electrical charges; rubbing together of two of these pipes results in current impulses which cause interference.) If the spark or arc is not essential to the operation of the device, it should be eliminated or reduced in intensity. Rubbing parts should either be completely insulated from each other or bonded together with flexible metallic braid or stranded wire.

When the sparking can neither be eliminated nor reduced, the logical procedure is to prevent the current impulses from flowing any distance away from the device. For this purpose filters consisting of condensers alone, or combinations of condensers with choke coils, are available and in general use. The correct sizes for these condensers and choke coils are usually quite difficult to determine in advance; it is necessary to try different values and use the smallest electrical sizes which satisfactorily stop the interference.

The most commonly used coil-and-condenser combinations for filtering or blocking impulse currents are shown in Fig. 2. That shown at A, consisting simply of a condenser connected across the power line as close as possible to the noise source, is often quite effective as a filter. The shunt capacity provides a low impedance path back to the noise source for the high frequency component of the impulse current, lessening the tendency for this current to flow out over the power line. When this condenser is installed on a vacuum cleaner, for example, it should preferably be connected to the terminals of the motor and not
across the outlet plug terminals on the wall. If possible, try grounding the metal frame of the offending device; a short ground lead oftentimes reduces interference appreciably. All condensers used for filtering purposes on 110- or 220-volt A.C. power lines should have peak voltage ratings of between 600 and 1,000 volts, for these units must withstand high voltage surges caused by impulse currents.

When trying various filter combinations, it is important that some one listen to the receiver to note the effectiveness of each combination when the disturbing device is not within “ear shot” of the receiver. Oftentimes the customer will be only too glad to listen to the receiver for you, but better results can generally be obtained with a trained assistant. If you are working alone, it is wise to set up a portable battery receiver near the location of the disturbing device, using headphones rather than a loudspeaker if the interference noise proves too annoying to those nearby.

With the filter shown at A in Fig. 2, there is no assurance that the impulse currents will pass to ground; the balanced condenser filter, having its center points grounded as shown at B, is therefore more effective.

When condensers of a reasonably high capacity, such as 1-mfd. units, fail to give satisfactory noise reduction when used alone, a combination condenser-and-choke filter like that shown at C should be tried. This is essentially a brute filter which allows only very low frequency currents to pass through to the power line. The higher the electrical values of the coil and condenser, the better is the filtering action. Always use the smallest commercially available size which gives satisfactory results, for purposes of economy. The condenser may be connected either to the load side of the choke coil (C) or to the line side of the choke coil (D). As a rule, however, the closer the choke is to the source of interference (D), the better is the impulse filtering action. Try the choke coil in one power lead first, then the other, to ascertain which position gives the better reduction in noise.

Two choke coils and one condenser connected either as at E or F will often give improved results, while the grounded combinations shown at G, H and I are even better filter combinations. Where several different parts of a device are sparking, such as in commutator type switches for signs or groups of contacts on a relay, then each line which carries impulse currents should be filtered in the manner shown at J. A choke coil is inserted in each line, and a suitable condenser connected from the load side of each line to ground.
Improved suppression of interference is often obtained by using a balanced filter having a ground connection which can be electrically varied in the manner shown at $K$; this circuit is otherwise essentially the same as those shown at $G$ and $H$. The same balancing scheme can be used with the simple two-condenser filter shown at $B$; a 100-ohm potentiometer, with its variable tap grounded, is connected between the two condensers.

![Circuit diagrams](image)

**FIG. 2.** Condenser filters and condenser-coil filters are here arranged approximately in the order of their effectiveness, the circuit at $I$ being the most effective for interference eliminating purposes. Circuit $J$ is used with devices which have three make-and-break contacts or with a three-phase load, while circuit $K$ is a variation of circuit $G$, which permits adjustment of the ground point. Grounding of $S$, the disturbing device, is optional in circuits $A$ through $F$.

**A TEST DEVICE FOR DETERMINING THE MOST EFFECTIVE FILTER**

Any serviceman, having located an interference-producing device, can almost always secure an effective cure by installing an expensive filter like that shown at $I$ in Fig. 2. But cost to the customer must also be considered in a successful noise elimination job. If noise-free reception costs too much, many people will forego the use of their receivers or endure the noise, rather than pay the price; this is clearly not an encouraging condition for the radio sales and service business. Experience has proven that a satisfactory job done at the lowest possible cost to the customer—a charge which gives a fair profit—is one of the most important requirements for success in radio servicing. This means that the simplest and lowest cost filtering devices should
always be tried first, working up gradually to the more complicated and more expensive combinations until the lowest cost unit is found which gives satisfactory filtering.

A variable filter combination system which gives a choice of circuit combination $A$, $B$, $C$, $D$, $G$ or $H$ in Fig. 2 simply by changing the setting of a rotary switch and changing connections to the unit is shown schematically in Fig. 3. All condensers used here should preferably have working voltage ratings of between 600 and 1,000 volts, while the choke coils should be capable of handling at least 5 amperes. Use non-inductive paper type condensers mounted in metal cases which can be grounded. Notice that two outlet receptacles, each having a plug-in cap with insulated alligator clips attached to flexible leads, are used for the input and output connections. A ground connection is made by means of a flexible lead having at one end a prong

![Circuit diagram](image)

**FIG. 3. Circuit diagram of a variable filter combination system which you can easily make for use in determining the most effective filter combination for an interference-creating device. Connections are made by plugging into standard electrical receptacles at $A$ and $B$, and by plugging test prong into pup jack $G$. Mount parts in box of convenient size. Be sure power is off before making connections. The filter circuits provided here are those most generally used.**

which plugs into a “pup” jack on the unit; at the other end of this lead is an alligator clip which is to be attached to the frame of the interfering device or to a grounded object.

When side $A$ of this variable filter circuit is connected to the offending device, the condensers are next to the source of noise; when side $B$ is connected to the device, the choke coils are closest to the source of noise. Single condenser connections and single choke and condenser filters are obtained by using one lead at $A$, one at $B$ and the ground connection. When using condensers alone, always start with the lowest capacity, increasing the capacity up to 1 mfd. before resorting to choke coils. In making this test filter, be sure to use only those parts which can be readily obtained from radio supply houses at any time, for once the best filter setting is found, you must duplicate the parts used at that setting.

The method just described for using a variable filter combination
system to determine the correct filter for a given job was first introduced by the Sprague Products Company; the interference analyzer which they developed for this purpose is shown in Fig. 4A, while the circuit diagram of their analyzer appears in Fig. 4B. This device is used in much the same way as that which was just described. The condensers and choke coils used in the Sprague Analyzer are exactly the same as the units supplied by the Sprague Products Company for use in interference filters; several of these are shown in Fig. 5. The choke coil is capable of handling currents up to 10 amperes; where larger currents must be filtered, larger capacity chokes can be obtained.

When the condensers and choke coils required for a noise elimina-

![Courtesy Sprague Products Co.]

**FIG. 4A.** This Sprague Interference Analyzer is one of the serviceman’s most effective weapons in the war against man-made radio interference. The knob on top controls the circuit-selecting rotary switch.

![Circuit diagram of Sprague Interference Analyzer]

**FIG. 4B.** Circuit diagram of Sprague Interference Analyzer. Numbers alongside condensers and choke coils refer to special interference elimination condensers and chokes sold by the Sprague Products Company, and shown in Fig. 5. A four-deck switch with six contacts per deck gives six different combinations of filtering units. Connections are made by inserting standard connecting plugs into the receptacles at A and B, and by plugging a test probe into the plug jack marked GND. Condenser IF-11 is of the dual-unit type, with the metal case serving as the common grounded connection. Positions 3 to 6 give balanced condenser filters.

- it is wise from the standpoint of eliminating fire hazards, securing a shock-proof installation and improving the general appearance of the installation, to mount the condensers and chokes in a standard electrical cut-out box such as is shown in Fig. 6. This procedure is compulsory for heavy-duty electrical devices which must pass fire underwriters’ specifications and the regulations of local electrical inspectors.

As you already know, a filter unit must be placed as close as possible to the source of sparking if it is to be effective in reducing noise. When a cut-out box is used, the leads connecting it to the source of disturbance should be run through BX conduit or iron pipes, this conduit being permanently clamped at one end to the cut-out
box and at the other end to the disturbing device; if necessary, a separate ground wire should be clamped or soldered to the conduit. This shielding procedure will prevent the standing waves, formed on the connecting wires, from radiating modulated disturbance waves of low wavelengths, which might cause interference in ultra high frequency receivers.

As a rule, interference filters have little effect upon the sparking or arcing itself, and serve only to prevent the current impulses from getting into the power line. Quite often the sparking at relay contacts, switch contacts and other make-and-break contacts can be greatly reduced by using a resistor in series with a single filter condenser connected across the spark source. This connection is especially worthwhile if you wish to prevent sparking contacts from pitting badly and producing even more serious disturbances at a later date.

THE NOISE DETECTIVE

You now know what to do once an interference-producing device is located; locating the offending device itself is another problem, however, and one which often calls for systematic thinking and even detective work. A well-trained interference-elimination technician can listen to the noise coming in over a radio receiver, ask a few questions of the customer and from these observations get clues which will permit rapid isolation of the offending device. Just as a detective asks questions when searching for a criminal, so should the Radio-Trician ask questions when on an interference job. When was the noise last heard? At what time of the day or night is it usually heard? Is the noise always the same in character? When was the
noise first heard? These are questions whose answers may give you clues to the solution of the problem. The opinion of the customer as to the source of the trouble is also of value. Ask if the noise began about the time that some one in the neighborhood bought an electric refrigerator, a vacuum cleaner, fruit juice extractor, or other electrical appliance; try to associate the beginning of the interference noise with the arrival of a new neighbor, the installation of traffic lights at the corner, or the installation of a new neon sign in a nearby store. Neighborhood gossip can provide useful tips for the noise detective.

The value of knowing the time when interference noises are heard can easily be demonstrated. For example, interference heard for a little while around breakfast time and perhaps occasionally late in the evening at a time when you know that the neighbors are having a party, may be produced by a fruit juice extractor; noise heard at intervals fifteen to thirty minutes apart may be due to an oil burner, a refrigerator, an air compressor in a nearby beer parlor or any other device which is operated only for short periods of time and is automatically controlled. Interference which is heard only when a street car or train is passing near the house gives an obvious clue; interference heard in apartments when the elevator is in operation proves that the trouble is in the elevator motor. Clicking noises heard when lights are turned on in the house tell their story at once. Your questioning of the customer, once you suspect a possible cause of the
trouble by listening to the noise yourself, should result in a quick isolation.

If the interference can be picked up by the radio receiver at the time when you call, give special attention to any peculiarities of the sound; note whether the interference is heard with the same intensity at all frequencies. With a little experience you will be able to make very good guesses as to the causes of different types of interference noises. Until you have gained this experience, use the following suggestions which have been prepared by the Tobe Deutschmann Corporation of Canton, Massachusetts, as your guide in recognizing the sources of interference noise which you hear.

*Whirring, crackling, buzzing, humming, droning* and *whining* sounds are characteristic of motors and generators. When motors start, the pitch of the whine increases until it reaches a steady value. This is especially true of commutator type motors. Repulsion starting single-phase induction motors may have a sputtering, whirring, crackling, buzzing or humming sound. When such sounds are heard, look for such electrically operated equipment as:

<table>
<thead>
<tr>
<th>Adding Machines</th>
<th>Farm Lighting Plants</th>
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<tbody>
<tr>
<td>Air Conditioning Units</td>
<td>Floor Polishers</td>
</tr>
<tr>
<td>Automatic Towel Rollers</td>
<td>Generators</td>
</tr>
<tr>
<td>Barber Clippers</td>
<td>Hair Dryers</td>
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<tr>
<td>Beauty Parlor Devices</td>
<td>Humidifiers</td>
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<tr>
<td>Billing Machines</td>
<td>Massage Machines</td>
</tr>
<tr>
<td>Cash Registers</td>
<td>Motor-Generators</td>
</tr>
<tr>
<td>Dental Engines</td>
<td>Portable Electric Drills</td>
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<tr>
<td>Dishwashers</td>
<td>Printing Presses</td>
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<tr>
<td>Dough Mixers</td>
<td>Sewing Machines</td>
</tr>
<tr>
<td>Drink Mixers</td>
<td>Shoe Dryers</td>
</tr>
<tr>
<td>Electric Addressing Machines</td>
<td>Small Blowers</td>
</tr>
<tr>
<td>Electric Computers</td>
<td>Telephone Magnetos</td>
</tr>
<tr>
<td>Electric Elevators</td>
<td>Toy Electric Trains</td>
</tr>
<tr>
<td>Electric Refrigerators</td>
<td>Vacuum Cleaners</td>
</tr>
<tr>
<td>Electric Vibrators</td>
<td>Valve Grinders</td>
</tr>
<tr>
<td>Fans</td>
<td>Washing Machines</td>
</tr>
</tbody>
</table>

*Rattles, buzzes* and *machine-gun fire* sounds indicate interference from buzzers, telephone dials or doorbells. These noises are usually intermittent, starting and stopping at irregular intervals. Short machine-gun firing sounds indicate telephone dialing interference. Look for such interfering devices as:

<table>
<thead>
<tr>
<th>Annunciators</th>
<th>Doorbells</th>
</tr>
</thead>
<tbody>
<tr>
<td>Automobile Ignition Systems</td>
<td>Elevator Controls</td>
</tr>
<tr>
<td>Buzzers</td>
<td>Sewing Machines</td>
</tr>
<tr>
<td>Dental Laboratory Motors</td>
<td>Switchboards</td>
</tr>
<tr>
<td>Dial Telephones</td>
<td>Vibrating Rectifiers</td>
</tr>
</tbody>
</table>
Violent heavy buzzing or rushing sounds are often heard over a large area or even a whole town, the sounds being at times so loud that they drown out the radio program. They may be louder at one end of the tuning scale of the receiver, indicating high frequency noise-modulated radiation; they may be heard only on certain bands of all-wave receivers. These sounds may be traced to:

- Air Purifiers
- Battery Chargers
- Diathermy Machines
- Doctors' Apparatus
- Flour Bleaching Machinery
- High Frequency Apparatus
- Insulation Testers
- Neon Signs
- Ozone Devices
- Spark Transmitters
- Spark Ignition in Oil Burners
- Violet Ray Apparatus
- X-Ray Machines

Crackling, sputtering, snapping, short buzzes or scraping sounds indicate loose connections; if in the house, they will be especially noticeable when walking about; if outside, heavy traffic or street cars may increase the intensity of the sounds. Look for:

- Defective Light Sockets
- Flimsy Elevator Controls
- High Tension Lines
- Power Lines Grounded to Trees
- Street Cars
- Wet Line Insulators
- Loose connections in floor lamps and appliance cords; broken heater elements in household appliances. Unbonded rubbing metal contacts in houses, such as adjacent water pipes.

Clicking sounds are a definite indication of some sort of make-and-break connection essential to electrically operated industrial equipment, such as:

- Elevator Controls
- Flashing Signs
- Heaters, Automatic
- Heating Pads
- Incubators
- Irons
- Mercury Arc Rectifiers
- Electric Typewriters
- Ovens
- Percolators
- Shaving Mug Heaters
- Soldering Irons
- Telegraph Relays
- Thermostats
- Traffic Signals
- Safe Time Clocks

Heavy violent buzzing sounds, usually of short duration, are characteristic of heavy sparking or arcing across a gap. Such sounds are traceable to:

- Arc Lights
- Automobile Ignition
- Breaks in Third Rails
- Electric Car Switches
- Electric Elevators
- Motion Picture Projectors
- Pole Changer (Telephone Interrupter)
- Street Car Switches
- Street Lights
- Toy Electric Trains
TRACING THE ORIGIN OF INTERFERENCE

After making a survey of an interference problem, the Radio-Trician is generally able to tell whether the interference noise heard is produced in the customer's house or outside the house; in an apartment building he can readily tell whether devices in the customer's apartment are at fault. When the source of noise can be quickly located, a simple filter will remedy the trouble at minimum expense to the customer.

When, however, locating the offending device involves a search through many apartments in a building or many houses in a neighborhood, by all means give the noise-reducing antenna first considera-

![Diagram of a receiver and dual-condenser filter](image)

Make this simple test with a condenser-choke filter of the plug-in type to determine whether interference is reaching the receiver directly over the power line. If interfering noises are still heard when the set is connected as shown, with the dual condenser line filter inserted in the power line, you have proved that the receiver itself is creating the noise; if no noise is heard but noises return with full intensity when the filter is removed, the interference is coming in over the power line. The remedy in this case is obvious: Install a line filter. The circuit diagram for the line filter recommended for this test is shown at the lower right.

tion, after you first try a line filter across the power leads of the receiver. Occasionally noise signals get in by this route.

Before actually installing a noise-reducing antenna, make sure that direct chassis pick-up of the noise signal is not involved, either by making the usual test with antenna disconnected and the receiver input terminals shorted, or by operating another receiver in the same location. Direct chassis pick-up is ordinarily encountered only in older types of receivers which have a number of unshielded parts.

Interference Originating in the Customer's Location. A quick test which will rule out the customer's location as the source of interference can be made with a portable battery receiver of the type which uses a loop or fish-pole antenna and no ground connection. The interfering noise should be heard on the battery receiver when it is placed in operation near the customer's receiver; if the noise is not heard, check the customer's antenna and ground system for poor joints and
exposed wires which are rubbing against a tree or building. Assuming that the interference noises are heard on the battery receiver, have some one open the main power switch which controls the entire electrical system in the house or apartment. If the noise disappears or is greatly reduced when this is done, at least one of the offending devices is in the place.

Locating the Noise-Producing Device in the Home. In small homes or in apartments this is easiest done by switching each of the electrical appliances off and on while the customer's receiver is in operation. In large homes this is done more quickly by having an assistant remove the fuses for each electrical circuit in the house in turn, while you note the effects on the customer's receiver. When the noise stops, you have isolated the defective device to one particular circuit; there remains only the checking of each part, device and connection in this circuit. The following procedure has proven very effective for isolating noise-producing sources:

1. Check the antenna, lead-in and ground for loose or poor connections.
2. Be sure that none of the service wires which enter the house are rubbing against the branches of trees or against the building.
3. Make certain that the service conduit containing the supply wires leading into the house is grounded.
4. The wiring in the house should be grounded as provided by the accepted local electrical code. Have a licensed electrician check this if there is any doubt in your mind.
5. Be sure that all switches in the distribution system make firm contact. All line fuses should be firmly in place, with clean contacts. No temporary fuses or fuse shorts should be allowed. Fuses should be checked, as a loose connection between the fusing material and the contact cap will create arcs.
6. Inspect all connections in switch boxes, distribution boxes and fuse boxes for looseness, tightening terminal screws where necessary.
7. Examine all lamp bulbs used in the house and make sure that they are firmly screwed into their sockets. Turn on each lamp and tap it on the side for loose elements and poor base connections. Question the socket.
8. Check all lamp extensions and attachment plugs to every appliance, looking for loose contact. Shake extension cords, listening to the radio for signs of poor internal connections while the device connected to the cord is turned on. Extension cords with knots and kinks, as well as worn cords, are prolific sources of interference.
9. Repair or replace snap switches which do not open quickly.
10. Water and gas pipes or electric conduit pipes rubbing against each other may discharge their electrostatic charges. Bond the pipes together at the rubbing joint or insulate the contact surfaces. Quite often the turning on of a water faucet, walking through the house, use of household appliances or the operation of oil burners or refrigerators will start such electrostatic interference. With experience you will be able to distinguish electrostatic noises from those produced by electrical apparatus.
In checking these items the receiver should be turned on, with your assistant or even the set owner listening to the receiver, while you check various things in the house. A broom handle may be used for probing or knocking against pipes; when the region surrounding the noise source is probed, noise will be clearly heard in the receiver.

*Interference Outside the Customer's Home.* When your tests show that the noise source is not in the customer's home, and the installation of a noiseless antenna proves inadequate, then the defective device must be isolated by means of an "interference locator." A portable receiver with self-contained batteries may be employed. The receiver should be sensitive, employing three to four R.F. pentode stages if a T.R.F. set; a portable superheterodyne may also be used. If the receiver is not already well shielded, it should be built into a heavy aluminum box. Inexpensive and sensitive portable battery receivers may be purchased from large radio mail order houses. In addition to the headphones used as an output indicator, a copper oxide rectifier type 0-5 volt voltmeter having a 1,000 ohm per volt sensitivity should be permanently connected to the output. Thus both aural and visual output indications are available. Whatever receiver is used, it must not have A.V.C., for this would tend to conceal changes in interference intensity.

The pick-up system may be a 7-foot collapsible aluminum pole or a loop antenna. In the latter case the antenna coupler in the receiver must be disconnected and the input condenser arranged to tune the loop. For an .00035 mfd. input variable condenser a box type loop containing 24 turns spaced 1/8 inch apart on a 20-inch square form will be needed to cover the broadcast band. Use No. 18 or 20 gauge D.C.C. wire. Both pole and loop antennas may be used by installing a D.P.D.T. change-over switch. The pole antenna is preferred where there are many overhead wires in the vicinity of the home; the loop antenna performs best in open spaces.

In locating a noise source, first listen to the noise signal on the portable receiver, with the tuning dial set to a frequency where broadcasts are not heard. Using the loop antenna, rotate the loop until a maximum output meter reading is obtained. The noise origin will be in the plane of the loop (along a horizontal line parallel to and passing through the top of the loop), but may be either ahead or behind the loop. Walk in the direction which gives increased output readings. Where overhead supply wires exist (we assume that the investigation is started outdoors, as everything in the house has been checked), the greatest noise signal will be evident when the loop is parallel to the overhead wires. This does not identify the source, however. Where
overhead wires do not exist, then the direction of interference may be identified from two positions about 200 feet apart and, by following the two directions to their apparent intersections, the approximate location is obtained.

With the pole antenna use the “hot-and-cold” method, walking in the direction which gives increased noise in the phones or an increased output meter reading. Where an overhead power line is involved, follow the line for maximum output. The loop antenna may also be used in the above “hot-and-cold” method. Always point the loop in the direction of greatest output and follow the direction of maximum output indication. Follow overhead lines with the loop parallel to the line.

If some indication of the direction of the interference is secured from the customer, increased output should be obtained when moving the interference locator toward the suspected point. For instance, if you are told that noise started when the neighbor installed a new refrigerator, walking to the neighbor’s home when the noise commences should show increased noise output.

The independent interference man must realize that in locating a fault he may have to trespass on private property. Where the trouble originates in a home or building, it should not be difficult to obtain permission, once he identifies himself. In case the trouble is traceable to power lines and line equipment, the power service superintendent should be informed; he will without doubt have his engineers cooperate in the matter and make the necessary corrections. Most power companies and public utilities have engineers who specialize in interference work. This text does not consider interference troubles peculiar to public utilities; where the trouble is traced to telephone equipment, street railway lines or other public service equipment, explain the situation to the customer and suggest that he notify the company in question.
Once the noise has been localized to some house or business establishment, first secure permission from the tenant or owner, then proceed to isolate the defective device in the same manner which you would use in the customer’s home. If the noise is traced to a point some distance away from the customer’s home, it is probably due to a device which draws considerable power; thermostat contacts, electric light switches, and electrostatic sources of interference can generally be ruled out in a case like this.

**COMMON INTERFERENCE CONDITIONS**

A study of a few common interference-producing conditions which may arise in various types of electrical equipment will help to clarify this important problem of interference elimination.

**Electric Motors and Generators.** Any electric motor, especially the D.C. and universal A.C. motors which employ commutators, should be suspected as a source of noise interference. Probable causes of trouble here include sparking at the commutators due to poor contact with the brushes, and dirty or uneven commutator segments. Sparking causes pitting and burning of the commutator segments, and the interference situation rapidly grows worse; before attempting to clean up the motor, connect the interference analyzer and determine whether a simple filter combination will completely eliminate the present interference. If the combination of filters required proves excessive in cost, repeat the analyzer test after you have remedied the sparking; a less-expensive filter should now prove sufficient. For motors try the filter combination shown at B in Fig. 2 first; if this is insufficient, add two choke coils as shown at G in Fig. 2, making sure that the coils used will carry continuously the full load current of the motor. For 110 volt motors figure 10 amperes per horsepower; estimate 5 amperes per horsepower for 220 volt motors.

No interference-remedying job on a motor can be considered complete unless the cause of the trouble is removed or at least rectified. The commutator should first be cleaned and made smooth with fine sandpaper, and the brushes then reshaped if necessary to fit the commutator better. It is common practice to smooth the commutator, where it is not too badly worn, by wrapping or tacking sandpaper to a flat block of wood and applying this while the motor or generator is revolving. Brushes can be reshaped with the motor or generator at rest; slip a piece of sandpaper under a brush, with the cutting surface facing the brush and the sandpaper pressed against the commutator. **Rock the commutator back and forth slowly until the brush takes its**
proper curvature. When you have finished, wipe off the brushes and the commutator carefully and apply a very small amount of vaseline over the surface of the commutator.

Oftentimes sparking at brushes can be reduced by shifting the positions of the brushes to get improved commutation. Rock the brushes slowly back and forth a very short distance until minimum sparking is observed; this should be done while the machine is operating at normal load if best results are to be obtained. With generators, moving brushes in the direction of rotation ordinarily reduces sparking; with motors the opposite holds true.

*Make-and-Break Contacts.* With simple make-and-break contacts such as are found in switches, temperature control thermostats, automatic electric irons, electric water heaters and similar devices, a filter consisting of a single condenser or a condenser in series with a resistor will usually prove sufficient to eliminate the interference. It is always a good plan to clean and adjust the contacts, in order to prevent a prolonged arc which would prove destructive to the contact points and cause even more severe interference than before.

*Oil Burners.* Interference produced by oil burners can usually be traced to the high tension ignition circuit, to automatic switching devices, or to the motor. Some burners use a gas pilot light, eliminating ignition systems as a possible source of interference; this you can easily confirm by inspection. If the interference noise is continuous for the period during which the burner is operating, the motor is clearly at fault; if the noise is heard only for a short period when the burner starts, the ignition system, one of the relay devices or the starting mechanism in the motor is at fault. Trouble at the motor can usually be eliminated by installing a filter as close as possible to the brushes.
Ignition system troubles are remedied by shielding all high tension wiring either with metal conduit, with flexible metal loom, or with metal braid, the shield being well grounded at each end in all cases. Some servicemen recommend that the frame of the oil burner be bonded to the boiler and to ground with heavy wire or metal braid, to prevent radiation. Try a coil and condenser type filter across the input leads of the ignition transformer; try simple condenser filters across thermostat contacts and relay contacts. Oftentimes it is necessary to place a wire shield around the ignition electrode in gun-feed type oil burners and ground this shield to prevent ultra high frequency radiation.

Here are a few practical suggestions concerning oil burners. If the noise elimination job on a burner appears at first inspection to be a rather involved affair, it is well worth while to contact the local distributor of that burner. Similar interference conditions will have been encountered in other installations, and often the distributor can make suggestions or supply special equipment which will remedy the trouble in short order. Once you prove that you can eliminate interference on that type of burner, the distributor may even refer similar jobs to you. Remember that all filters should be placed in metal housings to conform with underwriters’ regulations.

Electric Refrigerators. The motor is the usual source of trouble in electric refrigerators; its treatment has already been taken up. Static charges accumulating on the compressor-motor belt sometimes cause trouble; the remedy here is bonding the motor frame and the compressor frame either to the refrigerator frame, to some large metal mass in the unit or to ground. Refrigerator mechanisms are usually mounted on spring supports; occasionally you will find that a spring has weakened, allowing a make-and-break contact between the refrigerator frame and the part in question; in this case install a new spring. If the interference is traced to a sparking thermostat, it is wise to call in a refrigerator serviceman; adjustments on refrigeration control devices such as this require specialized knowledge.

Electro-medical Apparatus. X-ray machines, violet ray apparatus and diathermy machines can cause a great deal of annoying interference; these may prove the most stubborn cases which you will encounter. Most of the equipment now being marketed is designed to create a minimum of interference, but older models are trouble-makers. Modern vacuum tube type diathermy machines create interference at only one frequency in the short-wave region; this interference can be eliminated only by placing the machine in a screened room.
With medical apparatus in general, the first step involves placing a choke-condenser filter in the supply line to the device. If this is insufficient, the only recourse is to place the apparatus in a screened room. The frame of the room can be either of metal or wood; this is then covered with either iron or copper screening, preferably both inside and outside of the framing, and the screening is well bonded together at all joints. The door must be so constructed that it makes firm electrical contact with the remainder of the screen when closed. Filters should be placed on all power lines which enter or leave this screened room, for otherwise interference would be conducted outside and there radiated; the filters should be placed as close as possible to the exact points where the lines pass through the screen.

*Courtesy Tobe Deutschmann Corp.*

When electromedical apparatus is creating noise interference a grounding screen cage like this must often be used. All joints must either be soldered or continuously bonded in some way. Filter units must be attached to all power lines at the points where they enter the cage. All devices and filter units inside the cage should be grounded to the screen; connect the screen to a nearby ground if this gives an additional reduction in interference.

*Flashing Signs, Traffic Lights and Neon Lights.* In general, interference from these three sources can be spotted by visual inspection and by studying the nature of the sound heard in the receiver. For example, if a steady choppy noise is heard in synchronism with the flashes of a yellow blinker light up the street, the defect is immediately isolated. If a steady rolling or clicking sound is heard, and you note in the vicinity a sign having a continuous change of light, perhaps around the border, that sign is very likely the offender. Whenever there is some question as to the source of trouble, use the portable receiver to localize the trouble. The next step is a study of the device in question to determine the simplest filtering procedure.

Simple flashing signs which have a single make-and-break flasher require only a filter condenser connected directly across the contacts; the closer the condenser is to the contacts, the more effective it will be. Motor-driven contactors are generally used in signs which create the effect of motion; the first step with these is to filter the motor supply
leads, then the main supply leads to the electric lights. If this fails, it is then necessary to connect a filter to each contact on the contactor. The condenser should be connected from the contact to the common terminal for all circuits, which is ordinarily easily located. In severe cases of interference it is necessary to place a choke coil in each lead to the lights, the condenser being connected from the contact side of the coil to the common power lead. Short connections are essential here to prevent high frequency radiation.

Flashing traffic lights are treated in much the same manner, using condenser filters across the contacts and line filters where necessary. This work must naturally be done under the supervision of the proper authorities.

Moving belts and belt conveyers are sources of static discharges, not only creating noise interference but actually endangering the lives of nearby persons and the insulation in the machinery. A grounded metal comb with flat or coil springs rubbing on the belt will discharge this static electricity harmlessly to ground and at the same time stop the radio interference. Use a good ground which is carefully erected. In any industry where static electricity is produced, all fixed and movable parts of machinery should be grounded.

Neon signs of the non-flashing type do not as a rule cause interference troubles; where interference is positively traced to these signs, the only remedy is the installation in the primary leads of the high tension transformer of a condenser-coil filter of the type shown at G and H in Fig. 2. When a rotating contactor is used in the primary circuit of a number of neon transformers to switch from one group of tubes to another, filters must be connected to each contact and to the motor of the contactor. Where the rotating contactor is in the secondary circuit, switching high tension currents from one to another of a number of small sections of neon tubing, condenser filters are out of the question because of the high voltages involved. Try inserting 10,000 to 25,000 ohm spark suppressor resistors in each high tension lead; choke coils inserted in these leads may also reduce the interference. In certain severe cases the only remedy may be a complete change-over of the sign-operating mechanism, which will place
the rotating contactor in the primary circuit and provide a separate transformer for each section of tubing; such an arrangement is more readily treated for noise suppression, but the cost of making the change-over is generally so high that the job of filtering is given up.

Quite often neon tubing will accumulate an electrostatic charge which leaks off to the nearest metal objects or at points of support; try placing mica sheets at these points. The two chains which sometimes support neon signs in show windows often acquire a difference in potential; insulating each chain from the neon tubing or using a non-metallic type of support will effect a remedy. Neon signs should be kept as far away from glass windows as possible, to prevent ac-

![Examples of typical commercial filter units for interference-creating electrical appliances.](image)

Examples of typical commercial filter units for interference-creating electrical appliances. At A is a single condenser unit which may be slipped over the prongs of the appliance plug; B is a similar unit, but of larger capacity, for insertion between appliance plug and wall outlet; C is a dual condenser filter with a midpoint terminal which can be grounded; D contains a condenser and coil combination designed for use with larger appliances. These devices are generally carried by those servicemen who do not make a specialty of interference elimination; by trying each device in turn, they can generally find one which will give satisfactory noise reduction where there is only mild interference. Never connect condensers larger than 1 mfd. directly across an A.C. line for filtering purposes; the power losses in larger condenser units are often high enough to cause excessive heating on continuous duty, resulting in failure of the condenser.

accumulations of static charges on the glass. Quite often a general overhauling of the neon sign, done by a sign expert, will greatly reduce the interference and make ordinary filtering procedures effective. This involves cleaning of all insulators and all tubing, to prevent high tension currents from leaking over dust-covered glass surfaces.

Thus you can see that the elimination of interference, once the source has been located, calls for "horse sense" and a certain amount of "trial and error" work, as well as a knowledge of the causes of interference and the technique of filtering.

**Radio Noise Survey.** Although the results of any survey made of causes of radio interference noises will vary with the locality, the following data taken from one such survey gives a general indication of the frequency with which various noise complaints occur. Out of
9,000 complaints, about 30 per cent were traced to power companies and public utilities, about 30 per cent to apparatus owned by the general public, about 15 per cent to defective radio sets and the remainder to transient or unlocateable conditions. Of the 30 per cent traced to devices owned by the public, motors and motor-operated devices accounted for 10 per cent, defects in wiring of building—6 per cent, switches and interrupter apparatus—5 per cent, electro-medical apparatus and neon signs—3 per cent, and miscellaneous—6 per cent.

SECURING INTERFERENCE ELIMINATION BUSINESS

Noise is as old as civilized man, but radio has made the public more noise-conscious today than ever before. Noise ruins the entertainment value of radio programs, changing the radio receiver from a luxury to a nuisance in the customer's mind. Interference elimination is a community affair that many towns and cities have passed ordinances which compel those people owning interference-creating devices either to eliminate the noise or to cease using the offending device; as the public demands better and better radio programs, laws become more widespread. With laws such as this in your favor, the securing of interference elimination business is comparatively simple, but even if you must first sell the idea of noise elimination to a customer, there are enticing profits awaiting you in this field. In addition, this side-line of servicing will bring more regular service jobs to your shop.

Always make inquiries about possible interference on each radio receiver service call which you make; bringing noise interference to the attention of the public and stressing the fact that practically all noise can be eliminated, will eventually produce many interference elimination jobs for you. If you plan to become a specialist in noise elimination, it is a wise plan to select a certain section of your town, preferably in the immediate neighborhood of your shop, for complete noise elimination. It may take weeks or months to locate and remedy all noise sources in this section, but once all noise has been eliminated, your reputation will spread throughout the town, and your work in other sections will prove more easy. Then, too, the experience gained will be of great value in solving similar problems elsewhere.

Be sure to contact the trouble-shooting department of your local power company, and the same department in any other nearby public utility. These firms constantly receive complaints of interference; once you have shown that you can handle this work, they will welcome you with "open arms" and even send jobs your way. Whenever
you encounter an interesting or particularly successful job, always call your local newspaper; anything with a little human interest makes a good story for the newspaper and gives profitable publicity to you.

Having selected a six or eight-block square section of your town as a starting point, the best approach is to announce that you are making a "radio interference survey." Visit the homes and business establishments in this section, preferably during your spare time,

![Diagram of electrical devices]

Examples of filter installations on small electric appliances which are creating interference because of sparking or arcing. A—vacuum cleaner motor interference can generally be cured by inserting a dual condenser filter in the connecting cord, not more than six inches away from the motor, and grounding the midpoint terminal to the frame of the appliance; arrow points to filter. B—interference created by the blower motor of a small hair dryer can often be satisfactorily reduced by placing a condenser filter of the plug-in type between the wall outlet and the hair dryer plug. C—plug-in type dual condenser filter inserted in sewing machine motor cord, as close as possible to motor, gives a neat interference-reducing installation where it is not feasible to make connections directly to motor terminals. D—condenser filter connected directly to terminals of a small mixer; this is not an ideal installation, for the filter interferes with the use of the appliance. E—plug-in type condenser filter inserted in cord of barber clippers, close to motor, gives satisfactory elimination of interference in most cases. Always try plug-in filters at wall outlet first, to avoid unnecessary cutting of appliance cord. Ground midpoint of filter to frame of appliance or nearby ground whenever possible.

explain what you are doing and ask if they have noticed any radio interference noises. Secure their permission, if possible, to turn on their radio receiver so you can listen for the noise yourself. By starting in a section where you are known, opposition to such a survey will be at a minimum. Keep your eyes open for regular service jobs while making the survey, and put in your bid for the job either at the time of the call or at a later date.

After each call, when making the survey, write your observations on a small card, perhaps of the three by five-inch size. With these cards arranged in geographical order, a study of them should show you where interference is a maximum; your first efforts should be
concentrated in this region. Secure permission to check on all suspected devices, and apply the interference-isolating technique which has already been explained.

If you hesitate to make a sales talk in each home in order to explain your purpose, send printed post-cards or letters explaining what you intend to do; this will tend to offset possible objections or the need for lengthy explanations when you make your call. A cartoon or drawing on the card or letter will attract attention to the purpose of your message and thereby give better results for you. Literature like this can also be used to explain why certain devices cause interference and why this interference should be eliminated at its source; this literature, by calling to the attention of customers man-made interference situations which they may not have recognized as such, will make it easier for you to sell filters and interference-elimination services at a later date.

**LINE FILTER CONSTRUCTION DATA**

*Line Filters for Radio Sets.* Get two .5 mfd. tubular paper condensers rated at 600 volts D.C. working voltage, one bakelite coil form about 6 inches long and 3 inches in diameter, and a half pound of ordinary No. 18 bell wire. Unwind the wire and cut into two pieces of equal length.

Drill two holes (each about ⅛ inch in diameter) at one end of the coil form, locating them about a half inch in from the edge of the form and about one inch apart. Anchor each wire by looping it once or twice through its hole, leaving about 6 inches projecting for connections. Proceed to wind the two wires side by side on the coil form in a single layer, with turns as close together as possible. When all but about 6 inches of the wire has been wound in this manner, drill two more holes about one inch apart and loop the ends through these holes for anchorage. This will give you two coils of approximately 35 turns each, wound on a single coil form.

Insert this filter choke in the radio set power cord, either at the wall plug or at the radio set. In other words, cut the two wires of the radio set cord at the desired location, connect one pair of cut wires to the leads at one end of the choke, and connect the other pair of cut wires to the two leads at the other end of the choke. Now connect one terminal of each .5 mfd. condenser to one of the leads at the receiver end of the choke coil; connect the remaining two condenser leads together and provide a means for grounding this common condenser connection (to a convenient water pipe or to the radio set ground if you know that to be good). Cover all exposed connections with friction tape. This completes the filter itself, but you will probably want to mount it in a wood or metal box so no dangerous 110 volt A.C. terminals will be exposed. The circuit of this filter is like that shown on page 16 (with the receiver connected and plugged into the outlet on the filter), or like that in Fig. 2G if S represents the receiver.

*General Filter Construction Hints.* The same general filter construction described above will serve for practically any line filter application if the wire used in winding the choke is the same size as the power cord wire used for the appliance being filtered. In other words, if you are filtering an electric
motor having No. 14 wire in its line cord, wind the choke with about 35 turns per coil (70 in all) of No. 14 insulated wire; No. 14 tinned solid copper push-back wire will do nicely, or you can use the same size of double cotton-covered wire and apply a coating of insulating varnish to the completed choke. To get this number of turns, you will have to order about 60 feet of wire in whatever size is required. Naturally you will need a longer coil form for heavier wire, since the choke must be in a single layer. The condenser size specified is all right for all cases; in general, the condensers should be connected to that end of the choke which will make the interfering signals go through the choke before they reach the condensers.

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TEST QUESTIONS

Be sure to number your Answer Sheet with the number appearing on the front cover underneath the title of this text.

Place your Student Number on every Answer Sheet.

Never hold up one set of lesson answers until you have another ready to send in. Send each lesson in by itself before you start on the next lesson. In this way we will be able to work together much more closely, you'll get more out of your Course, and you will receive the best possible lesson service.

1. What effect does a tone control, which cuts off the high audio frequencies, have on static noises?

2. In locating the best position for the straightaway portion of a noise-reducing antenna, what three rules would you follow?

3. What type of filter would you try if simple condenser filters using 1-mfd. units failed to give satisfactory noise reduction?

4. What should be the peak voltage rating of condensers used on ordinary 110 or 220 volt A.C. power lines for filtering purposes?

5. What are the probable causes of noise interference in D.C. motors?

6. What type of filter would you use on a make-and-break contact?

7. When interference-producing apparatus is located in a completely screened room or cage, where should the line filters (which are placed on all power lines entering the room) be placed?

8. When using a pole antenna with an interference-locating receiver, how can you tell when you are approaching the source of noise?

9. If interference noise traced to an oil burner is continuous for the period of operation of the burner, what is the cause?

10. What should be the current-carrying capacity of a choke coil which is to be used in filtering the power leads to a 220 volt, one horsepower motor?
HOW TO CHOOSE AND INSTALL REPLACEMENT PARTS

47RH-2

NATIONAL RADIO INSTITUTE
WASHINGTON, D. C.
ESTABLISHED 1914
STUDY SCHEDULE No. 47

For each study step, read the assigned pages first at your usual speed, then reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind. Study each other step in this same way.

☐ 1. Replacement Parts .............................................. Pages 1-2
   The kinds of parts, what to stock, and where to buy parts are covered here.

☐ 2. Power Transformers ............................................. Pages 2-9
   Practical information on how to determine whether a transformer is just overloaded or has been damaged. The requirements for replacing and a discussion of how to replace power transformers conclude this section.

☐ 3. Iron-Core Chokes and Audio Transformers .......... Pages 10-14
   It is a problem to get a replacement if a duplicate is not available. However, once you know the characteristics which must be considered, it is possible to choose a satisfactory replacement.

☐ 4. R.F. and I.F. Transformers ................................. Pages 14-18
   These coils may be replaced by exact duplicates or you can have duplicates wound for you by firms specializing in this service. Also, replacement primaries can be used in some cases.

☐ 5. Replacing Condensers ........................................ Pages 19-21
   Next to tubes, condensers require the most frequent replacement. However, a relatively small stock will serve for most cases, as is pointed out here.

☐ 6. Replacing Resistors ............................................. Pages 21-24
   This section gives information on replacing both fixed and variable resistors. Practical hints are given on how to make a small stock do for most jobs.

☐ 7. Replacing Loudspeakers ...................................... Pages 25-28
   Speakers may be repaired by replacing the cone or field coil, or the entire speaker may be replaced. The best practice to follow depends on the condition of the original, the ease of replacement, and the availability of replacements. There are firms specializing in speaker repairs—many servicemen use these services.

☐ 8. Answer Lesson Questions and Mail Your Answers to N.R.I.

☐ 9. Start studying the Next Lesson.

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Replacement Parts

WHILE the final steps in making a repair—removing the defective part and obtaining and installing a replacement—are purely mechanical, it is possible to waste a great deal of time in taking these steps unless you know what to buy, where to buy it, and how to install it. We will give you this important information in this lesson, along with a number of hints on testing parts. Let’s start by learning something about the kinds of radio parts which are usually available.

KINDS OF REPLACEMENT PARTS

Replacement parts fall into three groups: exact duplicate replacements; universal replacements; and general replacement parts.

Exact Duplicate Parts. These parts are exact duplicates of the originals, both physically and electrically.

Universal Parts. There are a number of universal radio parts so designed that, with minor physical or electrical alterations, they can be used as replacements for a wide variety of radio parts. For example, volume controls come with extra-long shafts. Once you have chosen a control of the proper electrical characteristics, you can make it fit the receiver by cutting off the shaft to the required length. Thus, the same control can be used in any receiver which its electrical characteristics will fit.

As another example, output transformers come with tapped secondaries; by choosing the proper taps, you can match practically any loudspeaker to almost any output tube (or tubes).

General Replacement Parts. Finally, we have parts, such as tubes, resistors, and condensers, which can be used in any receiver as long as the proper electrical characteristics are chosen and as long as there is sufficient room for the parts.

We include, among these, parts not designed for the particular radio, but which can be used by making some slight change in the original circuit to “fit” the new part characteristics. Changes of this kind are rare, as the widespread distribution of exact duplicate and universal replacement parts generally makes it possible to make a direct replacement.

STOCKING RADIO PARTS

You can start a radio service business with a surprisingly small stock of parts. However, you will want to build up your stock gradually, both so you can cut down the number of trips or orders to the parts suppliers and so you can render the fastest possible radio service.

When you start in business, you will need a kit of resistors, a small number of electrolytic, paper, and mica condensers, a stock of tubes, an assortment of pilot lights, and a certain amount of hook-up wire and hardware. With this small stock as a beginning, you can increase gradually the amount and variety of these parts. Also, you can add items like universal output transformers, a volume control kit, i.f. transformers, tube sockets, dial cords and belts, and an assortment of knobs.

Some servicemen make the mistake of acquiring too large a stock. It is not wise to invest much money in slow-moving parts. Increase the quantity
and variety of your stock only as your service experience indicates the need for such expansion. At the beginning, ask your local distributor to help you choose parts which, according to his sales records, move rapidly in your area. This is particularly important in the case of tubes. There are about a thousand different types of radio tubes, yet perhaps in your district only seventy-five to one hundred types are widely used.

WHERE TO BUY RADIO PARTS

There are many sources of supply available to the serviceman. Perhaps the best known are the large mail-order radio parts suppliers, who carry very complete stocks of parts and who can usually obtain any special parts you may need. In large cities there are also radio parts supply houses and distributors who carry a wide selection of radio parts.

In addition, there are distributors scattered throughout the country who handle various popular makes of radio receivers. Exact duplicate parts for these receivers can be obtained through these distributors. Where there are no distributors, parts can sometimes be obtained directly from the factory. Also, many parts manufacturers (condenser and resistor manufacturers, etc.) deal directly with servicemen, although in recent years, mail-order and local parts supply houses have acted as distributors for these lines.

Collecting Service Data. All servicemen collect wholesale parts catalogs, both to locate sources of supply and to obtain information on the electrical and physical characteristics of different parts. Be sure to collect all the volume control guide booklets, vibrator replacement guides, transformer replacement guides, tube charts, and other service data which are available from your local distributors or supply house. Many of these are free, while others are sold for just a few cents.

While we are on the subject of collecting information—try to get all possible information on radio receivers themselves. You will find that your parts distributor will help you obtain service manuals.

Many set manufacturers publish their own manuals, which are kept up to date by supplements or come out in yearly editions. You may find it desirable to get those covering any particular brands of receivers which predominate in your locality.

Let us turn now to certain specific radio parts and learn more about the problems of obtaining the proper replacement and installing it quickly. We shall deal chiefly with transformers, condensers, resistors, and loudspeakers, as other parts—line cord resistors, tubes, batteries, etc.—are replaced most generally by exact duplicates.

Power Transformers

There are two ways in which a transformer can fail: 1, a winding may open; or 2, a short circuit may develop.

The first is rare, as electrolysis seldom occurs in a sealed transformer, and an open is seldom caused by an overload. If an open occurs, the fact is obvious, since one of the secondaries will not deliver voltage. A continuity check with an ohmmeter will lead you to the defective winding.

Watch particularly for an open center tap. You may still have continuity across the entire winding, so make a
check to the center tap from each end of the winding.

A short circuit is the usual transformer defect, and it is invariably caused by excessive heat. If too much current is drawn from a transformer winding—that is, if the transformer is overloaded—so much heat will be produced in it that the paper insulation between the layers of wire in the winding will char (carbonize). Since carbon is a fairly good conductor, the winding, or part of it, will be short-circuited. Once an internal short has developed, the transformer must be rebuilt or discarded. Under normal conditions, it is not economical to rebuild, so a replacement would be installed.

CHECKING FOR AN OVERLOAD

Your nose will first discover a short circuit or an overload. The pungent odor of burning enamel and paper insulation is unmistakable. Smoke may come from the transformer, and perhaps some tar or wax sealing compound will boil out of it.

These symptoms indicate that the transformer is overheated, but not necessarily that it is damaged. Simple tests will show where the trouble lies.

When you find an overheated or smoking transformer, turn off the set and remove all the tubes, including the rectifier. Now turn the set back on and wait to see if the transformer cools off.

If the transformer does cool and stop smoking, it has been overloaded but is probably not seriously damaged. The overload was probably caused by a B supply defect, the effects of which were stopped by removing the rectifier tube. Repairing the defect will usually restore normal operation.

On the other hand, if the transformer continues to overheat with the tubes out, it is being overloaded by shorted secondary leads or by a filament circuit short, or else it has been damaged by: 1, a B supply defect; 2, operation on the wrong power line frequency; or 3, by an internal transformer defect.

If you live in a district with 25-cycle power, check the receiver label to see if the transformer is meant for 60-cycle operation. Such a transformer will draw too much primary current from a 25-cycle line, and eventually will be ruined. The only cure is to replace it with a transformer designed to operate on 25-cycle power.

**Shorted Leads.** Next, remove the chassis from the cabinet. Then, with the tubes out, turn the set on and examine the bottom of the chassis for signs of arcing between the transformer secondary leads. Push the wires around with an insulated probe or stick. If you see or hear an arc, the insulation probably has become frayed on the wires. Tape or replace the leads in order to cure this.

Also, examine the rectifier tube socket. An arc may have occurred between a plate terminal and the chassis, between the two plate socket terminals, or from one of the rectifier filament socket terminals to the chassis. Often the arc can be seen or its charred path can be observed on the bottom of the socket.

If the rectifier has a wafer socket, the arc path may be between the two wafers of the socket and so may be invisible. If the transformer continues to overheat, disconnect the leads going to the rectifier socket to see whether this removes the overload.

Replace any socket which shows evidence of arcing, as the carbonized path is sure to give further trouble.

If there are no apparent shorts but the transformer continues to overheat, it is probably damaged.

**A Transformer Short Checker.** If you have taken the set to your shop, you can test the transformer with the
simple checker shown in Fig. 1. This device consists of a 60-watt lamp bulb in a socket wired in series with a power cord and an outlet. To use it, remove the tubes and pilot lamps from the radio and plug the set into the outlet. The lamp then will be in series with the primary of the power transformer and will indicate the amount of primary current.

Under these conditions, if the set is normal there will be so little drain on the power transformer that the lamp will barely glow—if it lights at all. If the lamp burns brightly, however, there is a short circuit between the high-voltage wires or in the rectifier socket, or else the transformer itself is defective. Examine the high-voltage wiring for shorts, then disconnect the leads going to the rectifier tube socket to see if the lamp glow decreases. If it does, the socket is defective. If not, the transformer itself is defective.

**Circuit Defects.** Whether you have an overheated transformer or a damaged one, be sure to clear up any overload conditions existing in the radio. Otherwise, the transformer (or its replacement) is certain to be damaged.

- Short circuits in the filament circuits are rare, as the low voltage is not likely to cause insulation breakdowns and there are usually no condensers in these circuits. One possibility of a short circuit is a grounded pilot light socket in a receiver which has a grounded center tap on the filament winding. This shorts half the filament winding. (However, many modern receivers use the chassis as one side of the filament circuit, so the pilot light is deliberately grounded to complete the circuit. Don't confuse this intentional ground with a short circuit.)

- On the other hand, the high voltages in the B supply cause frequent breakdowns, particularly of by-pass and filter condensers.

To check the B supply circuit, first test all the tubes, looking especially for shorted elements in the rectifier and power tubes. Then, with the set turned off, check the resistance from the cathode terminal of the rectifier socket to the chassis with an ohmmeter. Be sure to observe polarity. The positive ohmmeter terminals must go to the B+ side of the circuit. (When in doubt, reverse the leads after taking a reading. Then, the polarity permitting the higher resistance reading is the correct one.)

The receiver diagram will show what the resistance should be. Usually, the resistance between B+ and B— should be only the leakage of the electrolytic filter condensers (over 100,000 ohms).

![FIG. 1. A simple lamp device, used to check for shorted power transformer windings.](image)

(Some receivers have bleeder resistors which reduce the reading to some value between 5,000 and 25,000 ohms.) If the reading is less than that which is expected, look for a short. Such a short could overload the transformer and also could be the cause of a shorted rectifier tube. The electrolytic filter condensers, the most likely sources of trouble, should be checked first.

**CHOOSING THE REPLACEMENT**

When you have determined definitely that the transformer is defective and have cleared up any overload conditions, the next step is to make the replacement—preferably with an exact duplicate transformer. Such a transformer may be obtained from the distributor of the receiver, from the manufacturer, from a wholesale mail-order house, or from one of the large trans-
former manufacturers who specialize in exact duplicate replacements. When you order, give the following information:

1. The make of the receiver.
2. The model number of the receiver.
3. A complete list of the tubes used in the receiver.

If your customer does not wish to wait while you send for an exact duplicate, or if none is available, you must choose a universal or general-purpose replacement transformer which has physical and electrical characteristics similar to the original.

**ELECTRICAL REQUIREMENTS**

To choose a suitable universal replacement transformer, you need to know the ratings of the windings on the old transformer. You can usually learn this from the service information on the receiver, from your parts distributor, or from catalogs of transformer manufacturers. They list receivers by make and model number and recommend as replacements specific transformers of their line. (If you cannot obtain the recommended transformer, its characteristics, given in the manufacturer’s listing, at least will give you the information you want.)

Check these points to see that the proper transformer is obtained:

1. **Primary.** The transformer must be designed for the power line voltage and frequency. The frequency rating is usually given in the data on the primary winding.

   The original transformer may have had taps on the primary to adjust it for a power line voltage range of, say, 100 to 125 volts. Most universal replacements will not have these taps; if not, wire the replacement primary directly to the power line terminals.

2. **Wattage Rating.** If the proper voltages and currents are delivered, you need not worry about the wattage of a transformer. Just ascertain that each winding is properly rated for the load it must carry.

3. **Filament Windings.** There will be from one to four filament windings, each with a voltage and current rating. The voltage rating depends on the types of tubes to be connected to the winding, and the current rating depends on the total current drawn by them.

   Both the voltage and current demand for a winding can be found by determining which tubes are connected to it, whether they are in series or parallel, and (from a chart) what the requirements are for each tube. When tube filaments are in parallel, the filament winding must supply the voltage required for any one of the parallel tubes, while the current will be the sum of all of the current ratings of that group of tubes. When tube filaments are in series, the voltage required is the sum of all the voltage ratings (plus any series resistance drops), while the current rating will be that of a single tube in the group.

   Of course, the current rating of the winding can be any amount equal to or above the current drawn by the tubes—this rating just gives the maximum current the winding can deliver without overheating.

▶ Most universal replacement transformers come with center taps on some of the filament windings. If the original transformer has no corresponding center tap, just cut this tap off or wrap the end with tape.

▶ Some very old receivers used a center tap on the rectifier tube filament winding as the B+ connection. Generally, replacement transformers will not have such a center tap, but you can make the B+ connection to either side of the rectifier tube filament circuit.

▶ It is perfectly all right to use a transformer having extra filament windings.
Just tape the extra leads or ignore the terminals on the transformer.

4. The High-Voltage Secondary. The high-voltage secondary must furnish sufficient voltage to give the proper B and C voltages, and must have a current rating equal to or greater than the amount drawn by the tubes and any bleeder.

You'll have to be careful in figuring the proper voltage rating for this winding. If the voltage is too high, it may damage the filter and by-pass condensers and introduce excessive regeneration, while a voltage below normal may lower the sensitivity and output of the receiver.

Many universal replacement transformers are designed for average receiver conditions, and you need know only the number and types of tubes to get approximately the right transformer. For example, you can buy a transformer designed for a 5- or 6-tube receiver and the rating of the high-voltage secondary usually will be close to the requirements for the receiver.

It is better, though, to compute the rating from the normal operating voltages for the tubes used. Radios differ somewhat in their actual applied voltages, but a tube chart will show you the probable voltages used. Any set with 71A, 6C6, or 6A4 output tubes will need plate voltages of about 180 volts. If the output tubes are 42, 6F6, 6V6, or 6L6, the voltage may be 250 volts, but can be higher (depending upon whether the output is class A, AB, or B). Certain special class B tubes take voltages up to 400 volts. Most other common power tubes take 250 volts as the plate supply.

When the output tubes are triodes, the C bias voltage will be 40 or 50 volts and should be added to the plate voltage supply. You can ignore the bias requirements for pentode and beam power tubes.

If the speaker field is used as a choke in the B supply circuit, allow about 100 volts as the drop across it. Adding this value to the plate voltage requirement gives the d.c. voltage needed at the filter input, which is approximately equal to the a.c. peak voltage when a condenser input filter is used. Since transformers are rated in r.m.s. values, multiply this filter input voltage by .7 to get the approximate r.m.s. rating necessary to give this peak value. Then, add about 50 volts to this r.m.s. value to approximate the rectifier and transformer secondary drops. The total will be the r.m.s. rating for one-half the high-voltage winding.

For example, if we need 250 volts for the plate supply and 100 volts for the field, we need 350 volts d.c. The r.m.s. voltage needed is $350 \times .7$, or about 245 volts. Adding 50 volts to this gives a rating of about 300 volts for one-half the high-voltage winding, or 600 volts for the entire winding. This is a common rating for average size receivers.

The current requirement for the high-voltage winding can be found by adding the plate and screen grid currents of all the tubes except the rectifier. If a bleeder resistor is used, add about 20 milliamperes to this value. The total will be near the proper rating for the high-voltage winding. To be safe, you can choose a transformer with a current rating higher than this value if one is available.

MECHANICAL REQUIREMENTS

Several typical transformer mountings are shown in Fig. 2. The type shown at A is mounted above the chassis, with the leads going down through chassis holes. The important dimensions are the height (if the cabinet is small) and the mounting centers. (By mounting centers we mean the distance between the centers of the holes through which pass the bolts holding the transformer case to the
chassis.) This measurement may be made with a pair of calipers by spreading them until their points reach to the centers of each pair of holes, then checking the spread on a ruler.

The types shown in B and D mount through a large hole so part is above and part below the chassis. The dimensions of the cut-out area on the chassis are important, as well as the mounting centers. When the replacement listing does not give the “window” size needed, check the mounting centers and core sizes. If they are similar, then the winding dimensions will probably be similar.

Figs. 2A and 2C are two views of an above-chassis type with universal mounting brackets. These can be put on any of the transformer bolts in such a way that many mounting center distances can be accommodated.

**TRANSFORMER INSTALLATION**

Don’t remove the defective transformer until you’ve obtained the replacement. This will make it much easier to identify the connection.

When you are ready to take out the defective unit, make a sketch of the transformer, put the new one in place, and make the proper connections. If the new transformer has leads, they will be colored the same as those of the original, and you can easily find the proper connections from the leads you left in the radio. Remove identifying pieces of wire as you solder the new leads in place.

If the replacement has lugs, the lug positions will be the same as those of the original, and your sketch showing the colors of the wires connected to each lug will aid in your making the proper connections.

**Universal Replacements.** If the re-
will identify its terminals, and your sketch of the original transformer connections will show the proper connecting points. Any extra terminals can be ignored. Extra leads, such as unused center tap connections or extra filament windings, can be insulated with tape and tucked out of the way.

**IDENTIFYING LEADS**

If you have no lead identification slip for your transformer, you may be able to identify the leads from the standard R.M.A. color code for power transformers shown in Fig. 3—especially if it is a universal type made within recent years. However, there are many variations in the color codes used, particularly in transformers used in earlier receivers.

- If the color code is not helpful, you can identify the windings of an unmarked transformer with an ohmmeter and a simple lamp testing device.

  First, use the ohmmeter to discover which leads show continuity to each other; these leads go to the same winding. Next, put a 60-watt light bulb in series with a power cord and test leads, as shown in Fig. 4. Separate the transformer leads so that their ends do not touch, then touch the test leads to each pair of wires which show continuity. When you put the test leads across filament windings, you will have a full, bright light; across the high-voltage winding, no light; and across the primary, a faint glow.
Once you have identified the primary, connect it to the 110-volt a.c. line and measure the voltages developed by the secondaries. This will identify each winding. Since you know from the lamp test which is the high-voltage winding, you need not measure its voltage unless you want to know what it is.

Remember that the voltages produced by a transformer with no load connected to it may be somewhat higher than the rated voltages. Thus, a 6.3-volt filament winding may produce 7 to 7.5 volts with no load, while a 5-volt filament winding may produce 5.5 to 6 volts.

When the resistance values are given on a diagram, as in Fig. 5, you can identify the windings with an ohmmeter. Notice that the transformer shown in Fig. 5 has a tapped filament winding. (A check of the circuit diagram of the receiver in which it was used shows that the tube filaments operate from the 6.3 volts produced by the .074-ohm winding, while a special tuning circuit indicator uses the total voltage produced by this secondary.)

Also notice that the color code is not the standard R.M.A. code.

Another example is given in Fig. 6. Here, two of the filament windings have about the same resistance, so the ohmmeter reading only identifies them as filament windings. If you did not have a wiring diagram to show the connections or voltages, you would have to measure the voltages to identify these windings.

AUTO-SET TRANSFORMERS

As the transformer, vibrator, and buffer condenser of an auto radio are usually designed to "work together," and since the transformer is usually in a shielded compartment of limited size, it is best to use an exact duplicate replacement. However, universal types are available. The auto-set transformer has only one secondary, and its voltage and current ratings are the only ones to consider. Follow the same rules as for the high-voltage winding of a power line transformer. Remember that the plate voltages range from 180 to 250 volts, and that the speaker field is never used as a choke.
Iron-Core Chokes and Audio Transformers

FILTERチョックス

A properly moisture-proofed filter choke will rarely open or otherwise become defective unless subjected to a severe overload, such as one caused by shorted or leaky filter or by-pass condensers.

If possible, order an exact duplicate for ease in mounting. Simply ask for a filter choke for the make and model number of the receiver on which you are working. If you cannot get an exact duplicate, order one with about the same physical dimensions and mounting centers. If the original choke was shielded the replacement should also be shielded.

The resistance of a power supply filter choke is not important unless it is in the negative side of the circuit and the voltage drop across it is used for biasing. In this latter case, the resistance of the replacement should be approximately the same as that of the original part (which you may find on the wiring diagram).

Since high-capacity electrolytic condensers are now used universally in filter systems, an inductance of 10 henrys (or more) is satisfactory. Remember, however, that the choke inductance is figured for a particular d.c. current, and will be lowered if the current rating is exceeded. Thus, if a 60-ma., 10-henry choke is used in a 100-ma. circuit, the inductance may drop to 2 or 3 henrys, and the choke may overheat. You must use a replacement with a current rating equal to or somewhat higher than the actual current flow in the circuit to obtain a proper inductance value.

You can compute the current roughly by adding the normal cathode currents of all tubes (get these from a tube chart) with the exception of the rectifier. Add on 20 ma. if a bleeder system is used. Receivers using a power transformer and about six tubes will require a 60- or 70-ma. choke. Larger receivers will need a choke rated at 100 ma. or more.

In general, satisfactory chokes for a.c.-d.c. sets are obtained just by asking for a choke to use in an a.c.-d.c. set.

FIG. 7. A high-inductance choke, used as a plate load.

AUDIOチョックス

High-inductance chokes are sometimes found in impedance-coupled a.f. amplifiers and in stages where a coupling transformer or load device is isolated as in Fig. 7. These chokes must have high inductance to pass on low audio frequencies and, like other chokes, the amount of inductance will depend on the d.c. current flow.

If an exact duplicate is not available, use a choke intended to operate in the plate circuit of the particular tube used in the stage, or choose one which has a current rating above the normal d.c. value of that tube.

The higher the inductance of your choke, the better the low-frequency response will be. However, remember that a high inductance means many turns, and the distributed capacity may reduce high-frequency response.

If the original choke was shielded,
the replacement should be also (to minimize hum pick-up).

INTERSTAGE AUDIO TRANSFORMER REPLACEMENTS

An interstage a.f. transformer is one which is used to couple two audio stages together. The windings may open, short, or become noisy. In such cases it is best to use an exact duplicate, for then the response of the receiver will be unchanged and mounting difficulties will be eliminated. When ordering, state the make and model number of the receiver and the position of the transformer in the circuit (first a.f. or second a.f. transformer).

If a duplicate is not available, use an a.f. transformer with a step-up turns ratio of 3:1. The receiver tone quality and hum level may be affected by the change in transformer characteristics, but may even be improved if you use a good quality replacement. As a practical hint—don't deliberately try to change the tone quality unless the receiver owner indicates a desire for a change. He may like the tone quality and be dissatisfied with any change, no matter how much better it may sound to you.

To replace the defective transformer, first cut its leads close to the case (leaving the other ends of the leads connected in the chassis for identification), then remove it. The transformer may be held to the chassis by screws, bolts, or rivets, or by turned-over ears which project through holes in the chassis. Cut off rivet heads with side-cutting pliers, or drill out the rivets. Straighten turned-over ears with a heavy screwdriver.

Next, solder the leads from the new transformer into their proper places, removing each old lead when it has served its purpose as an identifier. The standard color code for a.f. transformers is shown in Fig. 8. Be sure that you follow any instructions accompanying the replacement.

Turn on the set and listen for excessive hum. Should the hum be abnormally high, see if you can rotate the new transformer to a position where the hum disappears or is at a minimum. If you find such a position, bolt the transformer to the chassis there—if not, bolt it in the most convenient location.

Bear in mind that hum might be caused by other defects—isolate the hum to the new transformer, as described elsewhere in the Course, before you consider special mounting angles or positions.

Emergency Repairs. If a replacement is not readily available for an a.f. transformer with an open winding, you can make a temporary repair by changing to impedance coupling. Shunt the open winding with a resistor and connect a coupling condenser between the plate and grid leads of the transformer. Fig. 9 shows this arrangement for a transformer with an open primary. The resistor used across an open primary should have a value of from 50,000 ohms to 100,000 ohms, while a 250,000- to 500,000-ohm resistor can be used to shunt an open secondary. A condenser of from .01 to .05 mfd., rated at 600 volts, will be a satisfactory coupling condenser.

This is usually a temporary repair,
to be used only while you are waiting for a replacement transformer. It will change the tone quality, and it may decrease the volume so much that the set will be unsatisfactory. Be careful to cut the leads going to the defective section (a and b for the primary, or c and d for the secondary in Fig. 9) to prevent it from “coming alive” and causing noise.

If the transformer is noisy, cut both the primary and secondary out of the circuit and use resistors in place of both windings. This, with the coupling condenser, gives ordinary resistance-capacitance coupling.

**REPLACING INPUT PUSH-PULL TRANSFORMERS**

Input push-pull transformers have the same troubles as interstage types—shorts, opens, and noise. If an exact duplicate is not available, you must first determine the class of operation of the output stage. Sometimes, looking up the output tubes in a tube chart will tell you this. Several tubes—type 46’s, for example—are used only in class B amplifiers.

If the output tubes are triodes and are not class B types, you are usually safe in assuming that the stage is a class A amplifier. However, if the output tubes are pentodes or beam-power tubes, such as types 42, 6F6, 6V6, or 6L6, they may be operated as class A, class AB, or class B.

You can sometimes tell the type of operation from the operating voltages. Sometimes, also, class B stages are driven by a power tube. For example, a pair of 42 tubes operated from a single 42 tube acting as the driver would probably mean class B operation.

If the secondary winding is not open, you can tell the transformer class by checking the secondary resistance. A class B input transformer will have very low secondary resistance, usually between 100 and 300 ohms. On the other hand, a class A input transformer may have a secondary resistance of from 1500 to 3000 ohms.

► Any high-grade input push-pull transformer which fits the space available will make a satisfactory replacement in a class A output stage. These transformers usually have ratings of 3 to 1.

On the other hand, a duplicate input transformer is necessary for a class AB or B stage. These transformers are designed to work from a particular driver tube into the grid circuit of particular class AB or B stages.

► Hum pickup caused by the transformer’s being too near the power transformer, speaker field, or filter choke may make it necessary occasionally to change the position of the replacement.

► Temporary repairs can be made if the primary of an input push-pull transformer opens. Simply shunt the open primary with a .5-watt resistor of between 50,000 and 100,000 ohms, and connect a .01- to .05-mfd., 600-volt condenser from the plate lead of the transformer to either (not to each) of the grid leads. The secondary will then act as an auto-transformer and will deliver equal signal voltages, 180° out of phase, to each push-pull grid.

**Phase Inverters.** If your customer is interested in improved tone quality
(at additional expense) and the output stage is class A, you may suggest using a phase inverter stage in place of the original input push-pull transformer.

The before-and-after circuits for making this conversion are shown in Figs. 10A and 10B. Since the phase inverter is self-balancing, no adjustments should be necessary. The phase inverter tube (a dual triode in a single envelope) must operate at the same filament potential as the original tube VT1.

OUTPUT TRANSFORMERS

The primaries of output transformers frequently burn out. While an exact duplicate replacement is preferable, universal output transformers will give very good results in ordinary class A output stages. These transformers are equipped with tapped primaries and secondaries, which make it possible to match either single or push-pull output tubes to any of the common voice coil impedances.

While complete instructions are packed with each transformer, the general procedure is to connect the primary first. For push-pull operation, the two outside leads go to the plates and the center tap to B+. For a single output tube, follow the instructions. In some cases, the center tap is not used, and one of the outside primary leads goes to B+ and the other to the plate of the tube. In other cases, half of the primary is used.

Then, solder the correct secondary leads to the points from which the original secondary leads were removed. If you know the voice coil impedance, the instructions will tell you which two of the secondary leads can be used. If you don’t, you can calculate the voice coil impedance roughly by measuring its d.c. resistance with a low-range ohmmeter. The impedance will be about 1.5 times the d.c. resistance. Modern speakers usually have voice coil impedances of 6 to 8 ohms, but others range from 2 to 15 ohms.

A slight mismatch is unimportant. However, a large mismatch will decrease the volume to some extent and will cause a noticeable change in response (the high notes will be either too weak or too strong). If the reproduction does not sound normal (with the receiver in its cabinet), try different taps, listening for best response.

If the set has pentode or beam-power output tubes, don’t disconnect the secondary leads while the set is turned on. Removing the load this way can damage the output tube.

Of course, turning off the set each time you try another secondary tap makes it hard to compare results, since by the time you’ve connected a tap, you’ll probably have forgotten how the set sounded when the previous tap was used. To avoid this difficulty, some servicemen use a test output transformer and “shorting” switch combinations like that shown in Fig. 11. This switch will not break contact with one point before making contact with another, so there will always be a load on the transformer. Clips are used to connect to the output tubes and to the
voice coil. The lead A also has a clip, and is fastened to one of the transformer terminals. The switch then can be rotated to various taps and responses can be compared rapidly. If optimum results are not obtained, lead "A" can be clipped to another terminal and a new combination tried. Thus, starting on 1, we can compare the outputs using terminals 1-2, 1-3, 1-4, 1-5, etc. Then, with A on 2, we can compare the outputs of 2-3, 2-4, 2-5, etc. (The set should be turned off while A is being moved, and the switch should not be set on the same terminal as A, to avoid removing the load.) The impedance of the taps which give the best output can be found on the chart furnished with the transformer. Now, knowing the proper impedance to use, you can install a transformer having this rating, or can find the proper terminals to use on another universal transformer.

A universal transformer should not be used in high-fidelity receivers, p.a. systems, high-power circuits (here the types of output tubes are a clue), or if class B operation is used. For such circuits, order a replacement (if you can't get a duplicate) by stating the type numbers of the output tubes, the model number and make of the loudspeaker, and its d.c. voice coil resistance.

In some receivers, the secondary of the output transformer is tapped to provide inverse feedback. Use an exact duplicate here if possible. Should howling occur when a duplicate is installed, the feedback is causing regeneration rather than degeneration; reverse the connections to the primary of the output transformer. This will reverse the phase of the voltage across the secondary.

If such a transformer must be replaced by a universal transformer, disconnect both ends of the lead which ran to the tap on the old transformer. If the lead is left connected in the circuit, even though it is not connected to the new transformer, it may provide a feedback path and so cause instability and howling.

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**R.F. and I.F. Transformers**

I.F. windings require more frequent replacement than r.f. coils, but their replacement is usually simpler. Both have opens, noise, and lowered Q as their troubles. Short circuits are not so common, but do occur in multi-layer windings. Let's run through the problems for both transformer types.

**R.F. Transformers**

In any receiver, the r.f. (tuning) coils must have equal inductance and distributed capacity values so that they will track when they are used with identical ganged condensers. This means the tuned secondaries must be held to close tolerance values, so most servicemen use exact duplicate replacement coils when anything is wrong with the tuning winding.

If a duplicate is not available, you can: 1, have one wound by a coil manu-
facturing company (your supply house will forward your order to one if you do not wish to order direct); 2, get a universal coil with an adjustable inductance; or 3, replace all the r.f. coils with a matched set. Let's consider these steps in order.

**Case 1.** In ordering coils, give the following information:

1. Name and model of radio.
2. Number of chassis and any chassis identification such as a series or code number.
3. Name of coil or of circuit in which it is used (antenna, 1st or 2nd r.f.).

This is usually enough information if the receiver is a popular brand, as coil winding firms know the specifications on these coils and can wind a duplicate. However, if the coil is removed from an “orphan” or unidentified chassis, send a schematic diagram of the section of the circuit in which it was used. If requested, send in the defective coil as a sample. (When ordering from a distance, it is a good idea to send the defective coil in as a model, whether requested or not.) If any lugs, leads, or mounting brackets are missing, make a note of their locations. If a shield is used, give its dimensions.

If the coil is one of an identical series of coils, such as are found in a t.r.f. set, a good coil can be sent in as a sample, but be sure to request that it be returned to you.

**Case 2.** The inductance of a universal coil can be varied over a wide range by an adjusting screw, so if its distributed capacity is not too far off, a universal coil can be made to track reasonably well. Fig. 12 shows a cutaway view of such a coil.

It's easy to adjust a universal antenna or r.f. coil in a t.r.f. receiver. Use a condenser of about 200 mmfd. (.0002 mfd.) as a dummy antenna; connect it between the hot side of a signal generator and the antenna connection of the receiver. Connect an output meter to the receiver, tune the signal generator and the receiver to 600 kc., and rotate the adjusting screw of the coil until you obtain maximum output. Next, tune the receiver and signal generator to 1400 kc. and align the circuits by adjusting the trimmers on the gang for maximum output. This may throw the adjustment at 600 kc. off somewhat, so retune the signal generator and receiver to this frequency and reset the screw for maximum output. Continued adjustment of the 600-kc. and 1400-kc. adjustments usually will give reasonable tracking over the band. However, if the responses are very unequal, or adjusting one throws the other completely off, the distributed capacity of the universal coil winding is widely different from that of the other coils; you should then use an exact duplicate coil or a matched set of coils.

▷ A slightly different procedure must be followed when you replace an antenna or r.f. coil on a superheterodyne,
since the 600-kc. and 1400-kc. receiver dial settings depend upon the receiver oscillator, rather than upon the coil being replaced. First install the new coil, then adjust the oscillator high-frequency trimmer and the low-frequency paddler, if one is used, so that the receiver tracks its dial. Next, adjust the new coil inductance at 600 kc. and its trimmer at 1400 kc. for maximum output. Repeat the low- and high-frequency adjustments for the new coil just as for a t.r.f. set. Do not adjust any of the other trimmers on the gang at this time—only the one for the new coil. Complete realignment can be made after the inductance of the replacement coil is satisfactory.

Case 3. When you have an a.c.-d.c. t.r.f. set using a single stage of r.f., in which either the antenna or r.f. coil is defective, simply order a matched antenna coil and an r.f. coil. Specify that they are to be used in an a.c.-d.c. midget. Get shielded coils if the originals were shielded. In most cases shields are not used, the antenna coil being mounted above the chassis and the r.f. coil under the chassis. The instructions which come with the coils will facilitate their installation, but of course you should identify each lead on the old coils before unsoldering them. Fig. 13 shows a typical pair of replacement coils.

Defective Primaries. The foregoing procedures are necessary if the secondary or tuned winding is defective. However, the secondaries seldom fail; it is the primaries, which carry appreciable amounts of plate current, that usually open up. The primaries of antenna coils (these are also called r.f. coils) are sometimes burned to a crisp by lightning or by a customer’s carelessly plugging the aerial and ground leads into a lower line outlet rather than an antenna outlet.

Most servicemen replace the entire coil if the primary is defective, particularly when a replacement is easy to obtain. However, the inductance of the primary is not critical, so it is possible to use a replacement primary winding which can be slipped on the coil form and wired in the circuit to replace the original.

Since r.f. coils vary in physical size you should have an assortment of these windings on hand if you intend to use them. Both low- and high-impedance types are available, but most modern radios use the high-impedance types.

The replacement primary is slipped over the secondary as shown in Fig. 14. Be sure to follow the manufacturer’s instructions carefully, for the position and direction of the winding are important. If it is necessary to disconnect the transformer to get the primary on, be sure to make a connection sketch.

**OSCILLATOR COILS**

An open primary is the usual defect of an oscillator coil. It is best to install an exact duplicate rather than a new primary, because the primary controls the amount of feedback.

If necessary, the coil can be repaired by one of the firms specializing in this business. If you send a coil off to be repaired, include a schematic diagram of the circuit in which it is used, give the make and model number of the receiver, the intermediate frequency, the type number of the oscillator tube, and state if the oscillator section of the con-

![FIG. 13. A cut-away view of a set of shielded, matched r.f. transformers. These are also available unshielded.](image)
denser gang has specially shaped plates.

Be certain that you draw and keep a diagram of the exact connections to the coil lugs—so you'll have no trouble in making the new installation.

Universal oscillator coils are available, but their installation and adjustment is quite a problem, particularly if the paddler adjustment has been disturbed. If you get a universal coil, be sure to follow the detailed instructions furnished by its manufacturer.

Multi-Band Coils. When the receiver has several wave bands and uses a tapped coil or multiple windings on the same form, either an exact duplicate or a rewinding job is necessary. Matched sets of coils with taps for the police band are available for a.c.-d.c. receivers, but these are almost the only exceptions.

1.F. Transformers

The i.f. transformer assembly includes the coils and their tuning condensers. Occasionally the trimmers will short or become leaky, or a high-resistance joint will develop. Generally, though, the trouble is an open coil, which you can repair by installing either a replacement coil or a complete transformer assembly. If a new transformer is available, use it in preference to new windings, since far less time will be spent in making the replacement.

Any large coil manufacturer can furnish either duplicate or satisfactory universal replacements. Sometimes, mounting holes for the new shield will have to be drilled in the chassis. When you order a replacement transformer, state the make and model number of the receiver, the i.f., and the position occupied by the transformer in the circuit. For example, if two i.f. transformers are used they are spoken of as the first (or input) i.f. transformer and as

the second (or output) i.f. transformer. If three i.f. transformers are used, the first is called the input i.f. transformer, the second is called the interstage i.f. transformer, and the third is called the output i.f. transformer. It is very important to get transformers which are designed to work with the number of i.f. stages used. Transformers designed for a single i.f. stage have very high gain, and if they are placed in a set using two stages of i.f. amplification, the amplifier will be unstable and may oscillate.

Try to get a replacement with the same physical dimensions—give the size of the transformer shield and the mounting centers.

The leads on the replacement transformer may have a different color code than the original transformer leads had. Replacement i.f. transformers use the color code shown in Fig. 15 unless otherwise specified.

The blue (plate) lead and the green (diode or control grid) lead should be kept as far as possible from each other, and away from other grid and plate leads. The position of the original leads is usually a reasonable guide, but if the new transformer has a higher gain than the original, it may be necessary to separate the leads more to prevent oscillation.

In general, the blue and green leads should be as short as possible. The
length and routing of the red (B+) lead is ordinarily unimportant. The length and route of the black lead is important only in output transformers; in them, this lead carries i.f. to the diode load and is quite “hot,” so should be kept short.

If you can’t get a replacement i.f. transformer which will fit on the chassis, install replacement coils—preferably exact duplicates. Again, your order must identify the receiver by make and model number and also identify the i.f. transformer. If an exact duplicate is not available, order a replacement designed to operate at the intermediate frequency of the receiver and equal or close to the original in physical dimensions. Typical replacements are shown in Fig. 16.

The spacing between the primary and secondary coils is important, but is usually factory-adjusted. Should you get a set of coils with adjustable spacing, however, measure the distance between the original coils before removing them, and use the same spacing on the replacements. This will give average satisfactory results.

An exact adjustment can be made by getting a response curve for the transformer with a c.r.o. and a frequency-modulated (wobbled) oscillator in the manner you learned earlier in your Course. If the coils are somewhat overcoupled the curve will be flat; if they are very much overcoupled it will be double-humped. If the coils are undercoupled the curve will be “low.” Overcoupling causes poor selectivity; undercoupling results in poor sensitivity. You should adjust the spacing of an ordinary i.f. transformer so that the response curve just starts to flatten at resonance. An adjustable band-expanding transformer should have no flattening of the response curve in the sharp or selective position, but in the broad or “fidelity” position the curve should have a flat top or even a double-

![Fig. 15. RMA color code for i.f. transformer leads.](image)

![Fig. 16. Typical replacement i.f. windings.](image)

hump. High-fidelity transformers (which are usually factory-adjusted) will be overcoupled and should have a broad flat-top or double-hump characteristic.

Once you’ve found the right spacing, cement the coils in place with coil dope.

Some i.f. transformers have three windings, for band-pass operation. Use a duplicate instead of a universal replacement for these.
Replacing Condensers

You may have to replace all kinds of condensers—even tuning condenser gangs. However, you will usually carry only an assortment of paper and electrolytic types, and perhaps a few fixed mica condensers in stock. Let’s take up condenser replacements according to type.

**PAPER CONDENSERS**

The most important ratings for any condenser are the capacity and the working voltage. The rating of the original part usually can be found from the schematic diagram or from the condenser label, but an exact duplicate replacement is seldom needed for a defective paper condenser. A wide variation in capacity is usually permissible.

If you don’t know the original capacity, use .01 mfd. to .1 mfd. for r.f. and i.f. by-passing, .25 mfd. to 1 mfd. for a.f. by-passing, .00025 mfd. for grid leak detectors, .002 mfd. to .05 mfd. for a.f. coupling condensers, and .001 mfd. to .05 mfd. for buffer condensers. This gives a clue to the sizes you should stock. A few each of the .01, .05, .1, .25, and .5-mfd. sizes will be adequate for practically all by-pass and audio coupling purposes.

A more important factor is the condenser working voltage rating, which should always be greater than the voltage across the terminals to which the condenser is connected. Many servicemen never use a paper condenser with less than a 600-volt rating (space permitting) even if the condenser is to be used in a low-voltage circuit. It costs only a few cents more and is excellent insurance against a call back. Buffer condensers in vibrator power supplies should be rated at 1600 volts or more. Filter condensers of the paper type (very rare today) should have a 600-volt to 1000-volt rating.

Sometimes one end of a tubular paper condenser will have a black ring on it and be marked “outside foil” or “ground.” The foil connected to the lead at this end of the condenser is the final outside layer and surrounds the rest of the condenser. If a condenser goes either directly or through a low-impedance path to ground, this ground connection should be made to the outside foil end of the condenser—the outside foil then acts as a grounded shield and prevents undesirable coupling between the condenser and other circuits. In most well-designed receivers, however, it won’t make any difference which end of a paper condenser is grounded. If the condenser is used for coupling (neither end grounded), ignore the outside foil marking.

**ELECTROLYTIC CONDENSERS**

Electrolytic condensers often prove puzzling to newcomers in the service business. When replacements are to be made, many questions about capacity, working voltage, and types come up.

Let’s consider capacity first. A replacement should not be much below the capacity of the original, but can be much higher. For example, a 10-mfd. output filter condenser should not be replaced by one smaller than 8 mfd., but a much larger condenser can be used and will give better filtering. However, do not replace an input filter condenser with one of more than twice the capacity of the original, for the peak current through the rectifier tube may increase to the point where the tube will be damaged. This is particularly true of a.c.-d.c. sets.

In replacing electrolytic by-pass condensers, never use a capacity lower than the original; a larger capacity will give better results. In replacing
condensers used across the filament strings of three-way receivers, stick to the original capacity if possible.

Here is a good rule to remember about working voltage. The working voltage of the replacement must be at least as high as the original; if it is higher there will be less chance that the new condenser will break down. If you are in doubt about the voltage applied to the condenser, check it with a d.c. voltmeter. When the set is first turned on, the voltage may be considerably higher than when the tubes start drawing current. It is this initial high voltage that the condenser must withstand. A working voltage of 150 volts is standard for filter condensers in a.c.-d.c. sets (voltage doublers use 250 volts), while 450 volts is standard for a.c. receivers. C bias by-pass condensers are usually rated at 25 or 50 volts.

Dry electrolytics usually—but not always—can be substituted for wet electrolytics. Remember the fundamental difference between the two. The dielectric of wet electrolytics can be broken down by an overload, but when the overload is reduced the dielectric film will reform. If dry electrolytics are overloaded for any length of time, their dielectric film breaks down permanently and the condenser must be discarded. In some sets using wet electrolytics, the initial starting surge breaks down the dielectric film each time the set is turned on. If you want to substitute drys, be sure to check this starting voltage. If it exceeds the working voltage of the condensers, either install wet electrolytics, or try a 50,000-ohm, 5- or 10-watt bleeder resistor across the output filter condenser. The resistor will draw current as soon as the rectifier tube starts passing current and usually will reduce the starting voltage to a safe level. Be sure to measure the voltage again after installing the bleeder, however, to be certain it does not lower operating voltages too much.

The type of can or container used for electrolytic condensers has nothing to do with replacements. For example, a condenser in an aluminum can may be replaced by a tubular paper type electrolytic with similar ratings. If there are a number of condensers in a case and only one is bad, you can connect a single-section replacement unit outside the case in the place of the defective section. However, it is best to replace them all, since the others will not last as long as the new one. Not only must the replacement contain the correct number of con-

![Fig. 17. Two styles of filter condenser blocks. These are NOT interchangeable, so be sure you get the proper replacement.](image)

densers, but also their leads must be arranged so that they can be properly wired into the circuit. As an example, look at Figs. 17A and 17B. Each condenser block contains the same condensers and each has three leads. Yet the blocks could not be interchanged—the block in Fig. 17A has a common negative lead for both condensers, while the block in Fig. 17B has a common positive lead for both condensers. If any of the leads in a block are common to two or more condensers, say so when you order a replacement. Two separate condensers, or two condensers in a block with separate positive and negative leads, could be used to replace the condensers in Figs. 17A and B.
MICA CONDENSERS

Mica condensers rarely go bad; when one does, it is best to use a replacement of the same capacity. Because different color codes are often used on micas, it is usually easiest to identify the proper size from the wiring diagram. If you have no service information, examine the original. You may find the capacity value is stamped on the condenser, or it may be marked according to the standard color code (see Fig. 18). Remember, private color codes are sometimes used, so if you come out to some unreasonable capacity value, the marking is probably not the standard code.

GANG TUNING CONDENSERS

In modern receivers the tuning condenser gang seldom becomes so defective it cannot be repaired. Even badly bent plates usually can be straightened with a thin putty knife. However, if they are beyond repair, the shaft is bent, or the bearings are damaged, a new gang—an exact duplicate—should be installed. Unless you order from the set manufacturer, remove the old gang and send it with your order to make certain you get the correct replacement. Be sure to give the make and model number of the receiver.

<table>
<thead>
<tr>
<th>COLOR</th>
<th>FIGURE</th>
<th>TOLERANCE</th>
<th>WORKING VOLTAGE</th>
</tr>
</thead>
<tbody>
<tr>
<td>BLACK</td>
<td>1</td>
<td>±1%</td>
<td>600 V.</td>
</tr>
<tr>
<td>BROWN</td>
<td>2</td>
<td>±2%</td>
<td>200 V.</td>
</tr>
<tr>
<td>RED</td>
<td>3</td>
<td>±3%</td>
<td>300 V.</td>
</tr>
<tr>
<td>ORANGE</td>
<td>4</td>
<td>±4%</td>
<td>400 V.</td>
</tr>
<tr>
<td>YELLOW</td>
<td>5</td>
<td>±5%</td>
<td>500 V.</td>
</tr>
<tr>
<td>GREEN</td>
<td>6</td>
<td>±6%</td>
<td>600 V.</td>
</tr>
<tr>
<td>BLUE</td>
<td>7</td>
<td>±7%</td>
<td>700 V.</td>
</tr>
<tr>
<td>VIOLET</td>
<td>8</td>
<td>±8%</td>
<td>800 V.</td>
</tr>
<tr>
<td>GRAY</td>
<td>9</td>
<td>±9%</td>
<td>900 V.</td>
</tr>
<tr>
<td>WHITE</td>
<td>10</td>
<td>±10%</td>
<td>1000 V.</td>
</tr>
<tr>
<td>GOLD</td>
<td>11</td>
<td>±11%</td>
<td>1100 V.</td>
</tr>
<tr>
<td>SILVER</td>
<td>12</td>
<td>±12%</td>
<td>1200 V.</td>
</tr>
<tr>
<td>NONE</td>
<td>13</td>
<td>±13%</td>
<td>1300 V.</td>
</tr>
</tbody>
</table>

FIG. 18. The RMA color code for mica condensers.

The plates of older condensers were often set in white metal castings. This metal may warp, throwing the condenser out of line and causing the rotor and stator plates to scrape against each other. Don't try to bend the plates, unless no replacement is available, as the casting will continue to warp and the trouble will reappear in a short time.

Replacing Resistors

Resistors fall into several classifications: fixed, semi-variable, and variable types. They may have carbon, a metallic deposit, or resistance wire as the resistive element. Let's take up each type in turn.

FIXED RESISTORS

You're usually safe in suspecting excess current as the reason for a metallicized or carbon fixed resistor's going bad, particularly if the resistor has a burned or charred appearance. (Wire-wound resistors rarely burn out—electrolysis at the junction of the terminal lug and the resistance wire is the usual trouble.) Look carefully for the cause of this excess current before installing a new resistor. A check from the low potential end of the resistor to the chassis with an ohmmeter will show whether a broken-down condenser or
some other short burned out the resistor. If the resistor is not changed in appearance and no short can be found, the element is probably cracked.

After you’ve repaired the short (or made sure there is none), determine the proper size for the replacement.

Resistance values are not critical and a variation of 20% is of little importance. You can find the value of the original resistor from the schematic diagram, or from the color code markings (if it follows the standard code). The color code for resistors is shown in Fig. 19.

The circuit in Fig. 20 shows some typical resistor value ranges. If you can’t determine the resistance of the burned-out resistor, install one that is shown by this figure to be appropriate for the circuit involved. If the set works satisfactorily and the voltages seem to be normal, leave the resistor in—otherwise, experiment with different values until you get the results you want.

Always use a replacement resistor with a wattage rating equal to or higher than that of the original—never lower. Otherwise, the replacement will burn out. You can use the physical size of the resistor as a guide if the replacement is the same type (carbon, metallized, or wire-wound) as the original. The replacement should be the same physical size, or larger.

If carbon resistors used as bleeders or voltage dividers are defective, replace them with 10- or 20-watt wire-wound types.

When sections of a candohm unit fail, it is generally best to replace the entire unit with a duplicate or with individual wire-wound units. Don’t use the lugs on the candohm as anchor points for individual resistors, because the defective unit may “come alive.”

Your stock of resistors should include a kit of carbon or metallized resistors in the 1/2-, 1-, and 2-watt sizes. You will usually find that values of 200, 300, 1000, 5000, 25,000, 50,000, 100,000, 250,000, and 500,000 ohms are used most. Then, you can add a kit of wire-wound 10- and 20-watt types. The most used sizes of these depend on the kinds of radios you service most, and they can be learned best from experience.

Most wire-wound voltage dividers have fixed, predetermined values. If a duplicate divider cannot be obtained and the section values cannot be determined from the service data, install
a 25,000- to 50,000-ohm, 50-watt semi-variable unit and adjust it to give the proper voltages. Then, measure the sections and use fixed resistors as replacements for them.

Some of the new molded resistors look like the small mica condensers. These resistors are ordinarily black, marked with three colored dots. Read these dots in the same order as you would those on a three-dot condenser; they then have the same meaning as the body, end, and dot colors respectively, on regular carbon resistors.

There are also condensers shaped like resistors. The condenser values are indicated by bands of color. Two groups of bands may be used, with the bands in each group being the same width, and the groups of bands being different in width. The bands of greater width indicate the significant figures of the capacity, while the bands of smaller size indicate the number of ciphers, the tolerance, and the voltage rating respectively.

**VARIABLE RESISTORS**

Volume and tone controls are the most important variable resistors. Exact duplicate controls are available and are the simplest to install. Some special dual control units can be replaced only by exact duplicates. However, a kit of universal sizes will permit replacement of most controls; sooner or later you will probably stock such a kit.

The physical size of a volume or tone control will not matter as long as it is not too large for the space provided. However, there are several types of

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**FIG. 20.** This circuit shows the locations of most of the resistors used in modern radio receivers. (The diagram is incomplete otherwise.) Here are typical values used:

- **R**₁ — a.v.c. decoupler — 50,000 to 250,000 ω (100,000 most common)
- **R**₂ — 1st det. bias resistor — 200 to 300 ω
- **R**₃ — osc. grid resistor — 50,000 ω for a.c., 100,000-200,000 ω for battery tubes
- **R**₄ — osc. plate resistor — 20,000 ω
- **R**₅ — screen dropping res. — 50,000 ω if no bleeder
- **R**₆ — a.v.c. decoupler — 500,000 ω to 2 megs. (1 meg. most common)
- **R**₇ — i.f. bias — 200-600 ω (usually 300 ω)
- **R**₈ — i.f. plate decoupler — 1,000 to 10,000 ω (usually 2000 or 5000 ω)
- **R**₉ — i.f. filter — 50,000 ω
- **R**₁₀ — diode load — 50,000 to 500,000 ω (100,000 ω most common)
- **R**₁₁ — 1st a.f. grid — 500,000 if biased; 10 to 20 megs. if convection biased
- **R**₁₂ — R-C plate res. — 50,000 to 250,000 (100,000 most common)
- **R**₁₃ — plate decoupler — 5000 to 50,000 (10,000 or 20,000 most common)
- **R**₁₄ — R-C grid res. — 100,000 to 500,000 (250,000 most common)
- **R**₁₅ — power tube bias — 150 to 600 ω (depends on tube, and whether bias is for single or push-pull tubes)
shafts, and if the wrong one is used the knob may not fit. Most shafts which are not exact duplicates are considerably longer than necessary and must be cut to the right length with a hacksaw.

The original control may have been equipped with an ON-OFF switch. If so, a switch can be attached to the back of a universal control by following the manufacturer’s instructions. Consult a control guide book if the original switch is a special type, such as may be found in battery sets; you may have to use a duplicate control.

The electrical size of a volume control depends on the circuit in which it is used. Some representative circuits are shown in Fig. 21. (Volume control guides show many more.) These guides will also prove helpful if you can’t determine resistance values from the schematic diagram or the original control. Actually, the resistance value is seldom critical.

Of the three types of connections commonly used today, the combination antenna-C bias control (Fig. 21A) may have any value between 10,000 and 25,000 ohms; the a.f. grid control (Fig. 21C) may be between 250,000 ohms and 2 megohms; and the diode load type (Fig. 21E) may be from 50,000 to 250,000 ohms.

More important than the resistance value is the control taper—the manner in which the resistance varies with the shaft rotation. You don’t have to worry about this, however; just name or sketch the circuit in which the control is used and your supplier can furnish the proper replacement. (Your kit of universal types will have a guide book showing the proper types.)

Some controls have taps for automatic bass compensation circuits. Be sure the replacement has similar taps.

Tone controls are ordered and re-

FIG. 21. Typical volume control connections.
Replacing Loudspeakers

Speaker repairs generally are made by the manufacturer or by firms specializing in this service, although occasionally you may find it profitable to replace a cone-voice coil assembly or a field coil yourself. Very often, particularly in small sets, the cheapest course is to buy a new speaker. Let's see just how you should go about ordering new parts, repairs, or new speakers, and how you should install replacements.

Replacement Cones. If you are going to have a cone replaced by the set manufacturer or by a firm specializing in this service, send in the entire speaker. It will come back with the proper cone installed.

If you decide to replace the cone yourself and can get the old cone out of the speaker intact, send it to the set manufacturer or to a firm manufacturing cones and request a duplicate. Include the make and model number of the receiver with your order.

If the old cone is completely torn up or missing, or if you are servicing some private brand or orphan receiver, you can send the speaker to a cone manufacturer and have an acceptable cone installed. Should you prefer to install the cone yourself, and don't know the name of the set manufacturer, examine the speaker carefully to see whether you can determine the name of the speaker manufacturer and the model designation of the speaker.

If you can't find this information, specify the diameter of the cone, the diameter of the voice coil, and the im-

*A private brand set is manufactured for department stores, chain stores, or small retail outlets. An orphan is one which does not have the manufacturer's name on the set, or one of which the manufacturer is out of business.

Field Replacements. What we've said about cone replacements applies also to the speaker field. You can return the original speaker to the set manufacturer or send it to a firm specializing in replacements, allowing them to install the proper type for you. If you want to do the work yourself, be certain that you specify the make and model number of the set, as well as any other numbers which may appear on the speaker itself.

If an exact duplicate replacement is unavailable, you must give the resistance of the field and its physical dimensions (length, and inside and outside diameter). Universal replacement speaker fields are available which have two windings; the resistance of these can be adjusted by making series or parallel connections, but the range of the adjustment is limited, so the field selected must be near the right size in the beginning.

You may wonder how you can give the field resistance when the original field is burned out. A service manual or a speaker field replacement guide usually will tell you what the resistance should be. If these sources fail, you can make a reasonable estimate of the resistance from the way the speaker is used, or you can find it by a resistance substitution method.
For example, you know that usually the speaker field of an a.c.-d.c. receiver either is connected across the output of the rectifier, as shown by coil $L_1$ in Fig. 22A, or is connected to a single diode as in Fig. 22B. In either case, the field value will be 2500 to 3000 ohms, and any value in this range will prove satisfactory.

Should the speaker be used as a choke in an a.c.-d.c. receiver, in the position indicated for coil $L_2$ in Fig. 22A, the resistance is low—usually 300 to 400 ohms.

In the standard a.c. receiver, the speaker field is usually used as a choke coil, illustrated as coil $L_1$ in Fig. 23. If this coil is burned out, a resistance substitution method will let you find its approximate resistance.

First, check the set to be certain that no short circuits have passed excess current through the field and caused the burnout. Repair any shorts that you find. Next, connect a resistor in place of the field as shown in Fig. 24. Use a 5000-ohm variable resistor, rated at 20 watts or more, which has one or more sliding taps. First move the slider to the end of the resistor, placing all the resistance in the circuit. Then turn on the set and measure the voltage between $B+$ and $B-$ . Compare this voltage with the voltage given in the service diagram or with the normal voltages usually applied to the output tubes. If the measured voltage is too low, decrease the value of the resistor by moving the tap (turn off the set before doing this). Experiment with the tap position until the correct $B$ supply voltage is secured, then disconnect the resistor and measure the resistance of the section finally used with an ohmmeter. This is approximately the resistance of the field.

A speaker in the negative side of the circuit may have a tap for bias connections, as shown in Fig. 25. If an open is found between the tap and ground, in the bias section, it is frequently possible to replace this section of the field with a resistor, allowing the remainder of the field to act as a choke coil.

FIG. 23. The speaker field is used here as the choke.

FIG. 22. Two methods of connecting speaker fields which are used in universal a.c.-d.c. sets.

FIG. 24. Finding the resistance of a burned-out field by resistance substitution.
Since the tapped section of the field usually has a resistance of only 300 ohms or so, a 500-ohm, 10-watt resistor with a slider tap can be used. (See Fig. 25.) To adjust the resistor, first put all the resistance in the circuit; then, with the receiver turned on, gradually reduce the resistance until the proper bias voltage appears across it.

If the open is in the main section \( (L) \), a replacement is necessary.

**ORDERING AND INSTALLING REPLACEMENT SPEAKERS**

There is, of course, no particular problem involved in ordering or installing an exact duplicate speaker. Simply order the replacement from the set manufacturer or distributor, giving the model number of the set and the make and model numbers of the speaker.

Some set manufacturers sell new speakers on a “trade-in” basis. When you send in the old speaker to get a new one on this plan, give the model number of the receiver. Any other information the manufacturer needs he can get from the old speaker.

▷ If you want to use a speaker which is not an exact duplicate, you must be sure the voice coil impedance, the speaker field resistance, and the physical size of new speaker are acceptable.

For instance, the new speaker should not have a cone diameter larger than the opening in the baffle of the radio cabinet—if it does, it will be necessary to cut a larger opening in the cabinet, which may not be practical. (Of course if the speaker is smaller, you can always mount it on a board which has an opening of the proper size and fasten this board over the original baffle opening.) And when you order a speaker for a table model cabinet, you must be sure to get one of such size or shape that it will fit into the cabinet with the radio.

As you have learned, the voice coil impedance can be found by measuring the voice coil resistance with an ohmmeter, then multiplying this resistance by 1.5. This is just an approximation, and it is possible for some mismatch to occur. Therefore, if you’re not positive that the voice coil impedance of the new speaker is the same as that of the old one, replace the output transformer as well as the speaker. You can usually buy an output transformer with the speaker which will match it to the output tubes used. Specify the make and model number of the speaker and the number and types of output tubes when you order the transformer.

You might use a universal output transformer, adjusting it in the manner you learned earlier in this lesson.

**Replacing Magnetic Speakers.**

Since magnetic speakers are far inferior in performance to p.m. dynamics, many servicemen use p.m. replacements for them. Usually, magnetic speakers are found in midgets, where space limitations are important. However, p.m. type speakers are made with diameters as small as 2 inches, and usually take no more space than the equivalent magnetic speaker.

Although a matching transformer is necessary with the p.m. speaker, those used with little speakers are generally small enough to fit into even a midget receiver without trouble. When you order your p.m. speaker, specify that it be equipped with an output trans-
former which will match it to the power tube used.

Replacing P.M. Speakers. You should always replace a defective p.m. speaker with another p.m., to avoid having to energize a field. You have to consider only the size of the replacement and the voice coil impedance. A new output transformer is necessary if the voice coil impedance differs from the original.

Replacing Electrodynamic Speakers. Should you wish to replace an electrodynamic speaker with a p.m. speaker, you must match the voice coil impedance of the p.m. unit to the set output (using a new transformer if necessary) and also make whatever set adjustments are necessary to compensate for the loss of the field coil.

If the field of the original speaker was in parallel with the voltage source (like \( L_1 \) in Figs. 22A and 22B), as it is in many a.c.-d.c. receivers, just remove the original field connections. If the field is used as a choke like \( L_2 \) in Fig. 22A or \( L_1 \) in Fig. 23, you will have to provide a choke coil to obtain equivalent filtering. For an a.c.-d.c. set, order an a.c.-d.c. filter choke, which is usually rated at 10 henrys and 50 ma. The resistance of this choke will be comparable to the field resistance.

You will have to use both a choke and a resistor in a standard a.c. receiver, since the average choke coil has a resistance of only 300 to 600 ohms, while a speaker field resistance may be anywhere from 1000 to 3000 ohms. The proper connections are shown in Fig. 26, where the choke \( L_2 \) and the resistor \( R \) replace the original field (shown as \( L_1 \)).

The replacement choke coil should have an inductance rating between 10 and 30 henrys and a current rating at least as high as the receiver current. A choke rating of 75 to 100 ma. is usually sufficient.

If the field was tapped and used to supply bias for the power output tubes (Fig. 27), you can follow the same general method of replacement used in Fig. 26, except that the resistor must have two slider taps. Connect one of these sliders to the point connected to the tap on the original field, and connect the other slider to the set chassis (see Fig. 27).

Make the resistance between points 1 and 3 of this figure approximately equal to the original field resistance. Then bring slider 2 toward slider 3 until the voltage drop across section \( R_1 \) delivers the proper bias for the output tubes.
Lesson Questions

Be sure to number your Answer Sheet 47RH-2.

Place your Student Number on every Answer Sheet.

Send in your set of answers for this lesson immediately after you finish them, as instructed in the Study Schedule. This will give you the greatest possible benefit from our speedy personal grading service.

1. Suppose a power transformer stops smoking and cools off when the tubes are removed, even though the on-off switch is still turned on. Was the overheating due to an overload, or is the transformer defective?

2. Suppose the tester shown in Fig. 1 is being used to check the primary current of a transformer having an internal short. Should the lamp light: 1, dimly; 2, brightly; 3, or should it show no light?

3. Suppose a power transformer is available which has a filament winding rated at 6.3 volts and 4.5 amperes. The set in which you want to use it has tubes rated at 6.3 volts, and they draw a total of 2.7 amperes. Can the transformer be used?

4. An input push-pull transformer has a defective primary, and a check of the secondary shows a resistance of 300 ohms. Is the transformer an input for a class A output stage, or for a class B stage?

5. Suppose a plate by-pass condenser in an i.f. stage becomes defective. Would you use 50 μfd., .01 mfd., .5 mfd. or 8 mfd. as the replacement capacity?

6. If a 10 mfd., 150-volt electrolytic condenser becomes defective, could a 16-mfd., 250-volt condenser be used as a replacement?

7. When you install a replacement r.f. coil having a variable inductance, is the coil inductance adjusted at 600 kc., or at 1400 kc?

8. Suppose a receiver using the a.f. volume control circuit of Fig. 21C has a defective control rated at 500,000 ohms. Could you use a 1-megohm audio type control as a satisfactory replacement?

9. If a receiver uses a dual electrolytic filter condenser having a common positive lead, can you replace it with a dual electrolytic condenser having a common negative lead?

10. If you do not know the voice coil impedance of a loudspeaker, how can you approximately determine its value? Multiply the resistance of the voice coil by 1.5.
SERVICING RECORD CHANGERS

48RH-2

NATIONAL RADIO INSTITUTE
WASHINGTON, D. C.
ESTABLISHED 1914
STUDY SCHEDULE NO. 48

For each study step, read the assigned pages first at your usual speed, then reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind. Study each other step in this same way.

☐ 1. Introduction .................................................... Pages 1-4
   The general characteristics of non-mixing and mixing record cha ngers are described in this section.

☐ 2. A Study of Changer Functions ............................... Pages 4-24
   Here you study each of the operations of a record changer in detail.

☐ 3. Servicing Record Changers ................................. Pages 24-28
   This section shows you how to service a defective changer.

☐ 4. Pickups and Their Servicing ................................. Pages 28-32
   A brief description of the 3 types of pickups and instructions for servicing them are contained in this section.

☐ 5. Motors and Their Servicing ................................. Pages 33-35
   This section contains servicing instructions for the popular types of phonograph motors.

☐ 6. Microgroove Records ......................................... Pages 35-36
   The newest developments in records and their players are briefly discussed here.

☐ 7. Answer Lesson Questions and Mail Your Answers to NRI for Grading.

☐ 8. Start Studying the Next Lesson.
AUTOMATIC record changers are the logical outgrowth of the return to popularity of phonograph records. Before the advent of radio, the phonograph was very popular. Then, for a time, the radio supplanted the phonograph in the home. Gradually, however, electrical record players that operated through the radio receiver or had built-in amplifiers became increasingly popular. As you know, these record players consist of a motor-driven turntable and a pickup arm. The latter, through either an electromagnetic pickup or a crystal pickup, converts the modulation on a record groove into an electrical signal that can be amplified and reproduced by an audio amplifier and loudspeaker.

You are undoubtedly familiar with the method of operation of such a record player. You place the desired record on the turntable, turn on the power, and lower the pickup arm into position on the outside edge of the record. When the record is finished, you remove the pickup arm, turn off the power, and remove the record, either to turn it over or to replace it with another. You can play any size record that is 12 inches or less in diameter—the standard 10-inch or 12-inch diameter records, or the smaller "specials."

The automatic record changer is designed to make it easier to play records. In all of the automatic record-changer systems, a collection of records is arranged in the order in which they are to be played. These records are stacked in some storage system. When the device is turned on, a record moves to the playing turntable, and the pickup arm is automatically placed on it. At the end of the record, the pickup arm is removed and the next record is put into playing position. This operation is repeated until all have been played. Most changers handle enough standard records to play for more than half an hour before it is necessary to handle the records.

Ordinarily, automatic record players play only one side of a record before going to the next record. As a result of the widespread use of such record changers, most symphonies and other selections requiring more than a single
THE NON-MIXING CHANGER

The most popular record changer is the non-mixing type, because it is somewhat less complex in its design and operation and therefore less expensive. Let's run through the operation cycle for a basic non-mixing changer:

First, a group of records, all of the same size (either 10-inch or 12-inch), are selected. Changers have different capacities, but most of them will play from eight to ten 12-inch records or ten to twelve 10-inch ones. Of course, any smaller number can be used at a time. Once the records are selected and have been placed in the order in which they are to be played, they are put into the storage mechanism.

The control switch or lever is now placed in the position corresponding to the record size. On many machines, this act also automatically turns on the changer mechanism.

When the changer starts to operate, the first record is dropped into playing position on the turntable. At the same time, the pickup arm is lifted from its rest beside the turntable and moved over the record, which by now is revolving. A positioning mechanism stops the pickup arm and lowers it so that the needle is on the outside plain edge of the record. Gravity, a spring, or the take-in groove on the record now swings the needle into the first playing groove.

To act in this manner, the changer must have some mechanism capable of separating one record from the rest. It must also contain a mechanism that can lift the pickup arm from its rest, swing it to the proper position, and then lower it onto the record. Once the pickup arm is placed on the record, it must be free to follow the record grooves. Therefore, the mechanism that moves the pickup arm must re-
lease it completely while the record is playing, then regain control of it when
the record is finished.

The eccentric groove cut as the last
groove on a phonograph record is al-
most always used to make it possible
for the pickup arm to be brought under
control again. This groove is shaped
with respect to the spindle about which
the record revolves so that the pickup
arm, when it engages the groove, is
forced to move rapidly back and forth.
A tripping mechanism is then actuated,
either because of the back and forth
motion or because the pickup moves
faster in this groove or because the
pickup is brought close to the spindle.
Once the trip is actuated, the pickup
arm is lifted from the record, then
swung to the side out of the way. The
next record in the group is now dropped
into playing position. Once there, the
pickup arm is returned over the edge
of this record and dropped into playing
position.

When the last record in the group
has been dropped and played, some
changers automatically move the pick-
up arm to the side and turn off the
mechanism. Others repeat the last rec-

ord over and over until they are turned
off.

In most cases, the control lever has
a reject position so that you can reject
any record in the group you don’t want
to hear. Pushing the reject lever to
the proper position actuates the trip
that the pickup arm normally actuates
at the center of the record. This auto-
matically causes the mechanism to
pick up the arm from whatever posi-
tion it is in and play the next record.

If you want to play 12-inch records
when 10-inch ones have been played
before, or vice versa, put the proper
records into position and move the con-
trols to the appropriate position. This
automatically moves a stop so that the
pickup arm will drop in the right pos-
tion. The remainder of the change
cycle is identical with the one we just
described.

INTER-MIXING CHANGERS

In an inter-mixing changer, 10-inch
and 12-inch records can be mixed up in
any order in the storage system. When
the mechanism is turned on, and the
first record is caused to drop, the drop-
ning mechanism makes use of a system
of fingers or feelers to determine the
size of the record that dropped. Auto-
matically, as a result of this, the stop
for the pickup arm is set to the proper
position so that the arm will land on
the plain outside edge of the record.
The playing cycle is identical with that
of the non-mixing kind except for this
additional automatic feature of the de-
vice’s determining the record size and
setting the pickup arm to drop in the
proper position.

All changers, whether non-mixing or
inter-mixing, can be played manually
by setting the control lever to the
proper position. Doing so takes the
record dropping mechanism out of
operation, and individual records can
be placed on the turntable and played just as on any single record player. This operation is necessary when any of the non-standard record sizes, such as certain children’s records, are to be played.

From the foregoing you can see that even the simplest record changer must be rather complex. It must separate one record at a time from the storage system and place this record on the turntable. It must lift the pickup arm, move it into position, and lower it onto the record. At the end of the record, it must remove the pickup arm so that the next record can be dropped. If it is an inter-mixing changer, it must determine the size of the record and from that properly place the pickup arm.

(Incidentally, some manufacturers refer to the pickup arm as the “tone arm.”) Finally, at the end of the group of records, it must either cut itself off or repeat the last record.

All of these operations are performed by a mechanical system driven by the same motor that operates the turntable. In a mechanical system as complex as this, there are almost endless possibilities for variations. In fact, a great many variations have been designed—far more than we can hope to cover in this one Lesson. We shall describe several of the most common record-changer mechanisms; studying these will make it easier for you to understand any other types you may meet.

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**A Study of Changer Functions**

Rather than try to study record changers as an entirety, we shall break down their operations into separate functions. This study of individual actions is desirable because it will show you how to concentrate on one particular action at a time. This is usually necessary in servicing these devices, because, in most instances, only one particular operation of the changer will be out of order. If you know how that operation should be carried out, it will be much easier for you to see just what adjustments are necessary to correct the difficulty.

**RECORD PLAYING**

Let’s start our study of a changer with one record on the turntable and the pickup arm on this record in a playing position. This is a logical place to start, because none of the changer mechanism is in operation. The motor is revolving the turntable and the record, and the pickup needle is following the record groove. To permit the needle to track properly on the record, the pickup arm is freed from the changer mechanism as much as possible.

The conditions while the record is being played are shown in Fig. 1A. The pickup arm is gradually approaching the center hole of the record because the spirally cut playing groove of the record gradually draws the needle toward the center hole. For several reasons, this spiral groove cannot be continued right up to the middle of the record. The most important reason is that the groove velocity would be entirely too high for the needle to follow the variations. Therefore, the actual recording ends about two inches from the center spindle. The recording is followed by a few more turns of the spiral groove containing no recording, then the last turn of the spiral groove feeds into an eccentric groove.

As shown in Fig. 1A, this eccentric groove is off-center with respect to the
center hole—the distance W is less than the distance X. In Figs. 1B, C, D, E, F, and G, we have shown what happens when the pickup needle enters the eccentric groove. Because this groove is off-center, the pickup arm is rapidly brought in toward the center spindle as shown in B, C, and D. Then, as the groove continues its rotation, the pickup arm is moved rapidly away from the center spindle. In other words, in E, F, and G, the direction of movement of the pickup is reversed from the normal direction that it has had throughout the playing of the record. This eccentric groove is endless, so the pickup arm oscillates back and forth in this same manner until it is taken from the record either by hand or by the record changer mechanism. As we said earlier, this motion of the pickup arm in the eccentric groove is used to actuate the trip mechanism that allows the changer mechanism to regain control of the arm.

**TRIP MECHANISMS**

Because the pickup arm is forced to move close to the center spindle by the eccentric groove, some trip mechanisms are arranged to trip when the pickup arm gets close enough to the center spindle. Others depend upon the fact that the motion of the pickup arm is reversed during a portion of the eccentric groove travel. Still others depend upon the velocity at which the pickup arm moves toward the center spindle. We'll describe all types.

Fig. 2 shows how the motion of the pickup arm is conveyed to the trip assembly. The pickup arm is mounted on a hub that is fastened to a hollow shaft. The weight of the pickup arm is carried on ball bearings above a support post that is a part of the motor shelf (or motor board, so called because the motor is suspended from it).

Attached to the end of the hollow shaft is a trip lever. This may be an individual lever, or may be part of the arm crank that is used to move the pickup arm back and forth when the automatic mechanism is operating. In any case, as shown in B, a motion of the pickup arm causes a similar motion of the trip lever underneath the motor board.

We cannot have a heavy trip lever, because the pickup arm, while playing the record, must be held back as little as possible so that it can easily follow

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**FIG. 1.** How the eccentric groove on a record swings the pickup arm in and out at the end of a record.
the spiral groove on the record. Therefore, the trip lever is an exceedingly lightweight arm that is used to actuate another arm or lever, thus starting the record-changing operation.

Incidentally, in a number of our drawings, we shall look down upon the operation as if we could see through the motor shelf from the top of the turntable. This helps in seeing just what goes on, but you must remember that the action will be reversed when you are watching the parts from underneath. We are also going to show some drawings of the mechanisms viewed from underneath so that you can become familiar with their appearance in this orientation.

**Eccentric Trips.** Now that you have a general idea of how the pickup arm can move a trip lever, let's see how the trip lever can be used to start the change action when the pickup needle gets to the eccentric groove. Fig. 3 shows one of the systems that operates when the pickup arm reverses its normal motion. As shown here, the arm shaft connects to a trip lever. At the end of this lever is a trip pawl. The pawl swivels on a bearing, and a spring holds a stop on the end opposite the finger against a cut-out in the trip lever.

While the record is playing, the normal motion of the trip pawl is to the left in this figure. We are looking upon this action from the top. During most of the playing of the record, the trip pawl is not even engaged with the teeth on the ratchet lever. However, as the end of the record is approached, the trip pawl engages the teeth on the ratchet lever. The spring holding the trip pawl is relatively weak, and the direction of the trip lever mechanism is such that the trip pawl tends to slide over these teeth (Fig. 3B). As long as it moves to the left, it will merely slide over the teeth. As the spiral groove brings the pickup arm closer and closer to the center, this pawl moves along tooth after tooth of the ratchet lever.

When the pickup arm enters the eccentric groove, it will move in the opposite direction during a portion of the rotation, as you learned from Fig. 1. When the motion is to the right, the trip pawl cannot escape the teeth. Therefore, the trip pawl is rotated toward a position in line with the trip lever (Fig. 3C). This increases the distance the end of the pawl projects beyond the trip lever. The pawl then forces the toothed end of the ratchet lever away from the trip lever. (The distance E in Fig. 3C is greater than D in Fig. 3B; and since the trip lever can-

![Diagram of pickup arm and trip lever](image)

**FIG. 3. One trip mechanism.**

not shorten, the ratchet arm is forced to move away from it.) The movement of the ratchet lever releases the mechanism, as we shall show a little later.

Another eccentric groove trip is shown in Fig. 4. This time we are showing the view as you would see it from underneath the record changer, so the direction of the trip lever is reversed. In this system, a trigger is held...
by spring pressure against a trigger ledge on the ratchet arm. If the end of the ratchet arm with the teeth is pushed upward, the trigger can escape from the trigger ledge, allowing it to drop downward and thus move the "bell crank" to engage the changer mechanism. The trip pawl, as it moves to the left (Fig. 4A) engages the teeth on the ratchet arm (Fig. 4B). Then, when the eccentric groove forces the trip lever to move in the opposite direction (Fig. 4C), the trip pawl straightens up and pushes on the ratchet arm. This moves the ratchet arm in the direction shown by the arrow, allowing the trigger to drop downward.

**Positional Trip.** Incidentally, this same changer also has a positional trip.

![FIG. 4. Another eccentric trip.](image)

That is, it is so arranged that should the eccentric trip fail to operate, the device will automatically trip anyway when the pickup arm reaches a fixed position from the spindle. The mechanism for this is shown in Fig. 5. An adjustable screw called a limit stop is fixed to the trip lever. If the trip lever moves far enough counter clockwise in Fig. 5, this stop will strike the lower end of a pivoted fork called a trip link. If this happens (it can happen only if the trip pawl has not yet moved the ratchet arm), the trip link will pivot clockwise, moving the ratchet arm and releasing the trigger.

**Velocity Trip.** Fig. 6 shows one of the velocity systems. There are many styles of these, but practically all of them depend upon some friction device that does not trip the mechanism until the pickup arm travels inward at high speed, as it does in the eccentric groove.

In Fig. 6, the pickup arm is coupled through a link to a friction plate. Therefore, the friction plate is pushed clockwise in this figure. This motion is transferred to the trip arm through a friction pad that is between the friction plate and the trip arm. The amount of friction here is very little, but it is sufficient to move the trip arm clockwise. There is a projection called a striker on the spindle at the center of the turntable. This striker moves clockwise in this illustration. As the trip arm moves during the playing of the record, the end of the trip lever is brought to where the striker can hit it.

![FIG. 6. A velocity trip.](image)

This feed-in of the trip lever is very slow. As a result, just the very tip end of the lever is eventually hit by the striker on one of the revolutions of the spindle. When the striker hits the trip lever such a slight and glancing blow, the trip lever jumps back away from the striker. The amount of friction be-
tween the trip arm and friction plate is such that the trip lever can thus escape from the striker during the normal playing of the record.

However, when the eccentric groove is reached, the pickup arm moves in toward the spindle very rapidly. As a result, the friction plate moves the trip arm in rapidly, so the trip lever is moved well over in front of the striker during a single revolution of the spindle. As a result, the striker now hits the side of the trip lever a full blow. The trip lever cannot now escape, because the pressure is no longer applied to its end but is applied to its side. Therefore it pivots at its junction with the trip arm, forcing its other end to the left in the slot in the trip bar. The trip bar is then forced to move to the left and thus engages the mechanism.

To sum up what we have learned: The eccentric groove that is cut on all modern records is used to notify the automatic mechanism that the end of the record is reached. Through a trip mechanism that depends on the waving back and forth produced by the eccentric groove, or on the velocity of travel during the eccentric motion, or on the fact that the pickup arm is brought within a preset distance from the spindle, some tripping mechanism is actuated that allows the automatic mechanism to go into action. In every instance, a trip lever or mechanism attached to the pickup arm (light in weight so that it puts no real restriction on the pickup arm motion) transfers the eccentric motion through other levers to a mechanism that allows the turntable motor to operate the changer mechanism.

Of course, as you might expect, there are many variations on these basic devices. There are even some types in which a switch is closed by the tripping mechanism, and the actuating system is electrical. However, regardless of the system, all tripping mechanisms have the purpose of initiating the changer action. And they all have their troubles—they fail to trip, or trip too early, depending upon their adjustment, the tensions of their associated springs, etc. We will discuss some of these troubles later in this Lesson.

**THE MAIN CAM**

It is necessary that all the changer actions be synchronized with each other—the pickup arm must be removed at the proper time, just before the next record drops into position; and, once the record has dropped, the pickup must be brought back. To control these actions simultaneously, changer mechanisms are arranged so that the entire cycle is controlled by a single main cam or by a single drive mechanism that operates all the cams simultaneously.

Fig. 7 shows the top and bottom views of a typical main cam. The top view is shown at the left. Notice that there are a number of grooves, rims, and raised ledges on this wheel, all of which are used to control, individually, some particular portion of the changer operation.

To prevent the main cam wheel from rotating when power is not applied, usually some form of detent is used to latch the wheel in the out-of-cycle position. In the example shown in Fig.

![FIG. 7. A typical main cam wheel.](image)
7, notice the detent notch at the left edge in the top view. A stud on the end of a lever fits into this notch until power is applied. This stud is withdrawn when power is applied so that the main cam can rotate and control the mechanism. At the end of the cycle, the stud is traveling on the rim, and falls into this detent notch just as at the end of the cycle to hold the main cam wheel motionless again.

There are almost as many ways of supplying power to the main cam wheel as there are record-changer systems. Fig. 8 shows the method used for the particular cam that we have shown in Fig. 7. First, if you will examine the bell crank assembly, you will find that the lower end has a stud that fits into the detent notch on the main cam wheel. The upper end of the bell crank assembly is fastened to a drive arm assembly. On this drive arm is a drive pulley that operates a small gear. This gear drives an idler gear that is meshed with the gear teeth on the main cam wheel.

When the ratchet arm moves away so that the trigger can fall downward, the bell crank moves in accordance with this motion. As a result, the lower end of the bell crank is moved outward so that the stud moves out of the detent notch; at the same time, the drive pulley and the whole drive arm assembly are forced over by the movement of the upper end of the bell crank so that the drive pulley touches the turntable rim. Since the turntable is being driven by the motor, the drive pulley starts to rotate, driving the main cam wheel through the gear train.

When the changer has run through its entire cycle and the pickup arm is being put back on the record for playing, the main cam wheel rotates to a position where a raised section on the wheel engages the trigger and forces it upward. When the trigger reaches the proper position, the spring on the lower end of the ratchet arm pulls the shelf under the lip of the trigger, resetting the trigger for the start of the next cycle.

The trigger is connected to the bell crank by a very heavy spring. When the trigger is reset, this spring would force the bell crank to pull the drive mechanism out of contact with the turntable rim, except that at that instant the stud on the end of the bell crank is riding on the rim of the main cam wheel, which holds the bell crank so that the drive mechanism still operates. When the detent notch comes around, however, the stud on the bell crank falls into this notch; this allows the upper end of the bell crank to pull the drive pulley away from the turntable rim. The main cam wheel does not turn again until the end of the next record or until a reject button is depressed to release the trigger again.

Another system is shown in Fig. 9. The tripping mechanism is that shown in Fig. 3, but this time the drawing is
such that we are looking at the mechanism from the bottom.

When the trip pawl engages the ratchet lever, the left-hand end of the lever in Fig. 9 is forced downward, which raises the right-hand end. This frees the drive cam pawl and also takes a stud on the ratchet lever out of a detent on the main cam, much as in the system we just described.

However, the drive mechanism is a little different here. The motor wheel is actually a form of gear having a number of slots cut in it. This turns all the time. The drive cam is a circular wheel mounted over the motor wheel. On the drive cam is a drive cam pawl, which has a latch-in stud on it that will fit into the slots on the motor wheel when the drive cam pawl is released. Attached to the drive cam is a small gear that has teeth engaged with the main cam.

Therefore, when the ratchet lever releases the drive cam pawl, the spring S pulls the drive cam pawl over so that its latch-in stud engages the motor wheel. When this happens, the motor wheel revolves the drive cam, which in turn revolves the small gear and drives the main cam.

Meanwhile, the ratchet lever is held up from the position in which it engages the drive cam pawl, because a bump on the lever is riding on the rim of the main cam detent. When this bump reaches the notch of the detent, the ratchet lever spring pulls the lever down into the notch. This brings the right-hand end of the ratchet lever down in front of the approaching drive cam pawl. As soon as the drive cam rotates sufficiently for the ratchet lever to engage the pawl, the pawl stud is pulled away from the motor wheel; this stops the driving of the mechanism.

Fig. 10 shows an electrical system for controlling the application of power to the main cam. In this system, there is a main drive wheel, practically the same size as the main cam, that is adjacent to the main cam. On the main cam there is the drive pawl shown in Fig. 10. This pawl cannot engage the teeth on the drive wheel until the armature of the relay R is moved away from the upper end of the drive pawl. When the tripping mechanism closes a switch, power is applied to the relay, which draws away the armature and allows the drive pawl to be pivoted so that it engages the teeth on the main drive wheel. The main cam is then driven by the main drive wheel through the pawl. At the end of the cycle, the armature, which is released by now, is in a position to engage the end of the drive pawl and thus withdraw it from contact with the main drive wheel.

Fig. 11 shows another type rather similar to that shown in Fig. 10 except that the trip mechanism is mechanical. This trip mechanism holds the drive dog up from the drive wheel (Fig. 11A) until the trip is actuated. The trip then
moves out of the way, allowing the drive dog to drop down (Fig. 11B). The drive wheel has a series of bosses or raised projections on it; one of these catches the drive dog and thus forces the rotation of the main cam. At the end of the cycle, the drive dog is lifted by the resetting of the trip mechanism, and the main cam is disengaged.

Now that we have arranged for the end of the record to signal the start of the automatic operation, and have learned how power can be applied to the automatic mechanism, let's go on and see how each individual action is carried out by the mechanism. Remember, we have to elevate the pickup arm and lower it, move it in and out, and drop the record to the playing surface. Although we shall break these down into individual actions, always remember that many of the processes may be combined so that a single master lever can perform several of these actions simultaneously. However, once you understand basically how each action is carried out, it will be rather easy for you to pick out and study that part of the operation of any record changer you may service.

**PICKUP ARM ELEVATION**

During each cycle, the pickup arm must be lifted and lowered. The arrangement for elevating the pickup arm is ordinarily separate from all the other functions. Fig. 12 shows two of the more popular systems.

In the system shown in 12A, the main cam has a cut-out space in it into which the end of the lift lever fits while the pickup arm is on the record. The lever is shown in this position. When the end of the record is reached, the main cam rotates. The end of the lift lever then rides up on a raised ledge on the cam. This depresses the left end of the lever, which means that the right end of the lever goes up. This right end presses against a bearing plate and so moves a push rod upward through the center of the hollow shaft. The pickup arm is fastened at bearing A. Therefore, when the push rod moves upward, the needle end of the pickup arm is lifted. Other mechanisms then move the arm out of the way and drop the next record. When the arm is moved back into the playing position, the main cam is completing one revolution, bringing the notch on the cam ledge back into position over the left-hand end of the lift lever. This end of the lever then rises, lowering the push rod and allowing the pickup arm to drop to the record. The main cam is then stopped so that it cannot rotate farther until the end of the record.

In the system shown in Fig. 12A, the push rod is threaded so that the bearing plate can be moved up or down.
This makes it possible to adjust the vertical movement of the push rod and consequently of the pickup arm. If the pickup cannot come down far enough to allow the needle to touch the record, then the bearing plate is too far down on the push rod. On the other hand, if the arm does not bring the needle up far enough to clear a stack of records on the turntable, the bearing plate is too high on the push rod. The latter condition is more common; it is corrected by lowering the bearing plate by screwing it downward on the push rod. In some systems, a nut follows the bearing plate and is used to lock it into position; in others, there are set screws in the hub of the bearing plate that are used to lock it.

The system shown in Fig. 12B is the same except that it is practically the inverse of that shown in A. Here, when the "lift cam" rotates, the left end of the lift lever is forced up, which pulls the lift rod down. Since the lift rod is bent at the top and attached behind bearing A, a downward movement of the lift rod pulls down on the rear end of the pickup arm and raises the needle end.

This mechanism can also be adjusted by moving the bearing plate up or down on the lift rod. Moving the bearing plate upward in this case provides a greater lift.

Fig. 13 shows two other basic systems of this kind. A cord is used in Fig. 13A to provide the lift. The cam in this case has an eccentric slot cut in it. The end of the lift lever moves in this slot. When the slot moves the lever toward the left, it pulls on the cord and so lifts the arm. Then, when the slot permits the lever to move to the right, the cord slackens and allows the pickup arm to drop. This system is usually adjusted at the point where the lift cord attaches to the lift lever. This end of the cord is normally attached to a threaded rod that can be run in or out of a bracket, effectively adjusting the length of the cord.

The system shown in Fig. 13B is the simplest of all. Here, the cam is directly under the push rod. As the cam rotates, the push rod is forced directly upward by the shelf on the cam edge, lifting the pickup arm. Simple systems such as this are found in the less expensive changers. They work as long as the tolerances in parts are carefully controlled, but there is usually little or no means for adjustment. In the style shown here, the only manner of adjusting is to bend the push rod at the top. Any such bending operation is rather critical, since the rod is liable to break.

**FIG. 13. Two more elevator mechanisms.**

**PICKUP ARM ROTATION**

Now that you know various methods of elevating the pickup arm, let's see how the arm can be carried into the proper playing position and then removed at the end of the record.

In Figs. 12 and 13, you will notice a projection at the bottom of the hollow shaft labeled "arm crank." Moving the arm crank to the right or left will move the pickup arm similarly,
because they are connected by the hollow shaft. Therefore, all we need is to arrange for the arm crank to be controlled by the changer mechanism so that the pickup arm is moved in the desired manner.

To see the general way that this control is exerted, study Fig. 14. The arm crank is connected at bearing A to the pickup arm. The other end of the arm crank has a finger that is in an eccentric groove in the main cam. (To get a general idea of what some of these grooves look like, examine Fig. 7.) Let’s suppose the eccentric groove has the shape shown in Fig. 14A. When the cam rotates around its bearing, marked C, it causes the finger on the arm crank to move so that it follows the groove. In Fig. 14A, the changer is in cycle and the finger of the arm crank is farthest from bearing C, which is at the center of the turntable. Therefore, the pickup arm is as far as it can be from the center bearing and is off the record completely.

Further rotation of the cam brings the groove to the position shown in Fig. 14B. The shape of the groove is such that the finger on the arm crank is now brought closer to the bearing C, bringing the pickup arm over the edge of the record. The elevator mechanism now allows the pickup arm to drop on the edge of the record just as the eccentric groove moves to the position where the finger enters the wide spacing of the groove. The finger is now released, because the cam ceases to rotate. The pickup needle now follows the record grooves, and except for moving the pickup arm, the trip, and the arm crank, the pickup is entirely divorced from the player mechanism.

As the needle is drawn toward the center of the record by the record groove, the arm crank moves through the free space. This space on our imaginary assembly is wider than the rest of the eccentric groove so that there is no interference with the movement of the arm crank.

At the end of the record, the trip mechanism goes into operation and starts rotation of the cam. The position of the arm crank just before this moment is shown in Fig. 14C. As the cam rotates farther, the finger on the arm crank enters the groove. When the elevator has lifted the arm from the record, continued rotation of the cam toward the position shown in Fig. 14A rapidly moves the pickup arm out of the way so that the next record can be dropped.

FIG. 14. A basic pickup arm left-to-right motion control.
The system shown in Fig. 14 is basically that used in all changers. However, the actual mechanism is considerably more involved than we have shown, because we must arrange for the pickup arm to be brought to the proper position for either 10-inch or 12-inch records.

If there were only one record size to be played, we could easily adjust the position at which the arm lands (the only critical factor) by adjusting the angle of the arm crank with respect to the pickup arm. However, we do have two record sizes. Let's look at some of the basic systems used to make it possible to shift from one to the other.

### 10-12 Landing Shift

Fig. 15 shows one way of adjusting the position at which the pickup arm will land. An additional T-shaped crank is used between the arm crank and the groove in the cam. This T crank can be moved to either of two positions by rotating the control cam attached to it.

![Fig. 15. A T-crank is used here to provide the landing adjustment for either 10- or 12-inch records.](image)

In Fig. 15A, the T crank is in one of its two possible positions. As the cam rotates, the groove moves the finger on the end of the T crank, which transfers this motion directly to the arm crank and thus to the pickup arm. The T crank is able to follow the eccentric groove because a slot on the crank permits it to move in all directions with respect to its bearing D.

To change the landing position from the 12-inch to the 10-inch position, the control cam in Fig. 15A is rotated 180°. This moves bearing D from one side of the control cam to the other, as shown in Fig. 15B. The T crank must now move along this new position. In Fig. 15B, the original positions of the T crank, arm crank, and pickup are shown in dotted lines. As you can see, the pickup arm now has a new position although the finger on the T crank is still in the same place on the eccentric groove. This means that the pickup arm will land in a different place when it is dropped by the changer mechanism. In this particular case, it will land an inch nearer the center of the turn-table, in the proper place for a 10-inch record.

A system of this sort usually has only one adjustment: the fastening between the arm crank and the pickup arm can be adjusted to make the pickup land properly on either a 10-inch or a 12-inch record. The pickup should then also land properly on a record of the other size when the control cam is turned to the other position.

Fig. 16 shows another basic system. Here, the cam has a raised ledge W that is eccentric with respect to the cam bearing C. The arm lever, which pivots about bearing E, is held against the side of this ledge by the spring S. The arm lever is therefore swung in and out by the eccentric ledge as the cam rotates.

Under the conditions shown in Fig. 16, the record is being played and the cam is motionless. This figure is drawn so that we are looking down through the top of the motor mounting board. The pickup (shown dotted here) is connected to the pickup arm crank at the bearing A, as in the systems we
have studied up to now. When in the position shown, the arm crank is not restricted at all, so the pickup arm is free to follow the record grooves.

The end of the arm crank has a finger that protrudes through the cut-out in the arm lever. As the pickup nears the end of the record, the mechanism is tripped, so the cam starts to turn. The pickup arm is elevated from the record by operation of the elevating mechanism, then the rotation of the ledge W begins to move the arm lever to the right. Soon end Y of the cut-out bears against the finger of the arm crank, thus forcing the arm crank to move to the right also. In turn, this swivels the pickup arm out of the way in the same direction.

When the next record has dropped, and the arm lever begins to return toward the position shown here, end Z of the cut-out presses against the arm crank finger and thus brings the pickup arm back in toward the edge of the record. At the proper point in the cycle, the elevator mechanism lets the pickup arm down on the record edge.

The operation just described is the one that occurs when a 10-inch record is played. Fig. 17 shows what happens when the mechanism is set to handle 12-inch records. The mechanism used to control the size setting of the changer has either an arm or a pin that can be dropped down in front of a finger on the arm lever. When this pin is up or out of the way, as it is when the size control is set for 10-inch records, the arm lever bears directly on the ledge W at all times. However, if the control is set for 12-inch records, the pin (labeled stop H in Fig. 17) is dropped into place while the arm lever has the pickup arm at its extreme right-hand position. Then, as rotation of the eccentric ledge W brings the arm lever to the left, the lever strikes stop H; it can then travel no farther to the left. This position is the proper one for the 12-inch record size while the pickup arm crank finger is against side Z of the cut-out.

Now let's see what the cycle of operation of this mechanism is when it is set for 12-inch records. During the playing of the record the pickup arm crank finger moves through the cut-out opening to approach side Y. When the mechanism is tripped, the pickup arm is elevated. However, the arm lever cannot move until ledge W comes over and bears against it. Then, however, it is carried to the right to the same extreme position as before. Then, when the next record drops, the ledge W allows the arm lever to return as far
next record can be put into place. In general, all systems use eccentric grooves, eccentric ledges, or eccentric screw mechanisms to make the pickup arm move through the proper motions. Some systems use a double groove on the cam, one groove for 10-inch and one for 12-inch records, and have arrangements whereby the arm crank finger can be switched from one groove to the other. Basically, however, all of them go through the actions we have just demonstrated.

**RECORD-DROPPING SYSTEMS**

So far, we have learned how the pickup arm is moved in and out and how it is raised and lowered. Next, let's see how records are fed one at a time from the storage system into the playing position. In general (except for a few complex types that play both sides of records), all changers made today drop records from storage above the turntable. The differences between them are in the means of separating the bottom record from the group and of supporting the stack.

There is probably more difference between changers in this particular item than in any other. Basically, there are two methods of separating the bottom record from a stack so that it can be put into playing position. In one system, support shelves originally hold up all the records. As the changer goes into operation, a set of knives is inserted in the record stack between the bottom record and those next above it. Then the supports are withdrawn, allowing the bottom record to drop onto the turntable. The knives then support the group.

Once the bottom record has been dropped into playing position, the support shelves are returned and the knives withdrawn, allowing the record stack to drop down onto the shelves.
The system is now ready for the knives to separate the bottom record of this stack on the next playing sequence.

In the other system, the bottom record is pushed off a supporting ledge, which then catches the remaining records.

Let's now turn to several typical changers and see just how they work. We can divide them into single-post, two-post, and three-post types.

**SINGLE-POST CHANGERS**

Figs. 19 and 20 show pictures of two basic single-post record changers. On these changers the records are supported by an offset ledge on the center spindle and by a single side post or platform. In the style shown in Fig. 19, the spindle is straight; the one shown in Fig. 20 has a “bent” spindle.

**Straight-Spindle Types.** Fig. 21 shows more details of a single-post straight-spindle changer. A section of the spindle at the center of the turntable is cut out to form a shelf. At the rear of this shelf is a guide trigger that makes the records move in the direction of the support head as they feed down the spindle. The records are thus supported at their center hole by the shelf on the spindle and at one outside edge by the support head.

![A single-post, bent-spindle record changer.](image)

When the record that dropped finishes playing, the cycle is repeated and the next record is dropped.

In this particular system, the adjustment for 10- or 12-inch records is made by rotating the support head. By com-

![Details of a straight-spindle changer.](image)
the 12-inch ledge is the right distance from the spindle to accommodate 12-inch records.

Some changers of this sort have a link mechanism down the support post column so arranged that rotating the support post head also adjusts the mechanism underneath to make the pickup arm land at the proper place for a 10-inch or 12-inch record. In others, it is necessary to set the support head and then throw a switch to adjust the landing point of the arm.

When the records have all been played, the support post is turned to a neutral position in which the shelves are out of the way. (Incidentally, this

paring the distances from the support post to each of the record ledges in Fig. 21, you will see that the ledge marked 10 is spaced farther from the post than the ledge marked 12. Therefore, the 10-inch ledge extends closer to the spindle, and is just the right distance away for 10-inch records. When the support head is rotated 180°,

is the position to which the support post is turned when the record changer is played manually.) Then, the stack of records is lifted up the spindle. On most models, the guide trigger on the back of the spindle slides into a slot in the spindle and thus out of the way as the records are lifted upwards.

Another style of single-post straight-spindle changer uses an eccentric cam to take the bottom record off the stack. Details of its construction are shown in Fig. 22. The spindle is hollow and has within it a drive shaft to which is fastened the cam. This drive shaft is offset toward the rear of the spindle

Three of the many methods of actuating a trigger to push a record off the ledge of a support head. At A, the cable $C$ is pulled by an arm traveling in a groove on the main cam, and this motion forces the trigger $T$ to protrude. At B, the cam $C$ is rotated by a shaft that runs down the support post; this forces the triggers $T$ to protrude because of the eccentric cut of the cam. At C, the gear $G$ is rotated to force the triggers $T$ to protrude, then the direction of gear rotation is reversed to pull them back in. The types at A and B are pulled back in by springs.

FIG. 22. Details of a cam system of aligning records with the spindle.
so that the cam will move in an eccentric fashion with respect to the spindle.

When the cam is in the position shown in Fig. 22A, it is lined up with the bottom portion of the spindle. It then acts as the top of the spindle to form a shelf upon which the records rest. As in the system just described,

The eccentric cam drive in Fig. 22 is attached to the gear G. This is driven here by the drive H, that in turn operates from a groove in the main cam. The drive works in one direction, then runs back to return the cam to its initial position.

the guide trigger directs the records, as they move down the post, onto the shelf furnished by the cam and onto the lip of a support head.

Let's suppose records have been loaded onto the mechanism as shown in Fig. 22A. When the changer first starts to operate, and the pickup arm is out of the way, the cam above the spindle is rotated by its drive shaft to the position shown in Fig. 22B. This brings it directly under the center hole of the bottom record, which then drops over the cam. The cam is not quite as thick as a standard record, so no more than one record can get onto it. The cam then rotates back to the position shown in Fig. 22A, dragging the bottom record with it and off the lip of the support head. When the cam is lined up with the spindle again (Fig. 22A), the record drops down the spindle to the turntable, and the next record is supported by the cam.

The storage mechanism of this type of changer, like that of the one previously described, is adjusted for record size by rotating the support platform.

The Bent-Spindle Changer. Fig. 23 shows a variation on the single-post changer. The spindle is like the others in that it has a shelf and usually a guide trigger. However, it has a bend in it that permits a somewhat different construction and action. The records feed down the spindle and over the trigger so that the shelf on the spindle supports them. The outer edges of the records are supported by a record head, which has a shallow notch into which the bottom record fits. The actuating means for getting the records to feed down the spindle is in the record head itself—the head and its support post move toward the spindle. When the head moves forward, the bottom record is pushed forward also by the back edge of the notch in the head; the rest of the records, however, slide back along the platform in the head just above the notch. When the center hole of the bottom record lines up

FIG. 23. A bent-spindle type.

with the spindle, the record feeds down the spindle. In going around the bend of the spindle the record is pulled away from the record head so that it falls free onto the turntable.

The next record then drops down on the shelf of the spindle. At the end of the playing cycle, the record head moves back away from the spindle, allowing the bottom record to drop into the notch in the head. It then
moves forward again, pushing the next record off the spindle shelf.

In the type shown in Fig. 23, the record head is revolved to play 12-inch records. Sometimes a switch or button must be actuated to cause the pickup arm to drop in the right position with this system.

![Diagram of bent-spindle type changer](image)

**FIG. 24.** This bent-spindle type is an inter-mixing changer.

To prevent the records from tilting, there is a hinged weight on the record shelf that is dropped on top of the record stack. (Most single-post changers use such a weight.) This weight holds the records steady while the shelf moves back and forth underneath the stack.

To load this changer, the weight is moved out of position, the stack of records is fed down over the top portion of the spindle, and the weight is then replaced. To remove the records, the weight, which protrudes somewhat, must be rotated out of position, or the entire record shelf must be turned 90° to clear the record stack as it is lifted off the spindle. In many of the bent-spindle types, the spindle can be lifted out of its socket so that the records can be removed without having to feed them up the spindle.

Some of the bent-spindle changers can play records that are mixed in size. Fig. 24 shows one system. The record head contains a trigger. When the records are placed on the storage portion of the spindle, 10-inch records will rest on the front edge of the head, ahead of the trigger, and 12-inch records will extend over the trigger. Let’s suppose we have the stack as shown in Fig. 24. When the first record, a 10-inch one, is to be played, the record head moves forward; the trigger, which is a square protrusion on the head, strikes the edge of the record and forces it off the shelf on the spindle.

When this record drops, the next 10-inch record drops down in front of the trigger. It, too, is pushed off by the trigger on the next operation of the changer. The next record, however, is a 12-inch one; therefore, it drops on top of the trigger instead of in front of it. The trigger is pivoted (see Fig. 24), and the weight of the record on top of it pushes it down flush with the top of the record head. As shown by the dotted lines, this raises the rear end of the trigger so that it is above a stop that up to now has prevented the record head from moving more than a certain dis-
FIG. 25. An inter-mixing bent-spindle changer.

tance from the spindle. When the rear end of the trigger is able to clear this stop, the record head is able to move considerably farther from the spindle—so far, in fact, that the trigger is brought out beyond the edge of the 12-inch record. The weight of the rear end of the trigger restores it to its original position once it gets out from under the record, so, when the record head moves forward again, the trigger is behind the edge of the 12-inch record and pushes the record off the spindle shelf.

This mechanism also automatically sets the landing position of the pickup arm for 10-inch or 12-inch records. The motion of the record head sets stops that control the pickup arm crank, with the result that the landing position of the pickup arm depends on whether the record head has moved back for a 10-inch or for a 12-inch record.

Fig. 25 shows a picture of a unit of this type. Notice the overlay arm that lies over the top of the records and straddles the spindle. This arm is necessary in this system to keep the records level while the record head moves back and forth and also to provide the force to keep the records moving down the spindle. It serves a purpose somewhat similar to the weight in the system shown in Fig. 23. However, because of the wider motion of the record head here, the overlay arm must be accurately positioned.

TWO-POST CHANGERS

Fig. 26 shows one of the two-post record changers. In this style, the center spindle is used only for guiding the record down to the turntable—it has no shelf on it.

The record is supported at two points by record-holder shelves that are attached to the record-holder posts.

When a record is to be dropped, the pickup is elevated and moved out of the way. Then the record-holder posts begin to rotate. Each of these posts carries a “knife” just above its rec-

FIG. 26. A typical 2-post changer, the RCA U-128.
The support posts may be rotated by means of a belt or gear system from the main cam. Here is a simple gear type; the arm A is moved from side to side by the eccentric groove in the main cam. It rotates gears C and D that are at the base of the support posts. Hence, the posts and accompanying knives are rotated about 180°, then are returned to their resting positions.

ord-holder shelf. This knife is a sharp-edged shelf that is spaced approximately the thickness of the average record above the record-holder shelf. Therefore, as each knife comes around and contacts the record stacks, the pointed tip of the knife is in just about the right position to go in between the bottom record of the stack and the one next above it. The mechanism rotating the knife is usually either spring loaded or allowed to have considerable play so that the knives can adjust themselves and slip in between the bottom record and the stack.

As the knife penetrates the record stack, continued rotation of the record-holder post will place the knife under all the records except the bottom one, and the record-holder shelf will be rotated completely out of the way. When this happens, the bottom record no longer has anything to support it, and the remainder of the record stack is supported by the knives on the two posts. Therefore, the bottom record drops onto the turntable. Before the pickup arm is placed in the playing position, the record holder posts rotate back to their normal positions. This withdraws the knives and drops the stack of records onto the record holder shelves, thus completing the changing cycle. The thickness of 10-inch records is somewhat different from that of 12-inch records. Although the knife is so loose that it will usually find the proper spacings and go in between the records, it is always possible for the knife to strike the edge of a record and cut into the record rather than separate it from the stack.

On some changers, the spacing is adjustable. Fig. 27 shows the details of one such system. The small screw marked G protrudes through the record-holder shelf. When 10-inch records are on the shelf, they do not extend over the shelf far enough to reach this screw. However, 12-inch records will lie on the screw and depress it.

When 10-inch records are on the shelf, the knife spacing above the shelf is that of the average 10-inch record. When the thicker 12-inch records are on the shelf, the screw G is depressed, and a lever arm connected to it raises the knife slightly so that the spacing between the knife and the record-holder shelf increases enough to clear the records.

This form of storage mechanism will usually permit the intermixing of 10-
and 12-inch records. Some of them require setting for the record size, in addition to setting the index lever to the proper position to control the pickup arm landing position. Others, of which the one we are discussing is an example, determine the record sizes automatically.

The mechanism used to do so in this changer is shown in Fig. 28. The lever 17 stands beside the turntable. Ten-inch records drop from the record support shelves to the turntable without striking lever 17. When this happens, the pickup arm automatically comes in for the 10-inch position.

However, a 12-inch record, because of its greater diameter, strikes lever 17 as it falls. This pushes the lever to the right in this drawing, moving its end out of the way of the pin marked V. This allows the pickup arm crank to move to the proper position for a 12-inch record.

Another form of two-post changer is shown in Fig. 29. In this, the spindle has a crook or hump in it. There are two heads on which the records are supported. When the device is actuated, the spindle is rotated by the main cam through a gear system.

In its initial position, the hump is at a point halfway between the two support heads. As it rotates, however, the hump approaches the left-hand head, thus forcing the bottom record into the slot in the left-hand head. This pulls the other edge of the record from the support ledge of the right-hand head and allows it to drop on the right-hand fork. (This fork is a movable support bar.) Then, as the spindle continues its rotation, the hump pushes the record to the right, into the slot provided by the fork on the right-hand head. This pulls the record from the supporting shelf on the left-hand head, allowing this edge of the record to drop similarly on the left-hand fork. The spindle comes to a stop in its neutral position, and the bottom record is now supported at both edges by the forks just below the two heads. The next records are down on the support shelves of the heads. Just before the record is to be played, the two forks are withdrawn by a linkage down the support post, allowing the bottom record to drop down the spindle.

Thus, the operating principle of this changer is that rotation of the spindle moves the bottom record from its support posts onto two forks, which are then withdrawn to allow the record to drop onto the turntable.

THREE-POST CHANGERS

The three-post changers are usually of the knife-blade type like the one shown in Fig. 26, except that they have three posts 120° apart instead of
two posts 180° apart. The same cycle of operation is used—three knives go in between the bottom record and the remainder of the stack, then the support shelves are withdrawn to allow the bottom record to drop.

It is possible to make either the two or three-post changers intermixing types by causing the record to strike a lever as it drops, or by using a trigger mechanism on the support shelves that is depressed by 12-inch records.

Servicing Record Changers

In the preceding section, we have analyzed the operations of changer mechanisms separately because, in general, you will find that only one thing is wrong when they are out of adjustment. That is, the trip mechanism may fail to operate or operate too soon, the pickup arm may not be picked up high enough or may not be let down low enough, the records may not be dropped, or more than one may drop, and so forth. Of course, it is always possible for the mechanism to jam completely, stopping all operation. Such jamming can be the result of a failure of some part or the result of mishandling of the changer. Most particularly, jamming can occur when someone moves the pickup arm while it is “in cycle”—that is, while the mechanism is trying to manipulate the pickup arm itself. (Moving the pickup arm while it is in cycle may also throw the changer out of adjustment.) Generally, however, the trouble is in only one portion of the change cycle. Therefore, when you are called on to service a record changer, first determine exactly what it fails to do or does incorrectly. You can then tell what adjustments need to be made to put it back in proper operating condition.

Even though you know just how a changer should perform a certain task, you may still have to spend a considerable period of time watching it go through its cycle again and again before you can see exactly which lever, gear, crank, or arm is incorrectly performing its duty. To save time, you should have all the information you can get on any particular record changer that you have for service. Fortunately, because of the complexity of changers, the manufacturers publish rather complete service manuals. These may be obtained directly from the manufacturers, like any other service information.

In addition, you can obtain much valuable service information in the Service Manuals of Rider and Howard
W. Sams. These Automatic Record Changer Service Manuals cover many different models. They are available from radio supply houses and local wholesalers.

However, if you have to service a changer on which you do not have service information, and you cannot get this information in time to complete the job, all you can do is run the changer through its operation several times and watch it carefully. By locating the apparatus that controls the faulty action, you will usually be led right to the proper adjustment.

Before you can seriously consider the servicing of automatic record changers, you must have some means available for supporting the changer on your workbench so that you can see underneath it as well as above it.

For example, the pickup arm is adjusted to land in the blank space at the outer edge of a record. Then, if the changer is level, a slight spring pressure or gravity feed will cause the needle to move over toward the spindle sufficiently to engage the first playing groove. If the changer is not level, the needle may not move into the playing groove, or may even jump the other way—completely off the record. Similarly, in cases where the spring tensions are critical, you cannot have the changer up on edge without putting excess tension on some springs or releasing others so that the changer cannot perform satisfactorily.

There are several support jigs for record changers that may be purchased from the radio supply houses. A typical one is shown in Fig. 30. If you prefer, you can make a jig like the one shown in Fig. 31. Of course, the spacing between the posts will be proper only for certain types of changers, so a home-made gadget of this kind is not quite as flexible as are some of the adjustable commercial jigs. In any case, the jig must fit the changer, must support it securely, and must hold it level. To be useful, the jig must hold the changer above the workbench high enough for you to see and adjust the changer mechanism from underneath. You don't have to put your head under the changer to watch its
operation—you can always use a mirror to let you see underneath—but you will usually have to get under it yourself when you make adjustments.

When you are servicing a record changer, it isn’t always desirable to have electric power applied. The changer may run through its operation too fast for you to watch everything sufficiently, or there may be some jamming condition that could actually damage parts if power is applied. For this reason, it is frequently necessary to disconnect the motor from the power line and rotate the turntable by hand to drive the changer mechanism slowly through its cycle of operation.

Sometimes you will find that support straps, mounting boards, or other objects obscure the view of the parts you want to watch. In such a case, removing the turntable may let you see the mechanism underneath. However, this won’t always prove helpful—in some instances the motor board is solid, and only the spindle comes through the board. If so, removing the turntable does not permit you to see underneath at all.

Now let’s describe a few basic troubles and learn their remedies before going on to examples of manufacturers’ service data.

**NON-STANDARD RECORDS**

A great deal of the trouble experienced with record changers is caused by the fact that the records involved are warped or are not standard in some way.

The standard 10- and 12-inch records made by the reputable manufacturers are all held reasonably close in their sizes. They are of the proper thickness and diameter, and in general the edges of the records are smoothly rounded. However, it is always possible for even a standard record to be outside tolerance in some way, and many of the records made by smaller companies are not standard at all. Here are a few of the troubles caused by non-standard records.

**No Eccentric.** You will find that many of the older recordings either do not have an eccentric groove at the center of the record or have one so shallow that it cannot trip some of the changer mechanisms. This is particularly true of some of the earlier classical recordings. Unfortunately, these classics are frequently the very records that appeal to owners of record changers. They may not realize that the eccentric groove is necessary for the trip, so you may very well get a call to repair a changer because it does not trip, when actually the trouble is this lack of an eccentric groove. Be sure to find out from the customer on such calls whether the mechanism fails to trip only on certain records—he may have noticed this characteristic, which will lead you at once to the trouble.

There are a few records, mostly foreign ones, that have the eccentric groove but carry the recording groove too close to the center spindle. This is perfectly all right as long as the trip mechanism operates from the eccentric groove. However, if used in a changer having a positional trip (in which the trip is actuated as soon as the pickup arm is brought within a preset distance from the spindle), these records may tend to trip too soon. If you find that the changer has a positional trip, check the manufacturer’s instructions to learn the distance from the spindle the device should be set. If it trips at the proper distance, the record is at fault.

**Thickness Variations.** Records that are too thick or too thin can cause trouble in the mechanism designed to feed the records from the storage sys-
tem to the turntable. For example, in all systems like that shown in Fig. 21, the record must slide under the guide trigger. A thick record may not be able to feed through here. On the other hand, if two very thin records get together, they may both try to feed through. The result could be a jamming on the shelf, or they both may drop at once. In the case of the thick record, of course, jamming results.

**Improper Diameter.** If the record is too large or too small in diameter, it may not feed through the storage mechanism properly, and naturally the pickup arm will not land properly on the record. Of course, a record that is off standard size this way is usually easy to detect because it can be directly compared with others in the stack. It is less easy to detect records that are thick or thin.

**JAMMING**

When a record changer is jammed, it is not advisable to try to force it to continue its changing cycle. Of course, just what is wrong depends upon whether the jam was caused by a defective part or by someone's trying to force the mechanism. In general, however, it is possible to clear jamming by rotating the turntable backwards. Of course the power must be shut off when you rotate the turntable by hand in this manner.

If the changer has a damaged part, it will jam at the same point in its cycle each time. In such a case you can rotate the turntable in the proper direction, by hand, until the jam occurs. You will then be better able to see just what has gone wrong.

Incidentally, it is possible for jamming to occur because the changer is not level or because it has shifted in its position in the cabinet. It may be that all the levers and gears originally cleared the interior of the cabinet but that a shift in position has permitted some lever to strike the cabinet. You can check the levelness of the changer with a carpenter's level.

**GENERAL DEFECTS**

Ordinarily, a changer is in need of repair because of the normal wear of some of the parts in it. This wear may be of such nature that an adjust-
ment, provided by the manufacturer, can clear up the difficulty. However, once a bearing becomes so loose that the lengths or positions of levers vary during the changing cycle, it will be necessary to make a major repair or to replace the changer.

Another common source of trouble is fatigue in the springs, of which changers have many. With use, these springs will eventually stretch so that they do not provide the proper tensions. When this happens, it is usually necessary to replace the offending spring; once in a while, however, you may find that the manufacturer has provided an adjustment for the spring by attaching one end of it to a movable terminal.

Incidentally, much of the trouble that is encountered with record changers comes about because of lack of proper oiling. Although oiling instructions usually accompany a changer, few owners remember to follow them—perhaps because the necessary oilings are infrequent and therefore easily forgotten.

The manufacturer’s instructions should be consulted when it is discovered that oiling is needed. It is usually safe to oil any metal-to-metal bearing, although often a light grease is indicated instead of regular oil.

There are some spots about a record changer that should never be oiled. Certain tripping mechanisms that depend upon friction may or may not require oiling, depending upon the materials used in them. For example, one changer has a cork washer to provide friction. Oil on this washer completely upsets the operation of the trip mechanism.

Similarly, it is desirable to keep oil away from all rubber parts. Many drive mechanisms are friction types, using rubber-tired pulleys. It is important to keep oil away from the rubber, but nevertheless to oil the bearings of such pulleys.

Pickups and Their Servicing

The pickup device itself has nothing to do with the automatic record changer other than the reproduction of the recording as an electrical signal. Nevertheless, when the output from a changer is distorted or sounds tinny or when there is no output at all, the serviceman is certain to get a call.

There are three types of pickups in common use today—the crystal, the magnetic, and a variable reluctance type of magnetic pickup. Let’s study their operation briefly.

As you know, all standard recordings used in the home are the result of modulating a groove so that it has “wiggles” from side to side in it. The record player needle fits in this groove and is forced to follow the variations. To reproduce the recorded sound, we have to have some means of translating this mechanical side-to-side motion of the needle tip into an electric signal.

Crystal Pickups. Probably the most widely used pickup today is one containing a crystal element. These units are inexpensive, easy to replace, and give a high output.

Fig. 33 shows the operating details of a crystal cartridge or pickup. The
FIG. 34. Several typical crystal cartridges. They vary in physical size, method of connecting leads, and whether or not permanent needles are used. Some have greater outputs, and others offer better fidelity.

phonograph needle is held in a chuck by a screw. In turn, the chuck is clamped to one end of the crystal element. The opposite end of the crystal is mounted in the case so that it cannot move. Now, as the record grooves force the needle to move from side to side, the chuck is twisted. This twists the end of the crystal, applying a mechanical stress to it that causes it to generate a voltage, which appears on its opposite faces. Foil plates on the crystal surfaces pick up the voltage and feed it through the leads.

The physical appearance of crystal cartridges vary somewhat, as shown by several typical ones in Fig. 34. However, they all operate on basically the same principle—the only differences are in the housings, the methods of connecting the cable to the cartridge unit, and the styles of needle mountings. We shall go into needles a little later.

**Magnetic Pickups.** Fig. 35 shows the details of the operation of a magnetic pickup. Essentially, this consists of a permanent magnet, a coil, and an armature that can be actuated by the phonograph needle. As shown in this figure, motion of the needle from side to side directs the flux in opposite directions through the armature. Therefore, since the coil is essentially around the armature, the flux variations in the armature cause voltages to be induced in the coil.

**Variable Reluctance Pickup.** The variable reluctance pickup, a typical example of which is shown in Fig. 36, is a variation of the magnetic type. However, the difference in the amount of needle pressure needed is appreciable. A magnetic head must be heavy
so that the head will not vibrate when the armature does; consequently, it presses rather heavily on a record and wears it out quickly. In the variable reluctance pickup, however, the vibrating device is a very tiny, lightweight reed; consequently, the head can be made light enough to cause practically no record wear.

This change in weight comes about because the coil no longer picks up the flux variations directly from the armature. Instead, the armature in the variable reluctance type merely varies the reluctance in a magnetic path, and the coil picks up its energy from the variations in flux in this path.

As Fig. 36 shows, there are two paths for magnetic flux in this pickup. In each path, flux flows from the magnet through half of the pickup coil and through the reed back to the magnet. The position of the vibrating reed in the air space between the two coil cores determines how much flux flows in each half of the coil at any instant. When the reed is near one of the cores, the flux through the half of the coil wound around the other core decreases. The two halves of the coil are wound in opposite directions around their cores, however, so a decrease in flux through one has the same effect as an increase in flux through the other. This doubles the effect of the movement of the reed in producing a signal.

**PICKUP DEFECTS**

**Crystal Pickup.** A crystal cartridge is rather easily damaged. If the pickup head is ever dropped, it is quite likely that the crystal will be cracked. Excessive motions of the needle may also crack the crystal. Moisture can also destroy a crystal. So can heat: a crystal should never be allowed to get hotter than 110°. Many pickups are ruined because the head is allowed to remain so long in the path of sunshine streaming in a window that the crystal becomes overheated.

A defect in the crystal practically always shows up as a severe distortion accompanied by weak volume. When you find a changer with such characteristics and have determined that the audio amplifier is not at fault, it is advisable to replace the crystal. Generally, the crystal is held in the pickup arm by two screws or by a simple clamp arrangement. Simply remove the defective unit and install a good one of the same type in its place.

When you replace a crystal, be careful not to overheat the terminals if you must solder a cable to the cartridge. Excessive heat from the soldering iron will destroy the crystal. Therefore, it is best to have the pickup cable terminals well tinned and coated with solder so that you can sweat the cable end to the crystal terminals quickly.

**Magnetic Pickup.** The armature of a magnetic pickup sometimes strikes the pole pieces, either because the rubber damping blocks have worn out or because the armature has shifted its position. When this happens, the device will chatter, and the output will be severely reduced. Occasionally an armature moves too far enough to stick to one of the pole pieces by magnetic attraction. When this happens, of course there will be practically no out-
put. No output may also be the result of an open coil.

It is impractical to repair the pickup; the only thing to do is to replace the head when you are sure it is at fault.

Incidentally, with any type of pickup, it is well to be cautious about the cable that connects the pickup electrically to the amplifier. The cable almost always consists of a center wire surrounded by insulation, which is surrounded in turn by a braided shield that is grounded to prevent hum pickup. This shield is one of the conductors. Rather often the insulation between the center conductor and the shield wears through, permitting the cable to short. This is particularly likely to occur at the point in the cable where it leaves the end of the pickup arm and goes down underneath the motorboard. The moving of the arm back and forth twists and untwists the cable at this point so that the insulation may be mechanically worn out. It is well to examine and check cables for both opens and short circuits before condemning the pickup.

The position of this cable is such that oil may get on it if anyone is careless in oiling surrounding parts of the changer. This will speedily destroy the rubber insulation, permitting the cable to short-circuit. Naturally, this will reduce or kill the output.

**PHONOGRAPh NEEDLES**

As you will realize, it is desirable for a needle to be as permanent as possible. If the needle wears excessively, it may not even be able to play a stack of records. This is particularly true of some of the non-metallic needles, such as the thorn or cactus types.

Needles should be made so that they fit the grooves of the record. If the needle point is too small, as shown in Fig. 37A, it can skid from side to side in the groove and thus introduce false frequencies. It may even ride out of the groove in such cases. In addition, a narrow needle may strike the bottom

![FIG. 37. Effects of needle-point sizes.](image)

of the groove and pick up a great deal of noise from the imperfections there.

On the other hand, if the point is too broad, as shown in Fig. 37C, it cannot fit down into the groove and will tend to escape and permit the pickup head to slide across the record. When the needle fits the groove properly (Fig. 37B), it will follow the modulations in the record grooves without introducing other frequencies and without escaping from the grooves.

The harder the needle point, the more important it is that it have the correct needle shape initially. Unless a hard point is made with extreme care, it may have imperfections that will wear down the walls of the record groove and so destroy the fidelity of the recording.

The standard steel needle is usually only an approximation of the right shape. However, it is made of a material sufficiently soft so that the abrasive contained in a standard record will quickly wear the needle down until it fits the groove reasonably well. This wearing causes shoulders to build up on the needle, however, soon shaping it so that it can damage records. For this reason, a standard steel needle should be replaced each time a record is to be played. Obviously, this makes such needles rather impractical for record changers.

Longer playing steel needles are
made for record players. These needles are tipped with alloys that make the tips very hard. Then the needles are carefully selected so only those having the correct shape are sold. For permanent-point needles, it is well to remember that it is a good idea to buy those made by a reliable manufacturer. Any imperfections will remain for the life of the point and will cause wear of the record grooves. Most such points are shadowgraphed by the manufacturer, which means that an enlarged shadow of the needle point is thrown upon a screen for examination. Any needle with an imperfection is rejected.

In addition to the steel needles, there are available needles tipped with sapphire or diamond. These are the longest playing of all. Of course, these needles are rather expensive; if handled carefully, however, they will last for many thousands of playings.

The standard needle is straight and has a relatively thick shank. The more rigid the shank of the needle, the better it will transmit high audio frequencies—including scratch noises. Many people find the elimination of the scratch more desirable than good fidelity; to do this, some needles are made thin and flexible; others are coated with paint; and, finally, many actually have a bend or knee in them. All of these changes in the basic shape of the needle result in a reduction of the high-frequency response and a corresponding reduction in scratch noise.

Most modern record changers have permanent built-in needles. If anything is wrong with the needle, the entire pickup cartridge must usually be replaced. There are a few exceptions in which it is possible to replace the needle, however.

One exception is shown in Fig. 38.

This is a view of a cartridge used on certain RCA changers. The sapphire playing tip is held in a tiny socket by rubber cement (such as Goodrich Plasticone). If this needle needs replacement, it may be grasped firmly with a pair of tweezers, given a few turns to loosen the cement, and then pulled out.

![Fig. 38. How to replace the sapphire tip on certain RCA changers.](image)

A new sapphire can be re-inserted, with just a drop of the rubber cement on it to hold it in the socket.

**RECORD CARE**

Many people do not know how to take proper care of their records. Whenever you find that the record collection of one of your customers is not in good condition, you can do him a service and create some good will for yourself by passing along these hints:

Records will become noisy if they are allowed to collect dust. They must be kept clean. It is best to store records in their original envelopes or albums, and then, if they do collect some dust, to brush them with a record brush.

It is necessary to store records properly to prevent them from warping. As you have learned, a warped record can easily get into trouble with the changer mechanism, because it may be impossible for the mechanism to separate warped records. Records should never be left resting on the support shelf of a record changer for a long time. They should always be stored carefully.
Motors and Their Servicing

The motors used on record players and changers must maintain their speeds accurately. The two basic motor types used in record changers—the synchronous motor and the induction motor using a governor—meet this requirement.

There are several forms of the synchronous motor. Most of them are of the eddy current type, but you will occasionally encounter shaded-pole or capacitor motors. Any of these are relatively constant in speed as long as they are not overloaded.

In general, little goes wrong with the motor itself as long as it is oiled properly. Once in a great while, you may find a changer in which the motor has a burned-out winding, but this is very rare. More commonly, any trouble will be with something related to the motor—the on-off switch may be defective or the motor may be overloaded. On the induction types, the speed governor may cause trouble.

A simplified drawing of a typical speed governor is shown in Fig. 39. This device contains a shaft that is coupled to the motor. Two weights are connected by springs to a collar on the end of the shaft. A wheel is also connected to the weights by springs. Let's see how the device works.

As the shaft rotates, the weights are thrown outward by centrifugal force. The pull exerted on the springs by the outward movement of the weights pulls the wheel to the right, bringing it up against two friction pads made of felt. As soon as the wheel strikes the friction pads, it is slowed down by the friction; since the wheel is connected to the shaft through the springs and collar, the shaft is slowed down also. Thus, the motor is retarded if it attempts to run faster than the speed for which the governor is set. This keeps the motor running at a fairly constant speed as long as nothing happens to make it run too slowly; the governor has no action that will speed up a slow motor. The speed at which the governor will start to slow down the motor can be adjusted by moving the friction pads toward or away from the collar; the farther they are from the collar, the slower the speed at which the governor acts.

These governors will not maintain the speed properly if the friction pads wear down or become hard because of a lack of oiling. Watch for this if the speed is uneven.

DRIVE MECHANISMS

The driving force of a phonograph motor is applied to the turntable either at the center spindle or at the rim. Fig. 40 shows one form of spindle drive, in which the motor drives a gear mounted on a shaft secured to the spindle.

When the center spindle is driven this way, considerable power is needed to get the turntable started. Once started, however, the inertia of the rotating mass of the turntable tends to keep it going.

Many changers use a less powerful motor and drive the turntable from its rim. In systems of this kind, as shown in Fig. 41, the motor turns a small
FIG. 40. A drive system that operates through the center spindle. This photo shows a record player; a changer would use a different spindle and would have the changer mechanism. These are omitted here to show the drive more clearly.

drive pulley. This is held against the rubber-tired idler pulley, the edge of which is against the rim of the turntable. Incidentally, it is necessary to use either gearing or an idler pulley system of this kind so that the motor can turn at a fair rate of speed, yet maintain the standard 78 revolutions per minute for the turntable.

There is another advantage to the rim drive system. In a center spindle drive, the motor is more or less directly connected through a gearing system to the turntable. Vibrations produced by the motor can travel to the turntable and be picked up by the pickup. On the other hand, with the rim drive system, the motor can be flexibly mounted in a spring suspension. Springs keep its pulley in contact with the idler pulley. With this arrangement, any variation up or down in the drive pulley does not transfer any motion through the idler pulley to the turntable.

Spring suspension of the motor is important to keep down what is called turntable rumble, a frequent cause of customer complaints. This noise consists of a low-frequency rumbling sound, somewhat similar to hum, that can be heard only when the record is being played. Generally you will find it is caused by the fact that the motor is no longer suspended on springs—someone may have tightened the mounting so much that the springs are no longer effective, or they may be weakened so that the motor can jar the motorboard and, through it, the turntable. Sometimes this condition is made worse by the fact that the entire

FIG. 41. Details of rim-drive systems.

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motorboard is not sufficiently spring-mounted, either because someone has screwed down the changer too tightly to the cabinet or because it has shifted in position so that the motorboard is bearing against the cabinet wall. Sometimes flexible couplings are used between the motor drive and the rest of the drive mechanism to cut down some of this transfer of noise.

Improper speed is perhaps the most common complaint involving the motor system and drive mechanism. A synchronous motor never runs too fast, but it may run too slowly if the motor or the changer mechanism needs lubrication. Usually, careful oiling and greasing will clear up a trouble of this kind.

An induction-type motor that has a governor usually also has an adjusting screw by which the speed of the motor can be changed. A typical example is shown in Fig. 42. To get the motor to the right speed, a stroboscopic disc is placed on the turntable. This is a disc having a special pattern on it; when the disc is observed under a light operated from 60-cycle power, the pattern will apparently stand still if the motor is turning at the proper speed but will appear to revolve if the motor is going too fast or too slow. The proper turntable speed is secured, therefore, by turning the adjustment screw until the pattern appears to be motionless. These discs are available from radio supply houses and from many record dealers.

Remember that improper line voltages or variations in line voltage may affect the speed of the motor. Of course, if any foreign particles have lodged between the armature and the field pole pieces, or in any of the gearing, the motor may vary in speed or may even jam and not run at all.

Overheating is another motor trouble. This of course can be the result of insufficient lubrication, but may also be the result of bearings that are too tight, of short-circuited coils in the motor, or of an excessive load on the motor, such as may be produced by improper lubrication of the drive mechanism or by off-center mounting.

Microgroove Records

Recently there have been introduced two different series of records having very fine grooves (generally called microgrooves because of their thinness); these records are intended to rotate at slower speeds than 78 r.p.m.

**LP Records.** One microgroove type consists of 10-inch and 12-inch records designed to be played at 33 1/3 r.p.m. The slower speed and increased number of grooves permit these records to hold much more recorded material; it is possible to get an entire symphony on both sides of one 12-inch record, whereas four to eight standard records would be needed for the same symphony.

The advantages of such a long-playing record are obvious—one no longer has to put up with the unnatu-
(long-playing) records have been developed. Of course, the new speed requires a change in gearing or size of idler pulley, and the fine record groove requires a special fine-tipped needle and light-weight tone arm.

Combination changers can handle both the LP and the standard records (as long as they are not intermixed) by having a switch to change the turntable speed, and either using separate tone arms or a switching arrangement that will change the arm weight and needle size. The latter is obtained generally by using a dual needle and a tilting or revolvable crystal so that the proper tip is put into play. However, there is a new 7-inch microgroove record for popular music; this requires an additional switch to set the pick-up arm landing position, plus an extension spindle or platform to provide proper storage support for this small-size record.

45-R.P.M. Type. Another microgroove system uses a record 6\(\frac{3}{8}\) inches in diameter and a turntable speed of 45 r.p.m. These records are radically different in that they are designed to operate on a changer having the “works” in the center post. Hence, the records have a center hole 1\(\frac{1}{2}\) inches in diameter, and are made thicker in the label area to provide a space for the separating knives.

These records were introduced along with the unique changer shown in Fig. 43, but some of the recent changers will handle not only these records, but also the LP and standard ones by having interchangeable center spindles, three speeds, dual needles, and a new landing position for the tone arm.

Fig. 43. A photo and two sketches of the player designed for 45 r.p.m. microgroove records. The upper sketch shows how the separator knives move in between records while the support shelves are withdrawn simultaneously to allow the bottom record to drop. In the lower sketch, the support shelves now hold the record stack and the knives are withdrawn into the spindle.

General break in the music that occurs when a record is changed in a conventional changer.

Both single-record players and changers designed only for these LP
Lesson Questions

Be sure to number your Answer Sheet 48RH-2.

Place your Student Number on every Answer Sheet.

Send in your set of answers for this Lesson immediately after you finish them, as instructed in the Study Schedule. This will give you the greatest possible benefit from our speedy personal grading service.

1. What feature is provided on standard records to make it possible for the changer mechanism to regain control after the record has finished playing?

2. What are the three main types of trip mechanisms?

3. What is the function of the main cam in most record changers?

4. In the elevator system shown in Fig. 12A, how would you adjust for the condition wherein the pickup arm is not lifted sufficiently to play a record on top of a stack?

5. What is the function of the guide trigger in the spindle of a single-post changer?

6. What is the function of the knives in a 2-post or 3-post changer?

7. What two things can happen if the pickup arm is moved when a changer is in cycle?

8. What precaution must be taken when replacing a crystal pickup?

9. Why should one be careful to keep oil off the electrical cable that connects the pickup to the amplifier?

10. What two things may cause a governor-controlled motor to run unevenly?
INTRODUCTION TO PUBLIC ADDRESS

49RH-3

NATIONAL RADIO INSTITUTE
WASHINGTON, D. C.
ESTABLISHED 1914
STUDY SCHEDULE NO. 49

For each study step, read the assigned pages first at your usual speed, then reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind. Study each other step in this same way.

☐ 1. Introduction .................................................. Pages 1-6
   This section contains a brief discussion of the requirements and problems of public address systems.

☐ 2. The Decibel and Power Ratios .............................. Pages 6-9
   The uses of decibel units in p.a. work are discussed in this section.

☐ 3. Amplifier Specifications ....................................... Pages 10-15
   Here the meanings of the various specifications given in manufacturers' amplifier catalogs are discussed.

☐ 4. Power Supplies, Output Stages, and Drivers .............. Pages 15-25
   The general characteristics of these stages in p.a. equipment are described in this section.

☐ 5. Voltage Amplifier Considerations .......................... Pages 25-31
   This section contains general descriptions of the various kinds of input couplings, mixing arrangements, and tone-control networks used in p.a. amplifiers.

☐ 6. Typical P. A. Diagrams ....................................... Pages 32-36
   The schematic diagrams of two typical amplifiers are discussed in this section.

☐ 7. Mail Your Answers for this Lesson to NRI for Grading.

☐ 8. Start Studying the Next Lesson.
INTRODUCTION TO P. A.

Radio servicemen constantly have opportunities to take on profitable side lines. Of course, a man who has so much radio service work that he does not have the time to do anything else may be uninterested in any of these extra sources of income. However, radio servicing is a seasonal business—there is much more repair work at certain times of the year than others, and a means of keeping up the income during the dull season is desirable. Also, to the man who is not overloaded with service work, because of competition or the smallness of his community, these side lines represent a means of augmenting the regular service income.

As you might expect, these side lines usually involve electrical apparatus or electronic equipment in one form or another. For example, it is quite common to find that the local radio serviceman also repairs home appliances, such as irons, toasters, and lamps. In an industrial community, he may work on a certain amount of electronic control equipment.

A profitable and logical side line is public address. It is a logical field because it uses loudspeakers and other devices with which you are already familiar. Servicing such equipment is just as profitable as servicing radios is; furthermore you can make additional profits by installing and selling equipment if you wish.

A lack of information about public address equipment prevents many servicemen from taking advantage of this field. Also, in many localities the opportunities appear to be limited. However, in most cases, this lack of opportunity is entirely a result of the fact that no one has taken the time and made the effort needed to create a demand for public address equipment, because there have been too few men trained to recognize the usefulness of the equipment, to recommend the proper installation, and to install it. The wide-awake serviceman can increase his opportunities by seeing to it that more use is made of this equipment.

Whether future opportunities cause you to enter the field only part way—
to the extent of servicing or perhaps occasionally doing installation work—or whether you eventually decide to specialize exclusively in public address, you will find these Lessons helpful. They will present the important details you need to know to succeed in this field.

WHERE IS P.A. USED?

Public address (commonly abbreviated “p.a.”) equipment is known to most people only as a system used where large numbers of people are to be addressed. As examples of occasional or seasonal uses, p.a. equipment is being used more and more at circuses and carnivals, political conventions or rallies, and at special events such as county and state fairs, rodeos, etc. There are other places, such as airports, railroad and bus terminals, etc., in which year-round use is made of sound-amplification equipment.

In addition to these applications, in which the sound systems are primarily used for amplifying speeches or giving information, there is an increasing use of p.a. systems in the entertainment field. Sporting events require systems for making announcements. Lecturers and speakers at dinner meetings also use sound systems to amplify their voices. Dance music in ball-rooms is now commonly fed through p.a. systems; in addition, such systems are frequently used for amplifying the music of soloists or even full orchestras at concerts.

Moving from the field of gatherings brought together for specific entertainments or functions, we find that sound systems are beginning to be widely used to provide entertainment in many factories—music is being played for the workers and apparently increases production. Even further from the conventional use of p.a. systems are the installations in hotels and hospitals in which individual speakers in rooms are used to bring entertainment to the hotel guests or to the hospital patients more or less individually.

Similar to these are intercommunicators, which are basically amplifier units designed for communication between just two people or between small groups of people. Typical uses are for interoffice communication between an executive and his secretary or his department heads, for communication from a service desk to a service department in a store, and for communication from lunch counter to cook in a restaurant, to mention just a few.

As this list shows you, there are a great many possible uses for p.a. equipment, and therefore there are a great many p.a. systems already in existence. All of these systems have to be serviced from time to time. Furthermore, many new systems are being installed all the time as new uses for p.a. equipment are developed. There is, therefore, an increasing opportunity for the serviceman in p.a. work.

P.A. REQUIREMENTS

Now that you’ve seen what some of the uses of p.a. systems are, let’s see what requirements the equipment must meet in these applications.

The basic p.a. system is shown in Fig. 1. It consists, as you can see, of an input device (in this case, a microphone), an audio amplifier, and a
loudspeaker. All p.a. systems contain these elements. Many systems are more complex than this, having extra input devices (other microphones, record players, and occasionally radio tuners) and multiple loudspeakers, but basically they are all alike.

When such a system is used for addressing a large crowd, the chief requirement made of it is that it must have enough power to make it possible for everyone to hear. If music is to be played over the system, it must have at least a reasonably good fidelity of response in addition to sufficient power. If the music is intended for a critical audience, the fidelity of the system must be excellent. Let’s discuss these requirements more fully.

One of the first things that must be considered in planning a p.a. installation is how much power is necessary to cover the audience properly. This problem can be solved only by having some knowledge of the acoustic problems involved in distributing sound. In a small living room, a power of two or three watts is entirely sufficient. However, in a large auditorium or at an outdoor gathering or sporting event, an electrical power of as much as 500 watts or more may be required.

There are many factors involved in the determination of the proper power levels. We’ll learn more about these later, but some of these factors are:

1. Noise Level
2. Acoustic Problems
3. Fidelity
4. Loudspeaker Efficiency

**Noise Level.** Whenever there is any appreciable amount of noise, any other sound tends to be masked. You are undoubtedly familiar with the fact that it is much easier to hear someone talking in a quiet room than in a noisy one. Conversely, a speaker must talk loudly in a noisy room to be heard. This fact means that the noise level at the location must be taken into account when a p.a. installation is planned. In general, it is necessary that the desired sound be amplified so that it is considerably stronger than the noise level. There are limits to this—if the noise level is too high, as it may be in a factory, it may be impossible to get above it, without making the amplified sound so loud that it is actually painful.

**Acoustic Problems.** The loudspeaker cone moves air particles directly before it, and these in turn move other particles at a distance. As this movement fans out, and as the distance between the loudspeaker and the listeners increases, a decreasing amount of sound power reaches individual listeners. Furthermore, much

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**FIG. 1.** This is the basic p.a. system—a microphone, an amplifier that builds up the signal from the microphone, and a loudspeaker that converts the electrical signal into sound.
of the sound power is absorbed by the cushions on chairs, by hangings on
the walls, carpets on the floors, and by the people and the clothing they
wear. Any soft material readily ab-
sorbs sound energy. All of these ab-
sorptions, plus that of any acoustic
treatment that may be placed in a
hall, will reduce the sound reaching
the rear of the hall appreciably. Out-
doors, sound is similarly absorbed by
people and dispersed by the wind. All
such effects increase the amount of
power a p.a. system must produce to
give adequate sound coverage.

One acoustic problem that occurs
only indoors is caused by sound
reaching listeners over two or more
paths. For example, if sound reaches
a listener directly from the loud-
speaker and indirectly by reflection
from a wall, the sound traveling over
the longer path will arrive later than
that over the more direct path. In an
extreme case, this can cause an echo
effect, with one sound heard sepa-
rately before the other. If the time
difference is too short to amount to
an actual echo, the sound arriving
over other paths may be sufficiently
out of phase to produce a muddled
response. This phase difference may
be exactly 180°, causing sound can-
cellation: in fact, it is quite common
to find that reflections from the walls,
floors and ceilings are such that there
are actual dead spots in the hall.

As we shall show later, the reflec-
tion problem can be partially solved
by acoustic treatment of the room,
but it is quite possible that severe
reflections will require the use of addi-
tional loudspeakers, so distributed
that sound energy will be put where
and only where it is wanted. Any such
multiple speaker installations will
usually require more power.

**Fidelity Requirements.** It is not
usually difficult to design a public
address system to handle only spoken
words. However, when music is also
to be handled, the fidelity of the sys-
tem enters into its design to a great
extent. The greater the fidelity
requirements, the greater the power
requirements. Low frequencies in par-
ticular require large amounts of
power to be heard at a distance,
because the human ear falls off in its
response characteristics at low fre-
quencies. Similarly, there is a drop-
off in the high-frequency response
because of the greater absorption of
these frequencies in the acoustic
materials of the hall. To make up for
these rather large drop-offs, it is
necessary to have high powers at the
low and high frequencies, and to de-
sign the loudspeakers and their baffles
to reproduce such frequency ranges
properly. Therefore, when high fidelity
is required, the power demand is in-
creased tremendously.

**Loudspeaker Efficiencies.** Once
the problems of noise, acoustic condi-
tions, and fidelity have been con-
sidered, it is possible to determine
about what acoustical power will be
needed to cover a certain area or
number of people outdoors or to
cover a certain room volume or num-
ber of people indoors. In fact, in later
Lessons, we will give tables that can
be used, once the necessary facts
about the installation are known, for
determining roughly the acoustical
power needed.

When the acoustical power is
known, you can find the electrical
power from the loudspeaker effi-
ciencies. The loudspeaker converts electrical power into sound power. Unfortunately, this conversion occurs with extremely low efficiency, so a considerable amount of electrical power is required to produce a small amount of sound power. At best, the ordinary cone-type loudspeaker of the sort used in home receivers has an efficiency of only about 2%. If this cone loudspeaker is placed in a carefully designed baffle, its efficiency rises to as much as 5%. Even the best speakers, using efficient diaphragm driver units in trumpets, have efficiencies of only about 15%, and this is obtained only at a considerable sacrifice in fidelity. In most cases, however, a surprisingly small amount of sound pressure is needed, so it isn’t necessary to go to extremes in electrical power to overcome this great loss in the loudspeaker.

Once we arrive at a reasonable estimate for the electrical power required, this sets at least one of the requirements to be made of our amplifier. Thus, if we find that we need 12 watts for a particular small installation, the amplifier must deliver at least this power output.

GAIN REQUIREMENTS

Turning now to the other end of the system, how much are we getting from the microphone? We shall find in other Lessons that this depends on the kind of microphone, and on the distance between the microphone and the person speaking, as well as on the sound energy delivered by that person. However, even at best, a microphone delivers a power that is only a fraction of a microwatt! Therefore, our amplifier must have sufficient voltage and power amplification to raise the output of the microphone to the power needed to drive the loudspeaker system. This gives a second requirement for the amplifier—it must have sufficient gain in addition to delivering the required output.

Once we have chosen the microphone, amplifier, and loudspeakers, we are faced with the problems of connecting them together. Often very short leads are all that are required, but sometimes we may have to put our loudspeakers several hundred feet away from the amplifier. As you will learn later, special impedance-matching methods must be used in this case.

Another problem rises when a sound system is used for amplifying music. To get fidelity, it is frequently necessary to use combinations of low-frequency and high-frequency loudspeakers. The power distribution problem is complicated by this, because we must not only match impedances properly, but also use frequency-dividing networks so that the speakers will get power at the frequencies they are designed to handle most effectively.

Further, we may not always want to use only a microphone with the sound system. Very frequently phonograph records are played over p.a. systems, for example, and occasionally radio programs are reproduced over them. The amplifier must therefore be capable of operating from a phonograph pickup or from the audio output of a radio receiver unit as well as from a microphone. These devices all have different output levels and are of different impedances. This brings up another problem in imped-
 ance matching, this time at the input of the amplifier.

Furthermore, the use of several input devices introduces the problem of switching from one to another. We can just unplug one and plug in the other, or just throw a switch, but, if we do, we will get a very loud click or pop from the loudspeaker. Most p.a. systems have some form of fading control, so arranged that the output of one or the other of the devices can be reduced to the minimum and then the output of the other can be raised gradually, or so arranged that they can be mixed together.

We are introducing you to these various public address problems so that you can better appreciate the material in the next several Lessons. Now that we have a general understanding of some of the problems, we can go on to a more detailed study of the amplifier itself.

The Decibel and Power Ratios

In public address work, we are dealing with extremely large power ratios. The acoustic power at the microphone is exceedingly small, whereas the sound output of the loudspeaker may be so loud that it is actually painful. The power ratio (output power divided by input power) is therefore so large that the figures involved become inconvenient to handle. It is not unusual to have gain figures representing power increases of as much as a billion times. For convenience, it is desirable to express the gains and power ratios involved in p.a. work in some way that will not demand such large numbers. This has led to the adoption of a special unit called the decibel, which we shall discuss in a moment.

Another factor that makes it desirable to use decibel units is the fact that the human ear responds exponentially to sound powers, rather than linearly. This means that if we double the sound power, we don't get twice as much sound as far as the ear is concerned—in fact, we can just barely detect the fact that the loudness of the sound has increased.

In other words, the human ear is so constructed that any complex sound must be doubled in power before it sounds louder. This is true at both low and high sound levels, provided the original sound is loud enough to be heard at all. For example, going from 2 to 4 microvolts produces a detectable increase in loudness; the apparent increase produced by going

<table>
<thead>
<tr>
<th>db</th>
<th>Power Ratio</th>
</tr>
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<tbody>
<tr>
<td>1</td>
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</tr>
<tr>
<td>2</td>
<td>1.6</td>
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</tr>
<tr>
<td>120</td>
<td>1000000000.0000000</td>
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TABLE 2

<table>
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</thead>
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<td>3.0</td>
</tr>
<tr>
<td>2.5</td>
<td>4.0</td>
</tr>
<tr>
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<td>4.8</td>
</tr>
<tr>
<td>3.5</td>
<td>5.4</td>
</tr>
<tr>
<td>4.0</td>
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</tr>
<tr>
<td>60.0</td>
<td>20.0</td>
</tr>
<tr>
<td>100.0</td>
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<tr>
<td>500.0</td>
<td>30.0</td>
</tr>
<tr>
<td>1000.0</td>
<td></td>
</tr>
</tbody>
</table>

from 200 to 400 watts is no greater.

This peculiar property of the ear is another reason why the use of decibel units in discussing sound power ratios is convenient, because the decibel system expresses these ratios in terms of what the ear can hear. Let’s go on now and learn what these important units are.

DECIBEL DEFINITION

The decibel (usually abbreviated db) is logarithmically related to the ratio of two powers by the formula

\[ db = 10 \log_{10} \frac{P_1}{P_2} \]

where \( P_1 \) and \( P_2 \) are the powers. To solve this equation, the two powers are inserted and their ratio determined. Then the logarithm to the base 10 of this power ratio is looked up in a table. Ten times this logarithm is the decibel gain or loss.

In this Lesson, we cannot go very far into the subject of logarithms. Briefly, however, a logarithm of a number is the power to which a base number must be raised to equal the original number. For example, you know that the second power of ten \((10^2)\) equals 100. In the common logarithms that use the base 10, 2 then becomes the logarithm of 100.

It is unnecessary to use the db formula because there are tables available, such as Tables 1 and 2, that give the decibels corresponding to certain power ratios. Furthermore, there are meters that are designed to indicate decibels directly. We’ll say more about these shortly.

USES OF DECIBEL UNITS

Although the decibel was originally developed purely from power ratios, careful tests have indicated that one decibel of power increase is just about the smallest change in power that can be detected by the average human ear. This change is detectable only when it consists of a single pure tone and only when the test is carried out under carefully controlled conditions. For complex tones—music, for example—a change of 3 decibels is ordinarily necessary to produce a detectable volume level change. Table 1 shows that a 3-decibel change indicates a power ratio of 2, meaning that the power must be doubled before we can tell that the complex sound is any louder. If we want to make it still louder, the power must be doubled again, and so on.

Since the decibel expresses the relationship between two powers, it is a convenient unit with which to measure power gains or losses. Furthermore, it can be used to express sound power or electrical power in terms of some reference value of power. The reference level commonly used when sound powers are given in decibels is
the sound power that is just barely audible to the average ear—in other words, the threshold of hearing of the average person. For convenience, technicians do not usually bother to mention the reference level when they talk about sound powers in db, but you should always remember that a sound level expressed in db is really the level with respect to the threshold of hearing. For example, the noise level in the average home living room has been found to be about 55 db; from what we just said, you know that this is 55 db with respect to the reference level, or about 300,000 times the power of the least audible sound.

Notice how much more convenient it is to say "55 db" instead of "300,000 times the power of the least audible sound." Obviously the decibel measurement is far easier to use in speech or writing. Furthermore, stating the noise level in db lets us get some idea of just how noisy the location is. Since each 3-db increase produces a barely audible increase in loudness, we know that the noise is 55 ÷ 3 or about 18 steps up the scale of comparative loudness.

Electrical powers are also often expressed in decibels in sound work. Here again, some power level must be used as a reference. In the past, considerable confusion arose from the fact that three different reference levels were used by different branches of the communications industry—the telephone company and the radio amplifier manufacturers, particularly, differing in their standards. Of these three older standards, a reference level of 6 milliwatts was the most commonly used; in fact, it still is in sound work. However, in recent years, there has been an attempt in the communications field to secure universal use of a new standard based on a 1-milliwatt reference level. This new unit is used throughout both the broadcast industry and the telephone companies. As a result, it is gradually spreading to sound equipment, and may eventually replace all of the older reference levels. Although the new unit is still a decibel, because the only change has been in the reference level, it is a common practice to indicate the new unit as a "VU" or "dbm" instead of "db" to avoid confusion.

In either case, the reference level is assumed to be the zero db level. Any power that is higher than the reference level is therefore a power increase above the reference level and is considered to be a plus db value. Power levels below the reference level are minus db values.

Table 3 gives some typical db levels based on the 6-milliwatt (.006 watt) and on the 1-milliwatt (.001 watt) reference levels. There is no need for you to try to memorize these values. All you need to do now is to learn how they are used. To that end, let's take a few practical examples of the use of decibels in sound work.

Let's suppose we have a case in which 60 watts of power fed through certain loudspeakers will produce sufficient audio power to cover an audience properly at the desired level. From Table 3, we see that this is an output of about 40 db above the reference level of .006 watt.

A typical microphone may have an output of -60 db, which means that its output is 60 db below the reference level of .006 watt. Therefore, we have to raise the microphone output of
-60 db to a plus value of 40 db. This means that the amplifier must have an over-all power gain of 100 db. The output power of the amplifier is therefore about one billion times that of the microphone!

An important point to remember is that we have to double the output power before we can get a noticeably stronger signal. If one amplifier is rated at 15 watts, and another is rated at 20 watts, their power difference is only slightly more than 1 db. Obviously, therefore, the 20-watt amplifier will not produce any appreciably louder sounds than the 15-watt one. This doesn’t mean that the 20-watt amplifier wouldn’t be the better choice of these two—we get somewhat less distortion by running an amplifier at less than its rated output, and of course one having the higher power rating would be better able to handle high power peaks without too much distortion. The 20-watt amplifier may therefore be the better of the two, on

<table>
<thead>
<tr>
<th>Watts</th>
<th>db</th>
<th>Reference Level: 0 db = 1 milliwatt</th>
</tr>
</thead>
<tbody>
<tr>
<td>1000.</td>
<td>+60</td>
<td>6000.</td>
</tr>
<tr>
<td>100</td>
<td>+50</td>
<td>600.</td>
</tr>
<tr>
<td>10</td>
<td>+40</td>
<td>60.</td>
</tr>
<tr>
<td>1</td>
<td>+30</td>
<td>6.</td>
</tr>
<tr>
<td>.1</td>
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<td>.6</td>
</tr>
<tr>
<td>.01</td>
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<tr>
<td>.001</td>
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<td>.000  01</td>
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<tr>
<td>.000 000 000 000 1</td>
<td>-100</td>
<td>.000 000 000 000 6</td>
</tr>
</tbody>
</table>

The basis of freedom from distortion, but it will not be any louder for complex sounds. If we had a 15-watt amplifier, we would have to go to a 30-watt amplifier to get a noticeable increase in loudness level. Similarly, we would have to go from 100 watts to 200 watts to get an appreciable increase in sound at a higher power level.
Amplifier Specifications

There are many types and sizes of p.a. amplifiers. In addition to differing in amount of electrical power output and in fidelity of response, they have different power-supply requirements, are capable of operating from different types or numbers of microphones or other inputs, and have different input and output impedance characteristics. All these factors must be considered in the choice of a particular amplifier for a specific job. To assist in making this choice, manufacturers' catalogs give the following information about each amplifier listed, either in the form of a complete description or as tabulated data:

- Power Output
- Gain
- Frequency Response
- Hum Level
- Input Impedances
- Output Impedances
- Power Required
- Tubes
- Physical Specifications

In addition, you may find a few other special features described, such as the kind of tone control.

Naturally, it is important for you to understand the real meaning of each of these specifications. Let's examine the important ones now to see just what they mean.

POWER OUTPUT

The power output is usually stated in watts, although you may sometimes find that the manufacturer also gives the output level in decibels above the 6-milliwatt reference level.

Some manufacturers give both a "normal" and a "peak" output rating. In these cases, the normal output level is the output for a certain specified percent of total harmonic distortion. The peak value is the maximum amount of power that can be obtained from the amplifier without regard to distortion.

It is common practice to select 5% total harmonic distortion as the acceptable distortion for normal output, because, at this level, the amount of third harmonic distortion is not so high that it is seriously objectionable. To obtain the power rating, therefore, the manufacturer increases the input while analyzing the waveform of the output. When the harmonic distortion reaches the value chosen, such as 5% (or 2% in the case of high-fidelity equipment), the output is measured. This becomes the normal power output. Then, the input is increased further until the point of maximum power output is reached. This too is measured. This becomes the peak rating.

If you find only one output value listed for an amplifier, you won't always know whether the manufacturer means normal output or peak output. The normal output is considerably less than the peak rating; therefore, if the rating given is close to the value needed for the installation in such a case, you would do well to determine just which is meant before purchasing the equipment.

Amplifiers intended for public address can be grouped into low-power, medium-power, and high-power classes. There is no strict
border line between these classes, however. In general, any amplifier under about 10 to 15 watts is a low-power type, those between this value and about 50 watts are medium power, and those above 50 watts are considered to be high power.

As we pointed out while discussing decibels, it takes a doubling of the power output to produce a noticeable increase in volume, so of course amplifier manufacturers do not make many different sizes in any of these groups. Usually a manufacturer makes only 4 or 5 amplifiers in each series—say a low-power amplifier of about 8 to 10 watts, a medium-power one of 15 to 20 watts, another somewhere between 35 and 50 watts, and then perhaps a high-power one. The outputs chosen are selected with the idea of having some amplifier in the line fairly close to any output that may be desired.

Manufacturers usually also make amplifiers for battery or a.c.-d.c. operation. These are not usually merely the standard a.c.-operated amplifiers with modified power supplies, because, for battery operation at least, it is necessary to make amplifiers as economical of power as possible, something that designers of a.c. equipment don’t worry much about. We’ll discuss this later.

**AMPLIFIER GAIN VALUES**

Because of the extremely high power ratios involved in public address work, it is standard practice to give the gain of amplifiers in decibels. Because these amplifiers are commonly used with phono pickups in addition to microphones, most amplifiers have input circuits for each. Since the output of a phonograph-record player is much higher than that of a microphone, less gain is needed for the phono channels. Therefore, the gain values are usually given for each input—some such value as 100 db gain for microphone and perhaps 40 to 60 db for phonograph. As you learned from Table 1, a db gain of 100 represents a power ratio of one billion.

Sometimes, in connection with the gain values, the manufacturer will list specific types of microphones or phonograph players that are suitable for the particular amplifier. If such information is not given, it may be necessary to make a calculation to determine whether a specific input device can be used with a particular amplifier. In such cases, the output power rating must be converted to decibels. Let’s suppose the amplifier is rated at 60 watts and has a gain of 100 db for the microphone channel. From Table 3, we find that a 60-watt output represents +40 db, based on a 6-milliwatt reference level. Since the output of our amplifier is +40 db, and the gain is 100 db, the amplifier will deliver its rated output of 60 watts if the input is −60 db. That is, a gain of 100 db will raise a level of −60 db to +40 db (100 minus 60 equals 40).

Microphones have different outputs ranging all the way from −40 db to perhaps −100 db. (This is from the 6-milliwatt reference level.) Naturally, if you have a microphone capable of giving −50 db, it has more than enough output to drive the amplifier we are discussing to full rated output. It will work satisfactorily with the amplifier because we can always
reduce the gain with the volume control. On the other hand, a microphone with an output of -70 db will not permit this amplifier to give full rated output. If we must use a microphone of this kind, we will have to have a preamplifier with a gain of at least 10 db to raise the microphone signal level from -70 db to -60 db so that the amplifier can be operated at full output.

This discussion shows you why the decibel is used in p.a. work. With its aid, it is rather simple to see just what will work with what. Any power losses or power gains in the systems can be taken into consideration simply.

**FREQUENCY RESPONSE**

The amplitude or harmonic distortion is given along with the power output rating, or is understood to be at some standard level when normal outputs are given. However, in addition, we can have frequency distortion—the limitation of the frequency range over which the amplifier will operate with a reasonable output. In public address work, the frequency response is rather important. If voice alone is to be handled, there is no need for very low notes, nor is there need for high notes above 5000 cycles. If the system is to have high fidelity, on the other hand, you'll want as wide a frequency response as is obtainable within the price range in which you are interested.

To arrive at the frequency response, the manufacturer determines the input at a reference frequency, usually 1000 cycles, that will produce the rated output. Then, the same input is fed into the amplifier at other frequencies. The amplifier output at each of these various frequencies is then expressed either in decibels or in terms of the number of decibels it is up or down from the output at the reference frequency.

Data on frequency response are frequently given in the form of response curves. In the type shown in Fig. 2, the output is given in terms of the rated output of the particular amplifier. A somewhat more common form is that shown in Fig. 3, in which the response at various frequencies is given in terms of its db variation from the reference frequency output. This curve applies to any amplifier having this response, regardless of its rated db output level.

A frequency response curve is ordinarily carried out only to the points at which the frequencies fall off 3 db from the reference value. Beyond these points, it is understood that the characteristic may have peaks, but in general, will be worse than 3 db off from the reference level. Therefore, whenever the manufacturer says that an amplifier is "flat within 3 db from
40 to 10,000 cycles," he means that the output will vary slightly but will remain within 3 db above or below the output at 1000 cycles between these limits. Notice that the curves shown in Figs. 2 and 3 are flat within 3 db from 80 to about 7500 cycles.

Many manufacturers give information on the effects of the tone controls on the frequency response. They will state that the tone control raises or lowers the output so many db at a given frequency. This will give a general idea of what happens to the response curve as the tone controls are varied.

**AMPLIFIER HUM LEVELS**

Naturally, the output hum and noise levels from an amplifier must be just as low as possible for best results. In any amplifier of reasonably good design, the noise level is far below that of the hum.

In high-fidelity systems, the hum voltage applied to the loudspeaker must be very small to prevent excessive hum output. The hum level is not quite so important in a low-fidelity system, however, because the low-frequency output is usually attenuated.

The manufacturer commonly gives the hum level as so many db below rated power output. A value around 35 to 50 db down is considered acceptable for general-purpose amplifiers.

When the noise level is given too, it is likewise given in terms of decibels down.

**INPUT IMPEDANCES**

When an amplifier is given a certain rating, it is assumed that its input and output will be properly impedance-matched so that the maximum power transfer will occur. Therefore, the number and impedance values of the input channels are important amplifier ratings.

The simplest amplifier may have only a phonograph input; very elaborate ones may have provisions for three or four microphones and perhaps two or three phonograph players. Because it may be desired to fade one signal out and fade another in gradually, without affecting the strength of any other signals, all of these input channels are usually fed through separate preamplifier stages, whose outputs are then combined in a mixing circuit arrangement. We'll study these amplifiers in more detail later.

Some microphones, such as the crystal types, are high-impedance devices that should feed into the grid input circuits of tubes. As a practical matter, you know that the grid circuit must have a d.c. path to the cathode. Since a resistor of around 100,000 ohms is commonly used to provide this path, it is standard practice to consider a high impedance input to be approximately 100,000 ohms for microphone services.

Microphones such as the dynamic types have low impedances, which are brought up to standard line impedances by means of matching transformers built into the bodies of the microphones. For low-impedance microphones, therefore, the input of the set has to be a transformer rated at some standard line impedance such as 250 or 500 ohms. Because high-impedance inputs are less costly, basic amplifiers are usually supplied with high-impedance inputs, with low-impedance inputs being available at a
slight extra cost. The type of input impedance is usually optional in the more elaborate amplifiers.

Phono channels are today practically all high-impedance types because it is standard practice to use crystal pickups. If magnetic pickups are used, it is expected that a matching transformer will be used to match the pickups to the grid circuit of a tube or to match from a standard transmission line to such a grid circuit.

**OUTPUT IMPEDANCE**

It is standard practice today for practically all amplifiers to have a tapped arrangement for matching various loudspeaker voice coil impedances. Values of 2, 4, 6, 8, and 16 ohms are usually available. In addition, most amplifiers also have provision for at least a 500-ohm line. Some of the more elaborate types have additional taps for 125 ohms and 250 ohms for use when lines are connected in parallel.

In addition to giving the output impedance values, the manufacturer will usually mention the method used for making connections to the output terminals of the amplifier. In some instances, these terminals are just brought out to terminal strips. In others, the terminals are brought out to sockets into which the loudspeaker lines are plugged, the proper impedance being selected by turning a switch. Such refinements as this latter are not absolutely essential, but they are helpful, particularly for amplifiers that are going to be set up and taken down frequently under conditions under which different types of loudspeakers may be needed. We shall go further into the subject of loudspeaker connections later (in another Lesson).

**POWER REQUIREMENTS**

Like radio devices, public address amplifiers operate from power supplies. It will do no good to find exactly the right amplifier for your installation if it will not operate from whatever power is available. Therefore, although the power requirement is usually far down on the list, it is one of the first things you should look for.

Of course, 115-volt, 60-cycle a.c. power is commonly available throughout the United States, and most p.a. amplifiers are designed to operate from such a.c. power lines. There is a wide variety of amplifiers available for such operation, so the choice of a particular amplifier depends on other considerations.

However, there are many cases where the proper power lines are not available. In some of the larger cities, for example, there are large districts in which only 110-volt d.c. power is available. In a few localities, the power lines supply only 25-cycle a.c. Special amplifiers are rarely available for such power supplies. The only thing that can be done in most instances is to obtain an inverter unit that will convert the available power to 60-cycle a.c. Such inverter units are available from radio supply houses.

Public address equipment used in a sound truck must operate either from storage batteries or from some form of power supply carried with the amplifier in the truck. In the case of high-power units, it is standard practice to equip the truck with a small a.c. generator driven by a gasoline motor. Because of the efficient cir-
cuits incorporated in modern amplifiers, however, it is practical to operate the small units from 6-volt storage batteries. Vibrator-type power supplies are used in such cases. Most such units supply enough 115-volt, 60-cycle power to operate a record player as well as an amplifier.

Naturally, when we are dealing with special units of this kind, it is particularly important that the required power levels be calculated accurately. Large sound systems drain storage batteries quickly and are rather costly. On the other hand, units that are too small are practically worthless. It is therefore necessary to select equipment that is adequate for the job but not more powerful than it needs to be.

**TUBES**

In practically all cases, manufacturers list the number and types of tubes used in p.a. amplifiers. This information is helpful if you find that some of the tubes listed are not the types that are commonly available in your locality, because then you can stock up on an extra set or so. The tube list will also give you a general idea of the circuits that are used, and from the power output rating, you can get an idea of how hard the tubes are being driven.

**PHYSICAL SPECIFICATIONS**

It is important that you know the dimensions and weights of public address units, particularly when they are to be permanently installed in a given location. The kind of housing, too, is frequently important. Sometimes you will want the amplifiers mounted in a standard rack. In other cases, you will want them to be enclosed in a metal shield or case, which is common practice for most amplifiers today.

The manufacturer may also describe the color and type of decoration on the housing, and, of course, he will usually show photographs of the general appearance of the amplifier. Naturally, it is always desirable to have a unit that is physically pleasing in appearance whenever it is to be located where it will be used by the public. Therefore, although such considerations are less important than getting the right technical equipment, they must be taken into account.

Now that we have a general idea of the data that can be expected in the manufacturer's literature, let's go on and briefly examine some typical p.a. amplifiers to see how they differ from standard audio amplifier equipment like that found in radio receivers.

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**Power Supplies, Output Stages, and Drivers**

As we have mentioned before, the public address amplifier is essentially like the audio amplifier in a standard radio receiver. As a matter of fact, the low-powered types are, for all practical purposes, identical with such amplifiers. Not only are the circuits similar, but also the same kinds of tubes are used. The only radical difference is that low-powered p.a. units usually require one more voltage amplifier stage so that they will
have sufficient gain to operate from the very low output of microphones. Higher-powered units differ more markedly from the audio sections of radio receivers, mostly because different tubes and circuits are needed to permit the handling of the increased power.

In the following discussion, we shall not go deeply into the basic theory of voltage and power amplifiers, because you have already studied this in other Lessons of this Course. (If you are hazy on certain points, review your Lessons on low-frequency amplifiers and on power supplies.) Instead, we shall point out the important differences between radios and p.a. systems. Let's start with power supplies.

**POWER SUPPLIES**

The smaller p.a. units operate from power supplies that are identical with those in standard radio receivers. The most common power supply uses a standard power transformer, a full-wave rectifier, and a filter, although you will find that a few of the small portable p.a. units use a.c.-d.c. supplies. The small mobile p.a. systems that are designed to operate in trucks use vibrator-type power supplies operating from a storage battery, almost identical with supply units you find in auto-radio receivers except that they are capable of delivering somewhat more power. If we consider devices like hearing aids to be public address-systems in miniature, we will even find batteries are used to furnish power directly.

Therefore, in all low-powered p.a. systems, we can expect to find power supplies that are identical with types we have included before in our study of radio receivers. It is only when we get up in the high-power units that we find much difference.

In high-power applications, it is standard practice to use a power supply with a power transformer, operating from 60-cycle a.c. If the equipment is to be used in mobile services, it is commonly operated from a gasoline-engine driven motor-generator that develops the necessary 110-volt, 60-cycle a.c. In districts with d.c. power or 25-cycle a.c., a motor-generator would be used to deliver the 110-volt, 60-cycle a.c. Hence, you will usually find that all high-power amplifiers are alike in their power supplies to this extent.

Voltages around 300 to 400 volts are easily obtained from a transformer power supply. Receiver-type rectifier tubes may be used; if the current requirements exceed the rating of a single tube, two tubes can be used with the sections in parallel, as in Figs. 4A or 4B. These two connections both deliver twice the current of a single tube. The only difference is that you will get only half-wave rectification, with consequent hum, if one tube fails in the circuit shown in Fig. 4A. The circuit in Fig. 4B will still give full-wave rectification as long as the remaining tube lasts. Of course, this tube will be heavily overloaded, so it won't last long.

As we shall soon see, some p.a. systems use power output tubes operated in class AB₂ or even in class B. Because of the very wide changes in current requirements between the no-load and full-load conditions, power supplies used with such output tubes
must have good regulation. Ordinarily, this means that the transformer and choke coils must have low resistance, and that a very high bleeder current must be drawn. This increases the current requirements.

Since the final stage requires more plate current than any other, its current is frequently taken directly from the rectifier output without passing it through the filter choke. This is permissible because there is no amplification beyond the output stage, so any hum developed is swamped by the desired signal. When the output current does go through the filter, a swinging choke is commonly used as the input choke to help keep the output voltage constant in spite of the high current changes between no-load and full-load conditions.

In some of the p.a. units of the highest powers—those rated well over 100 watts—the power output tubes are actually small transmitting tubes intended to operate on higher voltages than are applied to receiving tubes. The power supplies of such units must, of course, be designed to deliver appropriate voltages—around 800 to 1500 volts. This means that the secondary of the power transformer must have a higher voltage rating than is usual in p.a. equipment. To withstand the higher voltages, special rectifier tubes of the types that are more commonly found in amateur transmitting equipment are sometimes used. In addition, the filter condensers must be designed for these high voltages, which means that they are usually oil-filled paper condensers of the kind used in transmitters. The need for this special, expensive equipment makes high-power amplifiers disproportionately costly. For this and other reasons, high-power p.a. units of this kind are rather rare; when high powers are needed, it is common practice to use several amplifiers connected in parallel instead. The use of several smaller amplifiers is preferable because it is lower in cost, gives a more flexible arrangement (since the system can be expanded at any time by adding more units), and simplifies future servicing.

**POWER OUTPUT TUBES**

As you might expect, beam power and pentode tubes, which have high power sensitivity and high plate efficiency, are used as the power output tubes in practically all p.a. amplifiers. Obviously, if a triode tube requires 40 volts as the grid signal voltage for full excitation, and a pentode or beam tube is capable of giving the same power output with only 15 volts of grid drive, the latter is more desirable, since much less voltage amplification is necessary ahead of it. These tubes also have an advantage over triodes in that they convert
somewhat higher percentages of their plate power into usable power output.

The one advantage of the triode over the beam power and pentode tubes is that it has far less distortion. However, modern inverse feedback circuits make it possible to obtain reasonable fidelity from pentode and beam power tubes. Therefore, the triode power tube has practically disappeared from the p.a. field except for very high-fidelity systems.

In general, the types of tubes used in p.a. amplifiers are exactly like those in radio receivers, except that, because of its high power capabilities, the 6L6 tube is more commonly used in p.a. work than it is in radio receivers. Even the smaller receiver-type tubes are commonly used, sometimes in circuits that get more from them than is required in radio receivers.

**Class A Operation.** The power output stages of p.a. amplifiers are most commonly operated in class A, just as are those in radio receivers. In this class of operation, the operating point of the tube is set by the bias on the midpoint of the straight portion of its characteristic (see Fig. 5). The complete cycle of the incoming signal is reproduced in the plate current. As long as the input signal is not so high that it swings as low as the bottom knee of the characteristic or higher than the zero bias point, this class of operation is reasonably free from distortion. This matter of distortion is important because it places the real limit on power output. We can get only so much power output at a given distortion level from any particular tube once its operating condition has been specified. When the acceptable distortion level has been chosen, the drive or grid signal applied to the power output tubes can be increased only until this distortion percentage is found in the output.

In applications in which the tone quality is not very important and low powers are all that is required, the single-ended class A stage like that in Fig. 6 is sometimes used. If higher output levels are required and somewhat better tone quality is desirable, a push-pull circuit like that shown in Fig. 7 is used. Here, because the even
harmonics are cancelled in the output transformer, it is possible to get greater power output from each output tube than is possible in the single-ended connection shown in Fig. 6. As a matter of fact, properly increasing the grid drive permits about two and one-half times as much power to be obtained from a pair of tubes in push-pull as can be gotten from a single tube for the same relative amount of distortion.

Both the single-ended and push-pull class A stages are usually self-biased by a resistor in the cathode circuit. However, there are exceptions—the bias can be obtained from the power supply, making it a form of fixed bias. Such a system is rather commonly used with push-pull outputs, because it is desirable to balance plate currents of the push-pull tubes. Therefore, as we shall see later, the grid returns are split and brought back to separate adjustable bias resistors in the power pack; it is then possible to adjust the bias to produce equal plate currents.

The circuits in Figs. 6 and 7 use resistance coupling to the power tube grids. It is possible to use transformers, of course, but an input transformer is bulky and rather costly. Furthermore, unless it is of high quality it will introduce considerable frequency distortion and also pick up stray hum fields.

When resistance coupling is used at the input of the push-pull stage, a phase inverter must be used. Any of the types that you studied in your fundamental Lessons may be found in p.a. amplifiers. Several typical schematics of phase inverters are shown in Fig. 8. In each instance, the necessary 180° phase shift is obtained either by an additional tube or, as shown in Fig. 8C, by making use of the fact that the cathode voltage is out of phase with the plate load voltage.

If a transformer input is used for the push-pull stage, of course, a single-ended driver stage can be used.

**GETTING MORE POWER**

Once we have reached the maximum permissible output with a particular tube in class A operation, the only way of getting more power output is to change the conditions of operation or change the tube. Equipment designers usually prefer to use more efficient classes of operation, since transmitter tubes, the only types capable of giving more power output, are expensive.

Instead of class A, we can use classes AB₁, or AB₂, or even class B provided we use a push-pull circuit. The power output increases remarkably—if two tubes deliver 18 watts in class A push-pull, they may give 25 watts in AB₁, 45 watts in AB₂, and 60 watts in class B.

Fig. 9 shows the difference between these classes of operation. The class
A grid signal is limited so that the operation remains over the straight portion of the characteristic between the lower knee and the zero bias line. The plate current change for class A operation here reaches the peak value represented by 1. Naturally, the greater this amplitude can be made, the greater the amount of signal power output. Therefore, if we move the operating point down near the knee of the curve, we can apply a higher grid signal and produce AB₁ operation. The plate current swing for this class of operation is shown at the right at 2. Notice that amplitude 2 is higher than 1; this means a greater amount of power output is obtainable. However, the lower half of this plate current cycle is flattened out, meaning that a large increase in even-harmonic distortion has occurred. This distortion would make class AB₂ operation undesirable were it not that push-pull operation fortunately eliminates the even harmonics.

Increasing the grid drive more produces class AB₂ operation, in which the grid actually goes positive for a small portion of the cycle. This operation gives even greater power output, shown by the fact that peak 3 is higher than either 1 or 2.

Finally, when we move the operating point to class B operation, right at the cut-off bias level, only one-half of each cycle of the incoming grid signal is reproduced in the output. The plate current for this class is represented by peak 4, which is much greater than that of any of the preceding classes of operation. When two tubes are operated in class B push-pull, one tube furnishes power for one-half cycle, then it is cut off while the other tube is delivering power.

In class AB₁ operation, in which no grid current is permitted to flow, it is possible to use the same kinds of circuits as in class A operation. In class AB₂ and class B operation, however, the grids of the power output tubes draw current during small portions of the grid cycle. As a result, there is a power dissipation in the grid circuits of the tubes; this power must come from the driver stage. Furthermore, to avoid extreme dis-
tortion, the total grid circuit resistance must be kept very small so that there will be but a small voltage drop while grid current is flowing.

For these two reasons, resistance coupling is not used for class AB₂ or class B operation. Instead, the drive signal is applied through a specially designed input transformer that has a secondary winding with very low resistance or through a cathode follower circuit like that shown in Fig. 10. In the latter case, the "load" on the driver tubes VT₁ and VT₂ consists of the coil L and the cathode resistors R₂ and R₄. The low-resistance coil is in the VT₃ and VT₄ grid circuits, so grid losses are avoided. This connection provides a good impedance match between the drivers and the output tubes and thereby reduces distortion. Incidentally, the drivers VT₁ and VT₂ are actually small power tubes (operating in class A) that are driven by a voltage amplifier and a phase inverter.

For these classes of operation, it is desirable to have the grid bias of the power tubes adjustable so that the plate currents can be balanced. The balancing arrangement shown in Fig. 11 is commonly used. Adjusting the two potentiometers R₁ and R₂ makes it possible to get the plate currents equal and thus minimize the distortion that will naturally occur with these classes of operation. Incidentally, since the bias is at the cut-off value for class B operation, it is not practical to use self-bias for the power tubes—the bias must be supplied by a separate source such as the power supply.

FIG. 9. Four classes of amplifier operation.

FIG. 10. A cathode-follower circuit used to feed the output stage in amplifiers operated in class AB₁ or B.

FIG. 11. The plate currents of the two tubes are equalized by adjusting R₁ and R₂.
The problem of supplying an input signal to a class \( AB_2 \) or class B stage frequently means that the tubes preceding the power output tubes must be small power tubes themselves. The required grid input, although it may be only a fraction of a watt, is frequently more than the ordinary voltage amplifier tube is capable of supplying.

**Inverse Feedback.** Any of the forms of inverse feedback that you studied in your fundamental Lessons may be found in public address systems. The feedback may just be across the output stage—from the plate to the grid circuit, for example—or it may be over a loop of several stages. We'll see some typical diagrams later, in addition to the examples given in Figs. 12 and 13. To
refresh your mind—the feedback voltage is out of phase with the incoming signal and is of such nature that it decreases any distortion that is introduced between the point where the feedback occurs and the output. At the same time, the output level is reduced and the plate impedance of the output tube is brought down more nearly to that of the triode. The over-all result of this is that pentode and beam power output tubes can be used with nearly the fidelity obtained from the use of triodes. Although one of the advantages of the pentode and beam power tubes is lost in that the transformer tap arrangement. Each impedance value represents the impedance between that tap and the "common" terminal. Some amplifiers may have a few less taps and others may have a few more, but in general this is the basic arrangement.

Standard loudspeakers have voice-coil impedances of 4 ohms, 8 ohms, or 16 ohms. There are a few others, but these are the most common. If you are using a single loudspeaker of any of these values, all you need to do is to connect it between the proper taps to provide the desired impedance match to the output stage.

![Diagram of transformer taps](image)

**FIG. 14. The taps on a typical output transformer.**

power sensitivity is reduced, it is still better than that of triodes.

**LOUDSPEAKER COUPLING**

As you know, the ordinary radio receiver commonly has an output transformer designed specifically to match the particular loudspeaker used in the set to the output tube or tubes. In public address work, however, an amplifier may be used with any one of several types of loudspeakers or with a group of loudspeakers, depending on the installation, so the output transformer must have taps to accommodate different voice coil impedances.

Fig. 14 shows a common output transformer. If you are using more than one of these standard loudspeakers, the voice coils may be connected either in series or in parallel to equal some impedance value that the transformer can supply. For example, if you connect two 8-ohm loudspeaker voice coils in parallel, their net impedance will be 4 ohms, so you can use the 4-ohm tap. Connecting the same two 8-ohm loudspeakers in series would give 16 ohms net impedance, and the 16-ohm tap could be used; however, it is more common practice to connect the loudspeakers in parallel so that both will not be cut off if one of them should open or become defective.
Naturally, the more loudspeakers used, the more troublesome becomes the problem of impedance matching. We could connect four 8-ohm loudspeakers in parallel to get a net impedance of 2 ohms, which our transformer is capable of matching. However, connecting three such loudspeakers in parallel would give an impedance of $8 \div 3$ or 2.6 ohms, for which there is no transformer tap. When an in-between value like this is obtained, it is usually best to use the output transformer tap that is next lower in impedance, because doing so minimizes distortion and loss of power. Therefore, we should use the 2-ohm tap. (As a practical matter, although it is desirable to match within 10%, mismatching up to 25% is tolerable and causes very little power loss.)

Elsewhere, we will go further into this problem to show in more detail some of the difficulties met in coupling loudspeakers to amplifiers.

Returning now to our transformer, you will notice that there are two high-impedance terminals, one rated at 250 ohms and the other at 500 ohms. These are needed because the loudspeakers must frequently be at considerable distances from the amplifier. The loudspeaker voice coils have relatively low impedances, so even if you use rather large, low-resistance wires to connect them to the amplifier, there will still be considerable loss in the wire. For example, if we use No. 20 B & S wire to connect a 4-ohm loudspeaker to an amplifier, we cannot have the loudspeaker farther than twenty-five feet from the amplifier if we are to keep the line loss to a value of 15%. If the loudspeaker must be placed farther away from the amplifier, or if the power loss is to be kept less than 15%, we would either have to use much larger wire or, preferably, use a higher-impedance line. Such a line will also be discussed elsewhere, but for now let us say that a line will transmit power with a minimum loss if we connect a fairly high impedance to both of its ends. An impedance of 500 ohms is commonly used. With the higher impedance, we can have a higher terminal voltage and a much smaller current for the same power. Since the loss in the line depends upon the $I^2R$ value, reducing the current for the same power delivery means that the loss is decreased.

Therefore, if we connect one end of a line to the 500-ohm terminals of the output matching transformer, and connect the other end to a transformer that is designed to match 500 ohms to a voice coil, the line becomes relatively loss-free and can be run for considerable distances. For example, the No. 20 wire that we mentioned before, when used as a 500-ohm line, can be run for 1500 feet with a power loss of only 5%. As you recall, such wire has a 15% loss in a 25-foot run when it is used to feed the voice coil directly.

We'll go further into this problem of lines and impedance matching elsewhere. The important thing to know, as far as the amplifier itself is concerned, is that its output transformer has a number of secondary taps with which it is possible to match impedances under most ordinary circumstances.

Amplifiers vary considerably in the physical arrangement of their termi-
nals. Some have them brought out to a terminal strip to which the necessary loudspeaker connections can be made. In others, they are brought out to sockets into which the loudspeaker cables can be plugged.

**Voltage Amplifier Considerations**

An amplifier must have enough gain to raise the voltage level from that of the output of the microphone to whatever is required to drive the power output stage so that it will deliver the rated power output. By taking the ratio of these two voltage levels, we can determine the gain in decibels needed. From this, we can set up any combination of stages, the product of whose gains equals the necessary gain value. In practically all modern amplifiers, the voltage amplifier stages are resistance coupled and, in general, they duplicate receiver voltage-gain stages in their design. Triodes are commonly used; sometimes pentodes are used also. Figs. 15 and 16 show typical circuits.

The only major differences between p.a. amplifiers lie in the number of stages used and in the special features, such as the input coupling, the methods of mixing signals, and the tone-control network. We shall now take up these special items, leaving complete schematics for later.

**INPUT CONNECTIONS**

Standard practice is to bring the input terminals of the p.a. amplifier to jacks so that the microphones and other devices may be plugged in easily. From these points, the circuit goes to the grid of the first tube. There are three basic input arrangements, all of which are shown in Fig. 17.

Fig. 17A shows a high-impedance input, intended to operate from any high-impedance device such as a crystal phono pickup or crystal microphone. As you will learn in later Lessons, any signal source whose impedance is above, let us say 40,000 ohms is considered to be high impedance and can be fed directly to a tube grid as shown here.

Many microphones and the magnetic phono pickups are relatively low-impedance devices. For example, some dynamic microphones have as low an impedance as have many electrodynamic loudspeaker voice coils.

**FIG. 15. A typical triode voltage-amplifier stage.**

**FIG. 16. A voltage-amplifier stage in which a pentode is used.**
With devices of this kind, the proper impedance match must be made to the grid of the tube so that there will be sufficient voltage for proper operation. Also, since the microphone may sometimes have to operate at a distance from the amplifier, it is standard practice to use a matching transformer between the microphone or pickup and connecting line, which is almost always rated at 500 ohms. Then, at the amplifier, another transformer is used to match the 500-ohm line to the grid input of the first tube.

There are two basic arrangements for low-impedance inputs, which are shown in Figs. 17B and 17C. To set a fixed value for the grid input impedance, a resistance of some value around 100,000 ohms may be connected as \( R_2 \). Then, the transformer matches 500 ohms to the resistor value.

Fig. 17B shows what is known as the unbalanced line, in which one side of the line is grounded. The microphone cable used here (and in the high-impedance circuit in Fig. 17A) is a coaxial type consisting of an insulated conductor surrounded by a flexible braided shield, which acts as the other side of the line. In Fig. 17C is shown the balanced line. The basic difference here is that there are two separate conductors and that the ground is made to a center tap at a transformer at each end of the line. These two conductors can be and usually are surrounded by shielding braid that serves as a ground return. The advantage of the balanced system is that both lines will pick up an equal amount of noise or hum voltage and will feed these equal voltages in opposite directions through the input transformer of the receiver so that they will cancel. (The signal current sets up a circulating current throughout the entire system, however, so it is not cancelled.) Therefore, in applications where noise and hum are troublesome, the balanced input is used.

**MIXING AND FADING**

One of the important problems in p.a. work is the necessity of operating from more than a single source. Even the simplest of p.a. systems will have at least one microphone and one phonograph pickup, and will ordinarily have provisions for connecting
Another one in its place. And, as we mentioned before, the same is true of record players—when continuous music is desired, it is important to be able to fade out one player and run in another one without any appreciable break in the continuity of the program.

Therefore, p.a. systems have a number of input terminals, each with its own separate control to make it possible to adjust the levels individually. You will find that p.a. amplifiers differ widely in the number of such input channels provided, according to the uses for which they are designed. However, regardless of whether there are two microphone or phono input terminals or six, the following basic facts will apply.

**Resistance Mixing.** Fig. 18 shows two examples of what is known as resistance mixing. In Fig. 18A we have two microphone channels, each feeding into its own individual level control $R_1$ or $R_2$. By adjusting these controls individually, we can adjust the output from the corresponding channel to any desired level. Thus, it is possible to cut one off and the other one on, then to fade from the one that is on to the other one. Or, if desired, they may both be fed in at the same time at some predetermined level. From these controls, the signal goes through preamplifier tube $VT_1$ and then is resistance coupled to amplifier tube $VT_2$.

Potentiometer $R_a$ acts as a master volume control in that it controls the total signal level. With this form of control, it is possible to preset the mixer control $R_1$ and $R_2$ at some desired level and then use the master control to vary the volume as re-
quired. Placing the master volume control after amplifier tube VT₁ is desirable because all controls become noisy with use as poor contacts develop within them. Any noise signal caused by a control at the input of VT₁ will go through the entire amplifier and therefore receive maximum amplification. A similar noise caused by a control located at the input of VT₂ will produce far less noise output from the amplifier, because it will be amplified only by VT₂ and succeeding stages, not by VT₁ as well. In effect, then, placing a control at the input of VT₂ lengthens the life of the control, because it can get much noisier before it has to be replaced.

Going back now to the input: resistors R₃ and R₄ are used to prevent interaction between the two controls as much as possible. If these resistors were not used, and, for example, R₁ were set at zero, the grid of the tube would be grounded; there could then be no input no matter what R₂ was set. With resistors R₃ and R₄ in the circuit, however, the grid cannot be grounded by setting either R₁ or R₂ to zero; as a matter of fact, R₃ and R₄ are so large that adjusting either control throughout its range changes the resistance in the grid circuit very little. As a result, any adjustment of the control in one channel has little effect on the other channel.

The output from a microphone is always much less than that of any standard phonograph-record player. Therefore, there is always an extra triode or pentode preamplifier in the microphone channels. Notice that the phonograph outputs feed directly to the master volume control R₆ in Fig. 18A, whereas VT₁ acts as a preamplifier for all the microphone channels.

Although R₁ and R₂ get less use than the master volume control, they will still get noisy in time, and, because of the extra amplification, this noise will become objectionable very quickly. Furthermore, this particular form of resistance mixing always results in signal loss because R₃ and R₄ act as a voltage divider for any input signal. Since the signal is very weak at the grid of the preamplifier tube, very often the arrangement shown in Fig. 18B is used instead. Here, separate preamplifier tubes are used for each microphone channel, with the result that the very weak microphone signal feeds directly to the grid of its preamplifier tube and is boosted in volume at once. Then, each channel feeds into its volume control—R₃ for channel 1 and R₆ for channel 2. Resistors R₇ and R₈ are used to prevent too much interaction between these controls, just as R₃ and R₄ are in the circuit in Fig. 18A. Since the channel fader controls are now in the position occupied by the master volume control in Fig. 18A, it is common practice to eliminate the master volume control altogether and to use these fader controls as individual volume controls and as the fader-mixer control.

**Electronic Mixing.** Another input system is shown in Fig. 19A. This system is called “electronic mixing”; it is not the same as the electronic mixing with which you are familiar from your studies of radio, however, because the mixing does not occur in the electron stream of a tube. Separate amplifier tubes are used for each channel, both of which feed into a common-load resistor. This arrange-
FIG. 19. Two examples of electronic mixing.

ment makes it possible to adjust the input levels to either of the tubes without seriously affecting the other channels. The tubes thus act as decoupling devices that isolate the channels from each other.

Of course, if any channel is overloaded so that the plate resistance of its corresponding preamplifier tube changes, there will be an effect on the other channel, because each tube's plate resistance acts in shunt across load resistor \( R_3 \). This effect can be reduced by the arrangement shown in Fig. 19B, in which resistors \( R_5 \) and \( R_6 \) have been added to stabilize the two plate resistances. There is no appreciable interaction between the two channels when they are coupled this way.

Of course, this arrangement has the disadvantage of requiring that each channel be controlled at its input by a mixer control. As we mentioned, this is bad from the standpoint of noise production. Therefore, a combination consisting of two preamplifier tubes in each channel is sometimes used (see Fig. 20). Here, we still have the so-called electronic mixing in that tubes VT₃ and VT₄ feed into a common load resistor \( R_7 \). The controls are now not at the input—tubes VT₁ and VT₂ amplify their corresponding input signals so that the signals will be above any normal noise level produced by the control. A master volume control can be used at the input of VT₅ if desired, but in most cases the fader controls are used as volume controls.

When there are three, four, or more microphone channels, they can be connected in the same manner as two are. Usually all the microphone channels are treated alike.

**Phonograph Channels.** It is necessary to control the outputs of the phonograph-record players just as it is the outputs of microphones. If the system uses a master volume control like those shown in Figs. 18 and 19, it can be used to set the volume level. However, there is usually a separate control in each phonograph channel so that the average level can be set to correspond somewhat with the outputs from the microphone channels. Such a separate control is also necessary if phonograph music is to be used in the background behind programs.

FIG. 20. An improved electronic mixing circuit.
coming through a microphone channel. In such cases, it is necessary to balance the volume levels of the two channels so that they have the desired relative loudness. The master volume control can then be used to regulate the over-all volume.

Ordinarily, when there is more than one phonograph channel, a resistive mixing circuit like the one shown in Fig. 21A is used. As before, resistors $R_3$ and $R_4$ are inserted to prevent the controls from having too great an effect on each other.

A special fader control that is sometimes used for two phonograph channels is shown schematically in Fig. 21B. This control has a grounded center tap. As you can see, the output of each channel is applied across half the control. With this arrangement, there is zero output when the slider is set at the center. When the slider is moved toward one end of the control, the output from the channel connected to that half of the control is increased, but the other channel is cut off. If the control arm is moved in the other direction, the output of the other channel is increased and that of the first one is cut off.

This is called a fader control because it is possible to move from maximum volume for one channel down to zero for both and then gradually up to maximum for the other. Such a control has the worthwhile feature that only one hand is necessary to operate it.

Similar fading can be obtained with the controls shown in Fig. 21A, except that two hands must be used, one on each control. Since the operator may at that time have other duties, such as placing the pickup head properly on the record that is just starting, the one-handed control is desirable. However, it has an disadvantage in that you can only fade from one channel to the other, you cannot mix them. The control in Fig. 21A permits both record players to be operated at the same time, if this is ever desired.

Resistors $R_3$ and $R_4$ in Fig. 21A serve the same purpose they did in Fig. 18A—they prevent the controls from interacting on each other too much. Similarly, resistor $R_2$ in Fig. 21B acts as a decoupling resistor to prevent the control from grounding the grid circuit to which it connects.

**TONE CONTROLS**

Every form of tone control with which you are familiar in radio receivers is used in p.a. equipment. In addition, there are a few types found only in p.a. systems. Most of these involve some type of degeneration. A basic example is shown in Fig. 22. Here, when the switch SW is in position 1, all the a.c. components of the plate current must flow through $R_1$. Since the bias for the grid of the tube is developed across this resistor, all a.c. components are fed back equally.
so we have degeneration that is flat with respect to frequency.

When the switch is thrown to position 2, condenser C₁ is connected across R₁. If condenser C₁ is large enough, its reactance is so small that all audio frequencies are by-passed around R₁, and there is no degeneration at all. However, if this condenser is made rather small, its reactance comes into play. At low frequencies, it becomes a poor shunting path around R₁, so low frequencies are degenerated. On the other hand, since its reactance drops as frequency increases, it becomes a better by-pass at high frequencies, which are therefore not degenerated. Since degeneration reduces the output, this condenser now effectively reduces the bass response, because the bass frequencies are degenerated but the treble frequencies are not.

In position 3, a choke coil is substituted as the shunt across R₁. The large condenser C₂ is in series with the coil to act as a blocking condenser to prevent it from changing the bias by shunting R₁ by a d.c. path. However, the action is now the opposite to that when C₁ is in the circuit. Now, L₁ offers a low-impedance path for low frequencies, so there is no degeneration at these frequencies. It is a high-impedence path for high frequencies, however, so they are degenerated. Hence, the high-frequency response is reduced when the switch is in position 3.

The actual tone control circuits used are frequently more elaborate than this. We'll see some practical examples when we take up typical diagrams of complete amplifiers.
Typical P.A. Diagrams

In the following section, we are going to show two typical p.a. amplifiers. We have chosen these diagrams to illustrate some of the circuit ideas we have discussed. Other complete diagrams will be discussed elsewhere.

LOW-POWER AMPLIFIER

Our first example is shown in Fig. 23. An examination of the power supply shows that it is a standard a.c. type with a transformer, using a full-wave rectifier and a brute-force filter. There is nothing at all remarkable about the power supply.

This particular amplifier has one microphone and one phonograph pick-up connection. The microphone connection is of the high-impedance type, since it is arranged to feed directly into the grid of the 6J7 microphone preamplifier tube. The phono pickup is likewise of the high-impedance type and feeds into the grid of the second tube. The potentiometer \( R_4 \) acts as a volume control for the phonograph, and \( R_8 \) acts as the control for the microphone channel. No master control is used. Since \( R_8 \) is to be used as the volume control for the microphone channel, rather than just a level-setting control, it is in the grid circuit of the second tube that you would expect to find the master volume control. This arrangement permits the control to have a longer life, as you have learned, because any noise developed by the control is not amplified as much as is the signal from the microphone.

Resistors \( R_7 \) and \( R_8 \) are decoupling resistors used to prevent too much interaction between the two controls. It is possible to blend the phono pick-up in with the microphone signal if this is desired.

Whatever the signal source may be, the second (6SJ7) tube acts as the major voltage amplifier. Its output drives the grid of a 6L6 beam-power output tube.

The output transformer has a tapped secondary, the various taps of which are connected to a socket into which the loudspeaker line is plugged. Any of 5 output impedances can be selected by plugging the line into the proper terminals. Since this amplifier delivers only 8 watts, it is generally used to drive a single cone-type loudspeaker, although it can be arranged to drive two small loudspeakers at reduced output.

The output impedance of 4, 8, and 15 ohms provide for direct voice-coil connections, and the line impedance values of 250 and 500 ohms allow a transmission line to be used.

Inverse feedback is used to improve fidelity. The feedback path is from the 250-ohm tap on the secondary of the output transformer through \( R_{19} \) to the cathode of the 6SJ7 voltage amplifier. The inverse feedback voltage is developed across \( R_8 \), which is not by-passed. If the proper output transformer connections are made, this feedback voltage will be out of phase with the signal applied to the grid of the 6SJ7 tube; it will therefore reduce the over-all gain but at the same time will reduce even more any distortion developed within the voltage amplifier and output stages.
FIG. 23. Schematic diagram of the low-power Thordarson T-31W08 amplifier.

Courtesy Montgomery Ward
The tone control consists of resistor \( R_{13} \) and condenser \( C_5 \), which is connected to the slider of \( R_{13} \). As the slider is moved toward the grid end of the control, \( C_5 \) becomes more and more of a by-pass, thus reducing the high-frequency response of the amplifier.

**MEDIUM-POWER AMPLIFIER**

Fig. 24 shows a medium-power amplifier that has several unique features. There are two microphone inputs, each feeding into its own triode preamplifier. \( R_{11} \) is a gain control for microphone No. 1 and \( R_{10} \) a similar control for microphone No. 2. Notice that these are connected in an unusual manner—they appear to be backward from the way you are used to seeing volume controls. This connection makes it impossible for one gain control to short out the other when it is turned to zero, as you will find by examining the circuit. For example, if the slider on \( R_{11} \) is run up to the top, \( R_{11} \) is shunted by \( R_7 \) and by the plate impedance of the preamplifier tube for the No. 1 microphone. Therefore, it is never a complete short circuit. Resistor \( R_7 \) is necessary because the plate impedance of the preamplifier tube is not sufficiently large to make it a satisfactory shunt. \( R_9 \) is used similarly in series with the slider on \( R_{10} \).

There are two phonograph terminals, and the phono gain control is of the center-tapped type so that it can act as a fader from one to the other. An additional phono input is connected in parallel with phono input No. 1. However, this is for use with a built-in record player, which may be made a part of the amplifier cabinet. When this is used, phono input No. 1 is normally not used.

The phono gain control feeds into the grid of the 6SJ7 mixer tube, along with the microphone input. This is a resistance form of mixing, since all the signals are combined at the grid of this tube.

The plate of this tube is resistance-coupled to the control grid of the 6V6 driver tube. This driver tube is a beam-power tube but is connected here as a triode. It still furnishes considerable power through transformer \( T_1 \) to the grids of the actual power output tubes, which are two 6L6's connected in push-pull.

The tone control network consists of \( C_6 \), \( R_{14} \), \( R_{15} \), and \( C_4 \), which are connected in series from the plate to the cathode of the 6SJ7 mixer tube. When the slider on the tone control \( R_{16} \) is at the upward position (at the terminal connected to \( R_{14} \)), then \( C_6 \) and \( R_{14} \) are in series to ground from the plate of this tube. They act as a high-frequency by-pass. At the same time, all the resistance of \( R_{16} \) is in series with \( C_4 \), so this condenser is effectively no longer a by-pass across the cathode resistor \( R_{12} \). Therefore, complete degeneration occurs, which tends to flatten the over-all response.

When the slider on \( R_{16} \) is moved to the opposite end of the control, the full value of \( R_{16} \) is in series with \( R_{14} \) and \( C_6 \) is no longer an effective by-pass. At the same time, condenser \( C_4 \) is connected across \( R_{12} \). Since \( C_4 \) is a fairly small condenser, it is a very poor by-pass at low frequencies, so the low frequencies are still degenerated. It does become an effective by-pass at the high frequencies, however, thus reducing the degeneration at the
FIG. 24. Schematic diagram of the Airline Model 64BR-7320A, a medium-power portable amplifier.
high frequencies. Therefore, at this end of the control we are favoring the high-frequency response of the amplifier by reducing the effect of $C_6$ and putting $C_4$ in the circuit. At the other position, the high frequencies are reduced because $C_6$ is an effective bypass.

The output transformer has a tapped secondary, the taps of which are connected to a 4-position "speaker switch." Rotation of this switch connects the various taps to 4 paralleled sockets into which the loudspeaker lines are plugged. Thus it is possible to add or remove loudspeakers at will, provided the switch is set to give the proper impedance match. A separate socket is provided for use when a 500-ohm line is to be used.

Examining the power supply, we find that the B power supply is more or less standard. There is a direct connection, with no filtering except for the input filter condenser, to the plates of the output tubes. The supply to the output tube screen grid is filtered by an R-C filter consisting of $R_{22}$ and output filter condenser $C_{10}$. The plate supply of the 6V6 is filtered by $R_{22}$ and $C_{10}$ and is additionally filtered by choke $T_2$ and $C_{9B}$. Similarly, $R_{21}$ and $C_{9A}$ provide more filtering for the screen grid and plate of the 6SJ7 tube and the plates of the 12SQ7 tubes.

Because the preamplifier provides high gain, great care must be exercised to reduce hum. In this amplifier, the filaments of the 12SQ7 tubes are fed from a d.c. source; they are connected in series across $R_{23}$, which is in the B- lead of the power supply. Effectively, therefore, the plate current for all the tubes flows through these two filaments and through $R_{23}$. This means that the supply is nearly pure d.c., and is much more hum-free than an a.c. supply would be. Incidentally, the drop across this combination of $R_{23}$ and the two tube filaments also acts as grid bias for the 6L6 tube.
Lesson Questions

Be sure to number your Answer Sheet 49RH-3.
Place your Student Number on every Answer Sheet.

Send in your set of answers for this Lesson immediately after you finish them, as instructed in the Study Schedule. This will give you the greatest possible benefit from our speedy personal grading service.

1. If you had a 10-watt amplifier and another that was rated 3 db higher, what would be the power rating of the second amplifier?

2. What is the difference between the normal and the peak output ratings of an amplifier?

3. For what 3 reasons is it preferable to use several small amplifiers rather than one large one in setting up a high-power p.a. system?

4. What 2 features of beam power and pentode tubes make them better than triodes for use as output tubes in a p.a. amplifier?

5. Why can you get more than twice as much power from 2 tubes in class A push-pull as from a single tube in class A for the same relative amount of distortion?

6. Why is fixed bias commonly used with class A push-pull circuits in p.a. amplifiers?

7. What is the advantage of a balanced line compared to an unbalanced line?

8. Why is it desirable to insert a volume control after the first amplifier stage instead of ahead of it?

9. Why are the various input channels in a p.a. amplifier usually electrically isolated from one another?

10. In which of the four classes of operation (A, AB₁, AB₂, B) does grid current flow during part of the cycle?
ACOUSTICS IN
PUBLIC ADDRESS WORK

50RH-1

NATIONAL RADIO INSTITUTE
WASHINGTON, D. C.
ESTABLISHED 1914
STUDY SCHEDULE NO. 50

For each study step, read the assigned pages first at your usual speed, then reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind. Study each other step in this same way.

☐ 1. Introduction .................................................. Pages 1-3

The nature of the basic acoustical problems found in p.a. work are outlined in this section.

☐ 2. Microphones and Their Characteristics .................. Pages 3-16

Here the construction and operating characteristics of all the types of microphones now used are described.

☐ 3. Loudspeakers and Their Enclosures ....................... Pages 17-23

This section contains a description of the kinds of baffles used with loudspeakers in p.a. work and of the sound distribution patterns that each produces.

☐ 4. Practical Acoustics ........................................ Pages 24-31

Here you learn how the hearing characteristics of the human ear affect the design of a p.a. system and how the problems created by reflections of sound in an indoor installation are solved.


In this section you learn how to use the various factors discussed in the earlier parts of this Lesson to determine how much power is needed in a particular installation to produce the desired response.

☐ 6. Answer Lesson Questions and Mail Your Answers to NRI for Grading.

☐ 7. Start Studying the Next Lesson.
ACOUSTICS IN P.A. WORK

BEFORE it is possible to choose an amplifier for a particular location, it is necessary to have at least a general knowledge of some of the acoustic problems involved in public address work. Acoustics—the study of sound and its effects upon hearing—is considered to be a science, but is more of an art as it is practiced in p.a. work. That is, although it is possible to make carefully controlled scientific measurements of the conditions in a particular installation, such a scientific survey would be costly and would be of little use unless it were made under the exact conditions that exist when the equipment is in use. Therefore, in practice, acoustic problems in p.a. work are solved by using good judgment and past experience to a large extent. As we shall show later, certain tabulated information on acoustics is available that is helpful in planning an installation, but each job usually brings up its own special problems. Let’s see what some of them are.

Sound reflection and absorption cause trouble in indoor installations. Sound waves bouncing from wall to wall cause different effects, depending on the lengths of the paths traveled. Sounds coming from two directions to a particular spot may arrive 180° out of phase, with the result that the sound energy cancels, producing what is known as a dead spot. They may also arrive 360° out of phase, producing sound reinforcement. (There are several noted “whispering galleries” in which a whisper uttered in one spot can be heard at another spot perhaps 50 feet away, but nowhere else. This effect is the result of sound reinforcement.) Most commonly, the sounds are only partly out of phase; the result of this is usually that the sound is muddled and made hard to understand.

If the reflection path is long enough, there will be a complete echo—that is, the sound will arrive so much later over the longer path that it can be heard twice. This, too, is troublesome.

Another effect associated with reflections is reverberation. This occurs when there are many sound-reflecting surfaces in a room, as a result of which a sound is reflected many times and therefore takes a relatively long time to die out. The reverberation pe-
period of a room is measured by how long it takes a sound to drop 60 db from its original loudness. If this time is excessive, any continuous series of sounds produced in the room will seem extremely jumbled to a listener.

Another factor that varies from installation to installation is the surrounding noise level. This level plays an important part in determining the amount of power needed, because the p.a. system must have enough output to keep the average sound level well above the noise.

Absorption also creates problems. The system that sounds all right in an empty auditorium may not give enough power when the audience is present, because sound is absorbed by the clothing worn by the audience.

Outdoors, sound energy is rapidly dispersed even on a still day, because there are no containing walls to keep it in. If there is much wind, the sound dispersal is even more rapid. Noise is a problem outdoors also, of course.

All such factors must be considered before an installation is completed. As far as possible, they should be considered before the installation is even started; however, it is usually impossible to do much about reflections until the equipment is at least temporarily installed. (Reverberation, a special case of reflections, can be cured before installation of the equipment.) You can see, then, that far more is involved in making a p.a. installation than just setting up an amplifier, a few loudspeakers, and a microphone or two. The job must be carefully planned so that the installation will be adequate for its intended use but not so unnecessarily powerful that it is more expensive than it should be. Remember that the cost of an amplifier goes up directly with the power rating, because naturally more expensive power and output transformers must be used, as well as parts that have high wattage ratings.

**PLANNING A P.A. SYSTEM**

The purpose for which a p.a. system is to be used must be considered first of all when you are planning its installation. If the system is to be used only for paging or announcing, it should be designed to handle only the limited frequency range of the human voice; in this case, the system can be fairly inexpensive. If the system is to handle music, however, at least a fair degree of fidelity over a much wider frequency range will be necessary. This means that the microphones, amplifiers, and loudspeakers will have to be capable of delivering the required frequency range, and in general, that more power output will be required, as we shall show later.

Next, it is necessary to consider the location. It is possible to determine arbitrarily the amount of sound power that will be necessary to fill a certain cubic volume, so if we know the length, breadth, and height of a room, we can determine roughly what sound or acoustic power will be needed to fill it adequately with sound. To this basic amount, we must add enough power to overcome the effects of the average noise level plus enough more power to overcome the effects of reverberation. Then, once we have determined the acoustic power that will be needed, we can work backwards to find how much electrical power will be necessary. Certain specific kinds of loudspeakers and baffles may have to be used to meet the fidelity requirements, as we shall show later in this Lesson. Knowing the efficiency of these loudspeakers, we can determine how much electrical power output our amplifier has to have to produce the acoustical power needed. This sets the amplifier size.
Now that we have certain kinds of loudspeakers and an amplifier chosen, we must turn to the input. The number of microphones required depends on the conditions that are to be met. If the system is to be used for a large orchestra or to amplify the voices of actors who may be at different points on a large stage, a number of microphones may be needed. Very often, on the other hand, only a single one will be necessary. The types of microphones to use will depend on the fidelity wanted, on how rugged they must be, and on how necessary it is that they pick up only the desired sounds and ignore others.

Before we get into the acoustical problems of p.a. installations and learn exactly what must be done to solve them, we need to know more about the characteristics of loudspeakers and microphones. Let's take time out to study these two devices now.

**Microphones and Their Characteristics**

A public address amplifier may operate from a phonograph pickup, from a radio tuner that feeds a radio program to it, or from a microphone. The phonograph pickup and the radio tuner are covered elsewhere, so we shall consider only the microphone here. Incidentally, the microphone is the only one of these that brings up the problem of acoustic feedback, which we are going to study.

Any microphone is simply a device that will transform sound energy into electrical energy. Basically, all microphones contain some form of diaphragm—a movable cone, a plate, a ribbon, or the face of a crystal. When sound waves strike this diaphragm, the variations in air pressure cause it to move; its motion is used to set up an electrical current that varies correspondingly.

Let's examine the various types of microphones to learn something of their physical construction.

**CARBON MICROPHONES**

Essentially, the carbon microphone consists of a diaphragm that is in contact with either one or two “buttons” consisting of small packages of loose carbon granules or grains. Fig. 1 shows a cut-away view of a double-button type—one that has a button on each side of the diaphragm. A single-button type, of course, has only one button.

The diaphragm is a very thin metal plate, the edges of which are clamped in a ring assembly. The plate is so flexible that it vibrates when sound waves strike it. When it moves in on the package of carbon grains, they are pressed tightly together; when the diaphragm moves away from a button, the carbon particles separate or loosen up. When the carbon grains are pressed together, they make better...
electrical contact and the resistance through the button decreases. Conversely, the resistance through the button increases when they are allowed to be looser. In other words, the resistance of the buttons varies as sound waves strike the diaphragm;

![Diagram](image)

**FIG. 2.** How carbon microphones are connected to produce an output signal. The single-button type is shown in A, the double-button in B.

This varying resistance can be made to vary the current in a circuit by connecting the buttons in series with a battery.

The electrical connections for both single and double-button types are shown in Figs. 2A and 2B, respectively. In each circuit, the resistance $R_2$ is used to adjust the current to the desired initial value. Then the microphone causes the current to increase and decrease above and below this starting value in step with the sound waves. This varying current flowing through the primary of transformer $T_1$ induces a voltage in the transformer secondary; this voltage becomes the signal output of the microphone and can be fed to the grid of the first amplifier stage, either directly or through a transmission line. The transformer is necessary to match the low impedance of the microphone (200 to 500 ohms) properly to the transmission line or the grid of the first amplifier tube.

The double-button type is capable of giving better frequency response than the single-button. Both carbon microphones are relatively noisy compared to other types, however. Tiny sparks are formed as the carbon grains press together or loosen up, with the result that there is always an appreciable noise output. Although the carbon microphone gives a greater output than any other type, this noise trouble, and the need to use a rather large battery with it, have led to its almost complete disappearance from public address work. Today the only carbon microphones you're likely to find are certain hand-held microphones of the telephone type.

**CONDENSER MICROPHONES**

The condenser microphone, shown schematically in Fig. 3, is essentially a condenser whose two plates consist of a flexible diaphragm and a fixed plate. In Fig. 3, the diaphragm D is held in the clamp rings R, much as is the diaphragm of a carbon microphone. The plate P is very close to the diaphragm. The battery B furnishes a high voltage that charges the condenser formed by D and P. As the diaphragm is vibrated by sound waves, it alternately approaches and moves away from the plate P. This
increases and decreases the capacity. For a fixed voltage, the amount of charge that can be held by a condenser depends on its capacity, so this variation in capacity obviously changes the amount of charge stored in the microphone. Hence, a varying current flows through \( R_1 \) as the charge increases and decreases. The varying voltage drop across \( R_1 \) is the signal output of the microphone; this is fed out through the coupling condenser \( C_3 \).

Since the capacity of the condenser microphone is very small, the current change caused by movements of the diaphragm is measured in microamperes. As a result, \( R_1 \) must be very high in resistance for there to be an appreciable signal voltage. This means that the microphone must feed into a very high impedance for there to be an efficient signal transfer; as you know, any such high-impedance connection would be subject to hum and noise pickup if there were any considerable length of line between the microphone and the amplifier. Because of this fact, and because of the low output of the microphone, it is necessary to have a preamplifier right at the microphone; customarily, as a matter of fact, it is built into the microphone housing. Therefore, the housing must be rather large. Furthermore, the charging voltage for the microphone must be fairly high and must be pure d.c. if hum is to be avoided. Therefore, either batteries or an exceedingly well-filtered power supply is required.

Since a preamplifier is always a part of the microphone unit, it is customary to rate the output of a condenser microphone in terms of the preamplifier output. The condenser microphone therefore delivers a comparatively large output. However, its bulky nature and critical power-supply requirements make this a relatively unpopular type for p.a. use.

**CRYSTAL MICROPHONES**

Fig. 4 shows a cut-away view of a typical crystal microphone, and Fig. 5 shows its operational details. Once again we have a diaphragm that is clamped in a retaining ring. This diaphragm is coupled mechanically through a drive pin to a pair of Rochelle salt crystals. These crystals are very similar to the ones used in phonograph pickups. Two crystals are used, connected back to back. One terminal of the microphone unit is a tinfoil plate in contact with the two crystals where they join. On the outside of each crystal there is another plate; these plates are connected to form the other terminal.
Rochelle salt crystals exhibit what is known as the "piezo-electric" effect, meaning that a voltage will appear on the opposite faces of the crystal if the crystal is mechanically stressed in any way (or, conversely, that the crystal will be temporarily deformed if a voltage is applied to its opposite faces). In this unit, one edge of the crystal assembly is clamped tightly in the case and the other edge or corner of the assembly is secured to the diaphragm. As the diaphragm moves back and forth, the crystals are bent or twisted, which causes them to generate a voltage.

Fig. 6 shows another form in which a crystal microphone may be manufactured. In this unit, known as a "sound cell," groups of crystals are cemented into frames. The diaphragm and driving pin are dispensed with and the crystal units are acted upon directly by the sound waves.

Because the surface that is worked on by the sound waves is less in this microphone, the output is smaller than it is in one using a diaphragm.

However, the sound cell microphone is less affected by shock and vibration than is the diaphragm type, so it is popular in uses where it may be subjected to rough handling.

The crystal microphone is relatively rugged, and is less expensive than some of the other types. These factors make it one of the most popular of the microphones used in p.a. work.

It does have certain disadvantages, however, chief of which is that the crystals can be destroyed by very rough handling or by high temperatures. A crystal microphone cannot be used, therefore, in any location where conditions of high heat may exist. It is not a good microphone for use in a sound truck or for outdoor locations where the sun may get to work on it.

In the cut-away view in Fig. 4, there is a space in the microphone case marked "resonance chamber." We'll explain the purpose of this shortly.

**DYNAMIC MICROPHONES**

The dynamic microphone is almost the same as a p.m. dynamic loud-speaker, except that the cone is replaced by a diaphragm. Figs. 7 and 8 show the details of a typical one. A voice coil is placed in an air gap so that it is in a very strong magnetic field. When the diaphragm is actuated by sound waves, the voice coil (which is secured to the diaphragm)
is forced to move in and out through the magnetic field; as a result, a voltage is induced in the coil. This is passed on through a transformer mounted in the case to the output terminals.

As a matter of fact, a small p.m. dynamic speaker makes a relatively acceptable microphone—this idea is commonly used in intercommunication systems where the dynamic loudspeaker acts as a microphone when the appropriate switch is set in the "talk" position, but then is switched to be an actual loudspeaker at the output of the amplifier when the switch is allowed to return to its normal "listen" position. You'll learn more about this elsewhere.

The *dynamic microphone* is one of the most popular types used in p.a. work. It costs somewhat more than the average crystal microphone but is very rugged. It can be used where temperature and humidity conditions make the crystal type unsuitable.

Although the dynamic microphone is *not commonly* a high-fidelity microphone, it can be made to have a good frequency response, as we shall see.

**AIR-RESISTANCE LOADING**

In all of the microphones discussed so far except the sound cell, a diaphragm is used to convert motions of air particles into mechanical motion that may be used to generate the desired electric current. All such diaphragms contain sufficient material to have a certain amount of mass, and they are mounted so that the natural springiness of the material will tend to restore it to its original shape when sound pressure is removed. Since it has mass and springiness, which are the mechanical equivalents of inductance and capacity respectively, the diaphragm has a resonant frequency at which it will vibrate most readily. This resonant point is quite likely to occur in the audio spectrum, with the result that the microphone will exhibit a very undesirable peak in its response.

To a great extent, this peak can be ironed out by enclosing the back of the microphone so as to form an air chamber. A cut-away view of this arrangement in one form of dynamic microphone is shown in Fig. 9. A small tube, or vent, connects the air chamber to the outside air. You can understand the function of this vent readily if you have ever used a pump of the sort used to inflate footballs. Such a pump has a small, removable, hollow needle at one end through which the air being pumped out must pass. It is appreciably harder to pump air through this needle than it is to operate the pump with the needle.
removed. The reason is that the small opening offers considerable resistance to the movement of air through it.

By the same token, the small vent in the air chamber of the microphone in Fig. 9 does not pass air readily. Thus, when the diaphragm in this microphone moves inward, part of the energy of its motion is absorbed in forcing air out through the vent. If we again consider the diaphragm to be a resonant device, we can say that the air chamber and vent add resistance to the circuit. You know that adding resistance to an electrical resonant circuit reduces its output at the resonant frequency; similarly, the addition of this acoustical resistance to our mechanical-acoustical circuit reduces the tendency of the diaphragm to vibrate at its resonant frequency. As a matter of fact, it is possible to eliminate resonant effects almost completely by designing the air chamber and vent properly.

The cut-away views in Figs. 4 and 7 show the air chambers. Although Fig. 9 shows a dynamic microphone, the same general principle can be made to apply to others with diaphragms. Such microphones are called "pressure" microphones, because the voltages they generate are directly proportional to the pressures of the sound waves striking them.

![FIG. 9. Cut-away view showing the resonance chamber and vent in a dynamic microphone.](image)

**FIG. 10. A typical velocity microphone.**

**RIBBON MICROPHONES**

The ribbon microphone, shown in Figs. 10 and 11, is rather different from the types we have discussed so far, because it has no circular diaphragm. Instead, a very thin ribbon of an aluminum alloy is suspended between the poles of a powerful magnet. The ribbon is clamped at its ends, where connecting wires attach it directly to a matching transformer. The ribbon completes the primary circuit of this transformer and therefore acts as a 1-turn coil. When it moves in the magnetic field, a voltage is induced in it.

To permit movement of the ribbon, it is crimped or "accordion pleated." This ribbon has no springiness whatever, and very little mass—it is so light that it practically floats in air. When sound waves strike it, the ribbon moves back and forth in step with the air particles. The microphone is enclosed only by a perforated shield (which was removed before the picture in Fig. 11 was made) that offers no resistance to the free movement of air in and out.

Since the ribbon moves in step with
the moving air particles just as if it were an additional air particle, it is said to respond to the velocity of the air particles rather than directly to the actual pressure of the wave. For this reason, you’ll find that the ribbon microphone having both the front and back of the ribbon exposed to sound waves is called a “velocity” microphone.

**Pressure Type.** It is possible to make the ribbon microphone respond to sound pressure like other microphones, however, by enclosing the back of the ribbon in an air chamber. Fig. 12 shows the most common way of doing this. A pipe is used to enclose the rear surfaces of the ribbon completely. This pipe then leads down into a box at the bottom of the microphone where there is an air chamber. Enclosed on one side in this manner, the ribbon acts like a diaphragm, so the microphone becomes a pressure-actuated device.

The ribbon microphone is rarely used in p.a. work, because of its extreme delicacy. A single gust of wind, or a sharp puff of air from a person speaking directly into one, will un-

![Image](31x32 to 545x780)

**FIG. 11. Internal appearance of a velocity microphone.**

![Image](586.0x828.0)

![Image](586.0x828.0)

**FIG. 12. Front view (A) and side view (B) of the internal appearance of a pressure-operated ribbon microphone.**

crimp and straighten out the ribbon so that it sags completely out of position. This calls for a replacement of the ribbon, which can be done only at the factory. When these microphones are moved, they must be carried in a normal operating position—that is, with the ribbon in a vertical plane. Carrying the microphone in a horizontal position makes the ribbon sag or stretch. Jarring or rough handling may cause the ribbon to move far beyond its normal limits, with the result that it may be stretched out of shape or even stick to the magnet. In addition, rough handling may cause the magnet to move. The spacing in this microphone is very small to begin with, so even a slight change in the position of the magnet will restrict the air gap so much that the ribbon cannot move properly in it.

Despite all these difficulties, the velocity microphone is used in some high-quality installations, particularly when music is being picked up, because it offers higher fidelity than does any other kind of microphone commonly used. Should you encounter such a microphone, remember the
above characteristics. Shield it always from wind, and instruct persons speaking into it to stay well away from it and speak “across” the face of the microphone rather than directly into it. Always see to it that a velocity microphone is kept away from alternating current fields such as may be produced by power transformers and by power lines. If anything is the matter with such a microphone, don’t open it; it must go back to the factory for repair. Under factory conditions, in air-conditioned, dust-free rooms, it is possible to repair one. However, even taking the screen off to examine such a microphone in an ordinary service shop is quite likely to permit metal particles to get into the air gap and prevent it from working.

For that matter, it is not desirable to try to repair any kind of microphone. If you suspect the microphone of causing trouble, it is far better to try another in its place. If the substitute works properly, then something is the matter with the original microphone and it should be sent back to the factory for repair.

You have now learned basically how all the important types of microphones work, except for the cardioid types, which are combination microphones that we shall discuss a little later. Now let’s compare the characteristics of the various microphones to see what makes one type better than the others for different uses.

**FREQUENCY RESPONSE**

Practically any kind of microphone will prove satisfactory for voice pick-up. However, there is quite a difference in the responses of microphones to music. Furthermore, we can’t say that just because a particular microphone happens to be a crystal type or a dynamic type that it necessarily must have a certain specific fidelity, because it is quite possible to get a better response by careful design of the unit. For example, many of the more common dynamic microphones are reasonably flat over a frequency range of only 100 to 5000 cycles, but high-fidelity types are available that have flat responses from 25 cycles to 12,000 cycles. Other dynamics have responses in between these two extremes.

The same can be said for the crystal microphone, whose response may range from perhaps 100 to 7000 cycles to as much as 30 to 10,000 cycles. Velocity types are practically all high fidelity, with responses from 40 to somewhere between 10,000 and 15,000 cycles, depending on design.

The obsolete carbon types were all low-fidelity units, which is one reason for their disappearance from the p.a. field. The condenser microphone actually offers the widest frequency response of all, but, because of the disadvantages we discussed earlier, it is not used in p.a. Therefore, in general, if the conditions of use would permit either the crystal or dynamic microphone to be used, it is necessary to be sure that the one chosen has a frequency response that is suitable for the fidelity wanted. Naturally, the prices of microphones go up as their fidelity becomes better, because a high-fidelity microphone must be carefully made and uses costly materials. At the same time, high-fidelity microphones are usually more delicate than are low-fidelity units. Hence, it is common practice to choose a microphone that meets the fidelity requirements of the installation but does not exceed them much.

Microphones are like loudspeakers in that their response over a frequency range is not uniform but instead has many peaks and dips. In general, the
dynamic microphone is particularly subject to such variations and the velocity type is least subject to them. However, a well-made, high-quality microphone will have a smoother response than an inexpensive type.

**PICKUP PATTERNS**

Microphones do not respond equally to sounds coming from different directions. Some types exhibit definite directional characteristics.

All of the diaphragm types that we have studied are usually made with an enclosure at the rear of the diaphragm. Effectively, therefore, the diaphragm faces only one way in these units. As you might expect, they are much more sensitive to those sounds coming straight toward the front of the diaphragm than they are to sounds coming from other directions.

However, these types are classed as non-directional microphones because at low frequencies (below 1000 cycles) they do tend to respond to sound waves from all directions. This comes about because at these frequencies the microphone itself is rather small in comparison to a wave length, with the result that the diaphragm is operated upon by the pressure of a sound wave regardless of the direction of the wave. At higher frequencies, however, these microphones become at least semi-directional in that they respond better to sound coming from the front (see Fig. 13).

![Fig. 13. This graph shows how a nondirectional microphone picks up sound coming from various directions. The response at three different frequencies is shown. The front of the microphone faces the 0° line.](image)

**FIG. 14. A microphone that is relatively non-directional in its normal position (A) becomes even more so if it is turned to face upward (B). The response can be further improved by putting a shield above the microphone (C).**

If such a microphone is to exhibit good frequency response, then, it must be made to face the source of the sound so that its response will be approximately equal to all frequencies in its normal response range. Hence, the microphone and its stand must be placed so that the microphone faces the source of the sound that is to be picked up.

If sounds from several different directions are wanted, the microphone can be made much more non-directional by pointing it upward. For example, in Fig. 14A, the microphone faces the left, so sounds coming from this direction will be picked up best. The sound pickup will be poorest from the right in this drawing. However, if the microphone is swiveled on its stand so that it faces directly upward (Fig. 14B), it will receive sound best from directly overhead, but will pick up equally from all horizontal directions.

An improvement over this latter arrangement is shown in Fig. 14C. Here a metal shield is placed a short dis-
distance from the opening of the microphone. This prevents sound coming from directly overhead from being picked up much and improves the pickup from the sides.

The ribbon microphones that have their rear sides enclosed in a baffle, which makes them pressure-actuated, operate just like other pressure microphones as far as directionality of pickup goes. Of course, as you learned earlier, these microphones should not be turned upward because of the possibility that the ribbon will be damaged. Velocity ribbon microphones, which are open on two sides, are most sensitive from directly in front or directly in back, and least sensitive at the sides, as shown in Fig. 15. Sound is blocked off from the sides by the mass of the magnetic structure and by the wind shield that encloses the microphone. Therefore, response is greatest along the 0° and 180° lines in Fig. 15, and decreases gradually to a minimum at 90° and at 270°.

This bidirectional response can frequently be made use of when you have two different sound sources to pick up simultaneously. Suppose, for example, you want to pick up the music of an orchestra that is playing in a pit in front of a stage. The orchestra will be in two groups, with the conductor in the middle. You can get the desired pickup by placing the microphone in front of the conductor and orienting it so that the two halves of the orchestra are in line with the lines of maximum response of the microphone. This orientation will not only permit the orchestra to be picked up well but will also minimize pickup from the audience, which will be on either the 90° or 270° line of the microphone.

Incidentally, the problem of picking up unwanted sounds such as audience noise, is a severe one in p.a. installations. In fact, very often the possibility of noise pickup determines both the kind of microphone that should be used and the place where it should be located. We shall have more to say about this later in this Lesson.

**Cardioid Responses.** Several microphones have been developed that

![Cardioid Diagram](image-url)

**FIG. 16.** The cardioid response is produced by combining the responses of a velocity and a pressure unit.

are combinations of pressure-operated and velocity-operated units. These have pickup patterns like that shown in Fig. 16. This pattern is said to have a "cardioid" shape, because it resembles somewhat the shape of a heart.

A microphone having this response
picks up best from in front, less well from the sides, and very little from the rear. It is therefore very useful in applications where there is a single source of unwanted noise: the microphone can be turned so that its rear is toward the noise source, and pickup of the noise will be minimized.

![Cardioid Pattern](image)

**Fig. 17.** The response curve of one type of cardioid microphone. Notice the difference between this curve and the true cardioid shown in Fig. 16.

It is also possible to make a microphone having the modified pickup pattern shown in Fig. 17. As you can see, this pattern has two minimums. The microphone will pick up to some extent from the rear but nowhere near as much as from the front. At angles of about 130 and 230 degrees, it has minimum response. A microphone having these characteristics is particularly useful where there are two noise sources.

The cardioid microphone usually contains a ribbon velocity element in combination with something that will act as a pressure device. The kind shown in Fig. 18 has a ribbon element on top and a dynamic unit underneath it. A switch arrangement makes it possible to use the ribbon alone for a bi-directional response, the magnetic unit alone for a non-directional response, or the two in combination for a cardioid response. The amount of response from the two units can be varied to produce the response shown in Fig. 17, also.

Other combinations are also available, such as a ribbon and crystal unit. A third variety uses only a ribbon that has an air chamber behind half the ribbon and none behind the rest of the ribbon. With this unit, the half with the air chamber acts as a pressure-actuated type and the other half, of course, as the velocity unit.

Still another kind of microphone, known as the Super-Cardioid, has the directional effects of the 2-unit cardioid but contains only a single pressure-actuated unit (either crystal or dynamic). The cardioid effect is

![Microphone Cross-Section](image)

**Fig. 18.** Cross-sectional view of a microphone that can be used as a nondirectional, bidirectional, or cardioid microphone by turning the switch (lower right) to the proper position.
achieved by incorporating a special acoustic chamber in the microphone housing.

The cardioid reception pattern is obtained from a combination of pressure and velocity units because of the difference in the manner in which the two units respond to sound waves. When the waves come from the front of the microphone, both units are energized simultaneously. Their signal voltages are therefore in phase; and, when they are added in a suitable network, they produce an increased output. When the sound waves come from the back of the microphone, however, the action is not the same. The velocity unit is energized as soon as the waves reach the microphone, but the pressure unit is not energized until the waves reach the front of the microphone a short time later. The output signals of the two units are now out of phase; therefore, they cancel when they are combined, producing a minimum response to waves coming from the rear of the microphone.

Incidentally, the bi-directional response of the velocity microphone does not vary much with frequency: practically the same pattern is obtained for all frequencies to which the microphone responds. Some of this same effect is carried over to the cardioid, although here the pressure-actuated device can cause the pattern to vary somewhat with frequency.

Although it is never a true cardioid, the response of a non-directional microphone can be sharpened so that the response is mostly from the front by the use of an acoustic shield around the face of the microphone. Such a shield plate cuts down on the energy received from any direction except the front. Certain microphones are equipped with such shields; they are usually removable so that non-directional response can be obtained when it is desired.

**MICROPHONE OUTPUTS**

Microphones differ considerably in their output levels, even though all are low and require the use of high-gain amplifiers. The carbon microphone has the greatest output for a fixed sound level; the condenser microphone and its built-in amplifier have nearly as much; the crystal microphone has the next greatest output; and the dynamic microphone output ranges from about the level of the crystal microphone down to that of the velocity, which has the least power output.

Naturally, if you are to drive an amplifier to full output, the microphone you use with it must supply at least the minimum input power for which the amplifier was designed. As a practical matter, it is best to use the kind of microphone recommended by the manufacturer of the amplifier, if he makes any recommendation. If the amplifier manufacturer does not recommend a specific microphone, you must choose one that has a suitable output. If low-impedance dynamic and velocity microphones can be used with a particular amplifier, any other kind can also be used with it, because all other kinds have higher outputs.

Microphone sensitivity ratings are often confusing, because at least six different reference levels are in use. Most manufacturers rate their microphones in terms of the electrical output across a properly matched load at a reference frequency, with respect to a particular reference sound pressure. A few rate microphones unloaded, however; doing so gives an output that is 6 db more than it will actually be when the microphone is properly matched. (The unloaded voltage is higher because, when the
microphone is properly loaded by an impedance equal to its own impedance, half the source voltage is dropped across the microphone impedance.

Microphones are usually rated in terms of decibels down from either a reference voltage or a reference power, with the reference sound pressure given in dynes per square centimeter. (Sometimes the pressure is stated in bars; a bar is equal to one dyne per square centimeter.)

The reference voltage is usually 1 volt, but the reference power may either be 1 milliwatt or 6 milliwatts. Table 1 gives the six most commonly used reference levels. As a typical example, you may find the rating of a microphone given as "—50 db below 1 volt/1 dyne/cm² into a load of 1 megohm." When the complete rating is given this way, you know at least what reference level was used. On the other hand, if the listing is just "—50 db," as it frequently is in supply-house catalogs, you won't know what reference level was used; and you may be badly misled if you compare the output level of this particular microphone with that of another that was rated on the basis of a different reference.

For example, three different pressure reference levels are given in Table 1, each 10 times the pressure of the one preceding. A 10-times difference in pressure on a microphone increases its output by 20 db. Therefore, the same microphone could be rated at —70 db below 1 volt/1 dyne/cm², or —50 db below 1 volt/10 dynes/cm², or —30 db below 1 volt/100 dynes/cm².

Similarly, a power rating in terms of 1 milliwatt is 8 db higher than it would be if the microphone were rated on the basis of a 6-milliwatt reference level. In other words, a microphone rated at —50 db for the 1-milliwatt level would have to be rated at —58 db if the 6-milliwatt level were used as the reference.

All this means that we have to be careful to choose a microphone whose db output level is high enough to give full rated output from the amplifier used. Then, when we compare microphones made by different manufacturers, we must be careful always to make sure that their ratings are in terms of the same reference; otherwise, we may get the wrong idea of their relative outputs. If you cannot tell what rating standard was used from the information given, write both the manufacturer of the microphone and the manufacturer of the amplifier. One or the other will be able to tell you whether the particular microphone and amplifier you are interested in will work properly together.

Of course, once you have had experience with particular brands of microphones, you won't have to worry about the reference standards used, because you will know what their ratings are.

**MICROPHONE IMPEDANCES**

In general, microphones are classed as either low impedance or high impedance. The ribbon microphone has a very low impedance, and it nearly always has a built-in transformer that is designed to match the microphone either to a 500-ohm audio line or directly to the grid of an amplifier tube. Dynamic microphones have imped-
ances ranging from around 8 ohms up to about 50 ohms. Sometimes built-in transformers won’t be provided with those around 50 ohms, but the ones commonly used in p.a. work all have transformers designed to match them to 500 ohms or to a high-impedance input.

The only other common type—the crystal microphone—is usually a high-impedance microphone.

Amplifier inputs are generally designed either for high-impedance microphones or for 500-ohm transmission lines. One designed for a high-impedance microphone can be used with either a crystal microphone or a magnetic or velocity microphone that has an appropriate matching transformer.

When high-impedance inputs are used, the cable from the microphone to the amplifier cannot be very long. One reason is that there will be considerable frequency attenuation, as we shall learn elsewhere. Another reason is that if any point in the circuit is at a high impedance with respect to ground, very small stray hum and noise fields will introduce fairly large disturbing voltages. And, of course, the longer the section of the circuit above ground, the more likely there is to be trouble. It is therefore necessary to keep the microphone cable as short as possible—lengths are usually held to 10 to 25 feet at the most.

If the amplifier has a 500-ohm input, on the other hand, it is possible to use a 500-ohm transmission line, which permits cable lengths to be as much as 1000 feet. When a 500-ohm line is used, it is of course necessary that the microphone have a transformer designed to match it to the line and that the line be matched to the grid of the input tube of the amplifier by another transformer.

As a general rule, therefore, we can say that if the microphone is to be used within 10 to 25 feet of the amplifier, we can use a high-impedance microphone that is connected directly to the amplifier. This may be either a crystal microphone or a dynamic or velocity microphone containing a transformer that matches its impedance to that of the amplifier input circuit. A dynamic or velocity microphone that is matched to 500 ohms by its built-in transformer can also be used if it is connected to the 500-ohm input of the amplifier or if it is connected to another transformer that will match 500 ohms to the high-impedance input of the amplifier.

On the other hand, if the microphone is to be used at a greater distance from the amplifier, we must either use a low-impedance type matched to a 500-ohm line, which in turn is matched to the amplifier, or we must feed from a high-impedance microphone into a preamplifier that is a separate unit from the main amplifier. Then, this preamplifier can be connected to the main amplifier at a distance by proper matching through a 500-ohm line, as we will show later.
Loudspeakers and Their Enclosures

You have studied loudspeakers elsewhere in your Course, so we shall not have to spend time here to describe their operation. Instead, we shall discuss their use in p.a. work.

A few magnetic loudspeakers are used in p.a. installations, but dynamics are by far the most common. Permanent-magnet dynamics are almost always the kind chosen, because they do not require a field supply. Since the loudspeakers must frequently be mounted at a great distance from the amplifier, it would be impractical to furnish a field supply from the amplifier, because the extra pair of leads in the cable would greatly increase the cost and complicate the installation. Therefore, if an electrodynamic loudspeaker were to be used in such cases, it would have to have its own built-in field supply, which would have to be connected to a source of power. This would greatly increase the expense and would probably cause a higher hum level.

Therefore, the electrodynamic loudspeaker is commonly used only in small portable p.a. systems in which the loudspeaker is built into the amplifier assembly or is connected to it by a rather short cable.

The voice coil impedances of the loudspeakers used in p.a. work are similar to those of the loudspeakers used in home radio receivers: 4 ohms, 8 ohms, and 16 ohms are the most common.

Two basic loudspeaker types are used in p.a. installations. One is the familiar kind in which the voice coil drives a paper cone; in the other, the voice coil drives a metal diaphragm.

The paper-cone type is usually found in the lower-powered indoor installations and in high-fidelity in-
Fig. 19B shows one way this problem can be solved. As you can see, the diaphragm has a ball-shaped indent in it, and there is a plug in the center of the sound chamber whose rear edge is shaped like the indent in the diaphragm. The motion of the diaphragm forces air to flow around the plug and thence through the throat into the horn. This arrangement makes it practically impossible for any sound waves to be reflected from the walls of the sound chamber back to the diaphragm; instead, any reflected waves are channeled toward the throat by the sloping sides of the plug and the chamber. Many variations of this plug system have been worked out, but they all work on similar principles.

LOUDSPEAKER BAFFLES

A cone loudspeaker unit must be enclosed in some form of baffle to produce a reasonable coupling to the air. The shape and size of this baffle in a radio receiver depend on the fidelity and the efficiency desired. The same factors enter into p.a. work, and in addition, we have to worry about the possibility that sound from the outdoors without ample protection against weather.

These disadvantages of cone loudspeakers have led to the development of high-powered driver-type units that have metal diaphragms instead of paper cones. Such driver units are invariably used with horn enclosures, which we shall describe shortly. When the diaphragm is properly coupled to the air by a horn enclosure, it is possible to get an efficiency of 15% to perhaps 30% from a driver unit.

The basic structure of a driver unit is shown in Fig. 19A. For the horn size to be practical, the throat of the horn must be relatively small, considerably smaller than the diaphragm. Therefore, the diaphragm in this figure drives the throat through a sound chamber. Effectively this gives a very good coupling to the air, with the result that large amounts of air are moved at the throat. However, there is some difficulty with the frequency response, because, particularly at high frequencies, there are reflections within the sound chamber.

![Courtesy Allied Radio Corp.](image)

FIG. 20. These are box baffles of the sort commonly used in portable p.a. systems.
loudspeaker may travel through the air to the microphone. If sufficient energy can get from the loudspeaker back to the microphone, the system can become a self-sustaining oscillator, because this fed-back sound can replace the original sound and continue to repeat itself over and over through the microphone-amplifier-loudspeaker-air-microphone path. For this reason, loudspeaker baffles for p.a. work commonly have closed backs; this makes it possible to operate the loudspeaker near the microphone location without fear that the sound coming from the back surface of the loudspeaker cone will reach the microphone directly. An open baffle can be used only when the loudspeakers are located in such positions that feedback is unlikely.

Let's see what various common baffles are like.

**CONE-LOUDSPEAKER BAFFLES**

The simplest enclosure for a cone loudspeaker is the box baffle shown in Fig. 20. Two such box baffles are commonly used in portable p.a. systems, the two being so constructed that they can be secured together to form a closed box in which there is room for the amplifier when it is desired to carry the whole system from one place to another.

A baffle of this sort is not sufficient to give high fidelity, but it is adequate for voice or popular music. Since the back of this baffle is completely open, it must be carefully located with respect to the microphone to prevent feedback from the loudspeaker to the microphone.

Another simple baffle is shown in Fig. 21. This is a box that is intended to be hung on a wall. If enclosures of this sort are properly scattered around, well away from the microphone, it is possible to keep the feedback down to a satisfactory level. This baffle is actually enclosed at the back when it is mounted firmly against the wall, but since it is mounted so that it faces into the room, it can feed sound into the microphone unless the latter is carefully placed.

The larger cabinet baffles that are used where better tone quality is desired are generally completely enclosed at the rear. In most instances,
such units are of the bass reflex type, an example of which is shown in Fig. 22.

Any of the baffles described so far gives a relatively broad sound distribution somewhat like that obtained from a radio receiver. There are occasions, however, when it is desired to project sound in a more compact "bundle" to a distance, or when it is necessary to prevent sound from going in certain directions to eliminate feedback. With cone loudspeakers, projectors (sometimes called trumpets) are used for such occasions. An indoor type is shown in Fig. 23. Basically, this is a directional enclosure,

![FIG. 23. A cone loudspeaker mounted in a projecto housing.](image)

![FIG. 24. A cross-sectional view of a cone loudspeaker mounted in a weatherproof projector. Such an assembly can be used outdoors.](image)

![FIG. 25. This shows how a cone loudspeaker mounted in a box baffle (A) and one mounted in a projector horn (B) differ in their sound distribution characteristics.](image)

very similar to a horn in its directive effects.

Outdoors, a variation of the projector is the only kind that is practicable with cone loudspeakers. Cones must be protected from the weather outdoors, so a weather-proof projector like the one shown in Fig. 24 is used. This is so designed that rain and spray will not seriously affect the cone even if they enter the mouth of the projector directly.

**Sound Distribution.** Incidentally, the sound output from loudspeakers is rather peculiarly distributed. Fig. 25A shows the result of using a cone in any standard wall or cabinet baffle. As you can see, low frequencies are distributed rather uniformly from the front of the baffle over a wide area. Medium and high frequencies become more and more directional, however;
the sound distribution at the highest frequencies is practically a narrow beam straight in front of the cone. This unequal sound distribution presents quite a problem if we are interested in high-fidelity sound distribution. It is obvious that only the people who are directly in front of the cone will get all the frequencies with equal intensity.

The projector distribution shown in Fig. 25B is much more nearly uniform. However, here we run into the fact that the projector isn’t a very good baffle, because its low-frequency response is poor for reasonable projec-

To improve sound distribution, high-fidelity installations frequently use dual loudspeakers. In such installations, a large cone loudspeaker is used to give low-frequency coverage; the high frequencies are handled by a small loudspeaker unit (usually a driver type) that is designed to give an angle of coverage that approximates the medium-frequency coverage of the large cone. Fig. 26 shows one type of high-frequency loudspeaker, which consists of a pair of driver units arranged with dual horns at such an angle that a rather wide coverage is obtained. Fig. 27 shows a “cellular” construction in which the horn is broken into segments that disperse the sound to give a wide angle of coverage. This horn is driven by a single driver unit.

A form of dual loudspeaker that is commonly used in high-fidelity installations is shown in Fig. 28. This unit, called a coaxial loudspeaker, is used chiefly because it has a wide frequency range. In the immediate vicinity of such a loudspeaker, the fidelity is quite good, but it does not offer particularly wide-angle high-frequency coverage. Where a large area is to be covered with such loud-
In general, the horn must be rather long to have good low-frequency response. Since it should increase regularly in cross-sectional area as it increases in length, we must start with a very small throat if we are to have a reasonable mouth size in any practical horn length.

Horns that carry speech only need to handle only a limited frequency range; therefore, they can be, and commonly are, rather short. However, if music is to be carried through the horn, it must be long—so long, in fact, that the space required by the horn is quite a problem. One solution to this problem is to fold the horn up on itself as shown in Fig. 30. Even folded in this manner, the horn is rather large; a horn of this sort is generally used only in large auditoriums or theaters.

A more commonly used arrangement for getting a relatively long horn length in a small space is shown in Figs. 31 and 32. This device is known as the re-entrant or reflex horn. The name comes from the fact that the sound travels down an inside horn,

**HORN ENCLOSURES**

Some form of horn or trumpet enclosure is invariably used with driver units. Both the fidelity and the coverage angle are largely determined by the kind of enclosure chosen. A long, narrow horn with a small mouth tends to project sound directly in front of the mouth of the horn without allowing it to spread very much. On the other hand, if the horn flares outward rapidly, sound is distributed over a much wider angle.

From a fidelity standpoint, the rate of increase of the cross-sectional area of the horn is particularly important.
then is forced back toward the rear before it finally comes out of the mouth of the horn, as shown in Fig. 32. Because of this internal folding, it is possible to make the over-all dimensions of the horn rather short and yet have a fairly long air column. Furthermore, such a horn is weather-

FIG. 31. A typical reflex trumpet, much used for outdoor installations.

FIG. 32. Cross-sectional view of a reflex trumpet.

FIG. 33. A coupling of this sort makes it possible to use 2 driver units with one horn.

Courtesty University Loudspeakers, Inc.

This is an extremely powerful loudspeaker in which 12 driver units are used. It can handle powers up to 300 watts and can be heard for several miles, making it ideal for outdoor use.

Most drivers designed for use with horn units are rated at 25 watts but will work efficiently on 8 to 10 watts. If greater power is needed, extra loudspeakers may be used, or more than one driver may be used with a single horn. Fig. 33 shows a two-unit type; as many as twelve drivers are used on super-powered horns.

Now that you have a general idea of what the pickup patterns of microphones and the sound distribution patterns of loudspeakers are like, let's take up the practical problems of determining how much power is necessary for an installation.
Practical Acoustics

The amount of power needed for any particular installation depends on a number of factors. First of all, the hearing characteristics of the human ear must be considered. There must be a certain amount of power before the human ear registers any sound at all, the exact amount depending on the frequency of the source. At this threshold level, the ear is not at all a high-fidelity device; therefore, considerably more than this minimum power is needed to permit an audience to hear comfortably and with reasonably good fidelity.

As we have pointed out before, the noise level at the location of the installation must also be taken into account in determining the amount of power needed; the greater the noise, the greater the power that will be necessary. Indoors, we also have the problem of sound reflection from the walls and ceilings. Sound reflection is seldom a problem in an outdoor installation, but sound dispersal is. Let's make a complete study of each of these factors in turn to see how they affect the amount of power needed.

HEARING CURVES

The ear is very peculiar in the manner in which it responds to sound levels at different frequencies. It is most sensitive to sounds at about 2000 cycles. In other words, a very low-power sound at this frequency will be audible. At low or high frequencies, however, far more power is necessary to make a sound audible.

Fig. 34 contains a series of curves that indicate the average hearing ability of the human ear. Sounds having the intensities shown by curve A can just barely be heard, and sounds having lower intensities can not be heard at all: curve A is therefore called the “threshold of hearing.” Notice that this curve is very non-linear, illustrating what we just said about the ear being most sensitive at the minimum-loudness level to sounds around 2000 cycles and least sensitive to low-frequency and high-frequency sounds.

This variation in sensitivity with frequency becomes less marked at higher loudness levels. The dashed curves above curve A show the response of the ear at various loudness levels 10 db apart. (The threshold of hearing is used as the zero db reference.) As you can see, the response becomes much flatter as the loudness increases.

If a sound is made loud enough, the ear will feel pain instead of hearing the sound. The loudness level at which pain is felt (which is called the “threshold of pain”) is represented roughly by curve B in Fig. 34. Notice that this curve intersects the threshold of hearing at very low and very high frequencies but is widely separated from it at the middle frequencies. At frequencies around 1000 to 2000
cycles, the change is roughly about 120 to 130 decibels from the threshold of hearing to the threshold of pain.

You can see from these facts that the average person is able to hear only the middle frequency range if the sound level is very low; the low and high frequencies are completely inaudible. As the sound level is increased, higher and lower frequencies can be heard.

Obviously, the sound output of a p.a. system should be at least great enough to permit all the frequencies we are interested in to be heard comfortably. This means that the power required for a particular installation depends on what the system is intended to carry. If it is to be used for instrumental music, a wider frequency range must be handled than is needed if only voice frequencies are to be carried; consequently, more power is needed for the former kind of installation.

For convenience in comparing sound levels, it is standard practice to choose a reference frequency in the range where the hearing is most acute. The level necessary to produce an audible sound at this reference frequency is then considered to be the threshold of hearing, and other sounds and noises are said to be a certain number of decibels above this threshold level.

**EFFECT OF NOISE**

The ability to hear any sound is considerably affected by the noise level. Theoretically, even the weakest of the sounds in which we are interested should be at a level above the surrounding noise level if it is to be heard easily. Therefore, we need to know the noise level before we can choose the p.a. system.

Fig. 35 shows the sound levels of various common noises, and the noise levels that are found in typical places where p.a. systems may be used. Notice that the noise level in the average quiet home is about 35 db above the threshold; since the average conversation level is higher than this, we, of course, need no amplification to overcome the noise in a home. As a matter of fact, p.a. amplifiers are not needed to overcome noise until the noise level is above that of the desired sound. Acoustics standards state that the average sound level for speech should be maintained at least 10 db above the surrounding noise level. This is not practical, of course, when the noise level is up near the threshold of pain, because the sound level might then be over the threshold for some frequencies. It is therefore frequently impossible to keep the sound level above the surrounding

<table>
<thead>
<tr>
<th>Type of Sound Source</th>
<th>DB Level</th>
</tr>
</thead>
<tbody>
<tr>
<td>Threshold of painful sound</td>
<td>130</td>
</tr>
<tr>
<td>Hammer blows on steel</td>
<td>120</td>
</tr>
<tr>
<td>Riveting machine</td>
<td>100</td>
</tr>
<tr>
<td>Factory (very noisy)</td>
<td>90</td>
</tr>
<tr>
<td>Machine Shop (average)</td>
<td>90</td>
</tr>
<tr>
<td>Heavy street traffic</td>
<td>85</td>
</tr>
<tr>
<td>Printing Press</td>
<td>80</td>
</tr>
<tr>
<td>Ball Rooms</td>
<td>80</td>
</tr>
<tr>
<td>Restaurant (noisy)</td>
<td>80</td>
</tr>
<tr>
<td>Factory (average)</td>
<td>75</td>
</tr>
<tr>
<td>R.R. waiting room</td>
<td>75</td>
</tr>
<tr>
<td>Auditorium (average)</td>
<td>75</td>
</tr>
<tr>
<td>Office (busy)</td>
<td>65</td>
</tr>
<tr>
<td>Department store (average)</td>
<td>65</td>
</tr>
<tr>
<td>Auditorium (quiet)</td>
<td>65</td>
</tr>
<tr>
<td>Ordinary conversation</td>
<td>60</td>
</tr>
<tr>
<td>Quiet residential street</td>
<td>60</td>
</tr>
<tr>
<td>Restaurant (average)</td>
<td>60</td>
</tr>
<tr>
<td>Store (quiet)</td>
<td>60</td>
</tr>
<tr>
<td>Office (quiet)</td>
<td>60</td>
</tr>
<tr>
<td>Hotel lobby</td>
<td>55</td>
</tr>
<tr>
<td>Hospitals</td>
<td>55</td>
</tr>
<tr>
<td>Average quiet residence</td>
<td>35</td>
</tr>
<tr>
<td>Quiet garden</td>
<td>25</td>
</tr>
<tr>
<td>Average whisper</td>
<td>20</td>
</tr>
<tr>
<td>Rustle of leaves in gentle breeze</td>
<td>10</td>
</tr>
<tr>
<td>Threshold of hearing</td>
<td>0</td>
</tr>
</tbody>
</table>

**FIG. 35.** These are the levels in db above the threshold of hearing of various common sounds and noises. The figures have been compiled from several sources.
noise level to any great degree in installments in very noisy factories.

For ordinary music, it is desirable to have the average sound level 15 db higher than for voice, or a total of 25 db above the noise. High-fidelity reproduction of symphonic music requires another 10 db above ordinary music, or an average level 35 db above the noise level. Of course, there will be peaks that exist above the average levels; however, proper design on an average power basis permits the peak power capabilities of the amplifier to handle these.

One of the problems always facing the sound engineer, therefore, is the determination of the noise level at the location where a p.a. system is to be installed. This determination must, of course, be made under the conditions that will be present at the time the p.a. system is to be used. An empty auditorium is far quieter than one filled with people. This is particularly true at a sporting arena, where an enthusiastic crowd of spectators can make the noise level very high.

To determine the noise level, one must guess at it (a very difficult thing to do accurately), measure it with a noise level meter, or depend upon practical tables or charts like Fig. 35. Loudspeaker manufacturers give average levels in charts designed around their particular loudspeakers. We'll say more about this later.

**SOUND REFLECTIONS**

As we have already said, sound reflections from the walls, floors, and ceilings of a room are a major problem in indoor p.a. installations. These reflections provide additional paths over which sound waves travel from the source to the listener. Fig. 36 gives a simple example.

Such reflections occur because whenever sound waves strike a surface, some of the energy is absorbed and lost, some is transmitted through the material, and the remainder is reflected much as light rays are reflected by a mirror. How much reflection there is depends on the material; hard, smooth materials like plaster reflect far more than do soft materials like drapes. These reflections "save" energy by preventing it from escaping from the room. However, the re-

![Diagram of sound reflections](image)

Fig. 36. The direct sound wave between the source and the listener travels over path 1, which is the shortest path between these two points. Waves traveling over paths 2, 3, or 4 must go a greater distance to reach the listener, and consequently arrive somewhat later than those taking path 1.
turn of this sound energy is not instantaneous; it takes more time for sound to travel over a longer path, so sound waves that reflect from wall to wall do not arrive at a given point in step with sound waves coming over a more direct path. Such reflected waves may cause the sound at any particular spot to be louder, softer, or unintelligible. Let's study this last effect first.

**REVERBERATION**

When the surfaces of a room are hard and smooth, reflections occur and recur, with the result that it takes time for sounds to die out. Consequently, syllables or words traveling over direct paths are interfered with by earlier sounds traveling over the reflection paths. This prolongation of sounds, which is called reverberation, is the most common acoustic problem in auditoriums.

Unless a room is made absolutely dead by special acoustic treatment (by making the surfaces absorb energy instead of reflecting it), there will always be a certain amount of this reverberation. The actual amount depends on the size and shape of the room and on the characteristics of the materials used in the room. We don't want a room to have no reverberation —such a room sounds "dead," and music or speech is flat in it. A certain amount of reverberation makes a room "alive"; music, in particular, has more brilliance and richness of tone in such a room.

To determine what treatment may be necessary to make a room more nearly ideal in this respect, engineers assume that the period of reverberation is the time it takes for a sound to decrease in energy by 60 decibels. To measure this period, a short, sharp sound is made, and timing devices are used to determine when it has decreased by this amount. If the time taken is reasonable for the size of the room, no treatment is necessary.

In general, the larger the room, the longer the reverberation period that can be permitted. There is no exact agreement on the amount of time that is permissible, however, because this depends upon whether it is speech or music that is to be reproduced and upon what the installer thinks is an ideal "liveness" for the room. Usually periods of under two seconds are necessary. "Ideal" periods for music in rooms of various sizes are shown in Fig. 37. For speech, the ideal is from half to two-thirds the values given in this figure.

As an example of how excessive reverberation affects the ability to understand speech, a reverberation period of 5 seconds in a 5000-cubic foot auditorium reduces the number of recognizable syllables to only about 60% of the total. At least 75% recognition is necessary for intelligibility with very careful listening, and about 90% is needed for high-quality reproduction. In a room of 5000 cubic feet, a reverberation period of about .6 second is required for 90% intelligibility.
ACOUSTIC TREATMENT

Since the reverberation time is related to the volume of the room in cubic feet and to the absorbing ability of the surfaces of the room, there is a formula that can be used for calculating the approximate reverberation period of a room. It is:

\[ t = \frac{0.05V}{a} \]

where “V” is the volume of the room in cubic feet, “t” is the period in seconds, and “a” is the number of absorption units of the materials used.

We shall discuss absorption units in a moment.

This formula makes it possible to calculate the approximate period for a room. If the period is wrong, we can determine how much the absorption has to be changed to make the reverberation proper by restating the formula as:

\[ a = \frac{0.05V}{t} \]

which gives us the number of absorption units needed for a room of volume V to have the desired reverberation period t.

In general, acoustic treatment of a room is best left to an acoustical engineer. In a large auditorium, proper acoustic treatment involves a considerable expense, so it is far better to have the room treated by someone familiar with the materials that can be used for the purpose. If you have such a problem, therefore, you should call in an engineer or a representative of a company manufacturing sound-absorbing material. However, so you will understand what must be done, let’s see how such an expert would go about planning the acoustic treatment of a room.

![A small broadcast studio that has been acoustically treated with Acousti-Celotex tile on the ceiling and carpeting on the floor. The walls have been irregularly shaped to improve sound diffusion. Acoustical treatments in rooms served by p.a. systems are similar, though seldom so extensive.](image)

The number of absorption units in a room is computed by multiplying the area in square feet of each surface by a factor (called the absorption coefficient) that indicates the absorbing power of each square foot of the material. The total number of absorption units in the room is the sum of these, plus the units furnished by the audience and by the furniture.

As a general rule, any hard, smooth surface has very little absorption, so materials such as plaster walls will reflect sound and keep the reverberation period high. The same can be said for hard floor materials and for wooden seats.

On the other hand, soft, coarse ma-
terials absorb sound, so the period of reverberation can be reduced by the use of drapes or other cloth hangings, upholstering or pillows on the seats, rugs on the floor, etc. Even better sound absorption can be obtained through the use of special acoustic materials, which are commonly made of cane fibers. These materials either have a rough surface or have a surface with many small holes in it that break up the sound reflection and absorb much of the energy of the sound wave. Covering plaster ceilings and walls with such materials cuts down greatly on the reverberation and also reduces the noise (since it, too, is absorbed).

<table>
<thead>
<tr>
<th>Materials</th>
<th>Coefficients</th>
</tr>
</thead>
<tbody>
<tr>
<td>Floor Coverings:</td>
<td></td>
</tr>
<tr>
<td>Carpet</td>
<td>.20</td>
</tr>
<tr>
<td>Cork flooring</td>
<td>.08</td>
</tr>
<tr>
<td>Linoleum</td>
<td>.03</td>
</tr>
<tr>
<td>Rug, Axminster</td>
<td>.20</td>
</tr>
<tr>
<td>Wood flooring</td>
<td>.03</td>
</tr>
<tr>
<td>Hangings:</td>
<td></td>
</tr>
<tr>
<td>Fabrics:</td>
<td></td>
</tr>
<tr>
<td>Light</td>
<td>.11</td>
</tr>
<tr>
<td>Medium</td>
<td>.13</td>
</tr>
<tr>
<td>Heavy</td>
<td>.50</td>
</tr>
<tr>
<td>Hard Wall:</td>
<td></td>
</tr>
<tr>
<td>Brick, painted</td>
<td>.017</td>
</tr>
<tr>
<td>Cement</td>
<td>.025</td>
</tr>
<tr>
<td>Plaster on lath</td>
<td>.03</td>
</tr>
<tr>
<td>Openings:</td>
<td></td>
</tr>
<tr>
<td>Window</td>
<td>.5—1</td>
</tr>
<tr>
<td>Balcony</td>
<td>.5—1</td>
</tr>
<tr>
<td>Audience and Chairs:</td>
<td></td>
</tr>
<tr>
<td>People</td>
<td>3—4.3</td>
</tr>
<tr>
<td>Chairs, wooden</td>
<td>.17</td>
</tr>
<tr>
<td>Chairs, upholstered</td>
<td>1.6</td>
</tr>
<tr>
<td>Acoustic Materials:</td>
<td></td>
</tr>
<tr>
<td>Acoustil-Celotex C-2</td>
<td>.67</td>
</tr>
<tr>
<td>Acoustil-Celotex C-4</td>
<td>.99</td>
</tr>
<tr>
<td>Acousticone F</td>
<td>.87</td>
</tr>
<tr>
<td>Fiberglas Tile (1&quot;)</td>
<td>.97</td>
</tr>
<tr>
<td>Permacoustic (1&quot;)</td>
<td>.71</td>
</tr>
</tbody>
</table>

**FIG. 38.** The absorption coefficients of various materials. The figures given for audience and chairs are in terms of absorption units per person or per chair; the other figures are for absorption units per square foot. These units were determined at 512 cycles. The absorption at other frequencies differs somewhat, usually, though not always, increasing at higher frequencies.

**FIG. 39.** The computations needed to determine the reverberation period of the room described in the text before it is acoustically treated.

The presence of an audience may change the characteristics of a room considerably. Clothing is very efficient as an absorption material.

Fig. 38 gives a general idea of the absorption coefficients of several typical materials. (The figures given for people, wooden chairs, and for upholstered chairs are absorption units per person or per chair, not absorption coefficients.)

To take a practical example, let’s suppose we have a small hall 100 feet by 20 feet by 10 feet high, which has a volume of $100 \times 20 \times 10 = 20,000$ cubic feet. Let’s suppose it has a wood floor and plaster walls and ceilings. Let’s also suppose there are about fifty wooden chairs in the hall.

Fig. 39 shows the details of calculating the absorption units present in the basic hall, using the average coefficients given in Fig. 38. There are 2000 square feet of floor space, and wood flooring has an absorption coefficient of .03, so the floor has a total of 60 units. A plaster wall around the room has a total area of 2400 square feet; its absorption coefficient is also .03, making its absorption 72 units. The ceiling has a total
of 60 units and the chairs a total of 8.5 units. The sum of all these is 200.5, which we can round off to be 200 units.

The volume of the room is 20,000 cubic feet, so the time, as shown by the calculations, is five seconds. This is too long; Fig. 37 shows that it should be about 1.17 seconds for a room of this size if music is to be played in it.

An audience of fifty people present in the chairs will change matters, because the audience has an absorption of about four units per person or a total absorption of 200 units, which

<p>| | |</p>
<table>
<thead>
<tr>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Carpet:</td>
<td>(100 x 20 = 2000) x .2 = 400</td>
</tr>
<tr>
<td>Med. drapes on</td>
<td></td>
</tr>
<tr>
<td>walls:</td>
<td>2400 x .13 = 312</td>
</tr>
<tr>
<td>Plaster ceiling</td>
<td>2009 x .03 = 60</td>
</tr>
<tr>
<td>Upholstered chairs: 50 x 1.6 = 80</td>
<td></td>
</tr>
</tbody>
</table>

\[
t = \frac{.05 \times 20,000}{852} = 1.17 \text{ sec.}
\]

FIG. 40. How acoustical treatment affects the reverberation period of the room.

cuts the time in half, or to 2.5 seconds. Therefore, this hall will have much better characteristics with an audience than it has when empty. Even so, it still has too long a period. Using the formula for determining the absorption units needed, we find that to produce a 1.1-second period, we need:

\[
a = \frac{.05 \times 20,000}{1.1} = 910 \text{ units}
\]

(approximately) instead of the 200 to 400 we have.

Covering the floor with carpet, hanging medium-weight drapes on the walls, and using upholstered chairs produces the effects shown by the calculations in Fig. 40—the reverberation period is changed considerably. Our time of 1.17 seconds is now much better for a room of this size. With an audience adding 200 more units, the time is reduced to about one second, so this treatment is just about right.

Of course, a treatment that involves hanging drapes completely around the room, installing a carpet over the whole floor, and changing from wooden chairs to upholstered chairs cannot be described as a simple one. It may be less costly and more satisfactory in the long run to leave the floor and chairs alone and to have an acoustic material applied to the wall or ceiling. If we were to cover the entire wall with Acousti-Celotex type C-4, the number of absorption units for this treatment alone would be 2376 (2400 \(\times .99\)). This would be too much and would make the room rather dead, because the reverberation period would then be only about .4 second. To come out around 700 units, so that with an audience (200 units) the period will be about one second, we need only about 600 feet of this acoustic material on the wall. Therefore, it is possible to hang several panels of this material at various points along the wall and thus deaden the room just as much as it would be deadened if
we were to hang drapes over all the walls and put a carpet on the floor.

As you can see, there are a number of different things that can be done to change the reverberation period of a room. Initial costs, ease of application, and upkeep costs must all be considered in selecting a method of treatment. This is particularly true when a large auditorium is to be treated, because the cost of such a project may be very high.

An auditorium intended to seat several thousand people is a difficult problem to treat acoustically because of the fact that the audience may vary in size from just a few people to a capacity crowd. There will obviously be a tremendous difference in the absorption of the auditorium under the two extreme conditions; if the treatment is such that the reverberation period is correct when the auditorium is filled to capacity, the reverberation will be excessive when the audience is small. Usually the treatment for such an auditorium is calculated on the assumption that it is to be only moderately full. Then, as the audience varies around this average, the period is made slightly higher or lower, but never varies as much as it would if we assumed either zero or a capacity audience.

FOCAL POINTS AND DEAD SPOTS

Another factor that must be considered in planning a p.a. installation is the possibility that the shape and size of the room will cause unequal sound distribution over the floor area. An example of just such a room is shown in Fig. 41. The dome-shaped ceiling of this auditorium provides sound paths that tend to concentrate the sound from the origin to a spot in the balcony. At this particular spot, the sound will be excessive.

On the other hand, it is equally possible for the shape of the room to cause dead spots—points at which there is sound cancellation because the sounds arrive out of phase over two different paths. Such spotty responses are not likely in small rooms but are quite common in large auditoriums. In such cases, it is either necessary to treat the room acoustically to break up these reflection points or to place the loudspeakers so that the sound is more evenly distributed. The latter method is usually preferable, since it is less difficult and expensive than is changing the contour of a room.

![Figure 41. Sounds reflecting from various points on the curved ceiling of this auditorium are brought to a focus at a single small area in the balcony, making the volume level there considerably higher than it is elsewhere.](image-url)
Determining Acoustical Powers Needed

From the foregoing section, you can see that there are a number of factors involved in determining how much acoustical power will be necessary to obtain the desired performance from a p.a. system in a given location. Engineering methods can be used to calculate the exact power necessary, but the practical sound man seldom bothers to make such elaborate computations. Instead, he uses some table or graph that gives a general idea of the power that should prove suitable under average conditions.

Most such published tables, which, incidentally, may be found in the literature of the loudspeaker manufacturers, assume that the reverberation period of the room is normal for its size and for the conditions that will be met. If you find upon examination that the room is not normal in this respect, a correction must be made to make the room suitable for a permanent sound installation.

If the reverberation in the room is reasonable or is made so, we can find the power necessary if we know the volume of the room in cubic feet and the noise level that can be expected.

Unless you are going to go to the expense of purchasing or renting a noise level meter to check the exact level, you will have to depend on the averages that have been found for installations that are similar.

Tables, such as Figs. 42 and 43, give acoustic powers needed for particular noise levels and room volumes. To use them, you will have to estimate or determine the noise; the tables then give the approximate acoustic power needed for a room having the area of the one in which you are interested. Notice that the minimum powers for the reproduction of speech or music are given—more power may be needed if the room is dead or if it is necessary to overcome dead spots or reflections in the room.

Once you have determined how much power is needed, you will have to divide the number of acoustic watts by the speaker efficiency to get the electrical wattage the amplifier must supply. For example, if the acoustic power is .5 watt, and you are using ordinary cone loudspeakers, which are about 2% efficient, the amplifier must have an output of 25 watts of electrical power \((\frac{.5}{.02} = 25)\).

Generally, it is best to choose an amplifier rated somewhat above this minimum, to allow for losses in the transmission lines, and for possible increases in the noise level.

The tables in Figs. 42 and 43 are incomplete, since they cover only a

<table>
<thead>
<tr>
<th>Noise Level (db)</th>
<th>Area (sq. ft.) 500-2000 Assumed Room Height (ft.) 10-15</th>
<th>Area (sq. ft.) 2000-5000 Assumed Room Height (ft.) 15-20</th>
<th>Area (sq. ft.) 5000-10,000 Assumed Room Height (ft.) 20-25</th>
<th>Area (sq. ft.) 10,000-30,000 Assumed Room Height (ft.) 25-35</th>
<th>Area (sq. ft.) 30,000-70,000 Assumed Room Height (ft.) 35-50</th>
</tr>
</thead>
<tbody>
<tr>
<td>70</td>
<td>0.001-0.004</td>
<td>0.004-0.010</td>
<td>0.010-0.019</td>
<td>0.019-0.056</td>
<td>0.056-0.126</td>
</tr>
<tr>
<td>80</td>
<td>0.012-0.044</td>
<td>0.044-0.100</td>
<td>0.100-0.199</td>
<td>0.199-0.562</td>
<td>0.562-1.26</td>
</tr>
<tr>
<td>90</td>
<td>0.126-0.447</td>
<td>0.447-1.0</td>
<td>1.0-1.99</td>
<td>1.99-5.62</td>
<td>5.62-12.6</td>
</tr>
<tr>
<td>100</td>
<td>1.26-4.47</td>
<td>4.47-10.0</td>
<td>10.0-19.9</td>
<td>19.9-56.2</td>
<td></td>
</tr>
</tbody>
</table>

*Courtesy John F. Rider*
few noise levels. However, the trend of powers is obvious, so you can fill in for lower or higher noise levels by the simple process of dividing or multiplying by a factor of 10 for each 10 db decrease or increase in noise.

In attempting to estimate the amount of noise, you can use tables like that shown in Fig. 35 or charts that you obtain from loudspeaker manufacturers. Such loudspeaker charts give usual noise levels and the power needed for various room volumes when using certain particular loudspeakers. These charts apply only to the loudspeakers made by that manufacturer—you should obtain the one for the brand in which you are interested, because differences in efficiencies and coverage angles exist that make them wrong for other brands.

Let’s sum up what we have learned about calculating how much power is needed for an indoor installation. First, you must determine whether the room has the proper reverberation period or needs acoustic treatment. Then, from a table or chart, you must find how much acoustic power is needed for a room of the size of the one with which you are concerned, taking into consideration the average noise level of the room and whether music and speech, or speech alone, is to be carried by the p.a. system. Naturally, even more acoustic power is needed for the high-fidelity reproduction of music than is necessary for ordinary dance music or for speech.

To convert acoustic power into electrical power, you must know the efficiencies of the loudspeakers you intend to use. As we said earlier, cone loudspeakers are commonly considered to be 2% efficient in baffles and 5% efficient in projectors. By dividing the acoustic power level (in watts) by the speaker efficiency (expressed as a decimal), you will find the electrical power needed.

**SOUND OUTDOORS**

We have no reverberation problems outdoors but do have the problem of rapid attenuation of the sound. Since there are no walls to reflect energy back to the audience, sound power goes down 6 db for each doubling of the distance from the loudspeakers to the listener.

Horn loudspeakers are generally used in these installations. The horns may have either narrow or wide coverage angles, depending on the installation. Both types have their advantages and disadvantages. Horns with wide coverage angles cover a larger area, but since the sound is spread out over this area, it is weaker at any distance from the horn than it would be for a horn with a narrower coverage angle. On the other hand, if we use narrow-angle horns and must cover a wide area, we have to use more of the horns to cover this area properly.

---

FIG. 43. Minimum acoustic power required to override noise for normal p.a. requirements for speech and music reproduction in indoor coverage areas indicated. Areas are in square feet.
<table>
<thead>
<tr>
<th>Noise Level (db)</th>
<th>10-30 ft.</th>
<th>30-75 ft.</th>
<th>75-150 ft.</th>
<th>150-300 ft.</th>
<th>300-500 ft.</th>
<th>500-1000 ft.</th>
</tr>
</thead>
<tbody>
<tr>
<td>70</td>
<td>0.002-0.017</td>
<td>0.017-0.112</td>
<td>0.112-0.501</td>
<td>0.501-1.78</td>
<td>1.78-5.01</td>
<td>5.01-20.0</td>
</tr>
<tr>
<td>80</td>
<td>0.020-0.178</td>
<td>0.178-1.12</td>
<td>1.12-5.01</td>
<td>5.01-17.8</td>
<td>17.8-50.1</td>
<td></td>
</tr>
<tr>
<td>90</td>
<td>0.200-1.78</td>
<td>1.78-11.2</td>
<td>11.2-50.1</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>100</td>
<td>2.0-17.8</td>
<td>17.8-11.2</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Courtesy John F. Rider

**FIG. 44.** Minimum acoustic power required to override noise for reproduction of speech outdoors for coverage of indicated distance in feet. A coverage angle of 30° is assumed. More power is required if larger angles of coverage are used.

Fig. 44 shows a table for determining the sound power necessary outdoors. Notice that the table is for a certain specified coverage angle of the horn.

A comparison of Fig. 44 with Figs. 42 and 43 shows that the acoustic power needed outdoors is far higher than it is for indoor installations. However, since horns have a 15% efficiency, the actual electrical power increase needed is not as great as you might at first imagine. For example, if the indoor acoustic power needed is 1 watt, and 2% efficient loudspeakers are used, 50 electrical watts are necessary. With a 15% efficient loudspeaker, 1 acoustic watt is obtained from only about 6 electrical watts, however. Fifty watts delivered to 15% efficient loudspeakers will deliver as much as 7.5 acoustical watts, which is a respectable amount of sound power.

**PLACING LOUDSPEAKERS**

Either indoors or outdoors, once we decide on the electrical power that will be needed, we must then determine both from the power level and from the surrounding conditions the number of loudspeakers that will be needed. Cone loudspeakers are available in various power-handling capacities from as low as 1 watt to perhaps 40 watts. If the necessary amplifier power level is higher than one loudspeaker can handle, then obviously more than one must be used. Most driver units are rated at about 25 watts, but they operate satisfactorily from powers as low as 6 to 8 watts. However, if the output from the amplifier is greater than 25 watts, again more than one loudspeaker is needed.

Extra loudspeakers may be needed to give the proper coverage for the area. There are locations at which it is best to use a number of loudspeakers and divide the sound for better dispersion. In some instances, such as when sound is distributed to hotel or hospital rooms, this is a necessity—a small loudspeaker must be placed in each room, which of course means that there will be quite a number of loudspeakers.

Even when the major sound distribution comes from one or two large loudspeakers located near the source of sound, a few supplementary loudspeakers may be necessary to take care of spots that would otherwise be dead.

Incidentally, when the loudspeaker is in the room in which the performance is occurring, it is considered good practice to get the loudspeakers somewhat near the source of sound, so that the sound will apparently be coming from its source. This, of course, introduces the problem of feedback to the microphone through the air, which means that the loud-
speakers must be so enclosed or so positioned that the feedback will not be excessive.

Of course, this does not mean that all the loudspeakers must be grouped in one place. Even if the main ones are so grouped, there may have to be supplementary loudspeakers to feed sound into dead spots, under balconies, etc.

When you are feeding sound into rooms other than the one in which the performance is occurring, you do not have to worry about feedback to the microphone. It is possible to use a single cluster of loudspeakers in such a room, but because sound may be excessively loud near the loudspeakers and too weak farther away, it is more common in this kind of installation to scatter the loudspeakers about. The only problem here is to make sure that the coverage is approximately uniform over the entire room.

We will go into greater detail on loudspeaker placement when we take up typical installations elsewhere. For now, let’s cover a few general rules that will prove helpful in any installation.

**Loudspeaker Phasing.** When more than one loudspeaker is used in a cluster, it is important that the voice coils be connected so that the sounds from these loudspeakers are in phase. If the loudspeakers are outwardly identical, you can usually assume that the connections from the voice coil to the terminals on the loudspeaker are the same on each, and you can connect similar terminals together when the loudspeakers are in parallel, as shown in Fig. 45A. Fig. 45B shows the proper way to connect loudspeaker voice coils in series.

It is always possible, however, that the manufacturer has reversed one of the windings; if so, neither of these connections will be right for these particular loudspeakers. When the loudspeakers are out of phase, the sound from them tends to cancel. Therefore, the correct in-phase connection will be the one that gives the louder response for the same fixed input. If there is any doubt about this, you can make the simple test of listening to the loudspeakers while you reverse the connections to one of them.

You don’t have to worry about the phase of loudspeaker connections when the loudspeakers are very widely separated, particularly when they are outdoors.

![Fig. 45. Proper method of connecting loudspeaker voice coils in parallel (A) and in series (B).](image)

**Coverage Angles.** Loudspeakers in box baffles have fairly wide coverage angles at low frequencies, but the angle of coverage for the middle and high frequencies is more restricted. For this reason, loudspeaker placement may become rather critical. If it is desired to have the sound apparently come from the source on a stage, the loudspeakers should preferably be mounted above the stage and should be tilted downward to point toward the audience. If there are two loudspeakers, better results can be obtained by placing them to the right and left of the center of the stage, turning them so as to give the greatest coverage.

If a room is to be covered by a series of separated loudspeakers instead of by a centralized group, you
can either locate them along the longer wall or use ceiling loudspeakers that have 360° coverage. Incidentally, when loudspeakers are located along a wall, they should never be more than about 40 feet apart; if they are more widely separated than this, there will tend to be an echo effect as sound comes to listeners from different loudspeakers.

In general, outdoors, it is preferable to have the loudspeakers in a single cluster if possible. Of course, it may not be possible to use a single cluster. In football stadiums, for example, it may be necessary to string the loudspeakers around so that each covers a portion of the audience.

**MICROPHONE PLACEMENT**

The proper placement of microphones is often a problem. Of course, if voice is being picked up, it is common practice to have the microphone directly in front of the person speaking. Stage presentations, however, often require not only that the sound be picked up over a wide area but also that the microphone be concealed. Generally, in such cases, from two to four microphones are located in the footlight region of the stage, or several microphones are suspended from above the stage so as to cover as much of the area as possible.

When instrumental music is to be picked up, it is often necessary to locate the microphone or microphones very carefully with respect to the various instruments being used. Several typical examples are shown in Fig. 46. In cases of this kind, the only practical way to find the right microphone positions is to be present at a rehearsal and try various positions until the proper ones are found.

**FIG. 46. Practical microphone placements for picking up (A) a 2-piano team, (B) a small salon orchestra, and (C) a dance orchestra.**
Lesson Questions

Be sure to number your Answer Sheet 50RH-1.

Place your Student Number on every Answer Sheet.

Send in your set of answers for this Lesson immediately after you finish them, as instructed in the Study Schedule. This will give you the greatest possible benefit from our speedy personal grading service.

1. Which is more suitable for use in a location where the temperature may get high—a crystal or a dynamic microphone?

2. What type of microphone should be used in an installation where there is a single source of unwanted noise?

3. Arrange these three microphones in the order of their relative outputs, starting with the one having the highest output: dynamic, velocity, crystal.

4. What is the maximum distance at which a high-impedance microphone can be connected directly to an amplifier: 25 feet, 250 feet, 1000 feet?

5. What is the chief reason why p.m. loudspeakers are preferred to electro-dynamic loudspeakers in p.a. installations?

6. Why are cellular horns used on high-frequency tweeters?

7. What are the two chief advantages of a reflex horn?

8. How does the noise level affect the power that a p.a. system must furnish?

9. When several loudspeakers are mounted on a wall, what is the greatest distance they can be separated without danger of causing an echo?

10. What happens if two loudspeakers connected to the same amplifier are incorrectly phased?
the source impedance to give good low-frequency response. Hence a
transformer designed for "3600 ohms
to 4 ohms" is different from one de-
signed for 14,400 ohms to 16 ohms, al-
though both have the same 30-to-1
ratio.

For this reason, transformers aren't
listed by turns ratios; they are de-
scribed by the impedances between
which they are to work. Thus, one
crated at "500 ohms to 8 ohms" is de-
signed to match a source (or a line
matched to such a source) of 500 ohms
to a load of 8 ohms. However, it can
be used to match 250 ohms to 4 ohms,
because the source is lower in im-
pedance than the value for which the pri-
mary was designed. It can also be
used to match 1000 ohms to 16 ohms
with some loss in low-frequency re-
sponse. In other words, the secondary
impedance can be varied over a range
from one-half to twice the value for
which the transformer was designed
without causing too great a loss in
power (under .5 db) and without af-
flecting the frequency response too
seriously in any but high-fidelity
systems.

**IMPEDEANCE-MATCHING PADS**

A resistor network can be used in-
stead of transformers to make an
impedance match. A disadvantage of
using the resistor network is that it
always introduces a loss; however,
there are occasions when such a loss is
permissible or even desirable.

We can, of course, connect the load
and source directly together as shown
in Fig. 10A. As long as the difference
in their impedances is not too great,
there won't be a large power loss. For
example, in the case shown in Fig.
10A, the load impedance is one-half
the source impedance. From the curve
in Fig. 10B, we find that when the
generator impedance ($R_G$) equals
twice the load value ($R_L$) as in this
case, we have a loss of only about .5
db. This isn't much power loss. As
the difference between the source and
load impedances becomes greater,
however, the power loss also increases.
For example, if we have a 1000-ohm
source and a 250-ohm load, $R_G$ equals
$4R_L$, and, as the chart shows, we have
a 2-db power loss, which represents a
loss of about a third of the power.

With the impedance relationship
shown in Fig. 10A, we are not losing
much power, but the requirements
may be such that the frequency re-
sponse is very poor under this condi-
tion. If the poor response is caused
by the fact that the source is not pro-
perly loaded, we can improve matters
by adding a series resistor, as shown
in Fig. 11A, having a value such that
its resistance plus that of the load
equals the source impedance. In this
particular example, half the available
power is lost in the series resistor, so
STUDY SCHEDULE NO. 51

For each study step, read the assigned pages first at your usual speed, then reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind. Study each other step in this same way.

☐ 1. Introduction - - - - - - - - - - - - - Pages 1-3
   The kinds of lines used in microphone and loudspeaker cables are described in this section.

☐ 2. High- and Low-Impedance Lines - - - - - - - - - - Pages 3-11
   You learn the electrical characteristics of the lines used to connect microphones and loudspeakers to amplifiers.

☐ 3. Impedance Matching - - - - - - - - - - - - - Pages 12-16
   Methods of matching impedances with transformers and resistors are described in this section.

☐ 4. Microphone Connections - - - - - - - - - - - - - Pages 16-18
   Here you learn the solution to several problems you may meet in connecting microphones to an amplifier.

☐ 5. Practical Loudspeaker Connections - - - - - - - - - - Pages 18-25
   Methods of distributing power to various groupings of loudspeakers are discussed in this section.

☐ 6. Loudspeaker Switching; Equalizers - - - - - - - - Pages 25-28
   This section contains descriptions of constant-impedance switching networks, volume controls, cross-over networks, and equalizers.

☐ 7. Answer the Lesson Questions and Mail Your Answers for this Lesson to NRI for Grading.

☐ 8. Start studying the Next Lesson.
AMPLIFIERS, loudspeakers, microphones, and other components of public address systems may be purchased from their respective manufacturers, from local radio wholesalers, or from the mail-order supply houses that handle radio parts. It is possible to obtain a complete system as a "package" consisting of an amplifier together with suitable loudspeakers and a microphone. The portable units that are intended for temporary installations are almost always sold this way. These even come with pre-cut cables to connect the various components together.

There is no connection problem with such package units—all you need do is plug the cables into the proper outlets and place the components where you want them. If such a package unit is used in a permanent installation, you can conceal the cables and place the amplifier in an out-of-the-way location if you wish. In temporary installation work, you will probably mount the various components of the system in convenient places without making any great effort to conceal anything.

With such package units, there will be no problems of impedance matching nor of excessive line losses (provided you use the lines supplied). You can assume that the components of a particular assembly were chosen to operate properly together.

Of course, all sound installations aren't this simple. For most permanent and some temporary installations, you will have to assemble a sound system rather than use one that is offered as a unit.

One of the major problems you will meet in doing so is making the proper connections between the various components. You will have to match impedances to get maximum power transfer and normal frequency response, and you will also have to make sure that excessive power and frequency losses will not occur in the transmission lines used to connect the various components. This last is often a problem when a line must be run several hundred feet from an amplifier to a loudspeaker.

This Lesson is devoted to showing you how to connect the components of p.a. systems properly. We shall
study all parts of this problem. As the first step in our studies, let’s learn what kinds of lines are used in p.a. work.

**AUDIO CABLES**

Any set of conductors used to carry energy between pieces of equipment is called a transmission line. At power-line frequencies, parallel wires strung on insulators several inches apart can be used. However, such a line is not desirable for audio-frequency use, both because of difficulty in installation and because such an “open” line will pick up excessive amounts of hum, noise, and interference.

Such stray pickup is reduced by twisting the wires together so that they are separated only by their insulation. Close spacing and the twisting causes the stray pickup of one wire to be mostly cancelled by that of the other. Hence, such twisted wire—ordinary electric lamp cord, for example—is commonly used as an audio line where the power levels are high enough to make the losses unimportant and where the wire would not be subject to deterioration caused by weather or to wearing caused by excessive motion. Such lamp cord is readily obtainable: it is found everywhere that electrical supplies are handled, even in five-and-ten cent stores.

Radio supply houses carry a better wire for this purpose. A typical example is shown in Fig. 1A. This is a twisted pair of wires that is enclosed in a cotton loom that affords additional protection to the wire. It is possible to get wire like this with the loom specially treated to make it weather-proof. Such wire can be used outdoors.

Either of these two types is satisfactory for connecting loudspeakers to an amplifier, and one or the other is used for this purpose in most installations. These wires may be strung around the room in a temporary installation; in a permanent installation, they are frequently put in the walls, preferably in conduit. Outdoors, such cables are often enclosed in conduit or pipes and buried in the ground; this helps to protect the wire.

Incidentally, in installing any cable of this kind permanently, you will have to meet local electrical codes. Despite the fact that relatively low voltages and moderate power are be-

![FIG. 1. The three most common kinds of audio lines: A, twisted-pair line; B, unbalanced coaxial line; C, balanced coaxial line.](image)
a shield that acts as a second conductor. A typical example is shown in Fig. 1B. Because it is necessary that the cable be flexible so that the microphone can be moved about, the outer conductor consists of a number of fine wires braided together rather than a piece of copper tubing. Although the braid shield is not quite as effective a shield as solid tubing would be, it is satisfactory for all normal p.a. uses as long as it is not used near very strong fields.

For balanced microphone lines, two wires are enclosed in a coaxial shield as shown in Fig. 1C. The shield here acts as a third conductor, carrying the ground lead from the microphone transformer to the amplifier.

Shielded wire that has the shield on the outside can be obtained, but for better appearance and for ease in handling, microphone cable commonly has a rubber covering or cotton braid insulation over the shield as shown in the examples in Fig. 1.

Now that you know what kinds of lines are used for microphone and loudspeaker cables, let's learn what important characteristics of these lines must be considered in making an installation.

---

**High- and Low-Impedance Lines**

Regardless of the type of transmission line, it will have the following characteristics:

1. Resistance. No conductor is perfect; all have some resistance.
2. Capacity. Whenever two conductors are separated by an insulator, there is a capacity between them.
3. Leakage. Leakage is a measure of the quality of the insulation. Very good insulation has very little leakage; therefore, the current between the wires is very small. If the insulation is poor, however, it is possible for there to be an appreciable current between the wires.
4. Inductance. A wire also has a certain amount of inductance. This inductance is relatively small, however, so it is not appreciable at audio frequencies.

Inductance and leakage, then, can be ignored in considering an audio line if we assume that wire of good quality will be used. However, the resistance and the capacity of the line are very important.

The resistance of a transmission line depends on the length of the wire and on the wire size, increasing if the wire is made longer or if its diameter is reduced. (A wire table later in this Lesson will show you exactly how the resistance varies with each of these factors.)

The capacity between wires varies with the wire size, the length, and the spacing between them. The capacity increases if the wires are brought closer together or if the diameter or length is increased.

The resistance of transmission lines is what determines how much power loss there will be, and the capacity determines the frequency discrimination. The amount of this frequency discrimination and the amount of power loss depend upon the conditions under which the line is to be used. However, since both the resistance and the capacity of a line increase when it is made longer, it is obvious that a line should be kept as short as is practical.
LINE IMPEDANCES

When we speak of "low" impedance or "high" impedance audio lines for p.a. work, we are not referring to the resistance or capacity that is possessed by the line. If the line length is appreciable (a quarter-wavelength or more) with respect to the wavelength of the signal being handled, then the line does have a characteristic impedance of its own that is called its "surge" impedance. Telephone lines have such an impedance, and the impedance must be matched at each end of these lines for proper signal transfer. However, p.a. lines are at most only a few thousand feet in length, which is short compared to the wavelengths of audio signals. Hence, when we call a line a "low" or "high" impedance, or call it a "500-ohm" line, we are referring solely to the impedances of the terminating devices—the source and load that the line connects, and not to the actual line impedance.

As we shall show, what may be a low-impedance termination for one service may be high for another, so we qualify the impedance term by referring either to "microphone" lines or to "transmission" lines. The latter term is applied to lines carrying power, such as those that connect amplifiers to loudspeakers.

HIGH-IMPEDANCE LINES FOR MICROPHONES

Microphone lines are considered to be high impedance if they are connected between devices having impedance values above 10,000 ohms. Crystal microphones, for example, may well have impedances of 20,000 ohms or more. A crystal microphone having such an impedance can be connected directly to a tube grid circuit through a connecting line.

The crystal microphone is the only kind that has a high impedance of itself, but a dynamic or other low-impedance microphone is often made to have a high-impedance output by connecting it to a suitable matching transformer, which is frequently built into the case of the microphone. A short line can be used to connect such a microphone to the grid circuit of the preamplifier tube.

When a transmission line is used between two points of high impedance, the power loss in line resistance is negligible. If the terminal impedances are, say, around 10,000 ohms, a line having a resistance of 10 ohms or so will not be able to affect the current distribution appreciably. However, although power loss is no problem with these lines, frequency response and pickup of interference are.

Interference. Stray noise and hum fields are troublesome whenever the impedance to ground is high, because even a small field can develop an appreciable voltage across a high impedance. The impedance between the control grid of a tube and ground is usually 50,000 ohms or more, so the grid is particularly likely to pick up interference. Furthermore, the signal level at the grid of the first preamplifier tube is always low, so even a relatively low hum or noise voltage can be appreciable with respect to the desired signal. This difficulty can be minimized by keeping the impedance to ground low, by keeping physically small the amount of the circuit that is at a high impedance, or by shielding all portions of the circuits that are at a high impedance to ground. This latter method is used to minimize pickup in high-impedance microphone cables, which are always shielded coaxial lines. This shielding is always carried right inside the amplifier all the way to the grid of the tube, and sometimes even encloses the input resistor.
Even though it is shielded, however, a high-impedance line always has a certain amount of pickup per unit of its length. It is therefore desirable to keep the line just as short as possible to minimize this kind of interference.

**Frequency Response.** The shunting capacity of a microphone line always has an effect on the frequency response. The exact nature of the effect depends on the characteristics of the microphone impedance.

The average single-wire coaxial microphone cable has a capacity of 25 to as much as 75 mmfd. per foot. (It is possible to get lower capacities by increasing the spacing between the center wire and the braided shielding through the use of fillers made of threads or ropes. This increases both the bulk and the cost of the cable greatly, however; as a result, such low-capacity cable is found only in certain high-fidelity installations.) If we were to use a 20-foot cable that had a medium capacity value of 50 mmfd. per foot, the total capacity would be \(20 \times 50\), or 1000 mmfd. \((.001 \text{ mfd.})\), which is very appreciable.

Fig. 2 shows how a high-impedance microphone should be connected to the grid of a tube. The grid circuit is completed by resistor \(R_1\), which is chosen to match the impedance \(Z_s\) of the microphone. Since the grid of a tube is a voltage-operated device, we might expect \(R_1\) to be several times the microphone impedance so that most of the voltage generated by the microphone would be dropped across \(R_1\). However, it is desirable to load the microphone to minimize peaks in its response. For this reason, it is common practice to make the load into which the microphone works equal to the microphone impedance, even though this arrangement means that only half the microphone voltage is applied to the amplifier grid.

The capacity of the microphone cable, represented as \(C_c\) in Fig. 2, is in parallel with \(R_1\). Its reactance, of course, varies with frequency. At low frequencies, the reactance is so high that the capacity is a negligible shunt. It becomes an appreciable factor at higher frequencies, however, and has an effect on the frequency response.

As an example, let's assume that \(Z_s\) and \(R_1\) are 100,000 ohms each, and that we are using a 20-ft. cable having a total capacity of 1000 mmfd. The reactance of the capacity equals the resistance of \(R_1\) at about 1600 cycles. At this frequency, the net impedance of \(C_c\) and \(R_1\) in parallel is half that of \(R_1\) alone, so the voltage across \(R_1\) drops to two-thirds its original value. At a frequency twice this, 3200 cycles, the condenser reactance has dropped to 50,000 ohms, so the net impedance is one-third its former value; the voltage across \(R_1\) is now one-half what it was at the low frequencies, where the condenser reactance was too high to
matter. Obviously, therefore, the shunting capacity has considerable effect on the frequency response when the microphone impedance is essentially resistive.

If we reduce the values of $R_1$ and $Z_s$ to, say, 50,000 ohms each, the effects we have just described will occur at a higher frequency: the voltage is reduced to two-thirds at 3200 cycles and to one-half at 6400 cycles. Obviously, therefore, for a fixed cable capacity, the lower we can make the source and load impedances, the less shunting of high frequencies there will be.

Of course, we can't change the microphone impedance at will, so our choice of microphone fixes $R_1$. We must choose a microphone having a reasonably low impedance if high fidelity is wanted.

The effect of the capacity of the cable is made worse if the microphone has an inductive component in its impedance, as dynamic types that use matching transformers to give a high impedance often have. The inductance tends to make the microphone impedance rise with frequency, which produces an even greater voltage division with the line capacity.

**Crystal Microphone Cable.** Fortunately, the most common high-impedance microphone—the crystal type—has an impedance that is essentially capacitive, as shown in Fig. 3. This comes about because the crystal acts as a dielectric between the two terminal plates of the crystal element. Since it is capacitive, the impedance of the microphone goes down as the frequency increases. If the microphone cable is properly chosen, the internal impedance of the microphone can be made to act with the line capacity as a voltage divider of such a nature that the output is practically constant over the range that the microphone is intended to cover. That is, the impedance $Z_s$ goes down with frequency at the same rate as the reactance of $C_C$ does, so the output remains approximately the same even though the impedances decrease with frequency. Notice that this effect occurs only if the microphone cable has the proper characteristics. Therefore, you should neither shorten nor lengthen the cable that is supplied with a crystal microphone; if you do, the output will not vary uniformly with frequency.

In this case, the value of $R_1$ really sets the low-frequency response rather than the high-frequency response. As the frequency decreases, the impedance $Z_s$ increases. When it gets well above the value of $R_1$, an increasing amount of the signal is dropped in the internal impedance of the microphone. In this particular case, increasing the value of $R_1$ extends the low-frequency response—exactly the opposite of what happens in the circuit shown in Fig. 2. However, the necessity of keeping down hum and noise pickup places a definite limit on the value of $R_1$.

You can see that a line operated with its terminals at a high impedance must be specially chosen. Its length is critical when it is used with a crystal microphone, and it must be as short as possible when it is used with any other...
form of high-impedance microphone if reasonable frequency response is wanted. In general, therefore, high-impedance microphone cables are around 10 to 15 feet long; even the longest are no more than 25 feet in length. High-impedance microphones must therefore be placed close to the amplifier.

LOW-IMPEDANCE LINES FOR MICROPHONES

Whenever it is desired to locate a microphone at a distance greater than the allowable length of high-impedance microphone lines, it is necessary to use a line of lower impedance. As a matter of fact, low-impedance lines are generally used even for short distances when low-impedance microphones are used.

Terminating a line with lower impedance cuts down on the noise and hum pickup because the lower impedance to ground decreases the amount of voltage that can be induced by a fixed field. Another advantage of the low impedance is that, as we indicated in the discussion of high-impedance lines, a reduction in the terminating impedance decreases the effect of the capacity of the line on the frequency response. For example, you learned that the 20-foot cable began to be noticeably effective at 1600 cycles when the terminating impedance value was 100,000 ohms. This changes to 3200 cycles when the impedance value is made 50,000 ohms. If we can get the terminating impedances down to 10,000 ohms, we can go out to 16,000 cycles before the capacity of a 20-foot cable will have much effect on the frequency response.

Reducing the terminating impedances also permits the use of longer lines. If we increase the line length to 200 feet instead of 20 feet, the total capacity now becomes .01 mfd. To get out to 16,000 cycles, the terminating line impedance must now be 1000 ohms or less. Because it is sometimes desirable to run microphone lines for 200 feet or more, the industry has standardized the so-called low-impedance microphone terminations at 500 ohms, although a few manufacturers use 200 ohms.

The connections for such a line are shown in Fig. 4. If the microphone is a high-impedance type, transformer $T_1$ steps down its impedance to 500 ohms. If the microphone is low impedance, transformer $T_1$ steps up its impedance to 500 ohms. At the amplifier, transformer $T_2$ matches 500 ohms to the value of $R_1$, which is usually between 50,000 and 250,000 ohms. (This resistor is used because the transformer would be working into an "open" circuit of no definite impedance value if the resistor were not present. The use of the resistor gives a definite load for the transformer and therefore permits the turns ratio of the transformer to be fixed.) Transformer $T_2$ is usually located as close as poss-

![FIG. 4. How a low-impedance line is connected to a microphone and an amplifier.](image)

sible to the tube $VT_1$ so that the lead from the transformer to the tube grid can be as short as possible.

As we have mentioned, the line is called a 500-ohm line purely because it is used between transformer terminations that have this impedance value. The actual line resistance remains that determined by the length
and size of the wire, and the capacity is still determined by the size of wire, the spacing between wires, and the length of the line. Exactly the same kind of cable can be used as a high-impedance line in one case, and as a low-impedance line in another; it just depends on the terminating impedances.

A low-impedance line does not have to be a coaxial cable. The impedance is so low that hum and noise pickup is not usually a problem. However, if conditions are such that a.c. power lines must be near the microphone cable, it may be best to use a coaxial cable for a 500-ohm line.

As we said earlier, 200 ohms can be used as a terminating value if desired. It makes little difference whether a 500-ohm or 200-ohm line is used in most cases; the choice depends mostly on the transformers available for matching.

**PREAMPLIFIER LINES**

If a microphone must be located a long distance from the amplifier, the very weak microphone signal may be seriously attenuated by losses in the line and the transformers, and may be interfered with by even small amounts of hum and noise interference. In such cases, a preamplifier is used at the microphone location, then the amplified signal is sent down a line to the regular amplifier.

The preamplifier is essentially just a one- or two-stage amplifier, much like the first stage of the regular p.a. amplifier. The signal from the microphone is fed to this amplifier through either a high-impedance or a low-impedance line, depending upon the type of microphone; generally a very short microphone line is used. The output of the preamplifier is fed through a matching transformer or through a cathode-coupling connection to a 500-ohm line that runs from the preamplifier to the main amplifier. At the main amplifier, a transformer matches the line to the grid resistor of the input tube.

**LOW-IMPEDEANCE LINES FOR LOUDSPEAKERS**

At the other end of our p.a. system, we are faced with the problem of connecting the loudspeaker to the amplifier. Here, we are working from plate circuits, rather than into grid circuits. Also, we are dealing with low-impedance loudspeaker voice coils. In fact, the impedances with which we deal are so low that a loudspeaker line is not considered to be low-impedance unless it is below 50 ohms. In most instances, such lines terminate in values approximating voice coil impedances, ranging from 2 ohms to 16 ohms.

Obviously, since the signal power levels are high, we needn't worry

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**FIG. 5.** The resistance of the wire in a low-impedance loudspeaker line acts as a voltage divider with the load impedance.
of the line itself is important, however.

The line resistance is distributed in both sides of the line, one half in each, as indicated by $R_1$ and $R_2$ in Fig. 5. (Although we ordinarily think of copper wire as having practically no resistance, actually long lengths of wire run of this wire would therefore have a resistance of 1.28 ohms. If we tried to run a wire of this size and length from an output transformer to a 4-ohm loudspeaker voice coil, we would find that we would have appreciable line loss because the line resistance would be high with respect to the load impedance.

In general, engineers consider a line loss of 15% in the 400-to-1000-cycle range to be reasonable. Tables like that in Fig. 7 give the maximum length of line of any one size that can be used for various low-impedance values, assuming a maximum loss of 15%. In our example, the 100-foot run of No. 18 wire is too long for a 4-ohm load; if we were using this load value, we would have to restrict ourselves to 50 feet of No. 18 wire to keep the line loss at 15% or less.

Of course, using a larger size wire reduces the resistance and permits a longer run for the same load impedance, as you will observe from Fig. 7. However, large wire sizes cost considerably more money. Furthermore, if the load consists of a group of loudspeakers, its net impedance will be so low that the permissible length of the line will be severely limited. Hence, you ordinarily will not find low-impedance loudspeaker lines that are longer than about 200 feet and seldom one that is over 50 feet.

<table>
<thead>
<tr>
<th>B &amp; S GAUGE</th>
<th>D. C. RESISTANCE IN OHM</th>
</tr>
</thead>
<tbody>
<tr>
<td>10</td>
<td>0.0020</td>
</tr>
<tr>
<td>12</td>
<td>0.0032</td>
</tr>
<tr>
<td>14</td>
<td>0.0051</td>
</tr>
<tr>
<td>16</td>
<td>0.0080</td>
</tr>
<tr>
<td>18</td>
<td>0.0128</td>
</tr>
<tr>
<td>19</td>
<td>0.0161</td>
</tr>
<tr>
<td>20</td>
<td>0.0204</td>
</tr>
<tr>
<td>21</td>
<td>0.0256</td>
</tr>
<tr>
<td>22</td>
<td>0.0330</td>
</tr>
<tr>
<td>23</td>
<td>0.0407</td>
</tr>
<tr>
<td>24</td>
<td>0.0513</td>
</tr>
<tr>
<td>26</td>
<td>0.0806</td>
</tr>
</tbody>
</table>

**FIG. 6. This table shows the resistance per loop foot of various gauges of wire.**

These resistances act as voltage dividers with the voice-coil impedance $Z_L$; if they are relatively high in comparison with $Z_L$, there will be a considerable amount of power lost in the line.

For convenience in use for p.a. installations, tables like Fig. 6 give the resistance "per loop foot for various sizes of wire." A loop foot actually represents two feet of wire, since it is the amount of wire needed to connect an amplifier to a loudspeaker when the two are a foot apart. You can find the total resistance of a loudspeaker cable with the aid of such a table just by multiplying the resistance per loop foot by the number of feet of cable used between the amplifier and loudspeaker locations.

As an example, let's suppose you are using No. 18 wire, which is a common lamp cord size. Its resistance (Fig. 6) is 0.0128 ohm per loop foot. A 100-foot run of this wire would therefore have a resistance of 1.28 ohms. If we tried to run a wire of this size and length from an output transformer to a 4-ohm loudspeaker voice coil, we would find that we would have appreciable line loss because the line resistance would be high with respect to the load impedance.

In general, engineers consider a line loss of 15% in the 400-to-1000-cycle range to be reasonable. Tables like that in Fig. 7 give the maximum length of line of any one size that can be used for various low-impedance values, assuming a maximum loss of 15%. In our example, the 100-foot run of No. 18 wire is too long for a 4-ohm load; if we were using this load value, we would have to restrict ourselves to 50 feet of No. 18 wire to keep the line loss at 15% or less.

Of course, using a larger size wire reduces the resistance and permits a longer run for the same load impedance, as you will observe from Fig. 7. However, large wire sizes cost considerably more money. Furthermore, if the load consists of a group of loudspeakers, its net impedance will be so low that the permissible length of the line will be severely limited. Hence, you ordinarily will not find low-impedance loudspeaker lines that are longer than about 200 feet and seldom one that is over 50 feet.

<table>
<thead>
<tr>
<th>WIRE SIZE (B &amp; S)</th>
<th>LOAD IMPEDANCE</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>4 OHMS</td>
</tr>
<tr>
<td>14</td>
<td>125'</td>
</tr>
<tr>
<td>16</td>
<td>75'</td>
</tr>
<tr>
<td>18</td>
<td>50'</td>
</tr>
<tr>
<td>20</td>
<td>25'</td>
</tr>
</tbody>
</table>

**FIG. 7. Maximum loop lengths that can be used with various load impedances to keep line loss no more than 15% for audio frequencies below 1000 cycles.**
<table>
<thead>
<tr>
<th>WIRE SIZE (B&amp;S)</th>
<th>LOAD IMPEDANCE 100 OHMS</th>
<th>LOAD IMPEDANCE 250 OHMS</th>
<th>LOAD IMPEDANCE 500 OHMS</th>
</tr>
</thead>
<tbody>
<tr>
<td>14</td>
<td>1000'</td>
<td>2500'</td>
<td>5000'</td>
</tr>
<tr>
<td>16</td>
<td>750'</td>
<td>1500'</td>
<td>3000'</td>
</tr>
<tr>
<td>18</td>
<td>400'</td>
<td>1000'</td>
<td>2000'</td>
</tr>
<tr>
<td>20</td>
<td>250'</td>
<td>750'</td>
<td>1500'</td>
</tr>
</tbody>
</table>

**FIG. 8.** Maximum loop lengths for high-impedance loudspeaker lines if 5% power loss at the middle frequencies is allowed.

**HIGH-IMPEDANCE LINES FOR LOUDSPEAKERS**

The problem of power loss on the transmission line can easily be solved by operating the line at a high impedance. This we can do by using transformers at each end of the line, one to match the source to the line and the other to match the line to the load. For power transfer to loudspeakers, terminating line impedance values ranging between 100 and 600 ohms are called high impedances. Notice that these correspond to what we call low impedances for microphone lines.

The line loss is negligible when the termination is 500 ohms. Effectively, of course, going to a higher-impedance termination means that, for the same power, we have a higher voltage and lower current. The reduced current causes less drop to occur in the resistance of the line.

Fig. 8 gives line lengths that provide 5% power loss at the middle frequencies. Compare these with Fig. 7. Incidentally, 5% losses are used for these lines rather than the 15% value used for low-impedance lines because the impedance-matching transformers have some loss. The total loss including line and transformer loss will be kept to about 15% if the line loss is kept down to 5%.

Obviously, we can run much longer lines if we terminate them in high impedances. Also, we can use much smaller wire, thus saving something on the cost of the line; this saving is often worth while if a long line is used. Although, of course, we must use transformers at each end of such a line, such transformers may be desirable anyway to give the proper impedance matching and power transfer, as we shall show later.

Although going to a high-impedance line does solve the loss problem, it reintroduces the problem of the shunting capacity. The average capacity of a twisted-pair electric cord is about 50 mmfd. per foot, so the line length must be kept down if the high-frequency response is not to be seriously reduced.

**Line Lengths.** Fig. 9 shows a chart prepared by one loudspeaker manufacturer that permits you to determine the length of line that can be used for a particular frequency response and a fixed impedance, or permits you to determine the impedance that is neces-

![Diagram](image)

**FIG. 9.** Transmission line design chart prepared by Jensen Radio Mfg. Co. It is based on a 5% power loss in line and a 3-db loss at upper limiting frequency due to a line capacity of 50 mmfd. per foot across a typical moving-coil loudspeaker load.
sary at the end of the line when a certain length must be used.

As an example, let's suppose that the line is to be 1500 feet long. Also, let's suppose that the response is to go out to 7500 cycles before the power drops to half its normal value (3-db loss). Following the dotted line (at 1500 feet) from the left scale over to the 7500-cycle line, we find that we strike it near point A. Reading downward along the dotted line to the bottom, we find that the line impedance must be 250 ohms for this length and frequency response. The wire size necessary for minimum line loss and for the expected capacity value is the next larger above the point of intersection with the frequency line; in this case, it is No. 16 wire.

If we have a fixed impedance, we can determine the line length for a particular frequency response. For example, let's say that 200 ohms is our impedance value. Reading upward until we strike, let us say, the 10,000-cycle curve at point B, we find that we can again use a 1500-foot line. Notice that changing the impedance from 250 ohms to 200 ohms allows us to go from a 7500-cycle to a 10,000-cycle response for the same line length. If we go the other way—toward a higher line impedance—the fidelity will fall off if we must have a 1500-foot line length. For example, at 500 ohms, the response is something under 5000 cycles for a 1500-foot line.

This same chart can be used to determine the maximum line length for a particular size of wire and a particular frequency response. For example, let's suppose we want to use No. 18 wire. If the frequency response is to be within 3 db to 10,000 cycles, follow the No. 18 wire-size line to where it crosses the 10,000-cycle line. This is at point C. Reading now to the line length scale, you will find that we can use a line length of about 1050 feet, and reading downwards you will find that the impedance should be about 270 ohms. If we used the more practical impedance value of 250 ohms, which crosses the No. 18 wire size at D, we could use a line of 1000 feet of No. 18 wire and have a frequency response that would be somewhat less than 3 db down at 10,000 cycles.

Of course, the response will always be improved if any length less than these maximums is used. For example, returning again to our 250-ohm impedance value, point A shows a length of about 1500 feet and a frequency response out to 7500 cycles. Point D at 1000 feet and the same impedance gives a frequency response flat out beyond 10,000 cycles. If the length is only 800 feet, the frequency response goes out to 15,000 cycles.

To sum up: we see that if we terminate power transmission lines with high impedances, the line losses will in general be negligible, but the frequency response may suffer. On the other hand, the frequency response is good if we use lower impedances as line terminations, but we run into line loss difficulties. Therefore, it is the usual practice to choose some compromise impedance value that gives the desired frequency response without excessive line loss at the line length that is necessary.

Now that you have learned something of the basic characteristics of the transmission lines used, let's turn to the problem of impedance matching.
Impedance Matching

From your earlier Lessons, you will recall that it is necessary to match impedances whenever maximum power transfer is desired.

In public address work, the microphone does not, strictly speaking, require such impedance matching. Since the microphone eventually feeds into the grid of a tube, which is a voltage-operated device, we would ordinarily want the load impedance to be many times higher than the microphone impedance for maximum voltage transfer. If the microphone is operated this way, however, there will be peaks and ripples in its frequency response. It is best to load the microphone to smooth out these irregularities. As a compromise between the two opposing possibilities, it is common practice to terminate the microphone line with a resistor having a resistance equal to the impedance of the microphone. Hence, if the microphone is a high-impedance type, the load into which it operates will have the same impedance as itself. If it is a low-impedance type, impedance-matching transformers will be used to match the microphone to the line and the line to the grid resistor, or a single transformer will be used to match the microphone to the grid resistor. (In the latter two cases, the grid resistor is fixed by practical transformer design and by the need to avoid hum pickup at some value between 50,000 ohms and 250,000 ohms.)

At the other end of the system, we are interested in making an efficient transfer of power from the plate circuit of the power output stage to the loudspeakers. As you have learned elsewhere, we will get the maximum power transfer whenever the load impedance equals the source impedance. However, it so happens that because of the characteristic of vacuum tubes, the maximum undistorted power output is not obtained at exactly the same point as the maximum total power output. As a matter of fact, for triode tubes, the load should be twice the plate impedance for maximum undistorted power output. Fortunately, the power output secured with a load of this sort is only slightly less than that obtained when the impedances are properly matched.

In the case of pentode and beam power tubes, the impedance that gives maximum undistorted power is about 1/7 to 1/9 the plate impedance of the tube. Although these tubes give considerably less power output when they are operated with such loads than they would give if their loads were equal to their impedances, the distortion is so severe if they are operated in the latter manner that the power sacrifice is considered worth while. The load values chosen still give a reasonably high output.

It is safe to assume that the manufacturer of an amplifier gives load impedance values in terms of the maximum undistorted power output. In other words, when you must match a particular grouping of loudspeakers to an amplifier, the value specified by the amplifier manufacturer is the value you should match for maximum undistorted power output.

Let's learn something about how to match impedances.

OUTPUT TRANSFORMERS

Two kinds of output transformers are in common use in amplifiers. Each, of course, has a primary that is properly designed for the output tubes of the amplifier in which it is used. One kind is designed to match the am-
amplifier output stage to a particular loudspeaker or group of loudspeakers, and therefore has a fixed secondary impedance. If for some reason the loudspeakers chosen by the manufacturer are not to be used, others having the same voice-coil impedances may be substituted.

The second kind of output transformer is essentially a universal type in that its secondary has a number of taps that can be used to match to any common loudspeaker or group of loudspeakers. It is not unusual to find secondaries that are designed to match impedance values of 4, 8, 15 (or 16), and perhaps 30 ohms, and in addition have a 500-ohm and perhaps a 250-ohm tap for matching transmission lines.

Another type of output transformer that is occasionally used is primarily designed to match the amplifier to a line. A transformer of this type usually has taps at about 67, 125, 250, and 500 ohms.

Although the output transformer on the average p.a. amplifier does contain a number of taps, it is quite possible that the exact tap needed will not be available. For highest possible fidelity, the source and load impedances should be matched within 10%, but mismatches of up to 25% may be permissible in practice, depending upon the fidelity demanded. If a wide mismatch must occur, it is always better to connect the loudspeaker to the impedance tap next lower than the load impedance to minimize loss of power and distortion. Thus, if the combined loudspeaker load figures out to be, say, 6 ohms, you should use a 4-ohm tap rather than an 8-ohm tap.

**Line Transformers.** In addition to output transformers, there are line-to-loudspeaker matching transformers. These are designed to match the loudspeaker voice coil to whatever value is needed to terminate the line properly.

Incidentally, when we speak of the impedances of a transformer, we refer to the values the transformer is designed to match—not to the actual reactances of the transformer windings. The reactance of the primary winding (when there is no secondary load) should be about ten times the source impedance if good low-frequency response is to be obtained. Then, its turns ratio should be chosen so that the secondary load will appear as the desired “reflected” value in the primary. For example, if a transformer is supposed to cause a 4-ohm secondary load to appear as a 3600-ohm reflected impedance across the primary, its turns ratio should be 30.*

The actual impedance of the primary winding depends on the plate resistance of the tube to which it is to be connected. If it is to be used with a triode tube having a plate resistance of about 2000 ohms, the primary winding should have a reactance of 20,000 ohms (or more) at, say, 400 cycles.

A 30-to-1 transformer will also cause an 8-ohm secondary load to reflect as 7200 ohms; a 2-ohm load as 1800 ohms; a 16-ohm load as 14,400 ohms; and so forth. In other words, a transformer having a 30-to-1 turns ratio will always produce a reflected primary impedance that is 900 times as great as the impedance that is connected to the secondary. This does not mean that the same transformer can be used for any application in which an impedance ratio of 900 to 1 is wanted, because there is also the requirement that the primary reactance must be at least 10 times as large as

---

* The turns ratio equals the square root of the impedance ratio. Hence:

\[ N = \sqrt{\frac{Z_s}{Z_t}} = \sqrt{\frac{3600}{4}} = \sqrt{900} = 30 \]
we have a 3-dB power loss—much more than in the direct connection shown in Fig. 10A. If the ratio of source to load impedance were greater, the series resistor would have to be larger, and more power would be lost.

A bad feature of the arrangement shown in Fig. 11A is that the load \( R_L \) does not see its own impedance when looking back toward the source; in fact, the series resistor has made matters worse in this respect. (When we say that a source or a load “sees” an impedance when it “looks” in one direction or another, we are using an engineering expression that is often very convenient. The impedance “seen” is the effective impedance in the specified direction. For example, the source in Fig. 11A sees an impedance consisting of \( R_S \) and \( R_L \) in series when it looks toward the load; the load, on the other hand, sees an impedance consisting of \( R_S \) and \( R_L \) in series when it looks toward the source.)

If the load is entirely resistive, this won’t matter at all. However, if the load is a reactive one such as a loudspeaker voice coil or a long transmission line, it is quite possible for the load characteristics to be such that this mismatch affects the circuit response. For example, if a loudspeaker voice coil does not see its own impedance, it will have much higher peaks and valleys in its over-all frequency response.

If it is necessary that both the source and the load see their respective impedance values (that is, if they must both be matched), an L-type pad like that shown in Fig. 11B may be used. The formulas given in Fig. 11B are correct if the source has a higher impedance than the load. If the load impedance is higher than that of the source, \( R_L \) should be placed between \( R_S \) and the load rather than as shown, and the terms \( R_G \) and \( R_L \) in both formulas should be interchanged throughout. (That is, where \( R_G \) appears, use \( R_L \), and vice versa.) If we calculate \( R_L \) and \( R_S \) using the formulas given in Fig. 11B and the source and load impedances given in Fig. 11A, we find that they both come out to be about 706 ohms. In this case, \( R_G \) sees \( R_S \) and \( R_L \) in parallel, with \( R_L \) in series with the parallel group. The resistance of \( R_S \) (706 ohms) in parallel with the 500-ohm load is about 294 ohms; this resistance in series with \( R_L \) (706 ohms) makes a total resistance of 1000 ohms.

The load sees \( R_S \) in parallel with a series combination of \( R_G \) and \( R_L \) in series. The combined resistance of these three is 500 ohms, so the load is matched. However, there is now about an 11-dB power loss.

Such a power loss always occurs when pads are used to match imped-
ances. The amount of the loss depends upon the ratio of the source to the load impedance, but there is always at least some loss, and sometimes a very large one. Obviously, therefore, this form of impedance matching can be used only if the loss is permissible. If we cannot permit such a loss, we have to use transformers for impedance matching.

Microphone Connections

The input terminals on a p.a. amplifier provide for a certain number of microphones. The connections are usually all low-impedance or all high-impedance types, but occasionally a combination of these is provided. Obviously, if you use the kinds of microphones for which the amplifier input is designed, there is no problem. However, it may well be that an amplifier with exactly the input terminals wanted is not available, or you may be forced to use microphones with amplifiers that have the wrong kind of inputs. Let's see how to do so.

High to Low. First, let's suppose you have a high-impedance microphone that must be used with an amplifier having only low-impedance input terminals. There are two solutions to this problem.

One solution is to modify the amplifier. The fact that an amplifier has a low-impedance input means that it contains a built-in transformer that is designed to match a 500-ohm line to the grid of the first tube. It is possible to remove the transformer and bring the grid lead directly to the microphone jack. If high-impedance microphones are always to be used, such a modification may be worth while. However, if you believe there may be any need in the future for using a low-impedance microphone or for using a line to connect the microphone to the amplifier, such a change should not be made. It should not be made, either, if the additional length of wire connected to the grid of the first tube produces a marked increase in the hum level; unfortunately, you cannot find out whether this will happen without first making the modification.

If changing the amplifier seems inadvisable, the only solution is to use a transformer outside the amplifier that will match the high-impedance microphone to the 500-ohm input. In such cases, it is usually best to obtain a transformer that is designed for this particular microphone from the manufacturer of the microphone. The case of such a microphone often contains space for mounting a transformer right next to the microphone. If it is possible to install the transformer at this position without taking the microphone apart, this is the best place to put the transformer. However, if it is necessary to disassemble the microphone, it is better to send it back to the factory to have this work done, or to place the transformer at the end of the cable, right at the input of the amplifier. The latter method may or may not be desirable, depending on whether the transformer picks up too much hum and noise; again, only a trial will show.

Low to High. The opposite prob-
Problem occurs when the microphone is a low-impedance (500-ohm) type and the only jacks provided on the amplifier are at high impedance. In this case, the only solution is to use a transformer at the amplifier input that is designed to match 500 ohms to the grid of the first tube. Although such a transformer might possibly be mounted outside the amplifier, the chances are that the high-impedance lead from the amplifier connecting jack to the transformer would be so long that it would pick up excessive amounts of hum and noise. The most satisfactory solution is to mount the transformer as close as possible to the grid of the preamplifier tube. This is rarely a difficult problem, because most manufacturers usually provide space for the mounting of such transformers inside the amplifier on the assumption that he may be requested to supply the amplifier with low-impedance inputs. You can probably obtain the necessary transformer for making the conversion as well as mounting instructions from the amplifier manufacturer.

**Multiple Connections.** You will sometimes find it desirable or necessary to operate more microphones than the input jacks on the amplifier provide for. Although only one or two microphones are needed for most installations, high-fidelity music pickup or pickup from a stage often requires the use of a number of microphones. Since the average p.a. amplifier has only two microphone inputs and even the most elaborate types have only three or four, it is quite possible for you to need more microphone input terminals than the amplifier offers.

There are two common answers to this problem. If the microphones are to be located at some distance from the amplifier, the use of a preamplifier may well be justified. This will permit a considerable increase in the number of microphones that can be used, since a preamplifier commonly has four inputs, all of which feed through the preamplifier into only one input on the main amplifier.

Of course, if the microphones are to be used in a location near the main amplifier, the expense of a preamplifier may be unwarranted. In such cases, you can use commercially available resistor mixer boxes. Fig. 12 shows the schematic of one such box,

![Resistor Mixer Schematic](image)

**FIG. 12.** The schematic of a resistor mixer box used to mix the outputs of 3 microphones.

designed for three microphones, each of which feeds into its own mixer-control potentiometer (R₁, R₂, or R₃). These potentiometers are decoupled from each other by the resistors R₄, R₅, and R₆. The output of the box goes to one microphone input on the amplifier. Thus, like a preamplifier, a box of this sort permits the outputs of several different microphones to be fed into a single input jack on the amplifier.

The mixer boxes available are com-
monly designed for high-impedance microphone connections. If low-impedance lines are involved, transform-
ers will be necessary for each line, so a preamplifier will probably not be much more costly.

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Practical Loudspeaker Connections

At the other end of the amplifier, we are faced with the problem of connecting one or more loudspeakers to the amplifier. In the average job, usually at most only three or four loudspeakers must be connected together. More elaborate installations may require the use of a great many more, however, particularly when several rooms are to be covered. An extreme example of this is a hotel or hospital installation in which separate loudspeakers are wanted in a number of small rooms; as many as 100 or more may have to be connected in such an installation.

You have already learned how to determine the power ratings and the number of the loudspeakers to be used in an installation. Now we are going to discuss the problems involved in connecting these loudspeakers to the amplifier. Before we do, however, we must learn something more about the voice-coil impedances of loudspeakers.

Although voice-coil impedances are always given as some definite number of ohms, this fact does not mean that the impedance is the same at all frequencies. As a matter of fact, the impedance of a voice coil varies widely over the audio-frequency spectrum, reaching high peaks at some frequencies and falling to low values at others. The voice-coil impedance rating of a loudspeaker is therefore either a nominal value that is representative of the over-all characteristics, or is the impedance at some particular reference frequency, such as 400 cycles or 1000 cycles. There is no general agreement among manufacturers on how voice-coil impedance should be rated.

The variations in the impedance of a voice coil are minimized when the coil is properly matched to an amplifier. For this reason, you can compute the load on an amplifier with reasonable accuracy by using the impedance value given by the manufacturer of the loudspeaker.

Very often you will find that you can choose any of several different voice-coil impedances for a given loudspeaker that is usually equipped with an 8-ohm voice coil may instead be obtained on request equipped with a 4-ohm or a 16-ohm coil. It is particularly helpful to be able to make such a choice when you have to connect together a group of loudspeakers and must arrive at some reasonable combined impedance that can be matched by available transformers.

The same 4, 8, and 16-ohm values are almost standard today for driver units, although some that have a higher impedance (about 45 ohms) are also offered. High-impedance voice coils are useful when several loudspeakers are to be connected in parallel, because the total net impedance of the parallel combination will be very low unless the voice-coil impedances are fairly high to begin with.

Now let's study several practical examples of the problems you will meet in connecting loudspeakers to amplifiers and learn how they can be solved.
LOW-IMPEDEANCE CONNECTIONS

If the distance from the amplifier to the loudspeakers is small, it is possible, as you have learned, to run lines at the voice-coil impedance. In such a case, if we have more than one loudspeaker, we can connect the voice coils either in series or in parallel. If we connect them in parallel, and all have the same impedance, the net impedance will be equal to the impedance of any one divided by the number of loudspeakers. For example, the net impedance of the two 8-ohm voice coils in parallel in Fig. 13 is 4 ohms. We can use a low-impedance line to connect these two voice coils to the 4-ohm tap on the amplifier output transformer. In figuring the line loss, we must use the net impedance; in Fig. 13 it is 4 ohms, so the maximum line length is figured on this basis. If we used two 16-ohm loudspeakers, the net impedance would be 8 ohms; this would make it possible to use a longer line for the same wire size.

If two or more loudspeakers are connected in series, and all have the same impedance, the load will be the sum of the voice coil impedances. Fig. 14 shows an example.

It is also possible to connect the loudspeakers in series-parallel, as shown in Fig. 15. The impedance of this combination is the same as that of an individual voice coil as long as all the voice coils are the same and are connected as shown.

As long as the loudspeakers in each of these cases have the same voice-coil impedances, the power will divide equally between them. Thus, if two loudspeakers are connected either in series or in parallel and their voice-coil impedances are equal, each will receive half the power. If there are three loudspeakers, each gets one-third of the power; if there are four, each gets one-quarter of the power; and so on. Therefore, to make sure none of the loudspeakers in such a combination will be overloaded, all we need do is make sure that each has a power rating that is greater than its fractional portion of the amplifier output rating. If the amplifier is rated at 50 watts, for example, and we have two loudspeakers, each will get 25 watts of power and must be rated to handle it.

Unequal Impedances. If the loudspeaker voice-coil impedances are unequal, the power distribution will also be unequal. For example, when two unequal loudspeakers are connected in parallel, the resulting impedance is

\[ Z_T = \frac{Z_1Z_2}{Z_1 + Z_2} \]

As an example, if we have an 8-ohm speaker voice coil in parallel with a 16-ohm speaker, the result will be

\[ \frac{8 \times 16}{8 + 16} = \frac{128}{24} = 5.33 \text{ ohms} \]

A tap rated at 5.33 ohms will probably not be found on the output trans-
former of an amplifier; 4 ohms is about the closest we can expect. Connecting this group to the 4-ohm tap will result in a certain loss of power. There will also be an unequal power distribution; the loudspeaker having the lower impedance will receive the greater amount of power when they are in parallel. In this case, with an 8-ohm and a 16-ohm loudspeaker, the 8-ohm loudspeaker will get twice the power of the 16-ohm loudspeaker. (The same voltage is across both, and the power in each is \( P = E^2/Z \); hence, the lower the impedance, the higher the power.) You should keep with the higher impedance will get the more power: in our example, the 16-ohm loudspeaker would get twice the power applied to the 8-ohm loudspeaker.

You can see that it is desirable to have voice coils of equal impedances when low-impedance lines are used, unless you want to apply more power to some loudspeakers than to others. When a high-impedance line is used, the loudspeakers are matched to the line with transformers; in this case, it is not necessary to use loudspeakers having equal voice-coil impedances to secure equal power distribution, nor

![Diagram of matching impedances of voice coils connected in series-parallel.](image)

**FIG. 15. Matching impedances of voice coils connected in series-parallel.**

this fact in mind if you are connecting loudspeakers of unequal impedance in parallel, because otherwise you may accidentally overload one of the loudspeakers. If a 50-watt amplifier were connected to the 8-ohm and 16-ohm combination we just described, \( 33\frac{1}{3} \) watts would be applied to the 8-ohm loudspeaker and \( 16\frac{1}{3} \) watts to the 16-ohm one. The \( 33\frac{1}{3} \) watts would be a considerable overload if both loudspeakers were rated at only 25 watts, which is the maximum rating for all but the most powerful loudspeakers.

If these two loudspeakers were connected in series, the net impedance would be the sum of the two impedance values, or \( 8 + 16 = 24 \) ohms. In a series connection, the loudspeaker does the use of equal impedances mean that the power will necessarily be equally divided. Let's take up high-impedance lines now.

**HIGH-IMPEDANCE LINES FOR LOUDSPEAKERS**

One example of the use of high-impedance lines to match loudspeakers to an amplifier is shown in Fig. 16. Here there are two loudspeakers at a location remote from the amplifier. Since they are grouped together, however, it is practical to connect the loudspeakers together as a low-frequency grouping, then use a transformer to match this group to the 500-ohm line. The line is then terminated at the proper 500-ohm value at the amplifier. Operating this way, as
you have learned, provides far less line loss and permits the loudspeakers to be placed much farther from the amplifier. Because of shunting capacities, however, the permissible length of the 500-ohm line depends on the frequency range wanted. If the line has to be so long that a 500-ohm terminating impedance will not permit the desired frequency response to be secured, lower impedances must be used. Hence, in the example shown in Fig. 16, it might be necessary to use 250 ohms instead of 500 ohms at each end of the line. Transformer $T_1$ would then have to be designed to match the impedance of the loudspeaker group to a 250-ohm line. If the amplifier transformer did not have a 250-ohm tap, it would be best to replace it with one that did. Such a replacement would, of course, have to be designed to handle the power output of the amplifier.

Obviously, the arrangement shown in Fig. 16 can be used for practically any number of loudspeakers, provided that the loudspeakers can be connected in the proper series or parallel arrangement to give a terminating impedance that can be matched by transformer $T_1$. If four or more loudspeakers are to be connected in parallel, it is desirable to use 16-ohm rather than 8- or 4-ohm loudspeakers, because the net impedance will be higher and therefore more likely to be a value that can be matched by transformer $T_1$.

The statements made before about power distribution hold here; in fact, you can consider transformer $T_1$ to be the same as the amplifier output transformer. Therefore, if we use loudspeakers of equal voice-coil impedance, they will divide the power equally. If their impedances are unequal, they will divide the power according to their respective impedances and to whether they are connected in series or in parallel.

It doesn’t always happen that the loudspeakers are grouped closely enough together to make it practical to run a low-impedance connection between the voice coils. In such cases, the loudspeakers must be located wherever they are wanted, and then a high-impedance line must be run to each location. Each location must, of course, have its own matching transformer.

Fig. 17 shows an example. Here, each loudspeaker has a transformer that is chosen to match the voice-coil impedance to the line in such a way
that the net impedance of all the voice coils will equal the proper terminating impedance—500 ohms in this case. Since we have two loudspeakers, the transformers are chosen so that their reflected primary impedances are 1000 ohms; the net impedance of the two in parallel then equals 500 ohms. If we had four loudspeakers, each primary would have to have a 2000-ohm reflected impedance so that the net impedance of all of them would be 500 ohms.

In cases like this, where each loudspeaker is to get the same power, you can find the primary impedance each must have by multiplying the terminating line impedance by the number of loudspeakers wanted. Thus, if there are to be 6 loudspeakers and the terminating impedance is to be 500 ohms, $6 \times 500$ or 3000 ohms is the primary impedance that each matching transformer must have. Each transformer must then be able to match this impedance to that of the voice coil that is to be connected to its secondary. If each transformer meets this requirement, the power will be evenly distributed among the loudspeakers no matter what their voice-coil impedances may be, since the transformers effectively make them all equal as far as the amplifier is concerned.

**UNEQUAL POWER**

In many installations, we don't want equal power at each loudspeaker. One example is an installation in which high-powered loudspeakers are used in an auditorium and one or more smaller loudspeakers are used in side rooms to handle an overflow crowd. Obviously, an equal distribution of power would overload the smaller loudspeakers or under-drive the large ones, or perhaps do both.

To see how to create an uneven power distribution, let's suppose we have a circuit like that shown in Fig. 18, in which $L_S_1$ and $L_S_2$ are each rated at 25 watts, $L_S_3$ is rated at 10 watts, and the amplifier has an output impedance of 500 ohms. Our problem is to find the primary impedance for each transformer that will provide the proper power distribution.

To find these primary impedances, we must take these steps:

1. Find the total power.
2. Find the ratio between the total power and that needed for each individual loudspeaker.
3. Multiply the line or amplifier impedance by the power ratio to get the primary impedance each transformer must have.

The total power needed (Step 1) for our example is the sum of the powers of the individual loudspeakers. This is $25 + 25 + 10$ or 60 watts.

The ratio of the power (Step 2) of each of the 25-watt units to the total power is $60 \div 25$ or 2.4. The power
ratio for the 10-watt speaker is 60 \div 10 or 6.

The line impedance is 500 ohms. Therefore (Step 3), we must multiply 500 by 2.4 to find the impedance that the primaries of \( T_s \), and \( T_s \) must have: this turns out to be 1200 ohms. Multiplying 500 by 6 gives us 3000 ohms as the impedance of the primary for \( T_s \),

We can prove that we have found the correct ratios by computing the net impedance of these three primary impedances in parallel. The net impedance of the two 1200-ohm primaries in parallel is 600 ohms; this 600-ohm value in parallel with the 3000 ohms of \( T_s \) makes a net impedance of 500 ohms for the whole combination. Since this is equal to the amplifier output impedance, the loudspeakers are correctly matched to the line.

Effectively, therefore, if we use transformers that will have reflected primary impedances equal to those we have calculated, the power will automatically be divided so that each speaker will receive the proper amount. Again, it doesn’t matter whether the voice coils all have the same impedance or different impedances as long as the transformers match them properly to the calculated primary impedances.

Another way that we can get the same result, and incidentally prove that this power distribution will occur properly, is to calculate the source voltage needed and then to find the impedance of the transformer primary from the source voltage and the required power. In our example, we have a 500-ohm source and require a total of 60 watts. The source voltage can be found from the formula:

\[ E^2 = P_s Z_s \]

where \( P_s \) is the total power of the source and \( Z_s \) is the impedance of the source. Multiplying 500 by 60 we get 30,000 as the square of the voltage. By taking the square root of this value, we find that our source voltage is approximately 172 volts.

The impedance of each primary is given by

\[ Z = \frac{P_L}{E^2} \]

where \( P_L \) is the power needed for that particular load. As an example, loudspeaker LS1 requires 25 watts, and the square of the voltage is 30,000. Dividing 30,000 by 25 gives us 1200 ohms as the primary impedance, just as we calculated before.

As another and somewhat more difficult example, let’s compute the primary impedance needed for the transformers in the circuit shown in Fig. 19. This is a small hotel installation in which two 25-watt loudspeakers are used in a ballroom, four 5-watt loudspeakers are used in a dining room, and fifteen 2-watt loudspeakers are used in individual rooms. The loudspeaker groups therefore take respectively:

\[
2 \times 25 = 50 \text{ watts} \\
4 \times 5 = 20 \text{ watts} \\
15 \times 2 = 30 \text{ watts}
\]

making a total power of 100 watts. The power ratio for the 25-watt loudspeakers is 4 (100 \div 25). For the 5-watt loudspeakers it is 20 (100 \div 5) and for the 2-watt loudspeakers it is 50 (100 \div 2).

With an amplifier termination of 500 ohms, the primary impedance for the 25-watt loudspeakers should be 2000 ohms (500 \times 4); for the 5-watt loudspeakers it should be 10,000 ohms (500 \times 20); and for the 2-watt loudspeakers it should be 25,000 ohms (500 \times 50).

If we were to attempt to locate the
parts for this installation, we would find it difficult or impossible to obtain transformers for the 2-watt loudspeakers. Power-handling transformers rarely have turns ratios that would cause a loudspeaker voice coil to appear as a primary impedance of more than 10,000 to 15,000 ohms at the most. Therefore, it would be wiser for us to use some lower value of source impedance so that we can get this turns ratio for the 2-watt loudspeakers down to something reasonable.

If we use a source impedance of 125 ohms, the 25-watt loudspeakers will require primary impedance values of 500 ohms (125 \times 4), the 5-watt loudspeakers will require 2500 ohms (125 \times 20), and the 2-watt loudspeakers will require 6250 ohms (125 \times 50). Transformers having the necessary turns ratios to produce these impedances can be obtained easily.

Incidentally, while we are on the subject of transformers and the values that are available, there is one fact you should keep in mind when you are looking for transformers: a transformer rated to match two specific impedances can often be used for other impedances that are in the same proportion. For instance, a transformer listed to match 4000 ohms to let's say

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**FIG. 19.** A small hotel installation in which unequal powers must be applied to the loudspeakers.
an 8-ohm loudspeaker can also be used to match 2000 ohms to a 4-ohm loudspeaker or 8000 ohms to a 16-ohm loudspeaker. Line-to-loudspeaker transformers can be used this way because the source impedance is always less than the actual primary reactance of the transformer, so there is little frequency distortion. Notice that this is different from what you learned earlier in this Lesson about microphone-to-line transformers.

**Loudspeaker Switching; Equalizers**

In any installation involving a large number of loudspeakers, such as a hotel installation where loudspeakers are in separate rooms, it will always be necessary to make it possible for loudspeakers to be cut in and out at will to suit the desires of the listeners. As we have just shown, however, the loudspeakers are all matched to the line. If we attempt to cut any of them in or out simply by throwing a switch, we will upset the impedance matching and the power distribution. If even one loudspeaker is cut off, the power applied to the others will increase to some extent; if many small or one or two large ones are cut off, the power increase may be so great that small loudspeakers left in the circuit will be ruined. Even if the remaining loudspeakers are not damaged, the frequent changes in volume level as loudspeakers are cut in and out will be highly undesirable. To prevent such effects, it is common practice to arrange the circuit so that a resistor is substituted for the loudspeaker when the latter is out of the circuit. This keeps the total impedance of the circuit constant at all times and therefore prevents any variation in the power applied to the individual loudspeakers.

One such arrangement is shown in Fig. 20A. Resistor $R_1$ equals the voice-coil impedance. When switch $S$ is in the position shown, resistor $R_1$ is connected to the line in place of the loudspeaker; when $S$ is thrown to the other position, the loudspeaker is energized and the resistor is cut out. Effectively, therefore, there is a constant-impedance load on the line regardless of the position of switch $S$.

When transformers are used to match the individual loudspeakers, the arrangement shown in Fig. 20B may be used. Here the value of $R_1$ corresponds to the reflected primary impedance of transformer $T_1$. Again the line is not upset whether the loudspeaker is switched in or out.

**FIG. 20. Two ways of keeping the impedance of a line constant whether loudspeakers are switched in or out.**

It is often necessary to make some
provision for adjusting the volume level of individual loudspeakers as well as for cutting them in or out. A hotel room or hospital installation is a practical example of one in which a volume control for each loudspeaker is needed.

Again, it is necessary to be able to control the volume without upsetting the impedance match. Therefore, instead of using an ordinary volume control (which could not handle the power anyway), some kind of special attenuator is used for controlling volume at the loudspeaker. This attenuator is commonly either an L or T pad, so designed that it offers constant impedance at least to the source, and preferably to both the source and loudspeaker loads. Fig. 21 shows typical examples of the L and T connections. The resistor values are so tapered that the proper impedances are maintained. Sometimes these pads are continuously variable, sometimes they are switching units that use fixed resistors to produce a certain amount of attenuation at each position.

In either case, these attenuators are designed to operate between definite impedance values. In the case of the kind shown in Fig. 21, they must be designed to operate at the voice coil impedance.

**CROSS-OVER NETWORKS**

In high-fidelity systems, dual loudspeakers are used to give a good overall frequency response. One is a low-frequency or woofer type and the other a high-frequency or tweeter unit. A much better over-all frequency response can be obtained by the proper use of such combination speakers. The woofer speaker can be designed to handle the low frequencies exceptionally well, and the tweeter will give an extended high-frequency range.

However, it is necessary to prevent high frequencies from being fed to the woofer and to prevent low frequencies from going to the tweeter. Fig. 22 shows a typical cross-over network that is used to direct the various frequencies to the proper loudspeakers. It consists of a high-pass filter, $C_1-L_1$, and a low-pass filter, $C_2-L_2$. In the high-pass filter, condenser $C_1$ is small in capacity and therefore offers a high impedance to low frequencies. $L_1$ at the same time offers low impedance at low frequencies, with the result that practically all low frequencies are dropped across $C_1$ and are not applied to the tweeter. As the frequency goes up, however, $C_1$ drops in reactance and $L_1$ increases, so an increasing amount of power is applied to the tweeter.

The opposite action occurs with the
low-pass filter $L_2-C_2$ that is connected to the woofer. Here, only the low frequencies get through.

The exact design of the high- and low-pass filters that make up this network depends upon the "cross-over" frequency. The cross-over is the frequency at which the woofer response should begin to die out as the tweeter response begins to increase. The frequency at which cross-over occurs depends on the loudspeaker design. Some loudspeakers are designed for cross-overs around 200 to 400 cycles, others may have cross-overs in the range between 1000 and 3000 cycles. Therefore, the cross-over network used must be designed for the particular

![FIG. 23. Simple equalizer circuit used to correct high-frequency attenuation.](image)

loudspeaker combination that is being used. This means that you must select loudspeakers that are designed to work together—you can't just combine a small loudspeaker and a large one and hope to make them work well together. The design of the loudspeaker must be carefully worked out if a smooth overall response is to be obtained.

If the low-pass and high-pass filters are properly designed, the net impedance at the input terminals of the two loudspeakers will remain practically constant—effectively, as the impedance of one drops, that of the other will rise to compensate for it.

EQUALIZATION

Ordinarily, the over-all response of a complete p.a. system will be reasonably close to the expected design values. However, it is possible that peaks or valleys will appear in the response of a system when it is assembled. This may well occur if all of the components—the microphone, the amplifier, and the loudspeakers—happen to have peaks or dips in their response that occur at about the same frequencies.

The amplifier will usually have a tone control that will compensate for most of this kind of difficulty. However, there may be installations—particularly those in which transmission lines are used—in which it is not desirable to depend entirely on the tone control. For example, let's suppose

that a line somewhat longer than usual is required and that the high-frequency response has suffered accordingly. It may well be that the tone control of the amplifier is unable to make up this deficiency or is able to do so only by being turned to maximum treble gain, in which latter case there will be no reserve left for boosting the high frequencies in programs that need it. In either case, some other method of correcting the high-frequency attenuation should be used. If there is enough gain in the amplifier to permit us to throw away half the voltage, the equalizer shown in Fig. 23 can be used. Here, condenser $C_1$ has a capacity that is approximately equal to the total capacity introduced by the
line. Resistors $R_1$ and $R_2$ have equal resistances. Under these conditions, the over-all gain is reduced one-half, but the frequency response is extended considerably. The over-all response is also flatter. Incidentally, when impedance matching is important, the sum of $R_1$ and $R_2$ should equal the desired terminating impedance for this line.

Obviously, equalizers like this one can be used only where there is sufficient reserve gain to make up for the loss introduced by the equalizer.

Equalizers are also used for somewhat different purposes with phonograph pickups. A pickup may have a fairly high response in the region between 5000 and 7000 cycles, with the result that the normal record noises may prove annoying to some listeners. They may be willing to sacrifice fidelity to get rid of such noise. In such cases, scratch filters like those shown in Fig. 24 may be used. That in Fig. 24A is for use with a crystal pickup and that in Fig. 24B is for use with the magnetic pickup. Typical values of the circuit components are shown in each instance.
Lesson Questions

Be sure to number your Answer Sheet 51RH-2.

Place your Student Number on every Answer Sheet.

Send in your set of answers for this Lesson immediately after you finish them, as instructed in the Study Schedule. This will give you the greatest possible benefit from our speedy personal grading service.

1. Why are twisted pairs preferable to parallel lines as low-impedance audio lines?

2. When we refer to a "low-impedance" line in p.a. work, are we referring to any characteristic of the line itself or to the impedances connected to the ends of the line?

3. Give two reasons why the cable used to connect a high-impedance microphone to an amplifier should be shielded and short.

4. Why are low-impedance loudspeaker lines not used for long runs?

5. What would be the loop resistance of a 200-ft. loudspeaker transmission line made of No. 14 B & S gauge copper wire?

6. If loudspeakers of unequal voice-coil impedances are connected in parallel to a low-impedance line, which loudspeaker will get the most power—the one having the lowest impedance or the one having the highest impedance?

7. Suppose you are to connect 5 loudspeakers through line-matching transformers to a 250-ohm line, and each loudspeaker is to get the same power. What must the primary impedances of the transformers be?

8. Suppose you have to match 500 ohms to 8 ohms and have available a transformer designed to match 250 ohms to 4 ohms and another designed to match 1000 ohms to 16 ohms. Which of these transformers would give the better frequency response?

9. How are low frequencies kept out of the tweeter in a dual loudspeaker?

10. What is the purpose of the condenser-resistor unit connected across the output of a magnetic phono pickup?
STUDY SCHEDULE NO. 52

For each study step, read the assigned pages first at your usual speed, then reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind. Study each other step in this same way.

☐ 1. Introduction - - - - - - - - - - - - - - Pages 1-6
Here you learn what information you should gather to plan an installation.

☐ 2. Indoor P. A. Installations - - - - - - - - - - - - Pages 7-23
Low-power, medium-power, and high-power indoor installations are discussed here.

☐ 3. Outdoor P. A. Installations - - - - - - - - - - - - Pages 23-28
Various problems involved in installing p. a. systems outdoors are discussed in this section.

☐ 4. Mobile P. A. Installations - - - - - - - - - - - - Pages 29-36
This section contains a description of how to equip a sound truck for mobile p. a. service.

☐ 5. Answer Lesson Questions, and Mail your Answers to NRI.

☐ 6. Start Studying the Next Lesson.
YOU have already studied the equipment used in public address systems. In this Lesson, you will learn how to put this equipment to work in typical indoor, outdoor, and mobile p.a. installations.

Commercial p.a. installations are not usually very complicated from the technical viewpoint. Mechanical problems met in mounting loudspeakers or in running cables may make some jobs rather difficult, but the electrical and acoustical theory involved is generally fairly simple and straightforward. We shall, therefore, concentrate mostly on the practical aspects of p.a. installations in this Lesson.

We shall also restrict ourselves to discussing installations that are bought by the customer, rather than rented, except in the case of mobile installations. The typical rented installation is rather simple. If you go into the business of renting p.a. systems, you will undoubtedly have a stock of conventional medium-power amplifiers, some portable loudspeakers, a few dynamic microphones, and one or two record players. You will use this same equipment for all jobs as far as possible; obviously, you cannot afford to buy special equipment for one-time use. In other words, each job will be an adaptation of your existing equipment rather than an installation tailored for a particular problem. An installation that you sell, on the other hand, must meet the specific requirements of the location, and will therefore introduce problems in the selection and installation of equipment.

ADVANCE PLANNING

The simplest p.a. job, and perhaps the most common one, consists of making a temporary installation of a single microphone, a small amplifier, and one or two loudspeakers. These latter are usually housed in carrying cases that serve as baffles. Such an installation requires little or no advance planning if you are sure that a source of electrical power is available at the location to be used and that the equipment is adequate for the job. Any more complex installation, however, requires careful planning and preparation before the installation is made.
This planning consists of making a careful, detailed study of the location at which the installation is to be made and of reducing the results of your survey to written data that will show you exactly what is required to do the job. Naturally, the extentiveness of your study, and the completeness of your report, will depend on how complicated the installation is to be, but you should make an adequate study of every job except the simplest ones. Doing so makes it certain that you can set a price for the job that will be fair both to you and to your customer, and eliminates any chance of your running into unexpected difficulties.

Since proper advance planning is very important in a p.a. installation, let's spend a few moments now to learn just how you should go about making a job survey. There are three things to do:

1. Make a study of the location.
2. Make a sketch of the location, showing its shape, its important dimensions, and where the equipment is to go.
3. Put into writing other information gained from your study that will help you plan the installation.

To take care of this third step, it is a good idea to use a printed or mimeographed form that is complete enough to cover the most complicated job. Using such a form, rather than just taking notes, will keep you from forgetting to get some information that you will need.

A typical form is shown in Fig. 1. It is complete enough so that all the data required for a large, complex installation can be entered in it. On simpler installations, of course, you would not fill in the whole form—just the sections required for the particular job.

To show you the kind of information you need to plan a commercial p.a. installation, let's discuss each of the headings on this job survey form one by one.

The need for the four items above the line on the form—the file number, the date, and the customer's name and address—is obvious.

The first heading under the line is "purpose of installation." State here briefly what the installation is to be used for, such as "music and voice in dance hall" or whatever it is. Be sure to learn whether the installation is to be used for voice alone, or for both voice and music.

Next, list the acoustical facts. If it is an indoor installation, make the proper entry beside the headings "Room Volume, cu. ft.," and "Number of Seats." Also, put check marks in the appropriate places to indicate whether the floor is hard, medium, or soft; whether the walls are hard, medium, or soft; and whether the ceiling is hard, medium, soft, flat, or curved. These acoustical qualities of a room were discussed in earlier Lessons, to which you should refer if you have forgotten what each of the terms means.

Next, determine what the noise level of the room is when it is being put to its intended use. If possible, visit the room while it is in use; otherwise, estimate what the noise level will be. Put a check mark on the form in the proper place to show whether the room is very noisy, noisy, medium, or quiet.

Next, under the heading "Remarks," make notes about any special conditions that may require unusual treatment. For example, if the room is made noisy by sound coming from a motor room or a kitchen, you may want to recommend the installation of a wooden partition or a special door.
## JOB SURVEY FORM

**File No.**

**Customer's Name**

**Address**

**Purpose of Installation**

### ACOUSTICAL DATA

- **Room Volume, cu. ft.**
- **No. of Seats**
- **Area, sq. ft. (outdoor only)**
- **Walls: Hard □ Medium □ Soft □**
- **Floor: Hard □ Medium □ Soft □**
- **Ceiling: Hard □ Medium □ Soft □ Flat □ Curved □**
- **Noise Level:** Very Noisy □ Noisy □ Medium □ Quiet □
- **Remarks**

### EQUIPMENT

- **No. of Loudspeakers**
  - **Make**
  - **Model**
  - **Remarks**
- **No. of Microphones**
  - **Make**
  - **Model**
  - **Remarks**
- **Radio Tuner Input**
  - **Make**
  - **Power Supply**
- **Phono Input**
  - **Turntables: 1 □ 2 □ Make**
  - **Type**
- **Loudspeaker Wiring**
- **Microphone Wiring**
- **Power Wiring**

- **No. of Amplifiers**
  - **Make**
  - **Model**
  - **Power Output**
  - **Output Impedance**
  - **Frequency Response:** HI-FI □ Standard □ Special □
  - **No. Microphone Inputs**
  - **No. Phono Inputs**
  - **Power Supply**
  - **Remarks**

### COST ANALYSIS

- **Amplifier**
- **Loudspeakers**
- **Microphones**
- **Record Players**
- **Radio Tuners**
- **Cables**
- **Labor**
- **Total**

**Maintenance Agreement**

**Sound Engineer's Signature**

**Customer's Signature**

**Sworn To and Subscribed Before Me This Day of**

**Notary Public**

(My Commission Expires )

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**FIG. 1.** This is a sample of the kind of job-survey form you should make up for use in planning P.A. installations. To economize on space, we have left out some of the blank lines that you would normally use in a sheet of this sort. For example, you would probably want to use two or three lines for "Remarks" in each instance, although we have left but one.
to cut down the noise. If there are marked reflection effects, you may want to install a drape or curtain on one or more walls or put sound-absorbing material on the wall, floor, or ceiling. Make notes about any such conditions, because they will be important in your computations of the cost of the job.

If the installation is to be made in an outdoor location, enter the area in square feet in the proper place on the form. Determine the noise level at the location when a normal-sized audience is present, and check the form in the proper place to show whether the location is very noisy, noisy, medium, or quiet.

Next, under the heading “Remarks,” make notes about any steady or intermittent sources of noise, such as nearby trains or heavy traffic, or anything else that will play a part in determining the equipment you should use for that location.

The information thus far recorded will let you compute how much power will be needed to supply an adequate volume of sound. This, of course, has an important bearing on the number and type of loudspeakers to be used and on the amplifier to be selected.

The rest of the form can be filled out either when you make the first inspection of the location or afterward, depending on how complicated the installation is going to be. If it is a simple one, you can specify such things as the locations of loudspeakers and microphones as soon as you have inspected the location; on more complicated jobs, you will want to do some figuring first. In either case, the next thing to be entered on the form is information about the loudspeakers. Enter first the number of loudspeakers, then the make and model of each. (You will probably specify the make and model only after you have done the rest of your figuring and are making up final specifications for the job.) Finally, enter any remarks you want to remember about special locations.

Courtesy University Loudspeakers, Inc.

This is a typical church installation in which reflex trumpets are used instead of bells in a steeple. Louvered openings (shown in the back walls of the steeple) permit the sound to escape. This system is popular because it is far cheaper to play recorded bell selections through an amplifier and project the music through the loudspeakers than it is to install a set of bells. A great many recordings of bell music are available for such use.
and so forth. The actual placement of the speakers you should indicate on the sketch you draw of the location.

Similarly, enter the number of microphones, their makes and models, and any special remarks about them in the next section of the form. The use to which the installation is to be put will usually determine the number of microphones. Again, you will probably not specify makes and models until you are making your computations. The locations of the microphones should also be shown on your sketch of the installation.

If a radio tuner input is to be used with the installation, note that fact on the form by putting a check mark after the heading “radio tuner input.” Enter the make of the tuner, and the type of power supply required (separate or built-in).

Similarly, if a record player is to be installed, put a check mark on the form after the heading “phono input.” Also check the form to show whether one or two turntables are to be installed and enter the make and type (single-play, changer, mixer-changer) in the spaces provided on the form.

Next, specify the details of the wiring that will be necessary to connect the various parts of the installation. Space is provided for loudspeaker wiring, microphone wiring, radio-tuner wiring, and power wiring. In each case, record the type, size, and length of wire needed. For example, if you are going to need twenty feet of No. 14 BX cable to a high-power loudspeaker and a light twisted pair to a small loudspeaker, record these facts by saying simply “20 feet No. 14BX to 25-watt driver,” and “30 feet No. 18 lamp cord to single 5-inch p.m.” Similarly, list the microphone, record player, radio tuner, and power wiring by type, size, and length of wire or cable. (In many installations, of course, it will not be necessary to install special power wiring, since there will be outlets available into which the various pieces of equipment can be plugged.)

Information about the record-player wiring and radio-tuner wiring should all be concerned with the actual audio line from the equipment to the amplifier, not with any power wiring that may be necessary for them. This latter information should be listed under power wiring.

The general positions of all important wires should be shown on your location sketch. If there is apt to be any confusion because of the complexity of the installation, assign each circuit a number on your diagram and refer to the particular wire or cable by that number on your record sheet.

Once you have the facts concerning the speakers, microphones, and so forth, you will be able to select the type of amplifier needed. Space is provided on the form for you to show the number of amplifiers. If only one is to be used, list its make and model, its power output, its output impedance, the db gain of the microphone channels, the db gain of the phono channels, the number of microphone inputs, the number of phono inputs, whether or not a separate power supply is needed (you would fill this in as either “separate” or “built-in”), and the number of gain controls. There are also spaces that you can check to show whether the amplifier has a bass tone control, a treble tone control, or both, and to show whether the amplifier has a high-fidelity response, a standard frequency response, or a special response. (This last refers to an amplifier used for some special purpose requiring an extended low or high range of frequen-
cies.) Finally, there is a space for you to enter any special remarks that you feel will be helpful.

If more than one amplifier is to be used, and they are all identical, this listing will serve for all. If two different amplifiers are used, list the data on one above that on the other, using the same blanks for both. In this latter case, be sure you are consistent about keeping the data on one amplifier on top.

This completes the information needed to plan the installation. In addition, of course, you need to make an accurate estimate of the cost of the installation for the customer. Space is provided on the form for this under the heading "COST ANALYSIS." Here, enter the list price of the amplifier, the loudspeaker system, the microphones, the record players, the radio tuner, and the cables. In addition, list the cost of the labor to the customer.

You may, if you wish, charge a flat fee for the labor involved in the installation. However, if the work is to be done partially or wholly by someone else—say by a licensed electrician, as is required in many communities—you'll probably not have any very good way of estimating just what the cost of labor will be. It is better, in this case, to bill the labor as "time and material." This means that the customer is to pay for the material used and for the time actually spent in making the installation. The per-hour charge for labor should also be quoted; this figure should be high enough to cover the actual time charge of the workmen plus a reasonable profit for yourself. Be sure to remember this last item, because, on most jobs, the greater part of your profit will probably come from the labor charge.

In the space marked "Maintenance Agreement" write the details of any agreement you entered into with the customer about maintaining the equipment. For example, you might have an agreement to furnish one year's service at a cost to the customer of $10 per call plus cost of replacement parts, or you might offer free service for 3 months and future service for an annual fee.

When you have completed filling out the form, make a copy for your customer, sign both copies yourself, and have him sign both copies also. If you feel it to be desirable, you can have both signatures notarized by a notary public; space for this is provided in the form.
Indoor P.A. Installations

Now that we've seen the general procedure to follow in making advance plans for any kind of installation, let's study in detail several typical installations—one of low power, one of medium power, and one of high power. We'll take up the low-power installation first.

**LOW-POWER INDOOR SYSTEM**

Let's suppose someone has asked you to install a small, low-power p.a. system for a large business luncheon commercially available size, will provide enough power for the installation.

**Amplifier.** The Bogen E14 amplifier is well suited to this installation. Its schematic diagram is shown in Fig. 3.

This amplifier has two high-impedance microphone inputs and one phono input. Other models are available in which either or both of the microphone channels are low impedance. The amplifier delivers 14 watts at less than 5% distortion, and has a peak power of 25 watts.

![Diagram of an indoor p.a. system](image)

**FIG. 2.** This shows the sort of information you should put on a sketch of a proposed installation. You need not be as neat as this in your drawing, but be sure to take reasonable pains with it so that your finished sketch will be a recognizable plan of the installation.

where an audience of 100 people is expected.

The first step, as you know, is to inspect the location. Then you should make a sketch of the location and fill in the parts of the job survey form that apply to the particular job.

Let's suppose your sketch of the location is like that shown in Fig. 2. This sketch shows the general location of the mike, loudspeakers, and audience. As you can see, one microphone and two loudspeakers are to be used.

We'll assume that your studies of the location show that a power output of approximately 12 watts is needed to provide sound coverage of the room. A 14-watt amplifier, which is a com-

As you can see from the diagram, the microphone inputs feed into separate 7B4 voltage amplifier tubes, the outputs of which are fed to the grid of one section of a 7F7 dual triode. The output of this tube is fed to a voltage amplifier-phase inverter stage in which a 7N7 dual triode is used. The output of this last stage is fed to the control grids of the output stage, in which a pair of 6L6's in push-pull is used.

The phono input is applied to the grid of the other section of the 7F7 voltage amplifier, the plate of which is connected in parallel with the plate of the other section. All three inputs are therefore mixed before reaching
the control grid of the 7N7 voltage amplifier section. The volume level of each input is controlled earlier in the circuit by separate potentiometers.

A tone control circuit is incorporated in the grid circuit of the voltage amplifier section of the 7N7 stage.

The output of the amplifier is fed to two paralleled 5-hole speaker sockets. The output impedance of the amplifier at these sockets can be adjusted by connecting a flexible lead to any one of five terminals on a strip on the back of the amplifier. The available output impedances are 4 ohms, 8 ohms, 15 ohms, 250 ohms, and 500 ohms. A common (grounded) terminal is also provided on this strip so that you can, if you wish, connect speakers directly to the strip instead of to the speaker sockets.

**Microphones.** Either a crystal or a dynamic high-impedance microphone can be used with this amplifier. The microphone you choose should have a cardioid pickup pattern, since it is intended for use by someone speaking rather than for general sound pickup. The dynamic microphone is preferable in that it is more rugged than a crystal and less susceptible to damage if it is stored some place where the temperature becomes high. However, a good dynamic costs more than a good crystal microphone, so, if price is an object, it may be better to use the crystal type. A reasonably good crystal microphone will be perfectly adequate for the job it has to do in an installation of this sort.

Shielded lines must be used to connect the microphone or microphones to the amplifier. These should be not over 25 feet long, and preferably shorter. (If the microphone cable run must be longer than 25 feet, you should use a low-impedance microphone and the model of the amplifier that has a low-impedance input.) Standard shielded lines are available that are automatically grounded when the plugs at their ends are connected to the amplifier.

**Loudspeakers.** Two 8-watt loudspeakers will provide sufficient sound coverage in a room of this size. These can be mounted in wall baffles that are secured to the side walls of the room. One should be near the front of the room and the other near the rear. It is probably best to mount them both on the same wall, although it will be well to experiment to see whether it might not be better to mount them on facing walls. They should not, of course, be directly across from one another.

Alternatively, you can use cone loudspeakers mounted in projectors. These are provided with mounting arms that can be secured to the wall. The projector can then be aimed in any desired position. If you use these projector loudspeakers, you should mount one on either side of the room slightly ahead of the speaker's table, aiming them so that their sound patterns cover the room completely.

It would be possible to use one loudspeaker instead of two as far as power requirements are concerned. In an indoor installation of this sort, however, it is better to use at least two loudspeakers to secure even sound distribution.

Whatever loudspeakers you use, make sure that they are placed so that their sound output does not reach the microphone. Otherwise, there will be a feedback of acoustical energy that can cause howling.

**Installation.** The installation of a system of this kind is not at all difficult. Essentially, all you have to do is connect the microphone and the speakers to the amplifier, and plug the
FIG. 3. Schematic diagram of a Bogen E14 amplifier.

**SPEAKER SOCKETS**
Connect lug on flexible lead to impedance desired at Speaker sockets.

*Courtesy David Bogen Co., Inc.*
amplifier into the power source. You will, of course, have to make a careful check of the system in operation to be sure that feedback and howling will not occur even when the amplifier is delivering maximum output. If you find it impossible to prevent some feedback at maximum output, you have to determine the maximum output at which howling will not occur; announcement to speak louder than normal. When the luncheon is finished and scheduled speeches are being made, the noise level will be considerably lower and it will be desirable to reduce the amplifier output. Therefore, you should instruct the customer or one of his representatives in the manipulation of the volume and tone controls of the amplifier. If you have found it necessary to use less than maximum output to prevent acoustical feedback and howling, be sure to point that out to him.

In addition, teach the customer or his representative a few simple facts about the proper care and use of the equipment. If the microphone is to be stored when it is not in use, show him how to disconnect it and, if it is a crystal microphone, warn him about the effect of heat on the crystal. A little time spent in teaching someone how to use equipment properly may save you future complaints from the customer.

Now, let's see how you would go about installing a medium-power sound system.

**MEDIUM-POWER INSTALLATION**

The sketch of a typical medium-power p.a. installation is shown in Fig. 4. This installation is to be used to reproduce both voice and recorded music in an auditorium seating some 600 people. The reproducers are to be mounted on or near the stage. We'll assume that your preliminary acoustical studies have shown that a power of 25 watts will be needed.

A 25-watt amplifier like the Thordarson T-31W25A shown schematically in Fig. 5 will provide the necessary power. You can see from the schematic that this amplifier has two microphone inputs and one phono input. The amplifier is normally built with high-impedance microphone inputs, but
models with low-impedance inputs can be secured. Circuits for both types are shown in the diagram. The phono input is always high impedance.

**Amplifier.** Each microphone channel has a preamplifier stage in which a 6SJ7 is used. The outputs of both these stages and of the phono channel are applied to the grid of another 6SJ7 that is used as a voltage amplifier. The output of this stage is applied to a 6N7 driver stage that feeds the output stage, which contains two 6L6 tubes connected in push-pull. The 6N7 is a dual triode, but, in this use, its plates, grids, and cathodes are paralleled to double its power-handling capability.

The output transformer has a tapped secondary that offers output impedances of 4, 8, 15, 125, 250, and 500 ohms. A selector switch permits any of the taps to be connected to two paralleled sockets into which the speakers or the audio line is plugged.

Each channel contains a potentiometer by which the volume level of that channel is controlled. The amplifier has no master volume control with which the volume in all channels can be controlled simultaneously.

The frequency response of the amplifier is flattened by the use of inverse feedback in the 6SJ7 stage just ahead of the driver stage. There are two feedback paths, one from the secondary of the output transformer to the cathode circuit of the 6SJ7, and the other from the plate circuit of the 6SJ7 back to the grid.

The amplifier has a bass and a treble tone control. The bass control consists of $C_9$ and variable resistor $R_{19}$. Condenser $C_9$ is used in the coupling circuit between the 6SJ7 and the 6N7. Resistor $R_{19}$ is shunted across it. The influence of $C_9$ on the frequencies passed to the 6N7 can be varied by varying the resistance of $R_{19}$. When $R_{19}$ is adjusted so that

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*Courtesy Montgomery Ward*

This is a typical temporary p.a. installation. Notice the amplifier beside the music stand at the extreme left. Cone loudspeakers mounted in projector housings are used. These housings, which have flared openings, give the loudspeaker output a certain amount of directional effect. Notice that the projectors are located ahead of or beside the microphones; this eliminates direct acoustical feedback and lessens the danger that the system will go into oscillation.
its resistance is maximum, the lower frequencies in the signal are dropped in $C_9$; when $R_{10}$ is adjusted to have zero resistance, $C_9$ is effectively removed from the coupling circuit, and the low frequencies in the signal are passed on to the grid of the 6N7.

The treble tone control consists of potentiometer $R_{20}$ and condenser $C_{10}$. The two ends of $R_{20}$ are connected between the grid of the 6N7 and ground, and $C_{10}$ is connected between the slider of $R_{20}$ and ground. When the slider is run to the upper end of $R_{20}$, the higher frequencies are bypassed around the grid resistor, and cycles. When these controls are in their normal positions, at which they provide no attenuation, the frequency response of the amplifier is flat within 1 db from 30 to 15,000 cycles.

**Loudspeakers.** Assuming that your figure of 25 watts is based on the use of high-efficiency loudspeakers, you must select either folded auditorium horns or reflex trumpets for this installation. Trumpets are not particularly suitable for use in auditoriums, particularly when music is to be reproduced. For one thing, they have a directional effect, and would therefore not provide even sound cov-

![Image of Thordarson T-31W25A 25-watt amplifier.](image)

**Courtesy Thordarson**


the high-frequency response of the amplifier is therefore reduced. When the slider is run down to the lower end of the potentiometer, the condenser $C_{10}$ has no effect on the signal.

As you can see from this description, these tone controls are attenuators; that is, they can decrease the low-frequency or high-frequency response, but cannot boost it above the normal level. The bass control provides an attenuation of 20 db at 50 cycles and 12 db at 100 cycles. The treble control gives an attenuation of 40 db at 1000 cycles, 15 db at 5000 cycles, and 23 db at 10,000 cycles. When these controls are in their normal positions, at which they provide no attenuation, the frequency response of the amplifier is flat within 1 db from 30 to 15,000 cycles.

**Loudspeakers.** Assuming that your figure of 25 watts is based on the use of high-efficiency loudspeakers, you must select either folded auditorium horns or reflex trumpets for this installation. Trumpets are not particularly suitable for use in auditoriums, particularly when music is to be reproduced. For one thing, they have a directional effect, and would therefore not provide even sound cov-

The folded auditorium horn is superior to the trumpet in both respects. It offers wide-angle sound coverage and reproduces sound with much better fidelity. It has certain disadvantages, also: it is expensive, bulky, and heavy. There is, however, no better choice available where both high fidelity and high volume levels are needed.
High-power loudspeakers of the cone type mounted in console cabinets are sometimes used for sound reproduction in small auditoriums. Their fidelity, particularly when a tweeter-woofer combination is used, is very good, but they cannot handle volume levels as high as those handled by diaphragm-driven units, and their efficiency is considerably lower. In addition, such loudspeakers usually have considerable rear-end radiation, and they therefore produce acoustical feedback and howling unless they are very carefully placed with respect to the microphone. For this reason, they are usually not suitable for use in installations where the microphone may be moved about.

Let’s assume, therefore, that you will install two folded auditorium horns.

It is usually impractical to mount these horns on the wall or suspend them from the ceiling; they are usually so big and heavy that mounting them in either of these ways would be a major construction job. Probably the easiest way to mount them is to install them on the stage, one at either end.

You will probably want to conceal the horns. To do so without interfering with their efficiency, hang a drape in front of them as shown in Fig. 6. Notice that this drape is made of two kinds of cloth: heavy material is used at the top of the drape to make it hang properly, and lighter material that will pass sound readily is used for the part of the drape that hangs directly in front of the horn. An air space of at least 12 inches must be left between the drape and the mouth of the horn to prevent an undesirable increase in the loudspeaker loading.

Preferably, the horns should be secured to the floor to prevent their being accidentally moved out of position. If they must be left unsecured so that they can be moved out of the way when necessary, mark the proper locations on the stage floor so that they can be replaced properly.

The high-frequency response of the auditorium system can be improved by adding a pair of tweeters. These tweeters should be placed 10 or 12 feet above the stage, one at either end, and tilted downward at about 20° from the horizontal. If tweeters are used, of course, a suitable high-frequency cross-over network must be used to supply high frequencies to the tweeters and low frequencies to the horns.

**Microphones.** As you know, the amplifier used with this installation has two microphone channels, each of which can be either high impedance or low impedance. It is not, of course, necessary to use both inputs; however, it is probably a good idea to do so if the stage is to be used for plays or for other activities that will require sound pickup over a wide area.

Since there may be an orchestra in the pit playing while the micro-
phones are in use, it is best to use microphones having cardioid pickup patterns. These microphones will then pick up chiefly the voices of those on the stage, more or less ignoring the sounds of the orchestra. Both dynamic and crystal microphones having cardioid pickup patterns are available.

There are several possible locations for the microphones. If they are to be used for picking up voices during the presentation of plays or other stage performances, you will probably want to conceal them as much as possible. One way to do so is to install them in the footlights. Of course, it may not be possible to use this installation if a trial shows that it does not give sufficient voice pickup, or picks up too much foot noise. In this case, it may be practical to suspend the microphones from the ceiling of the stage if doing so will provide better pickup. If the stage is deep, it may prove impossible to use concealed microphones and still get adequate pickup from all positions on the stage.

**Amplifier Location.** The amplifier may be conveniently located either back-stage or in the orchestra pit. If a phono pickup is to be used with the amplifying system, it will probably be best to place the amplifier and the phono attachment back-stage. Records can then be changed by the person operating the amplifier.

An installation of this sort will require a certain amount of monitoring, since it will probably be put to a variety of uses that will require changes in the volume levels and perhaps the tonal characteristics. Explain the operation of the various controls on the amplifier carefully and thoroughly to the person who will be in charge of it.

**Tests.** You should, of course, test the operation of the system carefully after it is installed. In particular, make sure that there will be no feedback between the auditorium horns and the microphones. If a microphone is to be used on a stand, try it in all the positions in which it can be placed before concluding that there will be no feedback. If you find there is feedback, turn the auditorium horns slightly outward until it no longer occurs.

Now, let's see what procedure should be followed in installing a high-power indoor p.a. system.

**HIGH-POWER INSTALLATION**

The sketch of a large indoor installation is shown in Fig. 7. This installation is made in a large indoor arena. Its major use is for making announcements during sporting events, such as hockey games, prize fights, and basketball games, and for playing records before and after such events and during intermissions. Even though music is played over this system, intelligibility and carrying power are the chief requirements, not high fidelity. The fidelity should, of course, be as good as it is practical to make it without sacrificing power.

Let's assume your studies of the location show that adequate coverage will be given if an electrical power of 200 watts is used. This is to be divided among eight reflex loudspeaker trumpets, located in the center of the arena and each aimed at one of the sections of seats.

You can get the required electrical power using either a single high-power amplifier or a number of lower-power amplifiers. It is more common to use several amplifiers, since they are readily available, whereas single high-power amplifiers must usually be custom-built.
For example, 50-watt amplifiers are commonly available. Four of these amplifiers will supply a total of 200 watts, just enough for the installation. The schematic diagram of such an amplifier—the Thordarson T-31W50A—is shown in Fig. 8.

This amplifier has remarkably good fidelity for the power it handles. The manufacturer states that its frequency response is flat within 1 db from 30 to 13,000 cycles, and that there is less than 5% distortion at the full output of 50 watts. The 50 watts is a conservative rating; it will supply over 65 watts at peak power.

With two exceptions, which we shall discuss in a moment, the circuits of this amplifier are not unusual. There are three high-impedance microphone input channels and one dual high-impedance phono input channel, all of which feed into a 6SJ7 voltage amplifier stage. Each of the microphone channels has one stage of amplification ahead of this common amplifier stage. The output of the 6SJ7 stage is applied to a 6J5 voltage amplifier, the output of which is applied to a 6V6 used as a driver for the output stage, which contains four 6L6’s in parallel push-pull. The 6V6 used as a driver is triode-connected (the screen grid is connected directly to the plate). An input transformer is used to connect the driver stage to the output stage.

The output transformer is tapped

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**FIG. 7.** This set-up is typical of those used in large indoor arenas. The loudspeakers are mounted on a platform suspended from the roof in the center of the area, a location that makes it possible to have a uniform sound coverage for all seats. Eight reflex loudspeakers, only 3 of which are shown in this sketch, are used to furnish the sound output. Each loudspeaker is aimed at the center of one section of seats. Notice that a clock is also mounted on the platform. In many installations, a clock and a scoreboard are mounted on each of the four faces of the platform for the convenience of the patrons.
to provide impedances of 4, 6, 8, 16, 125, 250, and 500 ohms. A switch connects the desired tap on the secondary of the output transformer to half the receptacles in each of two parallel-connected octal sockets, from which connections are made to the loudspeakers.

We mentioned earlier that there are two unusual features in this amplifier circuit. One is the use of inverse feedback in the 6SJ7 voltage amplifier stage that is fed by all the input channels. The effect of this is to reduce the high-frequency response of the stage.

![Image](image.png)

*Courtesy Thordarson
The Thordarson T-31W50A 50-watt amplifier.*

The other unusual feature is the tone-control circuit, which is located directly below the 6J5 tube in the circuit diagram. This is a dual tone control that permits both the high-frequency and the low-frequency response to be adjusted above or below the normal position (at which the response is practically flat).

The circuit uses two dual potentiometers. These consist of two potentiometers, each of which is resistive for half of its circumference and conductive for the other half. Thus, when the slider is moved along the circumference of the potentiometer, the resistance between the slider and the end of the resistive element will increase until the slider has been turned to the center of the device; then the slider comes into contact with a section that has practically no resistance, and further rotation does not increase the resistance between the slider and the resistive end of the potentiometer.

To make up a dual potentiometer of the sort used here, two of these are mounted back to back so that a single control shaft operates both sliders. As you can see from the diagram in Fig. 8, with this arrangement the slider arm of one potentiometer is moving over a resistive portion while the other is moving over a conductive portion; then, when the midpoint of the control is reached, the slider of the first changes over to the conductive portion of the first control and the slider of the second starts on its resistive portion.

Degeneration is produced in the cathode circuit of the 6J5 stage of this amplifier because of the presence of $R_{22}$, which is inadequately by-passed for audio frequencies. (Although condenser $C_{13}$ and $C_{16}$ are connected in series across this resistor to ground,
their net capacities are so small—$C_{16}$ is only .001 mfd.—that the series combination has little by-passing effect even at fairly high audio frequencies.) Since this resistor is in the cathode circuit, the tube plate current must pass through it; therefore, a voltage is developed across it that is in phase with the a.c. signal current of the tube plate current.

By tracing through the circuit carefully, you can see that the voltage developed across the load of the preceding 6SJ7 stage, which is the source of the signal applied to the grid of the 6J5, is in series with the a.c. voltage developed across $R_{22}$. Therefore, the algebraic sum of these a.c. voltages is applied to $R_{21}$ as the grid signal. These two voltages are always opposite in polarity at any instant, so the voltage applied to the grid of the 6J5 stage is always the difference between them.

For example, when the a.c. voltage across $R_{20}$ is such that the upper or plate end of the resistor is positive, the voltage applied to $R_{21}$ will make the grid of the 6J5 more positive. This will cause an increase in plate current, causing a voltage drop across $R_{22}$ having a polarity such that the cathode end of $R_{22}$ will be positive. Trace the circuit—you will see that the voltage across $R_{22}$ will then oppose the voltage across $R_{20}$ (remember, we are talking about a.c signal voltages). Therefore, the voltage applied to $R_{21}$ will be less than it would be if $R_{22}$ were not present. In other words, the presence of $R_{22}$ tends to reduce the grid signal, with the result that for a given input signal, the amplification of the 6J5 stage is less than it would be if $R_{22}$ were not in the circuit. This effect, as you have learned in earlier Lessons, is called degeneration.

Since this is a resistance network, its effect is the same for all frequencies in the audio range. The tone control is designed so that it can reduce the amount of degeneration caused by $R_{22}$ for certain frequencies. Of course, if the amount of degeneration is reduced for any given frequency, the amplification of the 6J5 stages increases as far as that frequency is concerned. This amounts to a boost for this particular frequency.

First, let's see how the treble boost control operates. As you can see from the diagram; condenser $C_{14}$ is connected between the two sliders of this control. As the sliders are rotated from the center position of the control toward the "boost" end, a network consisting of $C_{14}$ plus part of the resistance of the potentiometer is connected in parallel with condenser $C_{16}$. As the control is advanced toward the boost end, the resistance in this network is decreased; when the control reaches the full boost position, all resistance is removed and $C_{14}$ is in parallel with $C_{16}$. Resistor $R_{22}$ is always by-passed by the series combination of $C_{13}$ and $C_{15}$, but the capacities of these two are so small, as we said earlier, $R_{22}$ is effectively by-passed only for the highest audio frequencies. However, when the treble tone control is advanced to the boost position and $C_{14}$ is put in parallel with $C_{16}$, the capacity of the by-pass network is increased to such an extent that practically all the high frequencies pass through the condensers rather than through $R_{22}$. As a result, there is little degeneration as far as these frequencies are concerned, and the high-frequency output of the amplifier is effectively boosted as a result.

The bass boost tone control works in much the same way, except that
a coil $T_6$ is connected between the two sliders. When this control is at the full boost position, the by-pass path around $R_{22}$ consists of $C_{13}$ and $T_6$. Since $T_6$ has a high impedance for high frequencies, and a low impedance for low frequencies, this path is effective as a by-pass for the lows. Thus, advancing the bass tone control to the full boost position reduces the degeneration in $R_{22}$ for the lows and effectively boosts the low-frequency response.

The boosts offered by these controls are not extremely great. The bass control gives a bass boost of $9\frac{1}{2}$ db at 80 cycles, the treble control gives a boost of $11\frac{1}{2}$ db at 8000 cycles.

These controls provide attenuation of the treble or bass in a similar manner by providing a variable by-pass for $R_{27}$, the grid resistor of the 6V6 stage. At the full-attenuation position of the treble control, condensers $C_{14}$ and $C_{15}$ are connected to ground in parallel across $R_{27}$. The high frequencies are then shunted to ground across this grid resistor, thus reducing the high-frequency part of the signal applied to the grid of the 6V6. When the bass control is advanced to its full-attenuation position, $T_6$ is connected to ground in parallel with $R_{27}$. Coil $T_6$ then acts as a low-impedance shunt for the low frequencies, reducing the proportion of them in the signal applied to the grid of the 6V6 and thus reducing the low-frequency output of the amplifier.

The attenuations offered by the controls are somewhat greater than the boosts they provide. The bass control gives an attenuation of 25 db at 80 cycles, and the treble control gives an attenuation of 25 db at 8000 cycles.

**Amplifier Location.** Usually the best place to locate the amplifier is the announcing booth. Then the announcer or technician can monitor the system when it is necessary to do so. The record player used with the system should also be located in the announcing booth so that the records can be changed by the announcer or technician in charge of the assembly. If there is no announcing booth in the arena, the amplifier should be located in some convenient place at which it can be monitored readily.

**Loudspeakers.** Eight 25-watt reflex trumpets will provide adequate sound coverage in this arena. You can see from the sketch that they are mounted on a platform suspended from the center of the roof. This arrangement, which is frequently used in indoor arenas, has the advantages that it permits each loudspeaker to cover a large area and also that all listeners are about the same distance away from the loudspeakers. This latter is an advantage because it permits the volume level to be about the same at all seats.

It is common practice, when this platform arrangement is used, to make the platform very substantial and to suspend scoreboards, lights, and perhaps a timing clock from it, as well as to use it as a mounting point for the loudspeakers. Such a platform must, of course, be installed by a construction crew; it is not part of the duties of the p.a. expert to build it or supervise its building.

A typical reflex trumpet suitable for use in this application has a sound dispersion angle of 90° and a low-frequency cut-off of 120 cycles. The fidelity of reproduction will not be too good with a cut-off characteristic of this sort, but it will be good enough for the uses to which the amplifier system is to be put. If the
arena is to be used for public ice skating, it may be desired to play records of organ music over the p.a. system; in this case, a somewhat larger reflex trumpet capable of reproducing lower frequencies should be installed.

A reflex trumpet is usually equipped with a mounting bracket that makes mounting it on a platform of this sort very simple. Just secure the bracket to the platform near the edge with screws, and the mounting job is done. Before mounting each loudspeaker, be sure to position its brackets so that you can aim the loudspeaker at approximately the center of the section of seats it is to cover.

**Power Distribution.** Since all eight loudspeakers are to be installed in approximately the same location, the most practical way to feed them is to connect them so that they form a common load for all four amplifiers. The chief problem involved in doing so is to connect these lines in such a manner that their net impedance will be a value that can be matched to the amplifiers with an available transformer.

A sketch of the connections in the distribution system is shown in Fig. 9. As you can see from the sketch, each loudspeaker is connected to the secondary of a matching transformer (T₁ through T₈), the primaries of which are connected in parallel to a single line. The four amplifiers, also connected in parallel, are connected to the other end of the line.

![Diagram of power distribution system](image)

**FIG. 9.** Power distribution system used to feed 8 loudspeakers from the outputs of 4 amplifiers. The 8 matching transformers, T₁ through T₈, are used to make the net impedance of the paralleled amplifiers. Notice that all connections are made to terminal boards for neatness and convenience.
Of course, the distribution system must provide proper impedance matches to prevent waste of power. In the system shown in Fig. 9, matching transformers T₁ through T₄ must provide primary impedances of 1000 ohms each when they are connected to the loudspeakers. The net primary impedance of all the transformers when they are connected in parallel will then be 125 ohms (1000 ÷ 8). At the other end of the system, the 500-ohm output terminals of the four amplifiers must be connected in parallel. The net output impedance of the four amplifiers is then also 125 ohms (500 ÷ 4). The paralleled speakers can then be connected to the paralleled amplifiers by an ordinary twisted-pair line. In theory, this line should also be 125 ohms in impedance, but, as a practical matter, standard p.a. cable, which has an impedance of approximately 500 ohms, can be used without there being any serious mismatch. To avoid excessive power loss due to resistance in the line, the wires in the cable should be 14 gauge.

Notice that the primaries of the matching transformers are brought to a terminal board before the parallel connections to the line are made. A similar arrangement is also used at the amplifiers. In each case, the terminal board should be very close to the equipment to which it is connected. This use of terminal boards is a good idea for several reasons: it makes a neat installation, it is of great assistance in helping you to identify individual circuits when you are servicing the installation, and it makes it easier to install a new component if one of those in use becomes defective.

This distribution system has two particularly good features. One is that it makes it unnecessary to use line-matching transformers; the only transformers used in the system are the impedance-matching transformers used at the loudspeakers, which would be necessary in any distribution system. The other advantage of this method of wiring is that only one line is run from the platform where the loudspeakers are mounted to the place where the amplifiers are installed. This, of course, represents a great saving in wire over what would be needed if individual lines were run from the loudspeakers to the amplifiers.

Microphones. Normally, only two microphone inputs and one phono input are used in an installation of this sort. One microphone is located at some point from which the announcer can see the floor area of the arena clearly. If the arena is used for boxing or wrestling, it is usual to have a microphone arranged so that it can be lowered to a point just above the ring for the making of announcements and can be taken out of the way during the progress of the bout. Sometimes provision is made for a third microphone that is located near the edge of the floor so that an official, such as the scorekeeper at a basketball game, can make announcements.

There are so many possible variations in the microphone set-up in an arena of this sort that we cannot give any definite rules for it. It is probably best, under almost any circumstances, to use microphones having cardioid pick-up patterns, since each microphone will usually be intended for use by only one person at a time.

You may find it necessary to use a pre-amplifier with one or more microphones if the microphone cable must run a long way to the amplifier. It is unlikely that the microphone cable
will be so long that the input signal will be too attenuated to operate the amplifier. However, it may well be that there will be enough hum pick-up in the microphone cable to cause trouble unless pre-amplification is used. The hum signal will not be harmful, of course, if the microphone output is built up to such an extent by a pre-amplifier that its level is much greater than that of the hum picked up. The only way to tell whether or not hum is going to be bothersome is to run the microphone cable in its proper location and make an operating test. If hum is then objectionably noticeable, you should consider using a pre-amplifier.

If a pre-amplifier must be used on a microphone that is to be lowered over a boxing or wrestling ring, probably the best place for you to locate it is on the platform where the speakers are mounted. Some care may be necessary in choosing the proper location for the pre-amplifier in this case, because some of the sources of hum, such as timing clocks, may also be located on the platform. Naturally, pre-amplification must occur before the hum signal is picked up by the microphone cable; otherwise it will be amplified along with the microphone signal, and the pre-amplification will be of no value. If the pre-amplifier is located on the platform, you will probably have to arrange some method of turning it on and off remotely unless the platform is so situated that it is easy to get to it.

If you find that a pre-amplifier is necessary for use with a microphone that is to be used on a judge’s or scorekeeper’s table on the edge of the arena floor, you will probably find it best to install the pre-amplifier on or under the table. It will then be possible to change the volume level of the pre-amplifier easily whenever it is found to be necessary. However, in most uses, and particularly when the pre-amplifier is to be located some place that is hard to reach, it is best to leave the volume level fixed and adjust the overall volume of the system at the main amplifiers.

If a microphone is to be lowered from the ceiling or from the platform to a ring, it will be necessary to provide a mechanical means for doing so. The simplest method is to suspend the microphone from a heavy cord, such as a sash cord, and run the cord through a guided pulley or two to some location from which an attendant can raise or lower the microphone as desired. A more elaborate method is to secure the cord to a small drum driven by a reversible electric motor; the microphone can then be raised or lowered by operating a switch by means of which the motor can be turned off or made to run in either direction. It is not advisable to suspend the microphone by the microphone cable alone.
Outdoor P.A. Installations

There are two chief differences between outdoor and indoor p.a. installations. One is that area, not volume, is the factor that you must consider in determining how much power is going to be needed to provide adequate sound coverage outdoors. The other is that the loudspeakers you use and the lines through which they are fed must be weatherproof.

![Diagram](image)

**FIG. 10.** A strong, permanent outdoor loudspeaker mast can be made by imbedding a pipe in concrete. The distance “d” should be at least one-third of the height above ground of the mast.

In its other essentials, however, an outdoor p.a. installation is like an indoor one. Similar amplifiers are used for both applications, although greater power is usually needed outdoors. Therefore, we shall not discuss the theory of the outdoor installation very much, since it does not differ basically from what you have already learned. Instead, we shall take up specific points that you would meet in outdoor work but are not likely to meet in indoor installations.

**MOUNTING LOUDSPEAKERS**

Very often, some support must be constructed for mounting a loudspeaker outdoors. A steel mast is frequently used for this purpose.

One method of installing such a mast is shown in Fig. 10. This mast is a heavy steel pipe, at least 3 inches in diameter, imbedded in concrete. The distance that the end of the pipe is below the ground should be at least one-third of the total height of the pipe above ground.

A mast of this sort may be used to support either a single loudspeaker or a cluster of them. The loudspeakers may be permanently secured to the top of the mast if desired; however, it will then be necessary for a serviceman to get to the top of the mast if one of the loudspeakers becomes defective. In many installations, the problem is solved by mounting the loudspeakers on a sliding collar that fits around the mast and raising or lowering the whole assembly with the aid of a rope that is secured to the collar, run over a pulley at the top of the mast, and brought down to a cleat at the bottom of the mast. This rope can then be used to raise
the loudspeakers to the top of the mast for use and to lower them to the ground for servicing.

If this method is used, the loudspeaker cables should be brought down through the hollow interior of the mast to a hole drilled near the bottom. The cables can then be fed into or pulled out of this hole as the loudspeakers are raised or lowered.

If the loudspeakers are to be fastened permanently to the top of the mast, the cables should again be let down through the inside of the mast and brought out through a hole in the side of the mast near the bottom. They should then be secured to terminals in a junction box, which should then be secured to the mast over the hole.

Even thick-walled steel pipes in a 3-inch or 4-inch size may not be strong enough to hold up three or four heavy loudspeakers if the mast is tall. The strength of the mast can be very considerably increased by pouring it full of concrete after it has been set in place. Of course, if you do this, you will be unable to run the cables from the loudspeakers down through the pipe. If you prefer to have the cables hidden when the installation is complete, run a piece of conduit through the pipe from the top to a hole near the base before you pour the concrete into the pipe. Be careful not to pour any concrete into the open end of the conduit. Then, when the concrete is set, you can thread the loudspeaker cable through the conduit and still have a reinforced mast. Alternatively, you can fill the mast solid with concrete and run the loudspeaker cables down the mast on the outside, enclosing them in heavy conduit.

Whether the loudspeaker cables are brought down the inside or the outside of the mast, they should go underground very near the base of the mast. In fact, it may be desirable to have them come down underground at the base of the mast. We will discuss this matter of underground cables in more detail a little later in this lesson.

Under some conditions, it may be desirable to use semi-permanent masts instead of permanent ones. This may be true, for example, when a football field is to be used during the summer months as an outdoor auditorium. In this case, masts will be needed for a period of several weeks, but cannot be left in place indefinitely.

![FIG. 11. This shows one way to build a sturdy wooden mast for temporary outdoor installations. Be sure to use 2 x 4 timbers. A height of 12 or 14 feet is about the maximum that is practical.](image)

A sketch of a temporary mast capable of supporting heavy loudspeakers is shown in Fig. 11. This mast is supported by heavy wooden braces at its base rather than by being sunk into the ground. Under ordinary conditions, such a mast may be self-supporting, but, to insure stability in high winds, wire guys should be used to steady it. These guys should
be brought to stakes permanently imbedded in the ground.

In installations such as football stadiums in which the seats rise in tiers from the ground, the loudspeakers are sometimes placed at ground level and aimed up at the seats. This is not perhaps the best system from an acoustical viewpoint, since the distance of the loudspeaker from the seat varies for each seat. However, it is one way of furnishing sound coverage without having the loudspeakers interfere with the view of the spectators.

In such an installation, the loudspeaker should be on some form of platform. A concrete block of suitable size, with one or two screws set into it to form a mounting point, is excellent for the purpose. The loudspeaker cable should be brought directly from the loudspeaker to the ground through a conduit.

If an installation is to be made in a field having covered stands, the loudspeakers can usually be secured to the roof of the stands or to the supports holding up the roof. There is nothing particularly unusual about such an installation in the technical sense. You must be sure, of course, that the speakers used will provide adequate coverage for all seats. In this as in all other outdoor installations, the loudspeaker cables should be run through conduit, both to protect them from the weather and to keep them from being cut by vandals.

All the mounting methods we have described so far are chiefly used when a considerable number of loudspeakers are to be used. Actually, for most purposes, it is better to use very few loudspeakers and concentrate them at one point if it is possible to do so. This procedure will minimize the echo effect that a listener gets from hearing the sound from two or more loudspeakers that are located at differing distances from him. Some sound engineers feel that the use of a great many loudspeakers enhances the brilliance of music, but, in the average installation, the problems created by the use of many loudspeakers more than outweigh the benefits gained by using them.

In a large outdoor installation, the use of only a few loudspeakers means that they must be very high-power units. A few types of loudspeakers of extremely high power are available; one kind, used on top of the Empire State building in New York, transmits the sound of a carillon for distances up to fifteen miles. Such a loudspeaker, which has a continuous operating capacity of 300 watts, is too powerful for any except a very large installation. However, there are smaller versions of this loudspeaker that can be used with less power.

Two or three such loudspeakers of suitable power, mounted on top of the center-field wall, can provide adequate sound coverage in a baseball field. One of the chief objections to the use of extremely high-power units is that the listeners near the loudspeakers must be subjected to an uncomfortably loud sound if distant listeners are to hear at all. However, in baseball parks, in particular, it is often possible to find a mounting place for the loudspeakers that no listeners will be very near; if this can be done, then it is perfectly practical to use very few loudspeakers of very high power.

**DISTRIBUTION LINES**

The audio lines used to feed loudspeakers and outdoor installations must be protected from the weather. Many owners of outdoor installations
have also found it necessary to protect the lines from people who maliciously or thoughtlessly cut them if they are left exposed. Cutting a cable seems to be a pointless form of destruction, but it does occur unless precautions are taken to prevent it.

Both problems can be solved by installing the cables in conduit. This conduit may or may not be buried in the ground, depending on the location of the loudspeakers. If a loudspeaker is on a mast that is standing by itself some distance from the amplifier, obviously the conduit running from it to the amplifier should be buried in the ground. If, however, the loudspeaker is mounted on the roof support of a grandstand, the conduit may be run down from the loudspeaker and underneath the grandstand to the amplifier location. In this latter case, there is no need to bury the conduit in the ground.

Rubber and lead-covered cables should be used for all outdoor audio lines. This cable, when it is enclosed in conduit, is very nearly proof against all forms of corrosion. It cannot readily be spliced, however, without introducing the possibility of corrosion at the joint. Unless you have had experience in running conduit and in using this rubber and lead-covered cable, you should have it installed by an electrician. In fact, the laws of many communities require that such work be done by a licensed electrician.

Cables that must run in the open should be buried 6 to 8 inches under the surface of the ground. You should draw an accurate map showing the locations of any buried cables so that they can be readily located in case one of them becomes defective. If possible, make sure that you do not bury a cable in any location that is apt to be dug up at any future time.

The most permanent form of wiring now available consists of rubber and lead-covered cable in a special gastight conduit that, after the installation is completed, is pumped full of nitrogen under about 80 pounds pressure. The presence of the gas prevents condensation from forming in the conduit, thus helping to preserve the cable.

![Image of a loudspeaker](Courtesy University Loudspeakers, Inc.)

The University Model MM-2 loudspeaker, a submergence-proof unit intended primarily for marine use. It is also well suited for applications involving extremely dusty conditions, since its method of construction prevents any foreign material from entering the mechanism. The loudspeaker has a flanged rim for mounting in bulkheads or walls. It drains automatically in the operating position.

Such an installation is, of course, expensive, and must be performed by a trained man; however, if an extremely long-lived installation is wanted, it may be worth while to go to the expense of using gas-filled conduit.

**MICROPHONES**

Microphones are never permanently installed out of doors. There may be,
and commonly is, one installed in the announcing booth if there is such a booth at the site of the installation, but microphones are much too delicate instruments to be exposed permanently to the weather. Therefore, in a permanent installation, you will run microphone lines to the places where microphones will be used and terminate the lines with a connector. These connectors have screw caps that permit them to be sealed weather-tight when they are not in use.

It may or may not be possible, depending on the installation, to run microphone lines to a number of points so that it will be necessary to use only a short coupling line from any microphone to the nearest permanent line. This can usually be done if the installation is in an outdoor auditorium or some other similar location where it is possible to know in advance where microphones are going to be used.

Permanent microphone lines, like permanent loudspeaker cables, should be rubber and lead-covered and installed in conduit. Again, this is a job for an electrician unless you are experienced in making such installations.

Generally speaking, dynamic microphones are best for outdoor work. They are both more rugged and more weatherproof than other kinds. It is possible to use other microphones when special requirements make it necessary; for instance, velocity mi-

crophones might be used when the best possible fidelity is wanted. As a general thing, however, it is best to plan on using dynamic microphones.

In setting up the microphones lines, you must be careful to use the microphone input of the amplifier that corresponds to the impedance of the microphones that will normally be used. Most dynamic microphones have output impedances of 200 to 250 ohms; if dynamics are to be used, then, the permanent microphone lines should be connected to the 250-ohm inputs of the amplifier. Of course, high-impedance microphones, such as crystal microphones, should not be connected to the permanent microphone lines unless the other ends of the lines are transferred to the high-impedance inputs of the amplifier.

Microphone lines should always be shielded to minimize hum and noise pick-up. It is not necessary to use shielded cable in the permanent microphone lines as long as you ground the conduit in which the lines run. It is inadvisable to run two unshielded lines in the same conduit, however, since there may be energy interchanges between them. The connecting line between the microphone and the permanent line should be shielded and should be grounded at the point where it connects to the permanent line. Usually the plug used to make the connection will complete the ground connection to the shield.
Mobile P.A. Installation

If you become a p.a. expert, very likely you will find it profitable to have a sound truck. Let’s see how you can equip a truck for use as a portable p.a. system.

The first question to settle is the kind of truck you are going to use. Most sound trucks are of the light or medium-duty panel delivery class. If you own a shop and already have a delivery truck, very likely you can convert it for use as a sound truck and still have enough room in it for deliveries too. It would be perfectly possible to use a station wagon; in fact, it would be desirable to do so, since it would be possible to ventilate the inside of the vehicle far more easily than it is when a panel truck is used.

Next is the problem of selecting the equipment to use. The fact that the equipment is to be mobile, and therefore cannot be operated from a power line, means it must be economical of power. This means that we must choose an efficient amplifier and use efficient loudspeakers. Assuming that you will not want to use a motor generator set for providing power, you must choose an amplifier that can be operated from a vibrator power supply powered by a storage battery. In most installations, the truck battery is used as a power source, although, of course, it is always possible to use a separate storage battery—perhaps installing an extra generator on the truck engine to charge it.

Since you will probably want to have sound coverage in all directions from the truck most of the time, you will want to use four loudspeakers, one mounted at each corner of the truck roof. Reflex trumpets are the most practical form of loudspeakers to use, both because they are highly efficient and because they are weather proof.

The rest of the equipment you will need to complete the sound truck installation is a microphone, a record player, and possibly a radio tuner. These are standard items, no different for mobile installation than for any other. Many mobile amplifiers are available that have record players already installed in their tops.

Now, let’s discuss the equipment and its installation in more detail.

Amplifier. The Airline 30-watt mobile amplifier is typical of those used in sound trucks. A schematic diagram is shown in Fig. 12. Notice that its power supply can operate from either a regular a.c. power line or from a 6-volt storage battery.

Aside from its power supply, this amplifier is conventional. It has two high-impedance microphone channels and two high-impedance phono channels. Each microphone channel feeds into a 6SQ7 voltage amplifier stage. The signals from these two stages are fed through individual volume controls, one for each channel, to a master volume control. Both signals are fed through this master control to the rest of the amplifier.

A potentiometer having a grounded center point is connected across the two phono input channels. As a result, a signal applied to one of the phono channels appears across half the potentiometer, and a signal applied to the other channel appears across the other half. Since there is only one slider on the potentiometer, the signal from only one phono chan-
nel can be fed to the amplifier at one time. As you rotate the slider from one end of the potentiometer to the other, the level of the signal from one phono channel will be reduced from full volume to zero; then, as rotation continues, the level of the signal from the other phono input will rise from zero to full volume. This arrangement permits smooth control of the input from the two phono channels.

A feature of this amplifier is the manner in which various output impedances are made available. As you can see from the diagram, the impedance of the complete secondary of the output transformer is 500 ohms. Taps make it possible to have 8 ohms, 4 ohms, 2.7 ohms, or 2 ohms impedance. These taps on the secondary are brought out to a 5-position speaker selector switch that can be rotated to furnish the desired impedance at the output terminals of the amplifier. At the 500-ohm position of the switch, the ends of the secondary are connected to a terminal board mounted on the rear of the amplifier. Other positions of the switch, marked 1, 2, 3, and 4 on a dial plate, connect the taps on the secondary to paralleled receptacles that are also mounted on the rear of the amplifier.

These receptacles are used when loudspeakers having 8-ohm voice coils are to be connected to the amplifier. When one such loudspeaker is to be used, leads from its voice coil should be plugged into one of these receptacles, and the speaker selector switch should be turned to position 1, thus connecting the 8-ohm tap to the receptacles. If two loudspeakers are to

The phono input signal, like the microphone input signals, is applied to the master volume control. Thus, if signals are fed in simultaneously from 2 or 3 channels, the master control can raise or lower the level of all the signals by the same amount, but cannot change the level of one signal with respect to that of another. Changes in the relative levels of the signals are controlled by the volume controls in the individual channels.

After passing through the master volume control, the signals are applied to a 6SQ7 voltage amplifier stage. Succeeding stages in the amplifier consist of another 6SQ7 voltage amplifier stage, a 6F6G driver stage, and an output stage containing two 807's in push-pull.

*Courtesy Thordarson

The 20-watt Thordarson T-31W20A-X mobile amplifier. It can be used on either 110-volt a.c. or 6-volt d.c.*
be used, each should be plugged into a receptacle and the switch set to position 2, thus connecting the 4-ohm tap to the receptacles. Similarly, if three loudspeakers are to be connected, the switch should be turned to position 3, connecting the 2.7-ohm tap to the receptacles; and, with four loudspeakers plugged in, the switch should be turned to position 4, connecting the 2-ohm tap to the receptacles.

Of course, this system can be used only with loudspeakers having 8-ohm voice coils, and then only when they are to be not more than 75 feet from the amplifier. If loudspeakers having other impedances are to be used, or if they are to be some distance away from the amplifier, the 500-ohm impedance of the amplifier should be selected by turning the selector switch to the position marked “500 Ohm.” Then the loudspeakers should be connected to the 500-ohm terminal on the back of the amplifier, using suitable matching transformers. In sound-truck use, where the lines are very short, it is quite practical to use the loudspeaker receptacles with 8-ohm loudspeakers.

Incidentally, notice that this selector switch does not permit loudspeakers to be cut in and out of the circuit. Turning the switch to any position furnishes a particular output impedance to all the receptacles, but does not cut off the power supplied to any of them. If you wish to cut out one loudspeaker, you must unplug it from the amplifier and change the setting of the selector switch to

A passenger car can be temporarily converted to a sound truck, as this was, by mounting cone loudspeakers encased in projector housings in the rear window openings. These loudspeakers are not weather proof, so they are not suitable for permanent outdoor use, but they are often satisfactory on a temporary basis. The amplifier can be in either the rear or the front seat, depending mostly on whether the driver or a passenger is to operate it. For safety, it is preferable not to have the driver do so.
This commercial sound truck is unusually well equipped with loudspeakers. The six loudspeakers mounted at the sides of the truck, three on either side, are used on most jobs. The large loudspeakers mounted fore and aft are used when extreme power is wanted.

The next lower number. If such a change is to be made, be sure the amplifier is turned off when you unplug the loudspeaker; otherwise the output stage might be damaged.

The amplifier is equipped with a standby switch for use when the amplifier is operated from a storage battery. Throwing this switch to the OFF position applies power to the filaments of the tubes, but cuts it off from the vibrator. The amplifier is then in the standby condition; when the standby switch is snapped to the ON position, power is applied to the vibrator and the amplifier is ready to operate. This arrangement permits the amplifier to be ready for instant use without drawing much power when it is not in use.

When this amplifier is operating from batteries, it draws 30 amperes. Therefore, it is a good idea to use two storage batteries to operate it to be sure of having plenty of reserve power. It would also be wise to install an extra generator on the truck engine to take care of charging these batteries.

This amplifier is supplied in two models. One is equipped with a 2-blade record changer installed in the top panel, the other has a single-play record player similarly located.

**Loudspeakers.** As we said earlier, reflex trumpets are the logical choice for use with a sound truck. Some sound trucks are equipped with only 2 loudspeakers—one pointing dead ahead and the other directly back. Since reflex trumpets have sound dispersion angles of only about 90° at most, arranging loudspeakers in this fashion will mean that no sound is projected to either side of the truck. If you want the sound to be audible on all sides of the truck, and, in most cases, you probably will, it is better to use 4 loudspeakers and mount them on the corners of the roof.
The loudspeakers may be mounted directly on the roof of the truck with machine screws passing through the mounting brackets. If you prefer not to cut holes in the truck roof, you can get a rubber-footed mounting platform resembling the luggage racks and ski racks that are used on passenger cars. These are held to the roof by suction cups and by mounting straps that hook under the rain ledge or window edge. If you mount the loudspeakers directly on the roof, make sure that the mounting holes are near some strong part of the roof, such as a roof bow. If you mount them in a part of the roof that is remote from supporting members, their weight may be enough to distort the roof metal when the truck starts or stops. Be sure to weatherproof the holes with sealing compound after the loudspeakers are installed.

If possible, the loudspeakers should be mounted so that they do not extend beyond the sides of the truck. This means that a loudspeaker mounted at one corner actually has its mounting point near the center of the truck, since reflex trumpets range from sixteen inches to twenty-nine inches in length.

Some sound truck owners find it desirable to have one trumpet pointing straight ahead. This lets the truck project a strong signal straight forward, thus attracting attention to the fact that it is coming. You may wish to do this yourself.

Reflex loudspeakers are usually equipped with 25-watt driver units. If we drive four of these units with a 30-watt amplifier, the individual loudspeakers will be supplied with only 7 1/2 watts apiece. This, of course, decreases the amount of sound that can be projected in any given direction, although it does provide uniform sound coverage in all directions. If you wish, you can use only 2 trumpets, feeding them with 15 watts each. The loudspeakers will then project the sound further, but, since you are using only two, you can cover only about 180° around the truck compared to the 360° you can cover using 4 loudspeakers. Of course, you can get both increased distance and complete angular coverage by using a more powerful amplifier. Your choice will depend mostly on what you expect to use the sound truck for. If its chief use is to be cruising the streets making announcements, it is probably best to use the 4 speakers and have the complete 360° coverage.

Installation. Fig. 13 shows one way you could arrange the inside of the truck to accommodate the equipment. The shelves shown are deep in the truck, right against the back of the driver's compartment. These shelves should be heavy wooden planks about 2 feet wide. They should be covered with heavy felt or rubber to dampen mechanical vibration. This is a convenient arrangement, but not the only possible one; you can use any arrangement you feel is best for you.

Be careful to get the wiring from the amplifier to the loudspeakers out of the way as much as possible. Run it directly up to the roof, then over to the loudspeakers—don't let it hang free in the space inside the truck, because it may get ripped loose accidentally. Be sure to waterproof the holes through which the loudspeaker cables are led inside the truck.

Remember that the inside of a closed truck can get extremely hot if the truck has been in the sun for a while. For the comfort of the operator of the equipment, and to protect the equipment from excessive heat, you must provide some means
of ventilating the truck interior if it has no windows. An electric fan or air scoops cut in the sides of the truck may be needed. Remember that any form of air intake or air circulator (such as a fan) must not create any noise within the truck, since such noise might be picked up by the microphone.

HIGH-POWER MOBILE INSTALLATION

The equipment we have just described is what is used in the usual sound truck. It is entirely adequate for a truck that is going to be used mostly to cruise streets making announcements, but it does not have the power to be heard for long distances nor to be used in addressing a large group of people.

Some sound trucks that are used for such purposes are in existence. These have a great deal more power—200 or 300 watts, in some cases—and they cannot, of course, be operated from a regular storage battery. When powers of this sort are required, it is necessary to use some form of motor-generator set as a power supply.

Often these high-power installations are made in a trailer, rather than in a truck. One reason for doing so is that these high-power mobile units are sometimes used in one lo-

FIG. 13. Suggested arrangement of equipment in a sound truck. Placing the equipment against the back of the cab in this manner uses up only about two feet of the depth of the truck body, so it will still be possible to use the truck for making deliveries.
cation for days or even weeks at a time, acting as temporary p.a. installations rather than truly mobile equipment. Such high power is seldom necessary for street-cruising work.

When several hundred watts of power are available, a variety of loudspeaker arrangements are possible. There is even one installation in which the entire front of a trailer has been converted to form a huge exponential horn. In other installations, portable loudspeakers are used that are set up on temporary masts around the truck or trailer when it is parked at a place of event. Such mobile systems are also sometimes provided with two sets of loudspeakers, one for use when high power and great coverage is desired, and the other, of medium power, for use when intense sound is not needed.

We shall not attempt to describe a typical high-power installation, because there are too few of them for any to be considered typical. The facts that you have learned about any high-power installation will help to guide you in designing such equipment if you should want to.
Lesson Questions

Be sure to number your Answer Sheet 52RH-1.

Place your Student Number on every Answer Sheet.

Send in your set of answers for this Lesson immediately after you finish them, as instructed in the Study Schedule. This will give you the greatest possible benefit from our speedy personal grading service.

1. When checking a location for a sound installation, why is it desirable to use a printed survey form rather than depend on taking notes?

2. Why is it desirable to use several small loudspeakers instead of one in an indoor installation of the sort shown in Fig. 2?

3. What are the two major advantages secured by grouping all the loudspeakers in the center of an arena?

4. What is the chief reason why it is desirable to use a preamplifier at the microphone if the distance between the microphone and the amplifier is fairly long?

5. What advantage is there in mounting loudspeakers on a sliding collar around a mast?

6. Why is it preferable to use rubber and lead-covered cable enclosed in conduit for the distribution lines is an outdoor installation?

7. What may be an objection to the use of extremely high-power loudspeakers in some outdoor installations?

8. What two reasons are there for mounting loudspeakers on top of a sound truck rather than, say, on its sides?

9. Of what value is a stand-by system like that used in the Airline 30-watt mobile amplifier in which the tube filaments are kept heated all the time?

10. If you wish to project sound to all sides of a sound truck, will it be necessary to use 2, 3, or 4 reflex trumpets to do so?
SPECIAL P.A. SYSTEMS

53RH-2

NATIONAL RADIO INSTITUTE
WASHINGTON, D. C.
ESTABLISHED 1914
For each study step, read the assigned pages first at your usual speed, then reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind. Study each other step in this same way.

☐ 1. Introduction ........................................ Pages 1-2
   The various kinds of special p.a. systems are briefly described in this introductory section.

☐ 2. Industrial Sound Systems........................................ Pages 3-10
   Here you study a high-power industrial sound system in which a Bogen E10 driver amplifier and two Bogen HO125 booster amplifiers are used.

☐ 3. Wired Hotel Systems........................................ Pages 10-13
   In this section you learn how a hotel installation that permits 4 different programs to be made available in each room is made.

☐ 4. Intercommunicators ........................................ Pages 13-20
   The types and methods of installations of “intercoms” are described in this section.

☐ 5. Specialized Sound Systems........................................ Pages 21-28
   This section contains descriptions of 3 specialized sound systems—the electric guitar amplifier, the juke box, and the home recorder.

☐ 6. Answer Lesson Questions, and Mail Your Answers to NRI.

☐ 7. Start Studying the Next Lesson.

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THERE are many specialized forms of public address systems that provide opportunities for men engaged in selling and servicing sound equipment. For example, sound systems have become common in industrial plants. Many of the earliest installations were intended only for paging (locating someone in the plant) and for making general announcements (talks, alarms, general messages, etc.). Systems of this sort, intended to reproduce voice only, are deliberately made to have poor low-frequency response. This effect is produced by using loudspeakers having short horns and amplifiers with deficient low-frequency response. As a result, such systems reproduce the human voice clearly but are not suitable for the reproduction of music.

Today, however, more and more industries are using their sound systems to entertain their workers. Sometimes this entertainment is given only at the lunch hour or for gatherings. There is, however, a growing tendency to play music continually throughout the day, because researches have indicated that, in some kinds of work, doing so will increase the output of the workers. Many plants, hitherto equipped with voice sound systems only, are experimenting with musical programs and consequently need systems having better fidelity. As a p.a. man, you may be called on to modernize and improve the response of an existing system that has proved inadequate for musical reproduction. Thus, the fact that a factory already has a sound system does not mean that it is impossible for you to get business there.

Another large field for sound equipment is in hotels. Here, there are several uses for p.a. systems: paging systems are used to locate guests; installations are used for voice and music in the dining room and in the ball room; and, in an increasing number of hotels, p.a. systems are used to provide radio programs in the guest rooms. In these last installations, master receivers are tuned to the desired stations and their outputs are fed through p.a. amplifiers to the rooms.
This arrangement is used in preference to installing individual receivers in the rooms because it is usually simpler than putting up a complex antenna system and eliminating interference. A somewhat similar system is installed in some of the more modern hospitals.

These major installations can be very profitable to a p.a. man, but, of course, they are not everyday occurrences. At the other end of the scale in complexity is the intercommunicator (usually called intercom), which is a low-cost device that is usually very simple to install. Although the profit from the sale and installation of a pair of intercoms is not great, many a p.a. specialist finds these devices are an important source of income because of their wide usefulness and consequent ready sale.

The typical intercom is a unit in which the speaker may be used as a microphone by throwing a switch. This “microphone” is connected to the input of an amplifier of low power, which is used to operate another speaker or several other speakers a fairly short distance away. Most intercom systems are intended for person-to-person communication over distances of not much more than 100 or 200 feet. Some intercoms work only one way—one station, called the master station, can talk to several others, but they cannot talk back. In others, the remote stations can answer; in fact, in the most flexible systems, any station can talk with any other station and sometimes with a group of other stations.

The coin-operated phonographs (juke boxes) that are found in many restaurants and drug stores can also be considered to be specialized forms of p.a. systems. Essentially, these are just record players that operate automatically through a small p.a. system. In localities where the servicing of these units is not handled by the dealer renting or selling them, the p.a. man may get not only service contracts but also installation and remodeling contracts as well.

Band instruments like the electric guitar are specialized p.a. applications that sound specialists are often called on to service. The electric guitar is essentially a stringed instrument with an electrical pickup through which the vibrations of the strings are converted into electrical energy that is fed through a small amplifier to a speaker.

Finally, the home recorder—a device with which anyone can make his own records—can also generally be used as a small p.a. system. This, too, is something a sound expert may be called upon to service.

Now let's take up these specialized applications in more detail.
Industrial Sound Systems

Let's see what problems are involved in furnishing high-power sound to a number of points on a large production line where the noise of machinery, tools, and shouting men must be overcome. We shall assume we are dealing with a conveyor-line factory for heavy units such as automobiles, refrigerators, washing machines, or stoves.

Let's take a moment to consider the problem of noise level. As you know, sound levels are rated in db. The threshold of audibility (0 db) is the level at which no sound can be heard. The soft rustle of leaves on a quiet spring day may be 10 db. Quiet conversation in an average office may be 25 db. Ordinary street traffic may be 75 db, heavy traffic may be 90 db, and the level of noise in a plant like the one we are considering may range from 70 to 85 db at various points.

This is a very high noise level to overcome with a sound system. As a matter of fact, if the noise level were somewhat higher—say over 100 db—it would be impractical to use a sound system at all, because the sound output would have to be so high that it would be distressing. (Sound actually becomes painful at about the 120-db level.)

You can see, then, that a high-output sound system is needed in a
plant of the sort we are discussing. 200 watts is not too much, even though the area of the factory is not very great.

A 200-watt amplifier can be secured on special order, but the installation can be made considerably less expensive by using smaller standard units. In this case, you can use two booster amplifiers rated at 125 watts each. The combined power of the amplifiers will then be 250 watts, giving you a margin of 50 watts to take care of line losses and to provide extra power that will be useful if some change in the factory makes it necessary to have more output.

The Bogen HO125 booster amplifier is suitable for use in this system. The schematic of this amplifier is shown in Fig. 2. Let's discuss its details.

**BOGEN HO125**

The output stage in this amplifier consists of two 807 tubes connected in push-pull. This stage operates in class B. Driving power for the grids of the 807's is furnished by a pair of 6SN7GT tubes. These latter tubes are normally dual triodes, but in this use, the plates, grids, and cathodes in each tube are connected in parallel so that the tubes act as single triodes. This arrangement permits the tubes to handle twice the power that a single section can. A 6SL7GT, also a dual triode, performs a double function in

![Schematic Diagram](image)

**FIG. 2. Schematic diagram of Bogen HO125 amplifier.**
this circuit. One half of the tube acts as a voltage amplifier, the other as a phase inverter. Plate power for the 807's is provided by a 5R4GY. A 5Y3GT provides the power needed for the other tubes and for the screen grids of the 807's.

This amplifier has several unusual features. One is the use of a third 6SN7GT as a regulator of the screensupply voltage. Its regulating action keeps the screen voltage at the proper value to produce correct plate dissipation in the 807 output tubes at all times, whether the signal input is large or small.

As you know, the screen current of any tetrode tube decreases when the signal applied to the control grid decreases. If the voltage applied to the screen grids of the 807 tubes were not regulated, it would increase when the voltage applied to the control grids decreased. This would occur because the regulation of the power supply is not perfect: its output voltage increases when the current drained from it decreases, and vice versa. Therefore, any decrease in the screen grid current would cause an increase in the power supply output voltage. The screen voltage would then increase as the grid voltage decreased, an effect that would tend to maintain the plate current relatively steady in spite of the control grid variations. This, of course, would cause distortion.

The 6SN7GT voltage regulator tube is connected, as Fig. 2 shows, across the screen supply (since it is connected between B+ of the 5Y3GT power supply and ground). The voltage regulator circuit, redrawn for greater clarity, is shown in Fig. 3. Notice that the grids of the tube are connected to a voltage divider made up of a 47,000-ohm and a 470,000-ohm resistor. One end of this divider is connected to B+, and the other to the negative end of a source of bias voltage \( V_1 \). The bias applied to the grids of the voltage regulator tube at any time is equal to \( V_1 \) minus the drop \( V_2 \) across the 47,000-ohm resistor. The latter drop is equal to one-eleventh of the total voltage across the voltage divider. If B+ increases, the total voltage across the divider will also increase; consequently, \( V_2 \) will also increase. Since the bias applied to the grids of the 6SN7GT is equal to \( V_1 \) minus \( V_2 \), an increase in \( V_2 \) means that the bias will decrease. This will allow the plate current of the tube to rise, creating a greater drain on the power supply and therefore lowering the B+ voltage it can furnish. Thus, the action of the regulator tube is to maintain a fairly constant current drain on the power supply regardless of variations in the signal. The constant current drain keeps the output voltage of the power supply constant, and therefore maintains the screen grid voltage at a fixed value.
Another unusual feature of this amplifier is the coupling between the driver tubes and the output stage. Notice that the plates of the two 6SN7GT driver tubes are connected directly to the B voltage supply. The loads for these tubes consist in each case of a resistor in the cathode circuit plus one-half of the coil connected to the control grids of the 807 tubes. (To see this, trace the cathode circuits; each is completed to ground for a.c. through the center tap of the coil.). The signal voltage output of each 6SN7GT driver tube is developed across the cathode load, and the part of the signal voltage that is developed across the halves of the coil is applied to the grids of the 807's.

The arrangement just described (called a cathode-follower coupling) is used because it provides direct, low-impedance coupling to the output stage, which, since it operates in class B, has a very low input impedance. Generally a stepdown transformer is used to couple to a class B stage. This method of coupling eliminates the need for such a transformer, which would be very expensive.

Another feature of this amplifier is that the application of high voltage to the plates can be controlled remotely if desired. This is made possible by a relay assembly incorporated in the power supply. When this relay is plugged into the relay socket shown in the schematic diagram, high voltage will not be applied to the tubes until the relay is actuated, which will not occur until a connection is made between two terminals on the relay box. A remote switch can be used to connect these terminals. If this remote switch is left open, the filaments of the tubes in the amplifier will be heated; thus, the amplifier will be ready to go into operation as soon as the switch is closed. This is a completely optional feature; the relay can be disconnected entirely, and a shorting plug can be plugged into the relay socket, in which case high voltage is applied to the amplifier tubes whenever the master power switch is closed.

The output transformer of this amplifier has four connections. One is ground, the other three are marked, respectively, "70V," "90Ω," and "140V." The 70V and 140V taps are called "constant-voltage" taps, so named because when the amplifier is delivering no more than its rated output, voltages between these taps and ground are essentially constant if the proper matching transformers are used to connect the loudspeakers to the taps. The 90Ω tap is used when the amplifier is connected to a high-output, multi-driver loudspeaker. Most such loudspeakers have input impedances of 90 ohms.

Loudspeakers may be connected in parallel to either of the constant voltage taps. With either tap, the correct impedance of the matching transformer is equal to $E^2 \div P$, where $E^2$ is the square of the tap voltage (which is approximately 20,000 for the 140-volt tap and approximately 5000 for the 70-volt tap), and P is the power to be applied to the loudspeaker. For example, if we want to drive a 25-watt loudspeaker driver from the 140V tap, the matching transformer must have a primary impedance of $E^2 \div P = 20,000 \div 25 = 800$ ohms. In other words, the primary impedance of the matching transformer used to couple
a 25-watt loudspeaker to the 140V tap must be 800 ohms. If two 25-watt loudspeakers are to be fully excited, and are connected to this output tap, the primary impedance of each matching transformer must be 800 ohms.

If we wish to connect a 25-watt loudspeaker to the 70V tap, the necessary primary impedance of the matching transformer must be \( E^2 \div P = 5000 \div 25 = 200 \) ohms. Again, transformers having this same primary impedance should be used when several loudspeakers of the same power are to be connected to the 70V tap.

Whether you choose the 140V tap or the 70V tap depends upon which permits the more readily available matching transformers to be used. In general, you should use the 70V tap when low power is to be taken, and the 140V tap when you are going to draw high power.

Notice that this amplifier is not provided with a tone control. The reason is that it is intended for use only as a booster amplifier driven by another amplifier; control of the tone takes place in the other amplifier.

This amplifier may be secured with either a low- or high-impedance input. Which you use depends on the driver amplifier you are going to use; with the high-impedance model, a driver amplifier having a 100,000-ohm output impedance is required; with the low-impedance model, the driver amplifier should have a 500-ohm output impedance. The schematic diagram in Fig. 2 shows the high-impedance type. The box at the lower left of the diagram shows the input section of the low-impedance model.

Fig. 4 shows the input circuit that should be used to couple a 500-ohm driver amplifier to this booster amplifier. The 500-ohm resistor shown in this diagram must be able to dissipate the full output of the driver amplifier. Resistors \( R_a \) and \( R_b \) must have a ratio such that approximately 5 volts will be developed across \( R_b \) when the driver amplifier is delivering about two-thirds of its rated output across the 500-ohm load. The combined value of \( R_a \) and \( R_b \) must be great enough so that no more than \( \frac{1}{4} \) watt will be dissipated in either \( R_a \) or \( R_b \) when the driver amplifier is delivering its full rated output to the load. Under these conditions, \( \frac{1}{2} \)-watt resistors may be used for \( R_a \) and \( R_b \).

Values of 12,000 ohms for \( R_a \) and 1000 ohms for \( R_b \) are recommended by the manufacturer when a Bogen E10 amplifier is used as a driver. Let's see what this driver amplifier is like.

**BOGEN E10**

A schematic diagram of the Bogen E10 amplifier is shown in Fig. 5. This amplifier has two input channels, one for a high-impedance microphone and the other for a phonograph. The microphone channel feeds into one half of a 6SC7 dual triode. The output signal of this half of the tube is then fed to the other half of the tube. The output of this section is then fed to a combination voltage amplifier and phase inverter stage, also using a 6SC7, the output of which is fed to a pair of 6V6GT output tubes connected in push-pull.
FIG. 5. Schematic diagram of Bogen E10 amplifier.

Courtesy David Bogen Co., Inc.
The phono input feeds into the grid of the second section of the first 6SC7, so the two input signals can be mixed in this tube. The volume level of each channel is controlled by a \( \frac{1}{2} \)-megohm potentiometer. The two input signals can be mixed in any desired proportion by adjusting the potentiometers. When only the microphone is to be used, the volume control in the phono channel should be set at zero; conversely, when only the phono is to be used, the volume control of the microphone channel should be set at zero.

The amplifier has a simple tone control, consisting of a \( 0.006 \text{-mfd.} \) condenser in series with a \( \frac{1}{2} \)-megohm variable resistor across the plate load of the amplifier section of the second 6SC7 stage.

The output of this amplifier is developed across two speaker sockets. A 5-hole impedance-selector socket is connected to taps on the output transformer of the amplifier. You can select any one of 4 output impedances—4 ohms, 8 ohms, 15 ohms, and 500 ohms—by inserting a connector in the proper socket. The diagram shows how this should be done.

When you are using the E10 amplifier as a driver for the HO125 amplifier, you should plug the connector into socket hole 5, thus connecting the 500-ohm tap across the speaker sockets. Insert pins in the speaker socket to make connections to the HO125 amplifier, using the connecting circuit shown in Fig. 4.

**INSTALLATION**

It is perfectly possible to connect the inputs and outputs of the two booster amplifiers in parallel, producing, in effect, a 250-watt amplifier. In one important respect, however, it is better to use parallel inputs and separate outputs for this installation. By doing so, you will have two sound systems, each consisting of one booster amplifier and its associated loudspeakers. Then, if either booster amplifier becomes defective, you can find out which one is at fault by determining which group of speakers does not operate.

A block diagram of the installation using this arrangement is shown in Fig. 6.

Eight loudspeakers, each with a 25-watt driver, are shown in this sketch. If desired, one more speaker could be

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**FIG. 6.** Block diagram of factory p.a. installation.
added to each line. Since high efficiency of reproduction is needed to overcome the high noise level in this plant, reflex trumpets should be used for the loudspeakers. You are already familiar with the use and installation of these trumpets from earlier Lessons. In a factory of this sort, it will probably be possible to mount the trumpets on girders used to support the roof.

Because of the relatively long runs, and the high power to be used, heavy wire should be used in the line. No. 14 or even No. 12 wire should be chosen and it should be enclosed in conduit or BX. Most likely, the local safety regulations will make it necessary to have the wiring installed by a licensed electrician, who may be a plant employee.

The Bogen E10 driver amplifier we have described is designed for use with a high-impedance microphone. Either a crystal microphone or a high-impedance dynamic microphone can be used. With either type, the microphone cable should be kept as short as possible and should be well shielded.

### Wired Hotel Systems

The ordinary sound systems used in hotel dining rooms and ball rooms are basically the same as those systems that you have already studied. Intercommunicators are covered later in this Lesson. Right now, let’s take up sound systems that carry entertainment to the guest rooms.

A block diagram of a typical system of this kind is shown in Fig. 7. Most generally three or four programs are made available to each guest room. A speaker, a volume control, and a selector are located in the guest room, making it possible for the guest to select the desired program and regulate the volume.

Because different programs are available, there must be a separate channel and a separate amplifier for each. These programs are fed into the amplifiers by radio tuners. Each tuner is normally adjusted to receive a local station and then remains fixed in its tuning.

As shown in Fig. 7, it is standard practice to provide an extra tuner and an extra amplifier for emergency use to replace any unit that happens to fail. Sometimes one or two channels are fed from a phonograph instead of a radio tuner. Sometimes, also, provision is made for plugging in a microphone so that announcements can be made over the system.

Let’s see what problems we would meet in setting up such a hotel sound system.

The amplifier in each channel must, of course, be capable of supplying the power needed to operate all of the speakers that can be connected to it. In other words, each channel amplifier must have enough power to operate every speaker in the guest rooms, even though it is unlikely that every one will be connected to the same channel at the same time.

To take a typical example, let’s suppose that the hotel is a fairly small
one having 90 rooms that are to be supplied with sound. Allowing one watt per room, 90 watts is needed for each channel. This is a minimum; to allow for the usual decrease in output caused by aging of the amplifier components, we should use a 100-watt amplifier for each channel.

Once the problem of how big an amplifier is needed is settled, designing the system becomes quite simple as far as the input and amplification sections of it are concerned. We select a suitable tuner and amplifier and make provisions for coupling each tuner to a master antenna system and each amplifier to one of the channels going to the rooms. To make the system flexible and to make it possible to substitute the stand-by equipment on any channel, we provide jacks at the outputs of the tuners and at the inputs of the amplifiers so that patch cords may be used to make the connection quickly between any tuner and any desired amplifier. A somewhat similar arrangement may be used between the outputs of the amplifiers and the wires going to the rooms.

So far, as you can see, there is nothing particularly unusual about the installation. We meet a new problem, however, when we come to plan the connections to the loudspeakers in the room. The problem is that we have no way of knowing how many loudspeakers are going to be connected to any channel at any time. This means that we cannot simply provide a switch for connecting each loudspeaker to a channel. If we did, the impedance of each channel would vary each time a loudspeaker was connected to it or disconnected from it, with the result that the volume would be constantly varying in level and the tone quality would be adversely affected. If it happened that only a few loudspeakers were connected to one channel, they would be heavily overloaded and probably damaged.

Instead, we must use a switching arrangement that provides a constant load for each channel. Such a switch-

**FIG. 7. Block diagram of typical hotel p.a. system.**
ing system must connect a load resistor to the channel whenever the loudspeaker is disconnected from it, thus keeping each channel fully loaded at all times.

The basic principle of the switching system is shown in Fig. 8. When switch S is thrown to position 1, the primary of line-coupling transformer T is connected to the line, and the signal is fed through the transformer and

![FIG. 8. Basic design of a constant-impedance switching system.](image)

through the volume-control pad to the loudspeaker. The dummy load resistor R is not used when the switch is in this position.

When we wish to disconnect the loudspeaker from this particular line, switch S is thrown to position 2. Resistor R is then connected to the line in place of the transformer primary. Resistor R has a resistance equal to the impedance offered by the primary of transformer T, so the line is not affected by the switch-over and for practical purposes is feeding into a constant impedance.

The transformer impedance, which determines the value of resistor R, is in turn determined by the number of loudspeakers on each line and by the impedance of the line. Usually the wattage rating of resistor R is somewhat above the value calculated for the loudspeaker so that it can safely handle the necessary power.

The switching system shown in Fig. 8 is all right for a single channel, but a more elaborate system is necessary where there are more channels. For example, if there are four channels, it is necessary for the selector to have an off position in which resistances are connected to all four lines and the loudspeaker coupling transformer is disconnected from all of them. Then, at the position for line No. 1, the first resistor must be cut out of the circuit and the transformer put in its place. At position 2, the resistor on line 2 must be removed, the transformer put in its place, and the resistor on line 1 must be reconnected to that line. Similarly, at positions 3 and 4, the transformer must be substituted for the resistances on those lines and the switching system must reconnect the resistors to the other lines.

There are several different switches made specifically for this purpose. The exact manner of operation will depend upon the switch used, so you should get from the manufacturer of the selector you install, a schematic showing the proper connections.

Some of the selectors are 2-pole switches having a very elaborate band-type switch arrangement for the second pole. Others, like the one shown in Fig. 9, have a pole for each line. This switch is a 4-pole, 5-position switch. At position O (the off position) the resistances \( R_1, R_2, R_3, \) and \( R_4 \) are connected across lines A, B, C, and D respectively. (Notice that the shielding on the line is used as a return circuit for these resistors and for the loudspeaker matching transformer.)

When the switches are moved to position 1, the transformer T is connected by \( S_1 \) between line A and the shield, but all other lines are still connected through their matching re-
FIG. 9. Multi-channel constant-impedance switching system.

sisitors. Similarly, at positions 2, 3, and 4, the transformer is connected by switches $S_2$, $S_3$, and $S_4$ to B, C, and D in turn.

To keep the line loss at a minimum, and to minimize the possibility of a line defect knocking out of commis-

sion too many of the guest-room loud-speakers, it is common practice to split up the distribution systems. In other words, each of the channels A, B, C, and D is broken into sections. For each channel, a small group of the rooms are wired in parallel to one line, which is then run to the amplifier. At the amplifier, this line is paralleled with other similar lines, each of which is connected to a group of rooms. The parallel combination of these lines then constitutes one channel.

In our example, assuming 90 rooms, a logical arrangement would be to split the rooms into 9 or 10 groups, whatever works out more satisfactorily for impedance-matching purposes. The simplest possible manner of dividing the rooms should be followed. If the hotel has 9 or 10 floors, lines could be run straight down, connecting to one room on each floor. In some cases, it may be preferable to connect all the rooms on one floor to a single line. The exact arrangement will depend upon the layout of the hotel. You should choose the one that will use the least cable and will involve you in the fewest difficulties in installation.

Intercommunicators

All the p.a. systems you have studied so far have been primarily designed to communicate with large groups of people. However, there is a large and rapidly growing field in person-to-person communication systems. Strictly speaking, that is not public address—it is more like an amplified telephone system. However, the p.a. man logically gets the job of installing and servicing such equipment, because, except for size and power, an intercom is basically the same microphone-amplifier-loudspeaker combination found in any p.a. system.

Intercoms sell readily because they are extremely useful in many applica-
tive at the master station can get in touch with any department of his business at the flick of a finger—he doesn’t have to wait for telephone connections to be made or undergo the annoyance of tied-up lines.

As shown in Figs. 11B and 11C, it is also possible for other sections of the same office to communicate with each other. Communications between the stock room and the purchasing office or accounting office are frequently of importance. Intercoms here permit a stock clerk to move about among the shelves and search for the required items or call out the inventory,
into two basic kinds—the direct-wire type and the wireless type. The direct-wire type utilizes audio lines between the units, and is completely an audio system. In the wireless kind, the sound modulates a local oscillator. The signal then goes out as an r.f. wave over the power lines to the receiving unit, where it is detected and amplified to operate a loudspeaker.

In either system, sound is picked up and must be reproduced through a loudspeaker. Intercoms are inexpensive devices because it was discovered that small p.m. loudspeakers will also serve as satisfactory microphones. You will recall that the dynamic microphone contains a diaphragm that, when subjected to sound pressure, drives a voice coil that is in a magnetic field. A loudspeaker contains the same items, except that the diaphragm is in the form of a cone. Therefore, a small p.m. loudspeaker that is used for voice reproduction can be used as a microphone if a switching system is provided to connect the loudspeaker to the input of

![Fig. 12. Intercom arrangement in a restaurant.](image)

without having to go back to a fixed position, as he would if a telephone were used instead. The ability of intercoms to pick up sounds over distances of 10 or 20 feet is very helpful in applications of this sort.

Other typical uses are shown in Figs. 12 and 13. In a small lunchroom, an intercom enables the counter man to give orders to the cook without having to yell, and the cook can hear instructions anywhere in the kitchen. The installation shown in Fig. 13 lets a repair man at a remote location communicate easily with the store counter.

These are only a few basic uses—many more similar applications can be found. The literature of the manufacturers is full of suggestions.

Intercom systems can be divided

![Fig. 13. Intercom arrangement in a service shop.](image)
the amplifier for use as a microphone, and to the output for listening.

This change-over is accomplished by means of a "talk-listen" switch. On most intercoms, this switch is held in the "listen" position by a spring so that the station can hear any other station that may be calling it. When the operator at a station wishes to talk, he presses and holds down the talk-listen switch, thus connecting his loudspeaker for use as a microphone.

To learn more about this switching system, let's now study the basic intercoms.

**WIRELESS TYPES**

Fig. 14 shows a typical wireless intercom. This is a simple type used for communication with one or more remote stations that operate on the proper frequency. The controls on the front panel are a volume control and a talk-listen switch. As we said, the talk-listen switch remains in the listen position until the operator desires to speak. In this particular instrument, this switch is a 5-pole, 2-position switch.

The schematic diagram of this instrument is shown in Fig. 15. A standard a.c.-d.c. power supply is used.

As you learned earlier, the signal for such instruments travels over the power line. The intercom can pick up a signal on the same frequency that is used for transmitting. Let us suppose that the carrier frequency is 100 kc. If any such signal comes over the power line, it will pass through coupling condenser $C_{11}$ and through the volume control $R_9$ to one section of the r.f. transformer $T_1$. From here, the signal is fed to the oscillator-detector tube. When the switches are in the listen position, this circuit will not oscillate, because the plate voltage is cut off—instead the control grid and cathode of the tube act as a diode rectifier. The resulting audio signal developed across detector load resistor $R_8$ is then passed through from point 2 to point 3 on the talk-listen switch assembly, and sent through coupling condenser $C_1$ to the grid of the first audio stage, which uses a 75 tube. This stage is resistance-coupled to a 43 power output tube, from which the signal goes to the loudspeaker.

If the operator desires to talk, he depresses the operating switch to the talk position. This disconnects the loudspeaker from transformer $T_2$ and connects it to the primary of transformer $T_3$. This transformer is now connected to the grid of the 75 tube through condenser $C_1$, because the talk-listen switch now connects terminals 3 and 4 together. The sounds are amplified by the 75 and passed on to the 43 tube, which now acts as a modulator on the oscillator. The other 43 tube now oscillates because it has plate voltage, and because clos-
ing positions 1 and 2 of the talk-listen switch has changed the grid resistance and thus produced a bias that will permit oscillation.

Plate voltage is applied to the oscillator through $L_1$ and through the primary of transformer $T_2$. Any audio voltage appearing across the primary of transformer $T_2$ as a result of someone's speaking into the microphone is in series with the plate supply voltage, and hence modulates the oscillator. The signal is transferred through transformer $T_1$ from the oscillator to the power line. It can now be picked up by any similar receiving unit that is tuned to the same frequency and has its switch at this moment in the listening position.

These instruments are quite easy to install—all you need to do is to plug them into a power outlet that is on the same power line as the receiving unit. There are no wires to run around in the building, and, if you should want to move the unit, all you have to do is to pick it up, move it, and plug it again into a power outlet.

These units would appear to be the ideal type, but they do have their limitations. To begin with, trouble is sometimes experienced if the transmitting and receiving unit are not on the same power-line branch. Sometimes it is not possible to get sufficient signal through the electrical wiring of the building. In addition, defects in the electrical wiring may cause excessive noise or cross modulation. This latter effect will produce mixing of signals having different carrier frequencies.

A further limitation is that a transmitter can be heard by all receivers.

**FIG. 15. Schematic diagram of wireless intercom.**

*Courtesy RCA*
tuned to its frequency. If there is only one receiver, everything is all right. If there are several, however, all the receivers hear all messages, even those not intended for them.

The only way out of this is to use different carrier frequencies. Doing so permits a setup like that shown in Fig. 16, in which three different systems are used in one office without excessive interference among them. However, this system is limited in its usefulness; the executive cannot call the bookkeeper, for example, unless some means is provided for changing the frequency of his unit. If the system must be flexible enough to permit the master station to call just one station if desired and also to call all or several stations, a wireless intercom cannot usually be used. Instead, a wired system must be set up.

We'll discuss wired intercoms in just a moment. First, however, there is another type of wireless intercom that should be mentioned. This intercom uses separate transmitters and receivers. These are single purpose units: the transmitter can be used only for transmission, and the receiver only for reception. Obviously, such an intercom can be used only in applications in which one-way conversation is all that is needed. The schematic diagrams of a typical transmitter and a typical receiver are shown in Fig. 17.

**WIRED SYSTEMS**

In wired systems, audio lines are used between the intercom stations. Each station consists of an audio amplifier with its accompanying power supply and a loudspeaker that can be connected by a talk-listen switch either to the output of the amplifier for use as a loudspeaker or to the input of the amplifier through an appropriate coupling transformer for use as a microphone.

A unit of this kind that has a selector system for picking out the station to which one wishes to talk is known as a "master" unit. If it does not have a selector, it is known as a "remote" or "slave" unit. A remote unit need not have an amplifier if it is at a reasonable distance from the master unit and is to communicate only with the master. In other words, if a remote unit is to communicate with the master station but not with any other remote unit, it can be just a loudspeaker feeding through a low-impedance cable to the master unit.

An example of a wiring of a system of this kind is shown in Fig. 18. Here, the talk-listen switch on the master and on each of the remotes is normally held in the listen position. The wiring arrangement is such that the master station can at all times hear any or all of the remote stations that may call it. The selector switch of the master station can be set to let the master talk to one particular remote or to all of them.

You can call the master station
from any of the remotes by holding
the talk-listen switch to the talk posi-
tion. You cannot, however, call any
of the other remotes. When the talk-
listen switch from a remote is in the
normal listen position, it can receive a
call from the master when the selector
switch at the master is turned to the
proper position.

If it is desired to have the remote

FIG. 17. Schematic diagrams of wireless intercom receiver and transmitter units.
FIG. 18. Schematic diagram of typical master wired intercom.

stations be able to communicate with each other without having to call the master, each must have its own amplifier and its own selector switch. In other words, a system in which each station can call any other station must be essentially a collection of independent master units.

An all-master system must also be used if the stations are at very considerable distances from one another. In this case, it is necessary to use master units to have enough amplification to get the signals through.

A third possibility in an intercom system is a combination of master and remote units. In an arrangement of this sort, each master can call each other master, but each remote can call only the master to which it is connected.

You can see that the wired units offer more flexible arrangements than the wireless type—of course, at the additional expense of installing audio lines. In addition, the wired types are more free from power line noises and hum troubles, which are frequently encountered with the wireless type.
Specialized Sound Systems

There are three specialized uses for p.a. equipment that do not fall into any of the categories you have studied so far. One of these is in the amplification of individual solo instruments in bands; another is in the familiar coin-operated phonographs or juke boxes; and the third is in the home recorder.

The electric guitar is an example of the first of these specialized p.a. systems. The original model of this instrument consisted of a standard guitar with a microphone attached to the sounding board. More recently, two electronic types have been developed; these are without sounding boxes and depend entirely on electrical pickup for sound output. Let's study both.

ELECTRIC GUITAR

There are several different makes of electric guitar amplifiers on the market. Each of them uses a circuit resembling the one shown in Fig. 19, which is a schematic diagram of the Gibson EH-150 guitar amplifier. The differences between this and other amplifiers are principally in the arrangement of parts in the chassis, cabinet construction, and other minor details.

This amplifier has two input channels, one marked "Instruments" and the other marked "Microphone" on the diagram. The Instruments input channel, into the jacks of which either one or two electric guitar pickups can be plugged, goes to the grid of a 6SQ7 voltage amplifier. The output signal of this stage is fed to a 6N7 voltage

![Diagram of Gibson EH-150 guitar amplifier](https://example.com/gibson_eh150_diagram.png)

FIG. 19. Schematic diagram of Gibson EH-150 guitar amplifier.

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amplifier and phase inverter stage, which drives the grids of the power output stage. This latter stage contains a pair of 6L6's in push-pull.

The Microphone channel is provided for use in making announcements and so forth. A signal fed into the microphone through this channel is applied to a 6SQ7 pre-amplifier stage that is used to bring the microphone output up to the same level as the signal from the guitar pickup. It is fed from this stage to another 6SQ7 voltage amplifier stage that is in parallel with the similar 6SQ7 stage used in the Instruments channel. The outputs of these two parallel 6SQ7 tubes are mixed by being applied to a common load.

The tone control of the amplifier is located in the grid circuit of the 6N7 stage. It consists of a .0075-mfd. condenser shunting a series combination of a .1-mfd. condenser and a 1-megohm resistor. When the 1-megohm resistor is set at its maximum resistance, most of the signal current is passed through the .0075-mfd. condenser; as a result, the response is high-pitched. Reducing the resistance of the variable resistor allows more signal current to pass through the .1-mfd. condenser, with the result that the tone of the amplifier deepens.

Each channel has a volume control consisting of a 500,000-ohm potentiometer, the variable arm of which is connected to the grid of one of the two paralleled 6SQ7 stages.

Since a microphone channel is provided in this equipment, it may be used as a small public address amplifier in small dance halls, night clubs, and similar places where great volume is not necessary. A typical set-up is shown in Fig. 20. To reduce acoustical feedback and prevent howling, the cardioid microphone is placed somewhat behind the front edge of the speaker.

The sketch in Fig. 20 shows the amplifier and loudspeaker in one unit. For ease in transportation, they are usually built into one luggage-style case. However, usually the case can be split into two parts when the equipment is set up for use, one part then housing the amplifier and power supply and the other part housing the loudspeaker. This is a desirable arrangement; if the amplifier and loudspeaker are in one case, the sound waves caused by the vibration of the speaker cone may cause physical vibration of the tubes of the amplifier, producing howling, noise, or distortion.

Usually the loudspeaker used in this equipment is placed nearer one side or the other of the stage, not in the exact center. This is done to allow room in the center of the stage, which is usually the place occupied by the vocalist or other entertainer. If the room is reasonably small, the amplifier will drive the loudspeaker with sufficient power to produce adequate
distribution of the sound waves all over the room. It is usually necessary to experiment somewhat with the position of the speaker to determine the direction in which it should be aimed to give the best results in tone quality and intelligibility of the sound output of the equipment.

THE JUKE BOX

Juke boxes (coin-operated phonographs) are a familiar sight all over the country. Like all electronic devices, they require servicing at least occasionally, so it is worth our while to spend a few minutes now to learn something about what is inside them. We shall confine our attention to the juke box amplifiers, which are usually simple and straightforward in design and construction. The mechanical systems of these record players are too varied and intricate to permit our studying them here. Fortunately, the mechanical systems do not often get out of order. If one does, you may find the information you need to fix it in the special service manual put out by the company that made the juke box. If you are working for a company that owns a string of boxes, it will undoubtedly have such manuals for all the types it owns. If you are an independent serviceman, you may be able to secure these manuals from the manufacturers—or the person owning the box may have one.

In a typical juke box, a coin deposited in an electro-mechanical push-button selector system permits selection of one or more records. The pushbuttons may be on the front of the juke box or on remote control boxes distributed through a room, at each table along a wall, or in each booth in a restaurant. The mechanical system causes the proper record to be selected and the pickup to swing in place, and turns on the amplifier. In some types, the amplifier warm-up period is eliminated by running the filaments of the tubes continually and merely switching on the plate supply. In others, the warm-up is shortened by momentarily applying a higher-than-normal filament voltage, then switching over to normal working filament voltage.

There are a number of different types of amplifiers available. In Fig. 21, a simple class B amplifier is shown. The signal voltage provided by a crystal pickup is applied directly to the grid of the 6J5. This tube functions as a standard class A voltage amplifier and is coupled through a step-down transformer to the grid circuits of a 6N7 duplex triode connected for push-pull operation. This tube is designed to work with zero grid bias for no signal input conditions. When grid excitation is furnished, the grids go positive on alternate half-cycles. The power sensitivity and power output are good, but the distortion is higher than for class A operation.

The circuit of a class A amplifier is shown in Fig. 22. A crystal pickup supplies signal voltage to the grid of a 6C5 through a tone-compensating network and a volume control.

The 6C5 supplies signal voltage to the tapped choke through a .02-mfd. condenser. The choke permits push-pull operation of the output stage without the necessity of using a phase inverter. Direct-coupled 6B5 output tubes are used. The tubes operate without external bias, but the grid of the input triode does not draw cur-
rent because a bias voltage for this grid is set up within the tube.

In Fig. 23, a more modern juke box amplifier is shown. The crystal pickup works into a 6J5G voltage amplifier through a special constant-impedance volume control. A tone control of the series type is part of the plate circuit of this stage. Choke-impedance coupling is used to permit push-pull operation of the 6L6G tubes in class A.

As you can see from the diagrams, both this and the preceding amplifier have provision for plugging in an auxiliary p.m. loudspeaker. These are sometimes used when the juke box is installed in a very large room.

The equipment we have described so far uses crystal pickups. In many juke boxes, such as the Wurlitzer 750 shown in Fig. 24, magnetic pickups are used. The magnetic pickup is
coupled through a tone-compensating network to the two-section volume control, which feeds into a 6J5 voltage amplifier. This tube works into a tone-control system, from which the signal goes to a 6SC7 functioning as a low-resistance triode, both triode sections of the tube working together in parallel. The 6SC7 supplies signal power through a choke-impedance coupling to the power output stage, voltage of 9.8 volts is applied to the amplifier filaments at first. With this above-normal voltage applied, the tubes warm up rapidly. When the filament current rises to the normal working level, a relay is energized and operates; its armature then acts as a transfer switch, disconnecting the 9.8-volt winding from the filament supply circuit and connecting the normal 6.3-volt winding in its place.

![Schematic diagram of Rock-Ola Model H juke-box amplifier.](image)

**FIG. 23.** Schematic diagram of Rock-Ola Model H juke-box amplifier.

which contains two 6L6G’s connected in push-pull. This power output stage is considered to operate in class AB; actually the operation is class A at low and moderate volume levels, approaching class B only at high volume levels.

This amplifier contains an arrangement that cuts the warm-up time appreciably. With the amplifier off and the primary circuit of the power transformer opened by a power switch, the 9.8-volt winding on the power transformer is connected to the amplifier tube filaments. No voltage is applied, of course, since the primary is open. When the equipment is turned on, a

**HOME RECORDERS**

The home recorder is another form of special p.a. system that the radio serviceman is sometimes called upon to repair. Such a recorder can be used both to make records and to play them. The recording section consists of a microphone, an amplifier, a cutting head, and a phonograph turntable. The playback section consists of the same phonograph turntable, a pickup head, the same audio amplifier, and a loudspeaker.

The general arrangement of these components in a recorder is shown in Fig. 25. As you can see from this block diagram, microphone signals are
fed through the amplifier and applied to the recording head, which is more or less the reverse of an ordinary phonograph pickup. In other words, electrical signals applied to the recording head cause it to move a special kind of needle, called a stylus, that cuts a groove in a blank record that is turned by the turntable. Once made, the record can be played back in the conventional manner through a phono pickup.

We shall not discuss the techniques of recordings and the types of recording heads in this Lesson. Here we are interested in the audio amplifiers used in home recorders. Such amplifiers are rather simple, but, since you may be called upon to repair one of them, it is worth while to discuss a typical example.

One feature of the recorder amplifier that is not usually found in other types of audio amplifiers is the volume level indicator. These indicators are used in recorders because it is necessary to keep the level of the signal applied to the cutting head below a certain maximum value and above a minimum level. The value differs for different types of equipment, but, for each type, there is a particular level that must not be exceeded; if it is, overloading and distortion, and perhaps even actual mechanical damage to the record, may occur. The minimum level is set by the noise level of the equipment; if the signal is too weak, you can hear nothing but noise. Some of the more elaborate professional systems use a dual indicator; one must record loudly enough to stay above the minimum, but not so loud as to exceed the maximum. However, on most home recorders only the
maximum is indicated; it is expected that the performer will speak loudly enough to stay above the noise level.

In professional recording equipment, a meter is used to indicate the level of the signal applied to the cutting head. A home recorder is not usually equipped with such an elaborate indicator; instead, it usually has a magic-eye tube that indicates the output signal level.

The schematic diagram of a typical home recorder is shown in Fig. 26. This device can be used as a recorder, as a record player, and as a small p.a. system. In this last use, the microphone, amplifier, and speaker are the only components that are used.

As you can see from the diagram, the microphone input is fed to the grid of a 6J7 voltage amplifier. The signal then goes to the triode section of a 6Q7 dual-diode, single-triode tube and from there to a 6K6 output tube. The output of this tube is applied through a transformer either to the driving mechanism of the cutting head or to the loudspeaker, depending upon the position to which a three-way switch in the output circuit is thrown.

The diode section of the 6Q7 furnishes a varying d.c. voltage that is used to operate the 6U5/6G5 magic-eye indicator tube. When the switch in the output circuit of the recorder is thrown to the cutting position (position 1), signal voltage is applied to this diode section from a voltage divider that is across the cutting head. Thus, the value of the voltage applied to the diode at any instant depends on the output voltage of the amplifier at that instant. The half-wave rectifying action of this diode produces a pulsating d.c. voltage across the diode load, which is essentially $R_{13}$ and $R_{14}$ in parallel. The d.c. portion of this voltage is applied through the a.c. filter $R_{18}$ and $C_{12}$ to the control grid circuit of the 6U5/6G5 magic-eye tube. The average d.c. value depends on the average audio level, so condenser $C_{12}$ is thus charged to a d.c. voltage, the amount of which at any time depends upon the value of the a.c. output voltage of the amplifier. This voltage across $C_{12}$ acts as a changing bias for the magic eye tube, closing or opening the shadow on the tube as the output voltage increases and decreases. If the audio input is properly controlled, the eye indication

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**FIG. 25.** Block diagram of home recorder system.

can be brought to a predetermined level that indicates full recording but not overloading. Any excessive peaks will over-close the eye, indicating a need for reduced input. Below-normal levels will not close the eye sufficiently, so the minimum level is also indicated.

As you can see from the diagram, switches SW₃ and SW₄, which are ganged, control the use to which the recorder can be put. When they are thrown to position 1, both the microphone and the cutting head are connected to the amplifier; this is, therefore, the switch position that permits records to be made. When the switches are thrown to position 2, the pickup head and loudspeaker are connected to the amplifier; this is the playback position. When the switches are thrown to position 3, the microphone and the loudspeaker are connected to the amplifier. In this last position, the recorder can be used as a small public address system.

The tone control for the equipment consists of a fixed condenser in series with a variable resistor connected from the plate of the 6Q7 triode section to ground.

We have seen how typical amplifier systems are constructed, planned, and installed. We have also learned something about the functioning of various special forms of p.a. equipment. In a succeeding Lesson, we shall learn how these amplifiers are maintained and serviced.
Lesson Questions

Be sure to number your Answer Sheet 53RH-2.

Place your Student Number on every Answer Sheet.

Send in your set of answers for this Lesson immediately after you finish them, as instructed in the Study Schedule. This will give you the greatest possible benefit from our speedy personal grading service.

1. Why is it impractical to install a p.a. system in a location where the noise level is well over 100 db?

2. What is the advantage gained by paralleling the cathodes, grids, and plates of the dual triodes used in the driver stage as in the Bogen HO125 amplifier?

3. What would be the effect on the output of an amplifier if poor power supply regulation permitted the screen grid voltage on the output tubes to increase when the control grid voltage decreased?

4. What benefit is gained from the use of cathode-follower coupling to link the driver stage to the output stage as in the Bogen HO125 amplifier?

5. In a p.a. system in which several amplifiers are to be used, is it better from the servicing viewpoint to use parallel inputs and parallel outputs or to use parallel inputs and separate outputs?

6. Why is it necessary to provide a constant-load loudspeaker switching system in a multi-channel p.a. system used to furnish several programs to hotel rooms?

7. Under what two conditions is it necessary to use master units at all stations of a wired intercom system?

8. How do signals get from one unit to another in a wireless intercom system?

9. For what reason do some juke-box amplifiers contain a circuit that applies an above-normal filament voltage to the tubes when the equipment is turned on?

10. Why is it necessary to keep the volume of sound to be recorded above a certain minimum level?
MAINTENANCE OF P.A. SYSTEMS

54RH-2

NATIONAL RADIO INSTITUTE
WASHINGTON, D.C.
ESTABLISHED 1914
STUDY SCHEDULE NO. 54

For each study step, read the assigned pages first at your usual speed, then reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind. Study each other step in this same way.

☐ 1. Introduction .................................................... Pages 1-2

This section introduces the three sections that are to follow, and explains how these procedures differ.

☐ 2. Renovating P. A. Systems ................................. Pages 2-14

After a theoretical discussion of distortion, hum and noise, and oscillation, this section shows what practical steps may be taken to improve amplifiers having these defects.

☐ 3. Preventive Maintenance ................................. Pages 14-17

The steps taken in regular inspections that prevent many breakdowns.

☐ 4. Servicing P. A. Systems ................................. Pages 17-28

Practical service procedures for oscillation and motorboating, hum, noise, low output, dead systems, distortion, and intermittent defects.

☐ 5. Mail Your Answers for this Lesson to NRI for Grading.

☐ 6. Start Studying the Next Lesson.
PREVIOUS Lessons have shown you how sound systems are planned and installed. Now we are going to discuss keeping sound systems in good condition.

This Lesson is divided into three sections. First, we shall discuss the renovation of existing equipment to improve its performance; next, we shall study preventive maintenance procedures that should be followed to keep a system from developing complete breakdowns; and finally, we shall learn how to service defective systems.

The need for renovation usually arises when new demands are made of the system. For example, it may be desired to play music over a p.a. system that was originally intended to carry only voice. In almost every such case, you will have to improve the frequency response of the system before it will be able to reproduce music with good fidelity.

Natural aging may make equipment deteriorate to such an extent that it no longer gives satisfactory service even though it is not actually defective. Many consider the restoration of such systems to be a renovation, although it may also be classed as servicing if the final results are not better than the original response.

Preventive maintenance involves making frequent tests and inspections of a system as a matter of routine with the object of anticipating possible part failures and thereby preventing them from happening. The procedures followed in preventive maintenance often seem to be unimportant actions, since they consist mostly of inspecting components, shaking wires, removing dust, and so forth. Their value is proved, however, by the fact that a system stays in operating condition longer when these procedures are carefully followed.
Repairing defects in a p.a. system is much like repairing the audio system of a radio. As you will learn, however, there are certain defects that are more apt to occur in p.a. systems than in radios, mostly because a p.a. system is worked more nearly at its maximum level, so slight changes in-tube characteristics or in operating potentials show up readily in reduced and distorted output. Also there are more connections between the amplifier and the input and output devices, and each joint is a possible source of trouble.

We shall consider both true p.a. systems and office intercoms in this Lesson. Although the office intercom is really just one form of public address system, the fact that it is low powered and is intended to carry only voice makes the general procedure of maintaining it somewhat different from that usually followed for a p.a. system.

Now, let's discuss the renovation of inadequate p.a. systems.

Renovating P. A. Systems

A p.a. system may distort, hum, be noisy, oscillate, or have insufficient volume without being defective in the sense that some part or parts have failed. Any one of these conditions may be bad enough (because of lower original design requirements) to require correction, particularly if greater demands are placed on the system than were originally made. Let's see why these conditions may arise, and what can be done about them.

DISTORTION

The output of a p.a. system is distorted when it differs from the original input in any way except in a uniform change in volume. There are four kinds of distortion—amplitude, frequency, intermodulation, and phase. Phase distortion is not important in practical sound work, but any one of the other three may be present to an objectionable extent. We have discussed these kinds of distortion in earlier Lessons, but, to refresh your memory, we shall describe briefly what each consists of.

Amplitude distortion is present if the shape of the output signal differs

Amplitude distortion is present when the output signal of an amplifier does not have the same shape as the input signal. Here, for example, a sine wave is applied to three different amplifiers. Amplifier 1 clips the negative half of the input signal, amplifier 2 clips the positive half, and amplifier 3 clips both halves. In each case, the output of the amplifier is distorted.
from that of the input signal. In other words, if we feed a sine wave into a p.a. system and do not get a sine wave out, the system exhibits amplitude distortion. Amplitude distortion usually results in a flattening out of one or both half-cycles of the sine wave. This means, as you have learned, that higher frequencies have been added to the original signal. If only one half-cycle is distorted, even harmonics have been added; if both are distorted, odd harmonics have been added.

and difference of the two audio frequencies. It is similar to amplitude distortion in that extra frequencies are added; these added frequencies are not harmonics of the original frequencies, however. For example, if an audio frequency of 300 cycles and another of 700 cycles are applied to some non-linear device—such as a saturated transformer—beat frequencies of 400 cycles and 1000 cycles will be produced, as well as frequencies resulting from the interaction of these beat frequencies with one another and

Frequency distortion is present when some frequencies are amplified more or less than the other frequencies in the input signal. An amplifier that is deficient in high-frequency or low-frequency response, or which has a peak in its response, exhibits frequency distortion.

Intermodulation distortion might also be called audio-frequency superheterodyning. It occurs when two audio frequencies are fed into a non-linear device. The result is the same as that achieved when a radio signal and an oscillator signal are fed into the first detector of a superheterodyne; that is, beat frequencies are produced that are equal to the sum with the original frequencies. These beat frequencies are not usually of great amplitude, but there are enough of them, when intermodulation distortion is pronounced, to cause fuzziness in the sound output. Intermodulation distortion is seldom measured for any amplifier, since there is no very easy way to measure it; in general, you can assume that a system having low amplitude distortion will also have low intermodulation distortion, although this is not true in every case.

A p.a. system exhibiting any one or any combination of these distortions may be satisfactory for some limited use, such as reproducing spoken words
only. However, such a system will almost invariably need renovating if it is to be used for music, since in that case the distortion will be far more noticeable.

**Amplitude Distortion.** Amplitude distortion may originate in the pickup device, the amplifier, or in the loudspeaker, but its most usual source is in the amplifier. In fact, any amplifier has a certain amount of amplitude distortion. Therefore, the problem is not to eliminate amplitude distortion altogether, but to reduce it so much that it is unobjectionable. A commercial amplifier of reasonably good quality usually has relatively low amplitude distortion if it is operated conservatively—that is, if its rated output is 20% or 30% more than the power actually drawn from it.

If you are renovating a system that exhibits excessive amplitude distortion, check first to make sure that the amplifier is not being asked to supply more output than it is designed to furnish. If it is, the only remedy is to install an amplifier that is easily capable of handling the load placed upon it.

In fact, whenever new demands make the amplifier of a sound system inadequate, about the only thing you can do is replace it with a better amplifier. We are assuming, of course, that the old amplifier is in good condition but is simply incapable of furnishing all the power needed.

Amplitude distortion may also be caused by a speaker cone that has become relatively inflexible. An old cone, or one that has been exposed to dry heat for some time, may have this defect. The only cure is replacement of the cone or of the speaker.

**Frequency Distortion.** Frequency distortion may be caused by the pickup device, by an audio line, by an amplifier, or by the loudspeaker. Each of these devices passes some frequencies better than others, even when it is in perfect condition, so the sound system may have marked frequency distortion without having any defective part.

You must always keep in mind the fact that the amplifier frequency response does not necessarily determine the fidelity of the sound system. Even if an amplifier has a perfectly flat frequency response throughout the
audio range; the sound system in which it is used may have severe frequency distortion if any of the other elements in the system has peaks or valleys in its response. Therefore, to secure high fidelity in a sound system, you must match the components so that the overall frequency characteristic has the desired shape. Usually it is best to design the system so that it is flat in response within reasonable limits for the desired range, but in some special cases it may be preferable to accentuate the lows or highs.

When you are attempting to improve a system that already exists, you do not usually have the freedom that the original designer had. The customer will generally expect you to make as few changes as possible to give the system the performance he desires. This means that, when a system exhibits frequency distortion, your efforts will be directed toward raising the valleys or lowering the peaks in the response as economically as possible. Very often this will mean that you will change a component of the system rather than attempt to improve its response, because the time cost of your work in making such an improvement would be more than a new component having the desired characteristic would cost.

Curves are very often used in discussions of the fidelity of sound systems. Such a curve usually shows how much db variation there is in the output of a sound system at different frequencies. To the p.a. man, however, these curves have only theoretical interest. As a practical matter, it is impossible to plot such a response curve for a complete installation. It is possible to do so under laboratory conditions, using very elaborate equipment, but it is never done in practice.

These curves do have some valuable information to give you, however. For example, if a frequency response curve furnished by the manufacturer shows that the microphone to be used has a deficient low-frequency response, you know at once that the low-frequency response of the amplifier will probably need to be boosted to give a reasonably flat output. Similarly, if the manufacturer's information on the loudspeaker to be used indicates that it is deficient in response at one end or the other of the frequency range, you know approximately what change you should make in the amplifier output to correct for this condition.

However, curves that apply to general types of microphones or loudspeakers do not necessarily apply to specific microphones or loudspeakers that you may have. These curves are usually plotted for average values, and the equipment you have may not follow the average exactly. For that matter, even if the manufacturer used the particular microphone or loudspeaker you are concerned with in gathering the data for these curves, you have no guarantee that the characteristics of the device have not changed appreciably since the curve was drawn.

Therefore, the only practical use of such curves is to give the man who is designing the system some idea of how well the various components will match. If it appears from the manufacturer's information that the equipment to be used will give a reasonably
flat frequency response in combination, the chances are that only small adjustments will be needed to get the desired frequency response after the installation is made.

Fortunately, it is not necessary to make measurements of frequency response when you are renovating a sound system. The object of the system, is, after all, to produce an amplified sound that will be agreeable, or at least acceptable, to the ear. Therefore, you can check the performance of the system adequately simply by listening to it. If it sounds good, it is good—regardless of whether or not a measurement of the frequency response would show it to be perfectly flat. As a matter of fact, a system that has a certain amount of frequency distortion may sound better than does one that is theoretically perfect.

There is one major difficulty you will meet, however, in testing a system by listening to it. This difficulty arises from the fact that people differ in their hearing ability and preferences. Many prefer accentuation of the low frequencies in music, for example. Consequently, you may find that a customer doesn’t like the reproduction a sound system gives even though you consider it excellent.

As a matter of fact, it is just about impossible to set up a sound system having a fidelity that everyone will consider satisfactory. Of course, extremely high fidelity is seldom required of a sound system unless it is to be used to amplify fine music for critical listeners. Installations of this sort are relatively rare, and usually highly trained sound engineers make them. We can, therefore, forget the problems of extremely high fidelity and instead consider only the sufficiently large problems of producing acceptable frequency response.

The process of compensating for lack of uniformity in the frequency response of a sound system is called “equalization.” Usually equalization involves increasing the low-frequency and high-frequency response of the amplifier to compensate for poor response in the microphone and loudspeaker at the ends of the audio range. Sometimes this can be done satisfactorily by adjusting the tone controls of the amplifier. This is possible, however, only if the system is well designed in the first place so that the components of it are reasonably well matched. If there are serious deficiencies in response in the microphone or the loudspeaker, or if the audio line connecting the various components discriminates excessively between frequencies, more elaborate correction is usually necessary.

A microphone may have weak low-frequency or high-frequency response or may have a peak at some frequency because of mechanical resonance. When this last defect is present, sound waves of the frequency at which the microphone is resonant will cause abnormally large mechanical motions within the microphone, producing an output with a relatively sharp, high peak. This defect is characteristic of low-priced microphones; the better grades contain damping arrangements that largely eliminate mechanical resonance. The only cure for a microphone that does exhibit excessive peaking from this cause is
to replace it with a better one. In some cases it is worth while to try a similar microphone in place of the one that is causing trouble, since there may be a considerable difference in their responses.

If a high-impedance microphone is used with a long microphone cable, there will be a very noticeable decrease in the high frequencies by the time the signal reaches the amplifier, because of the shunting effect of the distributed capacity of the microphone cable. For this reason, you should always use a low-impedance microphone and a low-impedance line when the microphone must be over 20 or 25 feet from the amplifier.

Weak low-frequency or high-frequency response, if it is not excessive, can usually be equalized by adjusting the tone controls of the amplifier. If not, you can insert an equalizer network in the line between the microphone and the amplifier. Such equalizers are made by several companies. Almost invariably, however, these are "losser" networks—that is, they compensate for a loss in high frequencies by reducing the low frequencies proportionately, or vice versa, with the result that the signal applied to the amplifier has the right proportion of highs and lows but is considerably weaker than the original output of the microphone. Such losser equalizers can be used only when the gain of the amplifier is great enough to overcome the attenuation they cause. Obviously, if the amplifier is working at its peak output, it cannot have sufficient reserve power to make up for the extra loss.

If the amplifier does not have enough gain to permit you to use an equalizer, probably the best way out is to use a better microphone—unless you intend to install a new amplifier anyway, in which case you may be able to get one having sufficient gain.
to make the use of an equalizer possible.

Frequency distortion may also occur in the output system of the amplifier. Any one or more of these causes for frequency distortion may exist:

1. The loudspeaker may not be linear in its reproduction.

2. The baffle system used with the loudspeaker may cause a loss of the low frequencies.

3. The line-coupling transformer may cause losses at the high and low ends of the audio range and may possibly cause resonant peaks in the middle.

4. There may be loss of high frequencies in the line from the amplifier to the loudspeaker because of the distributed capacity of the line.

Frequency distortion caused by non-linearity of the loudspeakers can be minimized by using loudspeakers of good quality and efficient design instead of inexpensive and inefficient units. If fidelity is particularly important, separate loudspeakers may be used for the high frequencies and for the low frequencies. As you know, a combination of two speakers used in this manner is called a tweeter-woofer combination. Modern coaxial loudspeakers, in which the high-frequency tweeter is mounted within the cone opening of the low-frequency woofer, will give very extended coverage; the best will reproduce frequencies from as low as 30 cycles up to about 15,000 cycles with a minimum of frequency distortion.

Of course, it is useless to use an extended-range loudspeaker system when the total range of the amplifying system is restricted. For example, if the amplifier system is capable of handling a range of 70 to 8000 cycles, a loudspeaker system that will reproduce from about 60 to about 10,000 cycles will give the system the utmost fidelity of which it is capable. There would be no point in using a loudspeaker system that had a more extended coverage than this.

When a tweeter-woofer combination is installed in this system, it is necessary to use a cross-over network with it to separate the highs from the lows and feed each to its proper loudspeaker. Most high-fidelity loudspeaker systems have such cross-over networks built into them; if not, the manufacturer almost always offers a suitable network separately.

It is practical to install better loudspeakers if the original ones are cone loudspeakers. If the original loudspeakers are exponential horns equipped with driver units, replacing them with cone-type loudspeakers is possible only if the consequent loss in sound power output is not objectionable.

There is usually a distinct loss in the frequencies below 250 cycles when horn loudspeakers are used. This makes them unsuitable for reproducing music when good fidelity is wanted. Folded auditorium horns will provide a greater frequency range but are not used very often, because they are large and expensive. Cone loudspeakers enclosed in suitable baffles offer a wider frequency range than any other kind of reproducer furnishes, and are therefore preferred for
faithful reproduction of music. They have several faults, however, mainly that they are limited in their power-handling capabilities and have low conversion efficiency. The fact that a cone loudspeaker has low conversion efficiency means that it must be fed considerably more power than must be applied to a horn loudspeaker to produce the same sound output. Since a single cone loudspeaker cannot handle much power, a great many of them must be used to get a high-level output.

There are various kinds of box baffles available for use with cone loudspeakers. Of these, the bass reflex baffle is the one most commonly used. A picture of one of these baffles is shown in Fig. 1. It consists of a wooden or fiber box that is closed on all sides except the front, in which there are two holes. A loudspeaker is mounted in one of these holes; the other is a port from which emerges the sound waves caused by the movement of the back surface of the speaker cone. The baffle is designed so that the sound coming out of this port reinforces the bass.

Several manufacturers supply dual loudspeakers with matching cabinet baffles for use when high-fidelity sound output is desired. These are, of course, considerably more expensive than an ordinary wall loudspeaker. If it is necessary to use reflex trumpets because high efficiency of sound conversion is needed, it may be possible to get better low-frequency response by installing larger trumpets, which have longer air columns and

FIG. 1. This is a typical bass reflex loudspeaker baffle. These come in three sizes to accommodate 8", 12", or 15" loudspeakers. The finished appearance of this baffle makes it suitable for installation almost anywhere without need for concealing it.

This is another type of bass reflex baffle. Two loudspeaker units are used in it, one for high frequencies and one for low. The frequency range the assembly can reproduce is from 75 to 12,000 cycles, and its power-handling capacity is 20 watts. The unit can be stood on the floor, or the feet can be removed to permit wall mounting.
correspondingly lower cut-off frequencies. It is possible to get a reflex trumpet that has an air column of 6½ feet and a low-frequency cut-off of 85 cycles. A trumpet having a cut-off frequency this low is adequate for all but the most demanding installations.

An important cause of frequency distortion in the output section of a sound system is the coupling transformer used in the loudspeaker circuit.

From the microphone to the amplifier does. If the amplifier has sufficient reserve power, some form of lossier equalization can be used that will attenuate the low and middle frequencies as much as the high frequencies are attenuated by the line, giving, as a result, a final output that is nearly flat. Such a method is particularly useful if the amplifier uses negative feedback in the output stage, and it is therefore not particularly critical

![Diagram](image)

**FIG. 2.** An example of a loudspeaker system in which one line is equalized but the others are not. This is sometimes necessary, as the text points out, when one line is considerably longer than the others. They are all drawn the same length here because it is not customary to indicate actual line lengths on a schematic diagram of this sort; however, the presence of the equalizer in one line indicates that it is different from the others.

Unless it is of very good quality, this transformer may introduce losses at the high and low end of the audio band and may cause excessive peaking in the middle of the band. If an undesirable amount of frequency distortion is caused by the matching transformer, you must replace the transformer with a better one unless the adjustment of a tone control somewhere in the system makes it possible to remedy the distortion.

The output audio line going to the loudspeakers may also cause frequency distortion, just as the line with respect to changes in the output circuit impedance.

When a parallel arrangement of loudspeakers is used, it is sometimes necessary to equalize the response of one but not of the others. This may occur, for example, if one loudspeaker is some distance from the others and therefore has a long audio line running to it (Fig. 2). The equalizer used will have very little effect on the amplifier load in such a case; its presence merely means that the impedance of one of the parallel branches is increased somewhat, and this, of course,
affects the net impedance of the parallel combination only slightly.

If equalization must be introduced in one of the lines of the parallel arrangement, however, it may be that attenuation will have to be introduced in the other lines to keep the outputs of all the loudspeakers at the same relative level. This may become necessary because of the power loss in the equalizer used in the line to the loudspeaker. The necessary attenuation can be secured by installing T pads in the lines of the other loudspeakers. These pads can be purchased commercially if you specify the db loss needed and the source and load impedances.

It is also possible that you may want to run the equalized speaker at a lower level than the others under some conditions. If the equalizer in the line does not reduce the output sufficiently, it may be necessary to use a T pad in the line also.

**HUM AND NOISE**

Any high-power amplifier has a certain amount of hum. Since this hum is low in frequency (120 cycles if the amplifier uses full-wave rectification), it may not be audible if the amplifier is used with loudspeakers that have poor low-frequency response. Installing loudspeakers with better fidelity characteristics during renovation of such a system may actually make the system sound worse by making the hum more audible. (Also, changing speakers may result in more acoustic feedback, thus causing a howl.) Whenever you improve the low-frequency response of a system, therefore, be on the lookout for an increase in hum. If it becomes objectionably noticeable, it may be necessary to improve the filter system of the amplifier, replace the power pack, or even install a new amplifier with a lower hum level.

Hum may also be produced during renovation if some unshielded part of the system is brought near a source of hum, such as an a.c. power line carrying heavy currents. Check over the installation to make sure this has not happened if the hum level is higher after a circuit change.

Noise in a system usually occurs only because there is some defect in the system. It is possible, however, that a change in microphones or in the conditions at the point where the microphones are may cause more noise to be picked up and reproduced than formerly. This may happen, for example, if you replace a sharply unidirectional microphone in a noisy location with one that has a more widespread pickup pattern. Choosing a microphone with a more nearly unidirectional pattern, or selecting some relatively noise-free location for the microphone, will clear up the difficulty.

**OSCILLATION**

Oscillation may be caused in a p.a. system either by a defect in the amplifier or by acoustic feedback from a loudspeaker to a microphone. When you have had a little experience, you will usually be able to tell easily which kind of oscillation is occurring. Acoustical howling caused by sounds from the loudspeaker feeding back to a microphone is usually in the middle register—around 2000 or 3000 cycles.
Oscillation caused by a defect in the amplifier is usually either very low or very high in frequency. If you can reduce the microphone input to the amplifier to zero and still hear the oscillation, and the loudspeaker is not so near the amplifier that it is likely to be causing actual physical movement of a tube within the amplifier, you can be reasonably certain that the defect is in the amplifier circuits. This is covered elsewhere under repair of defects. However, acoustic feedback and the resulting howling may be such that renovation of the system is desirable.

There is almost always a certain amount of acoustical feedback in any p.a. system. Oscillation occurs when so much sound energy is fed back to the microphone that the production of sound by the p.a. system becomes self-sustaining. Suppose, for example, that a person speaks into a microphone over a public address system. A certain part of the amplified reproduction of the voice will be reflected back to the microphone. If the reflection is so strong that the volume level at the microphone of this reflected sound is as great as the volume level of the original sound, the amplifier will produce an output equal to the original output, and the process will be repeated over and over again. The result is that the p.a. system will produce a steady tone or howl, probably with a frequency between 2000 and 3000 cycles, because the overall amplification and feedback tend to "peak" in this range. As you can see, this is similar to the production of oscillation in an electrical circuit except that here the feedback is caused by the reflection of sound waves from the output to the input.

You can readily see that both the amplification of the system and the amount of acoustical feedback determine whether such howling will occur. If the amplification is high, even a small acoustical feedback can cause oscillation; or, if the amount of feedback is large, oscillation can occur at relatively low amplifications.

To control this kind of oscillation, therefore, either the amount of feedback or the amplification of the system must be reduced. It is easier, but
less desirable, to reduce the amplification—easier because doing so merely involves turning down the volume control, undesirable because doing so puts an upper limit on the output of the system. In renovating a system, you should reduce the feedback to such an extent that oscillations will not occur even when the system is operating at full volume. We shall discuss means of reducing feedback in a moment.

Howling can sometimes also be stopped by adjusting the tone control of the amplifier. It is quite possible that a p.a. system will have peaks at one or more points in the frequency range—that is, some frequencies will be amplified more than others. It is also possible that some frequencies will be fed back to the input more than others; high frequencies are more directional than low frequencies, for example, and may therefore be concentrated toward a wall from which they are reflected very readily, with the result that the sound reflected to the microphone contains more high frequencies than low frequencies. As we said, oscillations may occur if either the feedback or the amplification is too high. If the amplifier response is greater for some frequencies than for others, or if the feedback is greater for some frequencies than for others, those frequencies may cause howling even though all other frequencies do not. If the frequencies causing oscillation can be reduced by adjustment of the tone control, it will not be necessary to reduce the volume of the whole system to prevent oscillation. Like the reduction of volume, this method of preventing oscillation is less desirable than is reducing the feedback, since it reduces the flexibility of the system. Hence, a correction of the frequency response may be the renovation step required.

**Reducing Feedback.** Assuming that the system does not have any loudspeaker pointing directly at the microphone, the ways to reduce acoustical feedback other than using different microphones or speakers are:

1. Move the speakers to a different location.
2. Move the microphones so that they pick up less feedback.
3. Reduce reflections in the room.

Most usually the latter is done by installing sound-absorbing material on the walls, floor, or ceiling. You learned in an earlier Lesson how acoustical material was used to reduce reflection in the room. Although we dealt with the matter there from the standpoint of the initial installation of p.a. systems, the principles apply just as well to a system that is already in existence.

Finally, the use of speakers having better back baffling and the use of modern microphones having controllable pickup patterns may both help to reduce the feedback.

**INTERCOM SYSTEMS**

Fidelity, hum, and noise are not usually matters of concern in an intercom system as long as speech can be understood easily over it. In fact, about the only difficulties in an intercom network that require system correction rather than servicing are lack of volume and inflexibility of arrangement.
Both these difficulties occur only in systems that have been poorly designed from the start, or that have been modified after the initial installation in the hope of extending their usefulness.

The usual reason for a complaint of low volume in an intercom system is that a remote station having no built-in amplifier is placed too far from the master. In this case, signals from the remote station may be so attenuated that they are at or below the noise level by the time they reach the master, and therefore cannot be amplified sufficiently to be intelligible above the noise level. Signals from the master will still be audible at the remote location, however, unless the line is extremely long.

The only cure is to install a master or a remote having a built-in amplifier in place of the original remote station.

In a wireless intercom system, a station may no longer receive or transmit signals when it is plugged into a different power outlet. If this happens, you can be reasonably sure that the new power output is not connected to the same branch as the other outlets used for the rest of the system. It is always possible that two outlets in the same room are not both on the same power branch, particularly in a large office building that has several main power lines. This complaint can be remedied rather simply by plugging the intercom into an outlet that is connected to the other outlets used for the system.

A part of a wireless system may transmit or receive poorly, or even fail to do so altogether, if it gets out of alignment. We will discuss the realignment of wireless intercoms later, in the section devoted to servicing.

If the system no longer provides the number of intercom stations that are desired, the renovation consists of adding the required units and using master units having the required number of switch positions.

Now, let's study the preventive maintenance procedures.

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**Preventive Maintenance**

Preventive maintenance is a procedure carried on for the purpose of preventing trouble by anticipating it. In p.a. work, it consists of regular, frequent inspection and testing of the equipment and the installation. Such work is generally done under a service contract, under the terms of which a serviceman agrees for an annual fee to make periodic inspections and to service whatever defects he finds. This fee may or may not include the cost of replacement parts—usually not, since the fee would have to be prohibitively high to take into account such possibilities as the failure of a large and expensive loudspeaker.

Preventive maintenance, if it is properly carried out, is extremely good insurance against breakdowns. To carry it out properly, however, you will have to develop an ability.
to notice little defects in performance or operation that are advance indications of future part failures. Much of this ability will come with experience. It is easy enough for any one to tell whether a resistor is hot, for example, but you must have had considerable experience in maintenance work before you can decide whether it is so hot that it will burn out quickly or is just operating at a high but safe temperature that it can maintain for a long time without damage.

MAINTENANCE PROCEDURE

The most important step in preventive maintenance is to set up a good maintenance procedure. The procedure will depend to some extent in its details upon the specific installation, but the main features of it should be about the same for all installations. We shall, therefore, discuss the basic principles you should follow in any preventive maintenance work.

There are three things you should do on each of your periodic visits to an installation you are maintaining: You should listen to it, inspect it carefully, and make various instrument tests on it. Let’s see what you can find out from each of these actions.

Listening Tests. Naturally, the first thing you should do when you make a service call is to ask the user of the equipment if there is any complaint. If he feels that the system has been performing poorly in any way, keep his remarks in mind as you go through your maintenance procedure. Whether or not he has any complaints, you should be on the alert to spot any possible causes of trouble.

Next, turn on the equipment and listen to it carefully. Listen to each of the loudspeakers to make sure that none of them is excessively noisy and that the hum level is not abnormal. If you find no such defects, run the volume and tone controls quickly up and down several times to see if doing so causes an excessive amount of noise. A noisy control should be re-

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This loudspeaker-baffle combination is designed for installations in which both high power and high fidelity are required. The unit has a frequency range of 50 to 11,000 cycles and is capable of handling powers up to 40 watts. As you can see, two coaxial loudspeaker units are used. These are powered by diaphragm-driven units.

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placed, since it will usually get worse very soon.

Having tested the system for background noise and hum, your next step should be to try it out in operation. Have an assistant speak into each microphone and operate each record player while you listen carefully to the output of each loudspeaker. After you have had some experience, you
can detect increases in distortion or loss in output that indicate the possibility of major defects in the future.

**Inspection.** While you are making your listening tests, you should also inspect the installation carefully.

![Loudspeaker](image)

*Courtesy University Loudspeakers, Inc.*

This is the University type MM-1 loudspeaker, a splash-proof marine unit that is unusually small. It has a continuous operating capacity of 8 watts. Its sealed construction makes it suitable for use in either wet or dusty locations.

Look over each loudspeaker while you are listening to it and make certain that it is firmly secured and that the cables going to it are in good condition. If a plug connection is made to the loudspeaker, make sure the plug is firmly seated. See that the shields in the loudspeaker cables make a good connection to ground.

Inspect the shielding on all other cables also. Make sure that the connections at both ends of each cable are firmly made. See that the knobs of all controls fit tightly on their shafts and that pointer knobs do not rub on their dial plates.

To make the instrument tests described a little later, you will have to open up the amplifier to some extent. When you do so, inspect all visible parts of it carefully.

See that the tubes are firmly seated in their sockets, that all shields are in place, that all soldered connections are good, and that all components appear to be in good condition. Dust out the amplifier case. Make sure all grid-cap connections are tight. In brief, use your eyes intelligently and carefully to spot any possible cause of trouble.

**Instrument Tests.** The most important single test on the system for you to make with instruments is a check of the tubes. This should be done each time the system is inspected. You should discard any tube that appears to be questionable. Tubes are one of the most common causes of trouble in an amplifier, so it will pay you to make a careful check.

You should also make certain voltage measurements as a matter of routine. On each service call, you should measure the output of the power supply and compare it with the reading secured on the last call. If there is a significant difference in the readings, you should discover the cause. Every now and then, also, you should make readings of the plate voltages of all tubes. How frequently you should make these voltage readings depends on the amount of use the
p.a. system gets. If it is in fairly constant use, plate voltage readings should be made every three months or less; if it is used relatively infrequently, six-month or even one-year intervals may be frequent enough. Plate voltage readings will be helpful, of course, only if you have other readings with which to compare them. Therefore, you should record all such readings and compare them with those taken previously to see if you can observe any marked change that indicates the possibility of future trouble.

While you have the amplifier opened up, you should, as we said earlier, make a visual inspection of all parts that you can see.

**SERVICE PROCEDURES**

The discussion we have just given of preventive maintenance procedures makes no mention of the possibility of your finding defects. Naturally, when you find a condition that indicates the possibility of a defect, you must check back through the circuit or part involved to see what might be causing the condition. If you find the hum level has increased, for example, the most probable causes are that cathode-to-heater leakage has developed in a tube or that the filter condensers are losing capacity. Cathode-to-heater leakage should show up when you test the tubes, but may not if the leakage is slight or your tube tester is not well designed for the test. You can check the filter condensers by disconnecting them and testing them with a condenser tester, if you have one, or, if not, by temporarily connecting a filter condenser you know to be good in place of the one you suspect. These, of course, are servicing techniques, which we shall discuss in the next section of this Lesson.

As a matter of fact, preventive maintenance becomes servicing as soon as you find any indication of an actual or possible future defect in the system. Now, let's see what techniques are particularly useful for servicing p.a. systems.
A public address amplifier is essentially the same thing as the audio amplifier of a radio, except that it usually has much higher power, and the servicing techniques used for the one are not much different from those used for the other. In our discussion of p.a. servicing, therefore, we shall make use of the knowledge you've already gained about the servicing of radios.

The most common defects of p.a. systems are oscillation or motorboating, hum, noise, low volume, no output, distortion, and intermittent operation. (These are not necessarily arranged in this list in order of relative frequency of occurrence.) Naturally, to service a p.a. system exhibiting any of these defects, you should isolate the defective stage, circuit, and part with the aid of the servicing techniques you have already learned. To assist you to do so, we shall now discuss each of the defects a p.a. system may be expected to exhibit and point out the most common causes for each.

Before we go any farther, we want to give you a word of caution. Never disconnect a loudspeaker from an amplifier while it is in operation. If there is only one loudspeaker in the system, disconnecting it while the amplifier is operating may ruin the output tubes because the removal of the load will cause a very high peak voltage to appear across them. If several loudspeakers are used, disconnecting one may cause the others to be overloaded and damaged. By the same token, you should never turn on an amplifier without making certain that the output stage is properly loaded.

In addition, don’t try to operate a system with the speaker driver unit out of the horn or baffle. The lack of proper loading on the cone or diaphragms permits overdriving, which results in a ruined cone or diaphragm.

**Oscillation and Motorboating**

Oscillation in a p.a. system can be caused either by acoustical feedback of sound from the loudspeaker to the microphone or by a defect within the amplifier. We discussed acoustical feedback in the first part of this lesson, since it is not really a service defect.

Correcting oscillation caused by a defect within the amplifier is done with the aid of the same methods you’d use to correct a similar defect in an ordinary radio. Oscillation in an audio amplifier usually takes the form of motorboating (oscillation at a very low frequency). Its most usual cause is a defect in a filter or by-pass condenser that causes an increase in gain or permits greater feedback. A condenser that is open or has lost capacity or has an increased power factor may cause oscillation. Any defect that changes bias, screen, or plate voltages so as to increase the gain of the stage may also cause
oscillation, but is not as apt to do so as is a condenser defect.

HUM

Hum in a p.a. system is most usually caused by a defective filter condenser or by cathode-to-heater leakage in a tube. In some amplifiers, hum may also become evident if the output stage becomes unbalanced because of differences in the characteristics of the tubes. This can happen, of course, only in an amplifier in which the balanced characteristic of

The output stage is depended upon to remove some hum.

Hum in the output of a p.a. system may also be caused by pickup from external sources. Microphone and loudspeaker cables are supposed to be completely shielded; if a poor connection develops between the shield and the amplifier chassis, however, hum may be picked up. Perhaps the most common cause of faulty shield-

example, a customer may complain that there is a great deal of noise in the system when one of his favorite records is played over it. You can almost be sure when you hear a complaint of this sort that the record has simply been played too often and is noisy. The system itself, of course, is not to blame, unless noise is present all the time.

The customer may also complain that
the system is noisy when the noise is actually picked up by the microphone. This, too, is a condition that cannot be blamed upon the system.

You yourself would not be likely to make a mistake in diagnosis of the difficulty in either of these cases. Remember, however, that the owners and users of p.a. systems are not necessarily technical men, and they often have a marked tendency to blame every difficulty upon the system instead of looking for possible outside causes.

Stage or by an open input circuit. Noise results from a defect of this sort because the grid of the amplifier stage is then able to "float"—that is, it develops a very high impedance to ground. Even very small noise currents developed through this impedance will produce an appreciable grid signal. This trouble may be caused by a poor connection at a microphone cable plug such that the cable shield does not ground satisfactorily to the amplifier chassis.

Noisy operation is often caused by

![At the factory, skilled technicians inspect damaged microphones minutely, using special equipment like the microscope shown here. A microphone is too delicate a piece of equipment to be prised into by someone who does not know exactly what he is doing.](image)

There are many defects that can cause noise in a p.a. system. A defective tube is a frequent offender, particularly one that has loose elements that may intermittently short-circuit if the tube is vibrated. A noise resulting from this cause usually consists of a loud crashing sound that occurs only at intervals.

A more or less steady, roaring sound may be caused by an open grid circuit in a voltage amplifier defective faders or variable attenuators. You can determine whether a control is defective by adjusting it throughout its range at varying rates of speed. If you hear a crashing noise when you do so, the control is at fault.

If you hear a rattling noise from the cone loudspeaker, the chances are that the cemented edge of the cone has come loose. It is, of course, easy to determine whether or not this has

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happened by inspecting the loudspeaker. If you are within the range of two loudspeakers, however, it is often rather difficult to tell from which the sound is coming. The easiest way to do so is to reduce the volume until you can hear a loudspeaker only when you are very close to it; unfortunately, doing so may mean that the loudspeaker will be driven so little that it does not rattle appreciably even though it is loose. In this case, disconnect one of the two loudspeakers from the amplifier. (Of course, the amplifier should be turned off before you do so.) Operate the system then at a fairly high volume level to be sure that rattling will occur if the cone is loose. Do not, however, allow the volume level to become so high that the loudspeaker will be overloaded; remember that there is now more power available because the other loudspeaker is disconnected.

With the loudspeaker removed, there will be an impedance mismatch at the output of the amplifier that may cause distortion. Pay no attention to the distortion, since it will be removed when the proper impedance match is made again.

**LOW OUTPUT**

Low output in a p.a. system may be caused by any of the defects that cause the audio amplifier section of a radio to be weak. In addition, there are some defects that may be found in a p.a. system that you would not ordinarily find in a radio. For example, many of the more complicated p.a. systems have terminal boards to which the leads from the loudspeakers are brought. It is always possible that one of the connections in this terminal board has become poor, either through corrosion or through someone's accidentally having loosened it. There may also have been some mechanical injury to one of the loudspeaker cables that broke some of the strands of wire in the cable and so increased its resistance considerably. The loudspeaker itself or the input device may also be to blame. Later on, in a section on microphones, we shall discuss some of the defects that may cause the microphone output to be reduced.

Remember, also, to look for some very obvious cause of reduced volume, such as a failure by the user to adjust the volume control correctly.

In an installation in which multiple speakers are used, a defect that causes a short circuit or partial short circuit across the line to one loudspeaker reduces the overall volume output of the system for two reasons: first, because the output from the shorted loudspeaker is lost, and second, because the impedance of the speaker combination is changed, causing a mismatch that wastes power. You can locate a defect of this kind by listening carefully to each loudspeaker to determine which has the lower output.

If a great many loudspeakers are used, it is sometimes helpful to disconnect them all from the amplifier and feed a signal from some other source, such as a signal generator, into each line individually to determine which one is at fault.

**DEAD SYSTEM**

When you receive a complaint that
a p.a. system is dead, your first step should be to make sure that the amplifier is plugged in and that the power is turned on. It is surprising how often such a simple and obvious cause for a dead system is overlooked by the owner.

Next, if the system has multiple inputs, check each of them. Sometimes the user of a p.a. system will assume that the whole system is dead when he gets no response from one input. If all inputs are dead, the trouble is in the amplifier or in some common connection to the loudspeaker system. If only one input circuit is dead, the defect must be in the preamplifier channel employed for this input device, in the input device itself, or in the cables connecting it to the amplifier. The easiest way to tell which is to blame is to try another input device (microphone or phono pickup) on the input terminals of the defective channel. If the test device works, then the original device or its cables is to blame. In this case, apply a test signal to the cable with the input device removed; if the system then operates properly, the original input device must be defective. Incidentally, it would be a good idea for you to include with your servicing equipment a test microphone that you know is in good condition, a record player, and perhaps an audio oscillator, for use when you want a test signal source.

If all the input circuits are dead, check the connections between the amplifier and the loudspeaker. If the loudspeakers appear to be connected properly to the amplifier, very likely the amplifier itself is defective. You can then follow the usual service procedure to locate the defective part. If the amplifier has a fuse, check it first before looking for other components that may be defective. If you do find that a fuse is blown, install another and then operate the amplifier for a reasonable length of time to make sure the fuse does not blow again. If it does, there is some other defect in the amplifier that you must locate. If, however, a replacement fuse does not blow within about five minutes, check the leakage of the filter condensers and the plate voltages of all tubes. If the filter condensers are not excessively leaky, and the plate voltages are not above normal, it is unlikely that there has been any long-time overload that has caused the fuse to blow. In this case, you can assume that some defect of a temporary nature, such as a power surge on the line, caused the original trouble.

If the fuses in a given place
blow rather frequently, you should check the line voltage to make sure that it is not excessively high. If the line voltage is above 130 volts, report the fact at once to the power company, because there is very possibly some defect in the power line itself. If the line voltage is slightly under 130 volts, the chances are that the amplifier is a little too sensitive to supply voltage variations, although it would work perfectly well in the usual supply voltage range of 110 to 115 volts. In such a case, you should consider installing a voltage-dropping resistor in the supply line to the amplifier if it appears that the high line voltage is to be present constantly. If you do use such a voltage-dropping resistor, be sure that its dissipating ability is sufficient for the power it must handle.

**DISTORTION**

We mentioned distortion earlier in the section on system renovation. Since we described there the defects outside the amplifier that might cause distortion, we shall deal here only with the internal defects that may cause a system to distort.

Distortion may be caused in a p.a. amplifier by any of the defects that will cause distortion in the audio amplifier of a radio. In addition, an unbalanced output stage, which would usually cause relatively little distortion in a radio, will produce very appreciable distortion in a p.a. amplifier because of the high power levels in such equipment.

Fig. 3 shows a typical output stage in a p.a. system. Any defect that could cause an unbalance in this stage could cause distortion. If condenser $C_3$ were to open, thus reducing the signal fed to one of the output tubes, the output of the amplifier would be very much distorted because of the loss of push-pull action. There would also be a loss in volume, although this would not be noticeable until the volume control was turned well up; at low volume levels, either output tube could supply sufficient signal.

A vacuum tube voltmeter can be used to determine how well the out-
put stage is balanced. While a test signal is applied to the input of the amplifier from a sine-wave audio generator, measure the voltages between the circuit components and ground at the points marked $E_2$, $E_3$, $E_4$, and $E_5$ on the diagram. If the stage is well balanced, $E_2$ should equal $E_3$, and $E_4$ should equal $E_5$.

An oscilloscope is also a useful instrument when you are trying to locate the source of distortion. The method of using the oscilloscope for this purpose was described earlier in your Course.

Again, the use of multiple pickups and loudspeakers gives a p.a. system possible sources of distortion that do not exist in a radio.

If a p.a. system uses two or more pickup devices and two or more loudspeakers, you will be able to localize the source of the distortion rather easily. If the distortion is heard when one microphone or record player is used but not when the other is used, then almost certainly the source of the distortion is in the first device. Similarly, if the distortion is heard on one loudspeaker and not on others, then the one on which it is heard must be defective. If the distortion is the same regardless of which pickup device or which loudspeaker is used, the amplifier is probably to blame.

**INTERMITTENT DEFECTS**

A p.a. system is subject to all the intermittent defects that the audio system of a radio has. Going dead intermittently is probably the most common defect, but any of the others we have described can also occur intermittently.

Loose connections, frequently the cause of intermittent defects, are more common in p.a. systems than in radio receivers. One reason is that there are usually many more connections of the sort that are frequently made and broken in a p.a. system—connections to loudspeakers or microphones, for example. Such connections are naturally much more likely to become defective.

Most intermittent defects have only one symptom: That is, a system hums intermittently, or goes dead intermittently, but does not do both. Occasionally, however, a system will have dual defects—for example, an intermittent loss of output accompanied by distortion. This is usually caused by a voice coil circuit that opens intermittently. If the voice coil is slightly off-center, it may rub against the pole piece enough to wear off the enamel insulation from the coil wire; the coil may then short circuit intermittently to the pole piece and, through it, to the voice coil terminal that is grounded to the frame of the loudspeaker. This is much the same thing as putting a short across the secondary of the coupling transformer feeding the voice coil. As a result, the output from the defective loudspeaker is reduced or killed completely. The changed impedance of this loudspeaker will also destroy the impedance match between the amplifier and the loudspeaker system, and will therefore probably reduce the output of all the loudspeakers to some extent.

Even though the output of each loudspeaker will be reduced by such
an occurrence, the output of the defective loudspeaker will be affected by far the most. Therefore, you can locate the defective loudspeaker by listening to each of them and determining which is the worst. This may, of course, be difficult if the intermittent operation occurs for only a short period of time.

If you have a sensitive ohmmeter, you can check the suspected loudspeaker rather easily. All you need matching transformer is removed from the circuit.

Alternatively, you can open the voice-coil circuit and place the ohmmeter in series with the voice-coil transformer and the voice-coil. In this case, the resistance you measure will be slightly higher than that of the voice-coil. Whatever method you choose, the indication of voice-coil trouble is a change in resistance as the voice-coil is moved.

![Image: Courtesy The Astatic Corp.]

Highly specialized equipment is needed to calibrate a microphone after it has been repaired. This picture shows a calibration run being made on a microphone at the Astatic factory. Don't attempt microphone repairs yourself—send the microphone back to the manufacturer.

As you recall, a wireless intercom uses the power line as a means of propagating a modulated carrier signal. The carrier frequency on most is around 100 kc. This frequency can be changed up or down by a factor of 25% so that as many as three systems can be used near one another without interfering with each other. For example, the units in one system may be tuned to 120 kc., the units in the second system to 100 kc., and
those in the third to 80 kc. Since all three carrier signals will be present in one power line, it is necessary for each unit to accept a bandwidth considerably less than 20 kc. to make sure it will not pick up a carrier that is not intended for it.

It is easy enough to align a wireless intercom to a specified frequency if you have a signal generator. Simply connect the signal generator to the input of the intercom, set the generator to the desired frequency, and turn the tuning trimmer condenser until you get a maximum response. (Most wireless intercoms use trimmer condensers to tune their input circuits.) Tune the other unit of the pair in exactly the same manner. Then, check your work by transmitting from one unit to the other and back from the second unit to the first. While one unit is operating, make small adjustments of the trimmer on the other to make sure that you have the point of maximum response. Do not adjust both units, however, since doing so may get you too far away from the frequency you want.

If you do not have a signal generator when you are attempting to align a wireless intercom, you can probably do a satisfactory job by turning the trimmers on both units all the way in, backing both off a quarter of a turn, and then adjusting one unit for maximum response while someone talks steadily into the other unit. When you have peaked one unit in this manner, reverse the procedure and peak the other while someone is talking into the one you have already adjusted. If you wish to add another system, turn the trimmers on both units on that system until they are a quarter of a turn from being all the way out and repeat the tuning procedure. Finally, if you wish to add a third system, set the trimmers of the two units half-way between the full-in and full-out position and again repeat the tuning procedure. In each case, be sure that you make only small adjustments during the alignment procedure so that you will stay fairly close to the point at which you initially set the condensers.

**MICROPHONES**

Microphones are extremely delicate devices and must be handled with great care. In fact, it is almost impossible for you to service one successfully unless you have had some special training in this delicate work. It is not even possible to test them successfully without special equipment. Therefore, never open a microphone on the assumption that you can poke around in it, find out how it works, and fix it. If the microphone is dead, return it to the manufacturer for repairs.

As a matter of fact, it is impossible to make a direct test of a microphone without special equipment normally used only in factories. About the only practical test that a serviceman can make is to install a microphone he knows to be good in place of one that he suspects of being defective. If the response of the system improves when the good microphone is installed, then either the original microphone is defective or its matching transformer and connecting cable are defective. Of course, the test microphone you install should be of the same type as
the one of which you are suspicious.

It is possible to test the microphone cable for continuity, but the microphone should be disconnected from the cable before you make such a test. An ohmmeter must never be connected to a microphone, because even the relatively small d.c. voltage of an ohmmeter can damage certain types of microphones severely or even ruin them.

Substitution of a good microphone is the only way you can tell whether or not a suspected microphone is deficient in its frequency response or in its output. These qualities of a microphone are tested by the manufacturer, but the method used is not practical for a serviceman. At the factory, the microphone under test and a standard test microphone are placed in a room that has had special acoustic treatment. Pre-determined amounts of sound power at various frequencies are then fed into both microphones, and the outputs of the two are compared as to amplitude and frequency response. No method that is less elaborate has been discovered for checking microphones accurately.

As you can see, a microphone that is bad is not something you can service. There are, however, certain practices that you can follow that will extend the useful life of microphones. We shall tell you what they are, and you can pass along the information to the owner of a p.a. system so that he can make his microphones last longer.

**Microphone Precautions.** Every velocity or dynamic microphone has a powerful magnet built into it. For this reason, none of these microphones should be placed on a workbench or any other place where it might pick up iron filings. The housing contains a metal screen around the diaphragm of such a microphone to prevent bits of metal from entering it, but iron dust or very small filings can work their way in if given a chance.
Care should be taken to keep any velocity or dynamic microphone away from alternating current fields, because such fields can partially demagnetize the magnet, thereby causing reduction in output and narrowing of the frequency range. For this reason, keep such microphones well away from power transformers and power lines, particularly lines carrying heavy current.

Velocity microphones of the ribbon type, which are normally used only when extremely high fidelity of pick-up is required, are very delicate. They must be handled with great care. For example, they should always be carried or moved in a normal operating position—that is, held upright so that the ribbon is in a vertical plane. Carrying such a microphone horizontally lets the ribbon sag and stretch. Any jarring or rough handling may cause the ribbon to move so far that it will be stretched out of shape or stick to one of the magnets between which it is suspended.

The magnets of a velocity microphone may themselves shift in position if the microphone is jarred or roughly handled. The location of these magnets with respect to the ribbon is very important to the proper operation of the microphone, so even a very slight shift in position of one or more of the magnets may destroy the fidelity of the microphone or even prevent it from operating altogether.

A strong blast of air on a ribbon microphone is certain to damage the ribbon and perhaps ruin it. Such a microphone must, therefore, be well protected from the wind if it is used outdoors. No one should be allowed to speak into it from very close range. In particular, you should never whistle directly into a ribbon microphone to test its operation.

The crystal unit in a crystal microphone can be damaged by heat, excessive humidity, or mechanical jarring. Therefore, such microphones should be kept away from extreme heat and dampness and should be kept out of direct sunlight as much as possible. They should be protected, also, from excessive shock and vibration.

Let us repeat one important warning. Do not try to repair a microphone; return it to the manufacturer for repairs. It is almost invariably less expensive to have a microphone repaired than to buy a new one.
Lesson Questions

Be sure to number your Answer Sheet 54RH-2.

Place your Student Number on every Answer Sheet.

Send in your set of answers for this Lesson immediately after you finish them, as instructed in the Study Schedule. This will give you the greatest possible benefit from our speedy personal grading service.

1. How does renovation differ from servicing?

2. How does intermodulation distortion differ from amplitude distortion?

3. Why would it be impractical to use a lesser equalizer network in a p.a. system that is working at peak output?

4. What two undesirable effects may result from installing loudspeakers with better fidelity when renovating an old speech amplifier?

5. When an acoustic feedback produces a howl, what three steps should be taken to cure the trouble, other than using other speakers or microphones?

6. What three steps are taken on each visit of a preventive maintenance schedule?

7. Why is it inadvisable to disconnect a loudspeaker from a p.a. amplifier using only one speaker while it is operating?

8. What may happen if a cone speaker is operated outside its baffle?

9. Why is acoustic feedback more likely to produce a howl around 2000 to 3000 cycles than at very high or very low frequencies?

10. What is the simplest practical test that can be made when the microphone is suspected?
REQUIREMENTS OF A TELEVISION SYSTEM

NATIONAL RADIO INSTITUTE
ESTABLISHED 1914
WASHINGTON, D. C.
STUDY SCHEDULE

For each study step, read the assigned pages first at your usual speed, then reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind. Study each other step in this same way.

☐ 1. Basic Principles of Television ................................................................. Pages 1 - 3

The uses and limitations of television signals and the basic equipment used to transmit and receive these signals are given.

☐ 2. Image Scanning in Television ................................................................. Pages 3 - 8

How, in a sequence of pictures, each is broken down into elemental impressions, and how these impressions and necessary synchronizing signals are transmitted.

☐ 3. The Cathode Ray Tube as an Image Reproducer ........................................ Pages 8-14

How a cathode ray tube in television receiver can be used to reassemble the elemental impressions to reproduce the television signal.

☐ 4. How Interlaced Scanning Reduces Flicker ................................................ Pages 14-16

How 60 pictures per second are used to reduce flicker within the bandwidth requirements of a 30-picture-per-second system.

☐ 5. Brightness and Contrast Controls .......................................................... Pages 17-19

The principles of operation and adjustment of these two important controls in television systems are considered.

☐ 6. Television Signal Standards ....................................................................... Pages 19-24

The technical standards of television signals and synchronizing pulses which affect both transmitter and receiver operation.

☐ 7. Television Receiver Circuits and Controls ................................................ Pages 24-28

The basic circuits in the usual type of the two-i.f.-channel superheterodyne television receivers are given.

☐ 8. Answer the Lesson Questions and Mail Your Answers to NRI for Grading.

☐ 9. Start Studying the Next Lesson.
Basic Principles of Television

With this Lesson you begin your special training in the field of television. This first Lesson will give you a comprehensive picture of a complete modern television system, showing how it is possible to see on the screen of a cathode ray tube in a receiver a scene which is at that same instant being viewed by the television camera many miles away in a studio.

In the NRI television Lessons you will find presented in a simple, logical, and understandable manner the important principles underlying all phases of modern cathode ray television systems. After mastering these Lessons, you will find it remarkably easy to keep in step with new developments in this exciting and rapidly growing field.

The process of scanning which breaks up a televised scene into successive signal elements results in a frequency range for picture signals of from nearly zero to more than

Courtesy Philco Radio & Television Corp.

The quality of the picture produced by a modern television receiver is evident in this image produced on the screen of a Philco television receiver (showing Mr. Larry E. Gubb, president of Philco Radio & Television Corp.). The actual image as viewed on the screen is much clearer, for considerable detail is lost in photographing the image and in reproducing the photo as a half-tone cut.
4,000,000 cycles (4 megacycles, abbreviated 4 mc.) per second. Radio waves constitute a logical carrier for bringing these picture (video) signals to a large number of people at any one time, but only very-high-frequency carriers are suitable for carrying through space a signal which has a frequency range of over 4 mc. Very-high-frequency signals are bent only slightly by moisture in the air, and are not satisfactorily reflected by the Kennelly-Heaviside sky layer; this means that reliable television reception is generally limited to points that can be reached by signals traveling in straight lines from the transmitter. With receiving and transmitting antennas at practical heights, the maximum range of reliable reception from a given transmitter is about 50 miles. The transmission of television signals over radio carriers is therefore essentially a local service. Each community must have its own local transmitting station, serving television receivers within a radius of about 50 miles. These local stations can be connected to one another and to a central source of television programs either by special coaxial cable or by microwave radio relay transmitters when national coverage is to be secured for a particular program.

Television must be accompanied by sound in order to be fully appreciated, just as everyone today expects movies to be accompanied by sound. Plans for television assume that sight and sound are transmitted simultaneously. Television will in no sense replace sound broadcasting; the reception of sound programs by radio will continue to expand as it has in the past, and television will simply be an added service to listeners for some time to come.

TELEVISION IS AN EXTENSION OF RADIO PRINCIPLES

A television camera is needed to pick up picture signals in a television studio, and a special reproducing device is required at the receiver to reproduce the transmitted picture; between these two special devices, however, we find a great many familiar radio circuits. At the television transmitter there is a master oscillator which generates the very-high-frequency carrier, together with r.f. power amplifiers, a modulator, linear r.f. power amplifiers and a transmitting antenna. At the receiving location the television signals are picked up by an antenna, and are amplified and selected in the preselector of the

Cubical transmitting antenna of General Electric 10-kw. television station W2XB, located in the Helderberg hills 12 miles outside of Albany, N. Y. The antenna consists of eight hollow copper tubes each four inches in diameter and about seven feet (one-half wavelength) long, arranged to form a perfect cube which will radiate horizontally polarized waves for both picture and voice carriers in the 66-72 mc. television channel. Being atop a 1500-foot hill, good coverage is expected for distances of 40 miles in all directions.
television receiver. This receiver, if of the superheterodyne type, will have a local oscillator, a mixer-first detector, an i.f. amplifier, a demodulator, and a picture-signal amplifier, all of which prepare the received signal for the picture-producing device. The sound accompaniment for a television program is handled in essentially the same way as in f.m. program broadcasting except that the frequency deviation is limited to ±25 kc.

Television equipment has its full complement of tubes, coils, resistors, condensers, transformers, and wires, just as ordinary sound radio equipment does. Television circuits may be identical with radio circuits, or they may be entirely new circuits developed to meet the special requirements of picture transmission and reception.

Sounds, no matter how complex, are inherently a succession of signal intensities. Unfortunately, a scene does not exist in this desired state; a scene must therefore be converted into a succession of signal intensities by a process of scanning, as the first step in sending images by radio or wire. The television camera provides this scanning, and feeds into the television system a signal corresponding to that fed into a radio system by a microphone. The succession of signal intensities in a television signal is handled by the transmitting and receiving systems in a more or less conventional manner. These varying intensities must be reassembled in proper sequence and position by an image-reproducing device at the output of the receiver in order to reconstruct the original scene. The image reproducer in a television system corresponds to the loudspeaker in a sound receiver.

To insure proper step-by-step reconstruction of the scene at the receiver, the circuit which controls the scanning at the television camera must also control the image-reconstructing process at the receiver; this control is referred to as synchronization. The synchronizing signals are produced by unique oscillator circuits, are sent out on the carrier along with the picture signals in a more or less conventional manner, and are separated from these signals at the receiver by special circuits which do not exist in the usual sound receiver. In the final analysis, however, all of these special circuits are based upon extensions of well-known electrical and radio principles.

Once the requirements of a television system are recognized, the special circuits in television transmitters and receivers will seem quite natural and obvious rather than something strange and new. By studying the process of scanning first, giving special emphasis to the synchronizing signals and the circuits which handle these signals, we can make television circuits seem just as logical and understandable as ordinary radio circuits. This Lesson is primarily intended to get you acquainted with the important problems in television.

Image Scanning in Television

When we look at a picture or scene, we see various colors and various shades in each color, arranged side by side or blended together according to the nature of the scene. An ordinary photograph, on the other hand, appears to be in various shades of one color. If we were to examine a photo-
A line drawing like that in Fig. 1A can be reproduced accurately by a printer without being broken up into large and small dots, and consequently we can use this drawing as an example and show how it appears originally and when broken up into various numbers of dots or lines. If we break up this drawing into 60 dots per inch in each direction, and make the size of each dot correspond to the average darkness over its corresponding 1/60th-inch square area in Fig. 1A, we secure the half-tone reproduction shown in Fig. 1B. The dots are arranged horizontally and vertically here for purposes of illustration, but in the usual photographic reproduction they are run diagonally so the screen pattern will not be so noticeable to the eye.

The dots in Fig. 1B are clearly visible at a normal reading distance, but if you hold this illustration about four feet away from your eyes, the dots will blend together to give the impression of a picture composed only of gray and black areas. Increasing the number of dots in a given area has the same effect as holding the illustration at a distance. Figure 2A has twice as many dots per inch on any one line as has Fig. 1B. Note that these dots blend together at a distance of about 2 feet from your eyes. The more dots there are per square inch in a photographic repro-

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FIG. 1A. A perfect printed reproduction of a line drawing made by means of a zinc "cut."

FIG. 1B. Line drawing reproduced as dots of various sizes, with 60 dots per inch.

FIG. 2A. Line drawing reproduced as dots of various sizes, with 120 dots per inch both horizontally and vertically.

FIG. 2B. Line drawing reproduced as horizontal lines of varying thickness, with 120 lines per inch.
duction, the better is the quality of reproduction. A large number of dots gives what is known in television as "high definition."

Lines of varying widths may be used in place of dots for photographic reproductions; an example of this is shown in Fig. 2B. The method of image reproduction used in television is essentially the same as this, except that in television the lines are of constant width and vary in intensity of illumination. There are 120 lines per inch in Fig. 2B, and there may be even more than 120 variations in line thickness for each inch of line length. Obviously, it is possible to get high definition with lines as well as with dot patterns. This is an important factor in the reproduction of television images.

When reproducing pictures by the process of printing, there are no real difficulties involved in securing as many as 200 dots per inch on a line, giving what is commonly referred to as a 200-line screen. In television, however, there is a limit to the amount of detail or definition which can be secured without exceeding the practical limits of the equipment. The number of dot elements per line, and the number of lines per picture are definitely limited in television by the maximum frequency range which can be handled by the system.

The larger the image size produced by the television receiver, the farther away the viewers must be if they are to see a properly blended picture rather than an assemblage of lines and line variations. For example, if we enlarged Fig. 1B to twice its size, giving 3600 dots in a 2-inch square illustration as shown in Fig. 3, we would find it necessary to move twice as far away (to a distance of about 8 feet) in order for the dots to blend together.

FIG. 3. When Fig. 1B is enlarged to twice its size, we get this result. There are now 30 dots per inch, but the total number of dots in the picture is the same as in Fig. 1B.

TRANSMISSION OF A SCENE

We have seen how a scene can be divided into elemental impressions which, when reproduced as a series of dots or variable-intensity lines, will show almost as much detail as the original scene. Now let us see how the line variations in Fig. 2B can be sent to a distant point in proper sequence, either over a wire or by means of a radio carrier signal.

Imagine a lens and photocell combination which can "see" only one small area of the picture in Fig. 1A at a time. (A photocell—often called an "electric eye"—is a device having a voltage output that depends on the amount of light falling on the cell.) Assume that this electric eye looks first at the upper left-hand corner of the picture, then moves gradually over to the upper right-hand corner, looking carefully at each elemental impression along this uppermost horizontal line of the picture. At the end of this line a mechanical force shifts the electric eye back to the left and down a little to the start of the second line. Assume that this "scanning" from left to right continues until the electric eye has looked
over the bottom line, at which time another mechanical force moves the electric eye back to its starting point at the upper left-hand corner. This action constitutes one complete scanning of the picture. The varying amounts of light reflected into the photocell by the elemental areas of the picture cause the voltage output of the cell to vary from instant to instant, and this varying picture signal voltage can be sent through space or over wires by a television system.

At the receiving end of our television system, let us imagine that we have a small nozzle which is spraying on paper a stream of ink which always covers the same definite area. This nozzle is designed so that the amount of ink which is delivered at any instant can be controlled electromagnetically; furthermore, the nozzle is so mounted that it will start spraying at the upper left-hand corner of the paper, and will travel horizontally to the right at a uniform speed corresponding as nearly as possible to the travel of the electric eye at the television scanner. The television signal which is picked up by the receiver is amplified and made to control the amount of ink flowing from the nozzle at any instant. A mechanical force returns the nozzle to the start of the second line at the same instant that the electric eye reaches the corresponding position on the original picture, and thus the nozzle delivers, for each elemental area of the paper at the receiver, an amount of ink proportional to the darkness of the corresponding elemental area on the picture. The result is that when the nozzle has completed the bottom line of the picture, it has painted with ink an almost exact reproduction of the picture at the transmitter. In a properly designed circuit, a low current would open the valve and deliver a large amount of ink; large currents would close the valve, reproducing the white portions of the original scene.

In this imaginary television system, it is essential that the electric eye and the ink nozzle start moving at exactly the same instant, travel at the same speed, and at the end of each line fly back to the start of the following line in synchronism with each other. This could, of course, be accomplished with automatic manual mechanisms, but there would be no assurance that the two devices would keep in step. Even if the nozzle happened to be only slightly slower or faster than the electric eye, there might be as much as half a line difference or error after a few lines. We therefore arrive at this conclusion: The television transmitter must, at the end of each line, send a signal impulse which will serve to swing the reproducing device back to the start of the following line in synchronism with the television camera. With this requirement met, we know that the transmitting and receiving devices will start each line at the same instant, even though they may vary in speed a certain amount during a given line. The impulse which is sent at the end of each line for reproducing-controlling purposes is called the line synchronizing impulse or the horizontal synchronizing impulse. In a practical transmitter this impulse is not produced by the electric eye, but rather by an impulse generator in the transmitter which directly controls the travel of the scanning eye and which by means of the connecting medium (wires or radio carriers), controls the travel of the reproducing device.

We must likewise provide means for returning the reproducing device from the lower right-hand corner to the upper left-hand corner at exactly
the same instant that the electric eye makes this movement. This means that the transmitter must send an end-of-the-picture impulse to the receiver along with the varying line signals and the end-of-the-line impulses. This end-of-the-picture impulse is called the picture synchronizing impulse, the frame synchronizing impulse, or the vertical synchronizing impulse.

The left-to-right scanning motion along a line is commonly called the horizontal sweep. The quick right-to-left return motion from the end of one line to the beginning of the next is called the line fly-back, horizontal fly-back, or horizontal retrace. The downward line-by-line movement from the top to the bottom of the picture is called the frame sweep or vertical sweep. The quick bottom-to-top motion is called the frame fly-back, the vertical retrace, or the vertical fly-back.

The mechanical picture-sending and receiving system just described corresponds to one practical scheme for picture or facsimile transmission (the sending of photographs from one point to another by wire or radio; also known as wire-photo). As you have just seen, the three important signals which must be transmitted on the picture carrier in an electronic television system are: 1. The picture signal or video signal, which is obtained by breaking up the picture into a number of elemental areas and scanning each of these in an orderly sequence; 2. The line synchronizing impulses or horizontal synchronizing impulses; 3. The frame synchronizing impulses or vertical synchronizing impulses.

**Actual Television Transmission.**

During the transmission of a television signal, the line impulse exists for an instant after each line has been scanned, and the frame (picture) impulse exists for a longer period after each frame has been scanned. The video signals need not exist while these impulses are being transmitted; in fact, it is wise to stop them entirely during these periods. The line and frame impulses must be sufficiently different in character to be readily separated at the receiver and applied to the proper control circuits. In actual television systems, this difference involves making one type of impulse last a longer time than the other.

The three essential components of a television signal (the picture signal, the line synchronizing impulses, and the frame synchronizing impulses) may be transmitted in a number of different ways, but the signal arrangement shown in Fig. 4 comes nearest to satisfying the requirements of the television receiver. The r.f. carrier will be considered later and hence is not shown in this diagram. First of all, notice that this television signal is a pulsating d.c. signal with all its

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**FIG. 4.** This diagram shows the three essential components of a television signal—the video signal, the line impulse, and the frame impulse. This is a modulated d.c. signal. Since the picture signal voltage swings in a negative direction with increases in line brilliancy, we have what is known as a negative picture phase.
components above the zero-voltage line, which is known as the WHITE LEVEL. The video or picture signal varies between the WHITE LEVEL and the BLACK LEVEL. The synchronizing impulses are all between the BLACK LEVEL and what is commonly known as the BLACKER THAN BLACK LEVEL. The frame impulse lasts about three times as long as the time for one line. The BLACK LEVEL is 75% of the maximum television signal amplitude.

Notice that points 1, 2, 3, 4, and 5 along the video signal, corresponding to elements along one line of the picture being scanned, are for increasing values of brightness, with point 1 corresponding to a black elemental area on the picture, points 2, 3, and 4 for gray areas, and point 5 for a white area. When increases in brilliancy make the picture signal voltage swing in a negative direction in this manner, we say that the signal has a negative picture phase. The synchronizing impulses are kept in a region not ever occupied by the video signal in order to make possible the use of a biased diode or triode tube for separating these impulses from the video signal. Notice also that before and after each impulse the television signal voltage remains constant for a short interval of time. These constant-voltage components of a television signal are known as pedestals.

When a very-high-frequency r.f. carrier is modulated with the television signal shown in Fig. 4, the white components of the video signal will exist as low carrier currents, and the impulses will exist as large r.f. carrier currents. This type of modulation is known as negative modulation, and is the exact opposite of the positive modulation scheme used in transmitting sound signals. (In radio broadcasting, the largest carrier currents correspond to the loudest sounds, and low carrier currents represent weak sounds). Negative modulation is used in television to insure having synchronizing impulse signals which are sufficiently strong to over-ride any interference noises which may be present. Furthermore, experience has shown that negative modulation gives more accurate synchronizing control at the image reconstructor in the receiver, and makes it possible to build into the television receiver a simple circuit for providing the highly essential automatic gain control action.

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The Cathode Ray Tube as an Image Reproducer

Although electromechanical methods of scanning and reproducing are perfectly feasible, these methods are far more cumbersome than purely electrical methods. Furthermore, the electrical methods, employing various forms of cathode ray tubes, are far more satisfactory for high-definition home television receivers than any mechanical system, hence, electronic systems are used exclusively.

The essential elements of one type of cathode ray tube being used for image reconstruction are shown in Fig. 5. They are: K—the cathode, which emits electrons when heated; F—the filament, which heats the cathode; A₁ and A₂—anodes which accelerate the electrons and focus them into a narrow beam; S—the fluorescent screen, which glows when hit by the electron beam; G—the control
The television signal must be applied in series with the negative grid bias in such a way that the spot will be dark each time a pedestal is transmitted; this condition is secured when the pedestals line up with the brilliancy cut-off point on the characteristic curve of the cathode ray tube. Video signals must make the control grid more positive than the cut-off voltage, thus varying the brightness of the spot on the screen. Synchronizing pulses must make the control grid more negative than the cut-off voltage, so the screen will be dark during the very short intervals of their duration (these intervals are, of course, too short to be noticed by the human eye).

The spot is in the center of the cathode ray tube screen only when there are no voltages on the vertical and horizontal deflecting electrodes.

FIG. 6. An electron beam passing between two oppositely charged metal plates is always bent toward the positive plate.

Now let us see how these electrodes can be made to move the spot to any desired point on the screen. Referring to Fig. 6, notice that we have an electron beam traveling between two oppositely charged metal plates. Remember that the electrons in this beam have negative charges; this means that the positively charged plate will attract these electrons, bending the beam upward and causing it to strike the fluorescent screen at point b instead of at a, the center. The greater the voltage between these two deflecting plates, the more bending of the electron beam there will be.

But we know that this electron beam must be moved in a definite manner if it is to produce an image on the television screen. You will remember that the scanning process
In the television camera involves analyzing the scene element by element in a manner exactly similar to that in which our eyes read a printed page. First of all, then, we require a means for sweeping the electron beam gradually from left to right in a horizontal line, then quickly back again to the left, with this horizontal sweeping motion being repeated continually.

We can secure horizontal sweeping of the beam by varying either the electromagnetic or electrostatic field in the tube. The magnetic method will be studied later. We will now study electrostatic sweep which is obtained by applying to the horizontal deflecting plates of a cathode ray tube a voltage having the characteristics shown in Fig. 7; this is known as a saw-tooth (or linear sweep) voltage.

![Fig. 7. Wave form of the saw-tooth voltage that is used for sweeping the electron beam back and forth in a television cathode ray tube.](image)

Observe that this voltage is zero at points 1, 2, 3, and 4, is positive at points 8 and 9, and is negative at points 5, 6, and 7. If this voltage is applied to plates x and y in Fig. 6, and plate y is grounded, plate x will be positive when the voltage is following path 1-8-2, and plate x will be negative when the voltage is following path 2-6-3. Plate y will always be at zero or ground potential. We can think of the voltage wave in Fig. 7 as showing variations in the charge on plate x. When this charge is at point 1, the deflecting plates will have no effect upon the electron beam and the spot will be in the exact center of the screen. As the charge on plate x approaches the positive value at point 8, the electron beam will be attracted gradually and uniformly toward plate x. As the charge drops to zero again at point 2, the spot will move rapidly back to the center of the screen. From point 2 to point 6, plate x will become increasingly negative, repelling the beam and bending it toward plate y. From point 6 to point 9 the beam will move gradually from plate y toward plate x, and from point 9 to point 7 the beam will move rapidly back toward plate y again.

We have seen that a saw-tooth voltage of the form shown in Fig. 7 will produce the desired sweep of the electron beam. If this saw-tooth voltage is applied to horizontal deflecting plates H in Fig. 5, it will cause the spot to sweep slowly from left to right across the screen, then return rapidly to the left again. If this voltage is applied to the vertical deflecting plates V in Fig. 5, it will cause the spot to move gradually from top to bottom and return rapidly to the top again.

In later television Lessons in this Course you will study the special vacuum tube circuits that are used to produce these saw-tooth voltages used for electrostatic control of cathode ray tubes. None of these circuits are absolutely steady in frequency, however; it is therefore necessary to send impulse signals along with the television signal for the purpose of controlling and stabilizing the sweep circuits. One saw-tooth oscillator circuit is required for the horizontal or line sweep and another for the vertical or frame sweep. The line sweep circuit builds up its voltage uniformly from point 5 to point 1 to point 8 in Fig. 7; at point 8, corresponding to the end of the line, a line impulse arrives with the television signal and causes this voltage to drop back to point 6 rapidly. The building up of voltage starts again, only to be stopped at
the drops in voltage are accurately controlled by the transmitter through the line impulses, we know that the electron beam in the reproducing device will be swept horizontally in exact synchronism with the scanning device at the transmitter. The vertical sweep circuit operates at a considerably lower frequency, and is controlled in the same manner by the frame impulses broadcast by the transmitter.

Now let us follow the movement of the spot on the screen of an actual television cathode ray tube as it sweeps back and forth and up and down under the influence of the impulse-controlled sweep circuits.

When the beam is under the control of the horizontal and vertical sweep voltages, we can consider its starting point to be point 1 in Fig. 8, at the upper left-hand corner of the screen. From this point the horizontal sweep voltage gradually allows the beam to "unbend" or return to the center of the top line, then gradually bends the beam in the opposite direction until the spot reaches the right-hand edge of the screen; during this action the vertical sweep voltage is gradually moving the spot in a downward direction a distance equal to the spacing between two lines.

At point 2 a line impulse arrives from the transmitter, causing the horizontal sweep voltage to move the spot almost instantly back to the left-hand side of the screen along the dotted-line path 2-3. This return motion is very rapid, but sometimes, if the receiver is not properly adjusted, it will produce on the screen a faint line which is known as the retrace or fly-back line.

This process continues for each other line until the spot is swept to point 36 at the end of the last line. At this time the first frame impulse

the gradual build-up of the vertical synchronizing voltage and causing the spot to move back up to the top of the screen. Even though this vertical sweep voltage drops back to its starting value at a rapid rate, the change does take more time than is required for a complete horizontal sweep. As a result, the spot actually takes a zig-zag path from side to side as it is being returned to the top of the screen. This zig-zag path is not ordinarily seen on the cathode ray tube screen, so for simplicity the ver-

FIG. 8. The path traced on the fluorescent screen of a television cathode ray tube by an electron beam under the influence of horizontal and vertical sawtooth sweep voltages is shown in this diagram. The waveforms of the sweep voltages are shown above and at the right of the screen; these voltages are applied to the ungrounded deflecting plate in each case. Thus, when the ungrounded horizontal plate is highly negative (at 1), the spot will be at the extreme left side of the screen at point 1; when this plate is at zero potential (a), the spot will be at a in the center of the screen, when this plate is highly positive (2), the spot will be at the extreme right side of the screen at 2; when the saw-tooth voltage drops suddenly back to the highly negative value (2 to 6 to 3), the spot flies back from 2 to 6 to 3 on the screen. Likewise, when the ungrounded vertical plate is highly negative (1), the spot will be at the top of the screen at point 1; when this plate is at zero potential, the spot will be halfway down the screen at point 18; when this plate is highly positive (36), the spot will be at the bottom of the screen at point 36; when the saw-tooth voltage drops suddenly back to the highly negative value 36 to 1, the spot flies up from 36 to 1 on the screen over a zig-zag path which for simplicity is shown here as a straight line.
path 36-1 in Fig. 8.

The scanning path just described, going from point 1 down to point 36 and then back to 1 again, constitutes one complete normal scanning of the scene. The entire process is repeated for each succeeding scanning.

No television picture signals exist while either a horizontal or a vertical impulse is being sent by the transmitter, hence the appearance of any retrace lines would only cause lines or diagonal streaks in the picture, marring the reproduction. The synchronizing impulses are applied to the control grid of the television cathode ray tube in the receiver in such a way that these impulses drive the grid highly negative, causing almost complete cut-off of the electron beam and thereby preventing either the horizontal or the vertical retraces from showing.

**IMAGE DETAIL**

A consideration of the processes of scanning and reproduction just described should make it clear to you that the video signal exists only while the spot is traveling from left to right along a line; at all other times the television transmitter is sending out pedestals and synchronizing impulse signals. The changes in the intensity of the video signal from one instant to another produce the essential picture detail; the more changes there are per line for an actual given scene being scanned, the greater will be the amount of detail in the reproduction.

Naturally it would be useless to have considerable detail in a single line if there were only a few lines in the complete picture. This means that if greater detail is desired, the number of lines per picture and the number of changes per line must be increased proportionately. Thinking up of a number of square dots, somewhat like the image in Fig. 3, we arrive at the basic fact that it is desirable to have as many dot impressions per inch along a line as there are lines per inch.

**Frequency Range.** We can now consider the maximum frequency involved in a video signal. Since frequency is expressed in terms of cycles per second, we must review the fundamental definition of one cycle:

* A cycle is a complete reversal or change. If the elemental areas along a line of a televised image are alternately light and dark like a checkerboard, it will take two elemental areas to give a change. This means that the shortest cycle in a television image is equal to the time duration of two elemental square areas along a line. We seldom have a perfect checkerboard pattern in television, and hence it may take a longer interval of time—a whole line, half a frame or an entire frame—in order to give the change which constitutes a cycle. We have a maximum number of cycles when the elemental square dots are alternately light and dark, so by assuming this condition to be present we can figure out the maximum video frequency. Let us see what this frequency is.

In this country, 525 horizontal lines are the standard for television. If the picture were square, there would be 525 times 525, or a total of 275,625 square dot elements in this square picture. But television studios use standard motion picture film for some transmissions, and the frames in motion picture film are never square. These frames are always wider than they are high; in fact, they are actually 1/3 wider than their height, so that a projected picture which is 3 feet high will be 4 feet wide. This ratio of width to height is known as
the aspect ratio. The standard 4/3 aspect ratio for motion pictures has also been adopted for television; since this will increase the length of each line by the factor 4/3, we must multiply the value 275,625 by 4/3. This gives us 367,500 square dot elements in a standard television picture.

Television standards also require 30 complete pictures per second. Multiplying 367,500 by 30 gives us 11,025,000 elements per second. Since 2 dots are required to give the shortest possible cycle, we divide 11,025,000 by 2, and get 5,512,500 per second as our maximum video frequency under the conditions so far presented.

This last figure assumes, however, that video signals are being transmitted all the time. We know that this is not true, for about 14% of the transmitting time is used for the horizontal flybacks. This leaves 86% of the total time for video signals, and means that our maximum frequency of 5,512,500 must be sent in 86/100 of a second. We must divide 5,512,500 by .86 giving about 6,485,000 cycles per second as the true maximum frequency for a 525-line image.

Investigation has revealed that for the average size of television image, the maximum frequency can be only about 60% of the value just specified without seriously impairing the quality of the image. Hence, a maximum frequency of 3.9 mc. is satisfactory for some television receivers. However, many high-definition receivers are capable of handling the full video 4.25-mc. band of broadcast television signals.

Theoretically, the lower limit of the video frequency to be transmitted is zero, corresponding to a scene which is all the same brightness (all white, all black, or all the same shade of gray). It is very difficult, if not impossible, to construct apparatus which will handle frequencies from 4.25 megacycles right down to zero, but practical experience has shown that a lower frequency limit of about 10 cycles per second will give satisfactory reproduction of ordinary scenes if the video frequency amplifier in the receiver is properly designed.

![A typical television camera in operation.](image)

**FLICKER**

The human eye is peculiarly sluggish in its response to moving objects, for it continues to see an object even after the object has disappeared. Motion pictures depend upon this persistence of vision characteristic of the human eye; 24 separate still pictures are flashed upon a motion picture screen each second in sequence, but the eye sees a continuous action rather than a series of separate pictures. The eye can detect individual views up to a rate of about 10 pictures per second, but above this value the scenes blend together, accompanied by pulsating light impressions which give the effect of flicker. At about 20 pictures per second the blending of pictures into motion is
almost perfect as far as the eye is concerned; flicker is greatly reduced at this rate but still is not entirely absent. Even at 24 pictures per second, the standard in the motion picture industry, flicker can still be noticed. It is for this reason that motion picture projectors have a shutter in front of the lens which breaks up each still picture into two separate views, giving the effect of 48 pictures per second although only 24 of them are different.

In television, the frequency of the available a.c. power has considerable effect upon the choice of a frame frequency (number of pictures transmitted per second). Since the power line frequency in this country is standardized at 60 cycles, ripple voltages at this frequency or some multiple of it will get into the video signal and the sweep voltages, tending to cause ripple effects, wobbling of the picture, and random movement of bright bands on the image if the number of pictures is increased to 48 or even 72 in order to eliminate flicker. By using a frame frequency equal to some sub-multiple of 60 (such as 30 or 20) or some multiple of 60 (such as 60, 120 or 240), these ripple effects can be removed or at least made stationary so they will be less objectionable. Frame frequencies of 20 or 30 are still too low to eliminate flicker entirely; on the other hand, a frame frequency of 120 pictures per second would increase the maximum frequency of the video signal to an extremely high value. There is left, then, a scanning rate of 60 complete frames per second, which imposes quite a burden upon the transmitting system insofar as maximum frequency range is concerned. With a 525-line image being scanned 60 complete times each second, the upper frequency limit for high definition becomes more than 8.5 megacycles. It is not impossible to make amplifiers which will handle a range of from 10 cycles to 8.5 megacycles, but the cost of these is so high that the production of inexpensive television receivers becomes a serious problem.

How Interlaced Scanning Reduces Flicker

A simple scanning trick which makes the maximum video signal frequency correspond to that of a 30-picture-per-second transmission while still keeping the scanning rate at 60 pictures per second is the solution which television engineers have developed for the problem of flicker. In this system, which is known as interlaced scanning, only half of a picture is transmitted during one complete scanning; the other half is transmitted in the next complete scanning. A simple scheme has been developed whereby lines 1, 3, 5, 7 and all other odd lines are covered during one scanning, and lines 2, 4, 6, 8 and the other even-numbered lines during the next scanning. Two complete scannings are therefore required to cover every elemental dot area on the scene being televised. At the receiver there must likewise be two complete scannings to give a complete reproduction of the image. With interlaced scanning, the frame or picture frequency is 30 per second since that is the number of complete pictures transmitted. For each complete picture the scene is scanned twice, so the field frequency
(vertical sweep frequency) is 60 times per second.*

The two requirements for double interlaced scanning of a given number of lines per second at a given frame frequency are: 1, an odd number of lines per picture; 2, a vertical scanning rate which is twice the frame frequency. This automatically gives scanning of the odd-numbered lines during one vertical sweep and scanning of the even-numbered lines during the next vertical sweep, with odd and even line scanning alternating automatically. An example will best illustrate how this is done; since an example based upon a 525-line image would be too cumbersome, a lower number of lines will be used to illustrate the principles involved.

Suppose that we divide our picture into 10 lines, as shown in Fig. 9A, and that we scan this complete scene 10 times per second (giving a vertical sweep frequency of 10 per second). This means that one complete scanning of the scene, starting at point 1, proceeding to 2, 3, 4, 5, ..., 18, 19, 20, and then returning to 1, will take 1/10th of a second. Assuming that fly-back time is negligible in these examples, we can also say immediately that it will take 1/100th of a second to scan one line, moving from point 1 to point 2 and back to the start of the next line at point 3.

Suppose, now, that we scan this same scene (having an even number of lines), 20 times per second by doubling the vertical sweep frequency without changing any of the other conditions in Fig. 9A. We will still be scanning the same total number of lines per second, and it will still take 1/100th of a second to scan one line, but now only 5 lines will be covered in one complete scanning from top to bottom. Referring to Fig. 9B, the scanning path starts at 1 and goes to points 2, 3, 4, 5, 6, 7, 8, 9, and 10 during one complete scanning of the scene. Vertical fly-back now brings us to point 11 at the upper left-hand corner and we cover exactly this same scanning path for the second scanning of the scene. A television system using an even number of lines per picture cannot secure interlaced scanning by doubling the vertical sweep frequency.

Now let us see what happens when we have an odd number of lines (11) per picture and we use a vertical sweep frequency of 10 per second again, as indicated in Fig. 9C. All 11 lines are covered in one complete scanning, and vertical fly-back takes us directly from point 22 back to the starting point at 1.

Next, suppose we double the verti-
cal sweep frequency, giving 20 complete scannings of the picture per second without changing the total number of lines transmitted per second. This doubles the speed at which the scanning spot is moved downward, so that we will arrive at point x in Fig. 9D (at the bottom of the picture) in exactly the same time it took to reach point x in the middle of the picture in Fig. 9C. In Fig. 9D, however, we have scanned only half the lowest line when vertical fly-back moves the spot up to point y for the following scanning. This time we scan along path 12, 13, 14, 15, 16, 17, 18, 19, 20, 21, and 22, midway between the lines scanned the first time; we are thus securing interlaced scanning of the complete scene. From point 22 the spot goes back to point 1 for the start of the next complete scanning.

Interlacing twice, as illustrated in Fig. 9D, is considered standard practice; to secure this without changing the total number of lines scanned per second (without changing picture detail), the vertical scanning frequency must be twice the frame frequency and there must be an odd number of lines per frame. If the vertical scanning rate is made three times the frame frequency and if the number of lines per frame is not exactly divisible by three, we secure triple interlacing. With the vertical scanning frequency increased to four times the frame frequency, quadruple interlacing can be secured.

Now let us consider interlaced scanning in terms of the standards in use in this country for television. With 525 lines per frame, a vertical scanning frequency (field frequency) of 60 per second, and double interlaced scanning, the total number of lines scanned per second must correspond to that scanned normally with a frame frequency of 30 per second. Multiplying 525 by 30 gives 15,750 as the total number of lines scanned per second in American television systems. This means that the frequency of the horizontal sweep is 15,750 cycles per second, and the vertical scanning frequency is 60 cycles per second.

The detail in the image will correspond to that of 30 complete scannings per second of all lines in a 525-line image.

In an actual modern television system, a few lines at the top and bottom of each picture are blanked out by the blanking signal associated with the vertical synchronizing impulse, for reasons which will be taken up later. The synchronizing impulse itself prevents vertical fly-backs x-y and 22-1 in Fig. 9D from being visible.

**Brightness and Contrast Controls**

It is essential that the television signal which is applied between the control grid and the cathode of the television cathode ray tube shall be **pulsating d.c.** and have a **positive picture phase**, so that synchronizing impulses will cause darkness, and video signals will give various degrees of spot brightness. Another requirement for faithful reproduction of a televised scene is that the pedestals shall all line up with each other at the input to the cathode ray tube despite variations in the brightness of a scene. For example, the pedestals at the output of the video demodula-
the same voltage for a brightly-lighted scene (Fig. 10A) as for a dimly lighted scene (Fig. 10B). Incidentally, with the exception of a reversal in phase, the signals shown in Fig. 10 have essentially the same form as those produced by television transmitters.

Now let us see how a cathode ray television tube reacts to signals of the type shown in Fig. 10 when the pedestals are lined up with each other. Remember that the various anodes in this tube have operating voltages which serve to focus the electron beam to a small spot on the screen, and that the negative voltage applied to the control grid of the tube determines the brilliance of this spot. The control which this grid has upon spot brilliance is more or less linear with respect to the applied grid voltage, except that complete cut-off or darkening of the spot occurs at a definite high negative grid bias voltage. The graph in Fig. 11 shows these facts; note that reducing the negative bias on the control grid (driving it in a positive direction) increases the spot brilliance. Points 2, 3, and 4 are increasingly brilliant, and correspond to increasingly positive control grid voltages. This \( E_g \)-BRILLIENCY characteristic is quite similar to the \( E_g - I_p \) characteristic curve of the average triode vacuum tube.

The negative bias on the control grid of a television cathode ray tube must be so chosen that the pedestals in the applied television signal will be at the brilliance cut-off point (point 1 in Fig. 11) on the \( E_g \)-BRILLIENCY characteristic curve of the tube. Under this condition the video signal will swing the grid more positive than cut-off, giving various degrees of brilliancy, and impulse signals will drive the grid more negative than cut-off (into the blacker-than-

When the video portion of the television signal shown in Fig. 11 is acting on the grid-cathode of the cathode ray tube represented by this characteristic curve, the instantaneous control grid voltage will vary between points 1 and 5 on this curve, and spot brilliancy will vary over the region indicated as B. The impulses associated with this television signal swing the grid beyond the apparent cut-off point (beyond 1) and hence cannot produce a spot on the screen. As long as the pedestals line up with the cut-off point, impulses will not produce a visible spot even with weak video signals, and weak video signals corresponding to a dim line or a dark scene will cause brilliancy to vary in the desired manner over the lower portion of the characteristic curve, such as between points 1 and 2.

Suppose that the television signal in Fig. 11 were applied in such a way that the pedestals lined up with point 2. The video signal would swing the

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**FIG. 10.** The television signal which is applied between the grid and the cathode of a cathode ray tube must have a constant pedestal voltage for scenes with all degrees of brightness, and must have a positive picture phase as shown here, so that the video signal will be positive with respect to the pedestal voltage, and the impulse signals will all be more negative than the pedestal voltage.
up along the curve to point 6, which is quite all right, but the impulses would only swing a small amount beyond cut-off and would not darken the spot completely. As a result, both the line and frame retraces would be clearly visible at the beginning and end of each line and frame. Obviously this is not a desirable operating condition.

Now let us consider another condition, that where the pedestals are beyond cut-off and line up with point 0. Portions of the video signal will now swing into the dark region beyond cut-off, causing dimly-lighted portions of a scene to appear black instead of gray. Obviously this operating condition is just as undesirable as that where the pedestals are to the right of cut-off.

The television signal at the input to the cathode ray tube in a television receiver can be shifted in two different ways in order to make the pedestals line up with the black level (cut-off) of the cathode ray tube. One method involves adjusting the fixed control in a television receiver which changes this bias is commonly known as the brilliancy control, for the most noticeable effect of changing the bias is a change in the brilliancy of the reproduced image. We can also shift the pedestals in one direction or the other to make them line up with the cut-off point by changing the amplification (gain) of one or more stages through which the television signal passes in the receiver. The receiver control which changes gain is commonly known as the contrast control, for its most noticeable effect is a change in the amount of contrast between bright and dark areas of the reproduced image.

We will want to decrease the amplification if screen brilliancy is too great or if the signal is so strong that it drives the control grid of the cathode ray tube positive (this causes the grid to draw current, narrowing the range of frequency response). If receiver amplification is too low, giving us a gray picture with insufficient contrast, we will want to increase amplification until we get the desired contrast between light and dark areas on the picture.

Another requirement for a clear image is that the electron beam be focused to a clearly defined spot of the correct size on the screen. An adjustable control called the focus control is usually provided to correct for errors in focusing due to natural aging of the cathode ray tube or to other causes.

The adjustable controls required in the sight section of a television receiver are thus the brilliancy control, the contrast control, the focus control, and the tuning control. These must be adjusted to give a reproduced image which has the proper brilliancy and the correct contrast between elements along a line, with no line and
in most cases, and vice versa, for there is some interaction between these two controls.

Television Signal Standards

In order for a television system to be successful, the receiver must be easy to adjust, the cost of the receiver must be relatively low, and the transmitter must have as much control as possible over the receiver. This last requirement means that the receiver and transmitter must be interlocked and synchronized. Furthermore, the type of transmission employed must be standardized to a certain extent, for radical changes in the method of transmission might make all existing television receivers obsolete. At the same time, it would not be advisable to set up standards in such a way that it would be impossible to make improvements in the transmitting and receiving circuits. Standards are essential for a successful television system, but these standards must be sufficiently broad to permit future improvements which might make interlock and synchronization more reliable or increase the definition of the reproduced scene.

A set of television standards which takes all these factors into consideration has been approved by the Radio Manufacturers Association (R.M.A.) for television systems in the United States and is required (as far as transmitters are concerned) by the FCC. There is no assurance that these standards will remain as originally set up indefinitely. Changes are bound to come, but immediate changes will not be so drastic as to make receivers obsolete in the near future. Minor changes in transmitters will require little or no changes in television receivers.

1. Television Channel Width; Channel Allocations. The present standards provide for essentially single side-band transmission and reception (partial suppression of one set of side frequencies results in vestigial side-band transmission), for with this method of operation, sufficient detail for a satisfactory image can be transmitted in a definite channel width of 6 megacycles. Twelve 6-megacycle-wide channels have been allocated by the Federal Communications Commission for television transmitters, as follows: 54 to 60, 60 to 66, 66 to 72, 76 to 82, 82 to 88, and seven other channels from 174 to 216 mc. A number of very-high-frequency and microwave channels have been allocated for television relay purposes such as linking the television studio to the transmitter by radio, linking the remote pick-up point to the transmitter by radio, or linking together television stations in different cities and towns to form a network.

2. Video and Sound Carrier Spacing. Obviously the audio and video signals which make up a modern television program cannot be modulated on the same r.f. carrier; each must have its own carrier. By agreement the sound carried must be exactly 4.5 megacycles higher in frequency than the picture carrier. To prevent interference between adjacent television channels or between a television carrier signal and services operating on adjacent carrier frequencies, it has been further agreed that there must be a .25-megacycle-wide guard band at the high-frequency end of each
television channel. All these facts are illustrated by the chart in Fig. 12, which shows a typical distribution of signals in one 6-megacycle-wide television channel.

3. Frequency Relation Between Video and Sound Carried. An example will best illustrate the frequency relationship existing in a television channel. Suppose that the 76- to 82-megacycle channel is assigned to a particular television station. To give the required .25-megacycle guard band at the high-frequency end, the audio signal carrier must be placed at 81.75 megacycles. According to the standards, the video carrier must be 4.5 megacycles lower, or at 77.25 megacycles. Since it is not as yet practical to remove all of the side frequencies below the frequency of the video carrier, a portion of the channel must be provided for those frequencies which cannot be removed. This portion is indicated by the cross-hatched lines in Fig. 12. With this arrangement of a 6-megacycle channel, the frequency range of television equipment can be improved up to a maximum of about 4.25 megacycles without making existing television equipment obsolete.

4. Type of Modulation; Black Level. Negative modulation of the picture carrier signal is standard for the United States. As we have already pointed out, negative modulation means that bright elements of a picture are transmitted at low carrier levels, and dark elements at high carrier levels. The R.M.A. standards further specify that the black level or pedestal level at the transmitter shall be at a definite carrier level which remains fixed regardless of variations in impulse signals or in video signals. The black level at any one point in a television system is the voltage which must exist at that point to give a just barely visible spot on the screen of a properly adjusted receiver.

Furthermore, the d.c. level of the video signal depends on the average brightness of the scene being televised. Hence it is not possible to refer to “percentage of modulation” in a television system in the same way as in a sound a.m. system. In television, the various picture and sync levels are given merely as percentages of the peak carrier output, not as percentages of modulation.

5. Impulse Amplitude. Both line and frame impulses must be transmitted as carrier values higher than unmodulated carrier level (black level). These impulses extend from 75% (black level) to 100% of the peak carrier amplitude. The video signals may vary in amplitude from the black level down to 15% of the carrier level or lower. The general appearance of a typical modulated video carrier signal as it is fed into the television transmitting antenna is shown in Fig. 13. When there is no modulation, the r.f. carrier will have amplitude A, corresponding to the black level. Any increases in carrier amplitude must be for the synchronizing impulses; and decreases in carrier amplitude must be for the video signals. Since we are primarily interested in the impulse and video signals in any study of television, we can neglect the r.f. carrier itself and concentrate our attention on its modulation envelope.

6. Line, Frame, and Field Frequencies. The establishment of standard
values for these three frequencies was based upon the need for high image definition with a minimum of flicker. A vertical scanning frequency (field frequency) of 60 times per second is now standard, for this value minimizes any trouble due to 60-cycle power ripple. (In England, where 50-cycle power lines are used, the field frequency has been standardized at 50 vertical scannings per second.) Since double interlaced scanning is used in the United States, two field sweeps are required to analyze all of the details once in a particular scene; these two vertical or field sweeps constitute a frame (one complete transmission of the picture), and consequently the standard for the frame frequency is 30 frames per second. As we have already seen, there are 525 lines per frame; this means that there are 262½ lines per field. With a 525-line picture being sent 30 times each second, the line frequency becomes 525 times 30, or a total of 15,750 lines per second.

Since it is desirable to obtain both the line and field signals from a single standard-frequency source, the value 525 was chosen as the number of lines per frame because it permits the use of comparatively simple frequency divider circuits in the master synchronizing generator at the transmitter.

7. Aspect Ratio. This ratio has been standardized at 4/3, corresponding to existing motion picture standards and giving a width-to-height ratio of 4 to 3.

8. Synchronizing and Equalizing Impulses; Blanking. The ability of a television transmitter to control the reproduced picture at the receiver depends entirely upon the synchronizing impulses. Many years of research have been spent on this problem, and many different forms of impulse signals have been tried. The standard synchronizing impulses shown in Fig. 14 have been found best suited to present and future requirements of television in this country. Pattern A shows, among other things, the synchronizing impulses recommended for the end of a frame; these will move the spot up to the top of the picture along the retrace path for the beginning of a new frame. Pattern B shows the impulse signal sequence recommended for the end of the first half-frame (the end of the first field); this moves the spot from the bottom to the top of the picture for the beginning of the second interlaced field scanning. A careful study of the diagrams in Fig. 14 will reveal five outstanding characteristics of a television signal:

I. The horizontal synchronizing impulse which is transmitted at the end of each line is not exactly rectangular. The enlarged diagram in Fig. 14D shows the exact shape of this synchronizing signal.

II. The video signal is blanked out for a short interval before and after transmission of the horizontal synchronizing impulse at the end of a line, in order to insure blanking out of the horizontal retrace. The total time for this horizontal blanking shall be about 14% of the time from the start of one line to the start of the

![FIG. 13. Modulated r.f. carrier signal, with the amplitudes varying in accordance with a television signal. A is the unmodulated, and B the peak carrier level.](image)
FIG. 14. Specifications for the standard television signal for 525-line pictures transmitted at the rate of 30 frames per second with double interlaced scanning, giving 60 fields per second. In these diagrams H is the time from the start of one line to the start of the next line, and is equal to 1/15,750 second. The time from the start of one field to the start of the next field is 1.60 second.

Diagrams A and B show blanking and synchronizing signals in regions of successive vertical blanking pulses. The black level is about 0.75 of the synchronizing pulse amplitude.

Horizontal dimensions in these diagrams are not drawn to scale. The receiver vertical retrace next line (this is designated as 14H at the right in Fig. 14D). Note that the horizontal synchronizing impulse occupies about half of this blanking time, and that the front (leading) edge of the impulse is near the start of the horizontal blanking. The two portions of this blanking signal which are on each side of the horizontal synchronizing pulse are known as pedestals, and are originally at the black level.

III. The vertical synchronizing impulse exists for an interval of three lines, but this impulse is divided into six small pulses, each acting for half a line. This serrated pulse is shown in Fig. 14A. Each vertical impulse is divided into six small pulses or serrations in order to maintain horizontal
impulses at all times. These serrations will be explained in detail later.

IV. Six equalizing impulses precede and six follow each vertical impulse period. The purpose of these will also be covered later.

V. The vertical blanking period starts slightly ahead of the first equalizing impulse and extends considerably beyond the last equalizing pulse; this vertical blanking period should take between 5% and 8% of the time for one vertical sweep. Note that horizontal synchronizing pulses are transmitted during the latter portion of the vertical blanking period.

Explanation of Standards. As long as we have 60 vertical sweeps per second, interlaced scanning will continue automatically throughout a transmission. The vertical fly-backs or retraces will be 1/60th second apart; they may occur either near the beginning or near the end of the vertical synchronizing impulse interval, but must occur at the same point in each impulse (this point is controlled by the design of the receiver).

Although the leading (left-hand) edge of the vertical synchronizing impulse in Fig. 14A is directly above the leading edge of the vertical synchronizing impulse in Fig. 14B, these actually occur 1/60th of a second apart due to interlacing. For this reason, the horizontal impulses at A and B in Fig. 14 are not in line.

Experience has shown that no matter what happens, the horizontal or line synchronizing impulses must not stop even for a single line. If the vertical synchronizing impulse were made three lines long without breaking it up, no horizontal impulses would exist for this period. To avoid the situation, the vertical impulse is serrated or separated into six smaller impulses.

To visualize why the vertical impulse must be broken up, let us first assume that it is broken up into three impulses as shown in Fig. 15, and see what occurs under this condition. For the moment we will forget about the equalizing impulses. Pattern A in Fig. 14 shows the last horizontal synchronizing impulse (just before the bottom of the picture) as being one whole line ahead of the start of the vertical blanking period, and pattern B shows this last horizontal impulse as only half a line ahead of the vertical blanking period; these are actual conditions for successive field sweeps, so we must consider them in Fig. 15. Line impulses must exist for the entire vertical blanking period; this means that there should be line impulses at points 2, 3, 4, and 5 in Fig. 15A. At each of these points there is a break or serration in the vertical impulse; since the leading edge of an impulse or serration is sufficient to control the horizontal sweep in the receiver, this will give adequate control of the horizontal sweep.

When we turn to pattern B in Fig. 15, however, we find that horizontal impulses should occur at points 2, 3, and 4. There are no steep leading edges at these points to control the line sweep, and consequently three serrations in the vertical impulse are not adequate for pattern B, which occurs for every other scanning of the picture. If the vertical impulse is divided into six parts as shown in Figs. 14A and 14B, we secure the desired steep front at points 2, 3, and 4 in pattern B in Fig. 15.

![FIG. 15. These diagrams tell why the vertical synchronizing impulse signal must be broken up into six smaller impulses.](image-url)
The vertical synchronizing pulse is chopped into segments by the application of a special signal having a rate twice that of the horizontal synchronizing signal. Because of the difficulty of synchronizing this signal exactly with the vertical pulse, this twice-normal signal exists somewhat before and after the vertical pulse as a series of horizontal synchronizing pulses at half-line intervals. Then, it is sure to properly cut up the vertical pulse. In Fig. 14A, these additional pulses are labeled "equalizing pulses." A pulse one-half a line from the proper one is ignored at the receiver; the sweep oscillator responds only to the pulse that occurs at the proper time to maintain the horizontal synchronization.

Television Receiver Circuits and Controls

Let us imagine that a radio wave having characteristics similar to those shown in Fig. 13 is being broadcast by a television transmitter, and consider just how this would be received and converted into an image by a television receiver having the essential sections shown in Fig. 16. First of all, the receiving antenna must be designed for efficient operation at the very-high frequencies employed in television. The radio waves radiated by the transmitter are generally polarized horizontally, and can therefore be picked up by a horizontal doublet antenna. Ignition interference from automobiles is a serious problem in cities and towns, and consequently the horizontal pick-up section should be located as far as possible from these sources of interference. For the same reason, the vertical transmission line must be shielded or otherwise designed so as to prevent pick-up of vertically-polarized interference signals. The antenna and its transmission line must be designed for efficient operation in the desired television channels and must have reasonably flat response over the entire 6-megacycle channel occupied by one station.

Either a tuned-radio-frequency circuit or a superheterodyne circuit could be employed for the r.f. amplifier section of a television receiver (ahead of the video demodulator), but when sight and sound signals are transmitted on carriers only 4.5 megacycles apart, good selectivity is highly essential. Superheterodyne circuits will provide this required selectivity and the necessary video pass band with a minimum number of stages, and consequently the superheterodyne circuit is used almost exclusively for simultaneous sight and sound (television) reception.

A practical superheterodyne circuit for a television receiver will have a preselector amplifying stage or at least one tuning circuit ahead of the mixer-first detector. This preselector must have essentially flat response over a 6-megacycle band, and must have sufficient selectivity to keep out television signals in other channels which could produce image interference.

To produce an i.f. signal, we naturally require a local oscillator; this must be reasonably stable in frequency. Since we feed two different r.f. carrier signals (one for sight and the other for sound) into the mixer-first detector, we get out two i.f. signals, one carrying the picture modulation and the other the sound modulation.

The local oscillator frequency will be higher than both the sound and
picture carrier frequencies, according to the accepted standards. The video i.f. value will be about 26.4 mc.; this means that if the incoming video carrier is 77.25 mc., the local oscillator frequency will be 103.65 mc. Since the sound carrier is 4.5 mc. higher than 77.25 mc., or is 81.75 mc., the i.f. value for sound signals will be the difference between 103.65 and 81.75, or an audio i.f. value of 21.9 megacycles. By using two separate i.f. channels, one tuned to 21.9 mc. and the other to about 26.4 mc., we automatically separate the two signals. Notice that the video and audio i.f. signals differ by 4.5 megacycles, just as did the video and audio carriers. Since the maximum deviation of the f.m. audio i.f. signal is 25 kc., this i.f. amplifier can be quite selective as compared to the preceding circuits. Too much selectivity is not desirable, however; the sound i.f. channel must be sufficiently broad to allow for a certain amount of drift in the local oscillator frequency.

You will recall (Fig. 12) that the entire band of side frequencies above the video carrier frequency is transmitted; this upper sideband extends for 4.25 megacycles according to present-day standards, so we have side frequencies in the range from 77.25 megacycles to 81.50 megacycles in the example we are analyzing. In the video i.f. amplifier this band will extend from 26.4 mc. (the video i.f. carrier value) to 22.15 mc. The video i.f. amplifier should therefore be flat in response from 22.15 mc. to 26.4 mc., and the sound i.f. amplifier which is tuned to 21.9 mc. must be sufficiently selective to eliminate 22.15-mc. video i.f. signals.

The sound i.f. amplifier will be followed by limiter stages (if used), a discriminator, an audio amplifier, and a loudspeaker, and will probably have automatic volume control on its i.f. stages. Since this sound i.f. amplifier is highly selective and is interlocked with the video i.f. amplifier through the common oscillator and mixer-first detector, the tuning of a television receiver simply involves tuning for maximum volume and clarity of the sound signal; this automatically tunes in the video signal properly.

Automatic gain control (a.g.c.) is a very desirable additional circuit in a television receiver. Like automatic volume control in an ordinary sound

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**FIG. 16.** Essential sections in a modern cathode ray television receiver. The amplitude separator is also known as the clipper, and the impulse separator is often called the frequency separator.
receiver, a.g.c. compensates for fading and also serves to supply the demodulator with an essentially constant signal. Of course, normal fading due to interaction between ground and sky waves does not exist in a television system, but it is perfectly possible for an effect like fading to occur due to swaying of the receiving antenna or the transmission line in the wind, or to reflection of radio waves by a moving automobile in the path between transmitter and receiver. If there are two or more television stations in a given locality, one may provide a stronger signal than the other at a given receiving point, causing different signal levels at the demodulator. Automatic gain control can compensate for all these effects. The circuits associated with or following the video detector are in some cases adjusted according to the amplitude of the television signal; this means that for reliable reception it is important that the video demodulator be fed with a reasonably constant signal. This condition can be met if automatic gain control is provided.

In sound receivers, the a.v.c. system is actuated by the average carrier level; in a television system, however, the average carrier level varies with the nature of the video signal being transmitted at any instant. The one fixed characteristic of a television signal is the black level; for a given station this is fixed and corresponds to a definite carrier level. The synchronizing impulses, which are transmitted at amplitudes above the black level, are likewise fixed from instant to instant, so by feeding the television signal from some point in the receiver where the pedestals line up with each other (such as at the output of the video demodulator) and using a filter which makes the output follow the peaks of the impulses, we can secure for the a.g.c. system a d.c. voltage whose value varies with the true carrier level of a television transmitter.

The video demodulator will usually be followed by one or more video-frequency amplifier stages. The exact number of stages used depends upon the amount of gain required and the phase of the television signal at the input of the video amplifier. Stability becomes another important consideration when a large number of stages is employed and amplification at low video frequencies is essential. Video-frequency amplifiers will invariably be of the resistance-capacitance type, capable of providing uniform amplification for all signals from about 10 cycles up to about 4 megacycles. These amplifiers should really be of the d.c. type, in order to maintain the d.c. characteristic of the television signal, but this is impractical. A d.c. amplifier which will provide the required gain is not only excessively high in cost but also has a tendency.
to produce undesirable low-frequency oscillations. A conventional a.c. amplifier with resistance-capacitance coupling is ordinarily used instead; with this, the d.c. component is temporarily removed from the signal, and we amplify an a.c. signal.

Up to the video demodulator stage, the television signal is a modulated carrier having negative modulation, in which the video components of the television signal are negative in voltage with respect to the black level. The television cathode ray tube requires a modulated d.c. signal with a positive picture phase, in which the video components of the signal voltage are positive with respect to the black level.

The television signal can be removed from a diode detector as a d.c. signal with either positive or negative picture phase, as desired. We have this additional fact to keep in mind, that a resistance-capacitance coupled amplifier will reverse the phase of a signal voltage (reverse the picture phase). This means that if we supply to the diode detector a carrier signal having negative modulation, and remove from the detector a positively-modulated d.c. signal (a signal with a positive picture phase), we must use two video amplifier stages in order to secure a positive picture phase for the cathode ray tube. If we remove the television signal from the detector as a negatively modulated d.c. voltage (a signal with a negative picture phase), we must use either one or three video amplifier stages to give the proper phase at the input of the cathode ray tube.

If the video amplifier amplifies only the a.c. component of the television signal, as is usually the case, a d.c. restoring circuit should be used just ahead of the television cathode ray tube to restore the d.c. component. This d.c. potential must be restored in such a way that the pedestals will all line up with each other again, for they may be thrown considerably out of line by the video amplifier stages. All of the components in the television signal, including the video signal itself, the horizontal and vertical synchronizing impulses, the equalizing impulses and the pedestals, are applied to the control electrode of the cathode ray tube.

In order to make the electron beam in the cathode ray tube sweep both horizontally and vertically, we need two saw-tooth sweep oscillators; these must be of such a nature that they can be controlled by the horizontal and vertical synchronizing impulses in the television signal. The impulses must be separated from the video signal before they can be applied to these sweep circuits; this is accomplished by the stage known as the synchronizing separator, the amplitude separator, or the clipper. The television signal voltage which is fed into the amplitude separator must be a modulated d.c. signal voltage with the pedestals lined up. When this signal is fed into a negatively-biased diode tube or into a triode tube which is negatively biased so that only the impulses can get through, the desired separation of impulses from video signals is secured.

After the impulses have been separated from the video signal, there will remain the problem of separating the horizontal impulses from the vertical impulses. A separate circuit is required for this job; this circuit is called the impulse separator or frequency separator, and supplies the synchronizing impulses to the line and frame sweep oscillator circuits.

Power packs are an essential part of a television receiver, since the
various tubes used will require both a.c. and d.c. operating voltages.

Adjustments must be provided in the video section of a television receiver for controlling the amplitude separator, the impulse separator, the horizontal and vertical sweep oscillators, and the d.c. restoring circuit; these controls may be of the screw-driver type, however, for once they are set properly, they will remain in adjustment for long periods of time. There is also need for a control which will adjust the beam of the cathode ray tube to the exact center of the screen when no sweep voltages are applied to the horizontal and vertical deflecting plates, for even with modern tube-making machinery it is not possible to align the various electrodes with sufficient accuracy to keep the spot at the exact center of the screen. Centering of the spot is accomplished with simple circuits which introduce adjustable biasing voltages in series with the horizontal and vertical deflecting plates. The size of the picture and the aspect ratio can be varied by changing the magnitudes of the sweep voltages.

Although the controls just described are essential for preliminary adjustment, they are rarely if ever used by the owner of a television receiver. The essential picture controls which require adjustment by the owner include a tuning control (either a manually rotated knob or a push-bottom tuning system) for simultaneously tuning in the video and audio signals from the desired station when there are several television stations in a locality, a focus control, a contrast (gain) control for the video amplifier to adjust picture contrast, and a background brilliancy control which adjusts the d.c. bias applied to the control grid of the cathode ray tube and thereby places the pedestals of the television signal at the cut-off point on the grid voltage-brightness characteristic curve of the tube. In addition, the sound section of the receiver will have a volume control and sometimes a tone control. The on-off switch is usually combined with the volume control.

LOOKING FORWARD

In this first introductory Lesson on television, we have surveyed the important needs of a television system. In some cases brief explanations of these needs have been given, and in other cases we have simply made statements because the explanations would be lengthy and not essential to the clearness of this "bird's-eye view" of the entire modern television set-up. The various methods for producing saw-tooth sweep signals, for providing interlocks and for separating impulse signals will all be taken up in later Lessons, along with typical circuits for the various other sections described in this Lesson.
Lesson Questions

Be sure to number your Answer Sheet with the number appearing on the front cover underneath the Lesson title.

Place your Student Number on every Answer Sheet.

Send in your set of answers for this Lesson immediately after you finish them, as instructed in the Study Schedule.

1. What is a scene converted into by the process of scanning?

2. If the size of a television image is increased, should you move closer or farther away from the receiver to see a properly blended picture?

3. What three important signals must be transmitted on the picture carrier in an electronic television system?

4. One requirement for double interlaced scanning of a given number of lines per second at a given frame frequency is a vertical scanning rate which is twice the frame frequency. What is the other requirement?

5. What three adjustable controls are required for the sight section of a modern television receiver in addition to the tuning control?

6. In a standard 6-megacycle wide television channel, what is the frequency relationship between the sound carrier and the picture carrier? (State how many megacycles higher or lower the sound carrier is than the picture carrier.)

7. What is meant by negative modulation of the picture carrier signal?

8. Why is each vertical synchronizing impulse divided into six serrations?

9. If the video amplifier in a television receiver amplifies only the a.c. component of the television signal, what circuit should be used just ahead of the television cathode ray tube?

10. What should be the nature of the television signal voltage which is fed into the amplitude separator section of the television receiver?
Dear Mr. Smith:

My NRI training gave me confidence to apply for a job as a radio repairman with a large concern here. Later I became Radio Service Manager of the firm. Now I am Chief Engineer of a broadcast station, in charge of four assistant engineers. I owe all I know about radio to NRI.

C.J.B., South Carolina

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Equipment Used by Servicemen

Equipment for testing suspected parts is something every radio serviceman, beginner or expert, must have. You do not need to buy any test equipment immediately, however, for your second RK Kit will contain parts and instructions for building a multimeter you can use to get started in servicing. Later, profits from your early servicing jobs can be used to buy professional equipment. The purpose of this booklet is to describe the various test instruments you will eventually need and to show you how to use them. We shall give most space to the all-important multimeter, in preparation for the Booklets on testing radio parts, but there will also be sections on other instruments.

The Multimeter

The multimeter is a four instruments in one, consisting of a voltmeter to measure d.c. voltage, a voltmeter to measure a.c. voltage, an ohmmeter to measure resistance, and a milliammeter to measure current. Some multimeters have, in addition, an ammeter to measure large values of d.c. current.

Widely different voltage, current, and resistance values exist in radio receivers. D.C. voltages may range from a fraction of 1 volt to as much as 400 volts; a.c. voltages from 2 to 700 volts; resistances from a fraction of an ohm to as much as 20,000,000 ohms (20 meg-ohms). It is impossible to read such widely different values on a single range, so most multimeters have a
system of overlapping ranges to provide full coverage
of the values to be read.

One of the major differences between multimeters is
the means used to convert them from one use and range
to another. Some use a series of jacks into which the
test leads are plugged, others use selector switches,
push buttons, or combinations of these methods.

Several typical multimeters, all of which are ade-
quate service instruments, are shown in Fig. 1. Whate-
ver type you choose, you must know three things be-
fore you can use the instrument to test a circuit or part:
1, where to connect it; 2, how to read the meter scales;
and 3, how to interpret the readings you get.

Interpreting the readings is sometimes simple, some-
times difficult, depending on what you are testing. You
will learn all about this important subject in future les-
sons and in other RSM Booklets. Let us concentrate
here on how to handle the equipment, how to connect it
to obtain proper readings, and how to read the meter.

HOW TO READ MULTIMETER SCALES

Before you connect a multimeter to anything, you
must be sure you can read the meter. The pointer moves
over a card on which are printed the various scales pro-
vided for the meter. Reading such a meter is really less
difficult than telling time by a clock, once you have had
a little experience.

Here is the right way to read a meter. Figs. 2A, 2B,
and 2C show three typical meter scales. They could be
for either voltage or current values.

Naturally, you have no trouble reading the values
that are marked, but there is not room enough to place
the proper numerical value opposite each division on
the scale. Thus, you must find out what each division
represents before you can read values that fall between
the numbers. To do so, count the number of division
lines between any two marked divisions, starting with
the line after one marked division and continuing
through the next marked division. Then, divide this
number into the numerical difference between the two
divisions. This will give you the value of each division.
In Fig. 2A, for example, there are ten divisions from

the one marked 20 up to and including the one marked
30. The numerical difference between 20 and 30 is also
10. Hence, each of the line divisions represents 1 (10
divided by 10 equals 1). If you want to find, say, 23 on
this scale, you need only count three divisions past 20.
Similarly, 12 is two divisions past 10.

In counting the marks, you will find that every fifth
one is a heavier (thicker) line. This makes it easy to
find points like 5, 15, 25, etc. Practice on this scale by
finding various values.

In Fig. 2B, we have a somewhat different scale. Let's
see what each division represents, following the rules
we just developed. Between 50 and 100 there are ten
marks (including the mark for 100), and the numerical

FIG. 1. Typical multimeters made by (top left and right) Triplet;
(bottom left) RCA; and (bottom right) Weston.
difference between 50 and 100 is 50. Dividing this difference by the number of scale divisions, we find that each division represents 5 (50 divided by 10 equals 5). Thus, 65 is three divisions past 50, 205 is one division past 200, etc. Notice that every other division is made longer—so that it is easy to find numbers like 60, 70, 80, and 90.

Now see if you can figure out the value of the main divisions and the value of each scale division in Fig. 2C before reading on.

Using the same method as before, we find the difference between two numbered values, such as 60 and 90. The difference between these is 30. There are fifteen divisions from 60 up to and including 90. Dividing 30 by 15 gives 2, so each division represents 2. Thus, reading each division from 60, we have 62, 64, 66, 68, 70, 72, 74, 76, 78, 80, 82, 84, 86, 88, and 90. The two heavy division lines are at 70 and 80.

**Reading In-Between Values.** Once you know what each division on a scale represents, it is easy to estimate readings with the pointer in between two divisions.

Suppose, for instance, the meter pointer moved to a position halfway between 90 and 92 (the first division to the right of 90 in Fig. 2C). Although there is no division line there, you know the reading must be 91 (since 91 is halfway between 90 and 92).

As we will show, you don’t need to read a meter too closely for service work. In fact, it is all right to estimate meter readings roughly when the pointer does not fall directly on a division line. Close meter readings are unnecessary because the value of voltage, current, or resistance, in most cases, may be off as much as 20% from the rated value without affecting the operation of the circuit very much.

**Multiple Ranges.** In Fig. 3 we have the same scale as in Fig. 2C with a new 0-75 range added. (It is very common to find two or three ranges used with each scale in a multimeter.) To find the value of each scale division with the new range, proceed exactly as before, forgetting all ranges except the one in which you are interested.

There are fifteen divisions between 30 and 45 (on the 75-volt range), and the numerical difference between 30 and 45 is also 15, so each division represents 1. Thus, to find 34 on this range, you would count four divisions past 30. Each heavy division line represents 5.

**Scale Multiples.** Here is another point you should understand clearly. In Fig. 2A we have a scale marked 0-to-50, but the multimeter using it may have a 0-500 range in addition. Will there be another scale of 0-500? No, because this would unnecessarily clutter up the meter dial. You can use the 0-50 scale for the 500 range simply by mentally adding a zero to each reading. This

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**FIG. 2.** D.C. voltage and current scales of multimeters frequently use markings like these.

**FIG. 3**
is the same as multiplying each reading by 10. Thus .0 becomes 100, .20 is 200, .30 is 300, .40 is 400, and .60 is 600. The “in-between” values are similarly stepped up; each division now represents 10 instead of 1.

In much the same way the 0-150 scale in Fig. 2C may be used for 0-15. Here you should “knock off” a zero from your reading or, as we say, “move the decimal point one place to the left.” Thus each division, formerly equal to 2, is now equal to .2 (2/10), and the 30, 60, 90, etc. readings now become 3, 6, 9, etc. Starting at 0 and going to 3, the values now are .2, .4, .6, .8, 1, 1.2, 1.4, 1.6, 1.8, 2, 2.2, 2.4, 2.6, 2.8, and 3.

The scales in Figs. 2 and 3 are called linear scales because the divisions are spaced equal distances apart. That is, the distance between 100 and 150 in Fig. 2B is the same as that between 200 and 250 or between 0 and 50 on this same scale.

This is not always true, particularly in the case of ohmmeter scales on which the readings will be crowded or bunched at one end of the scale. This is clearly shown in Fig. 4A; here the readings are spread out on the right half of the scale and are bunched together at the left-hand end. Such a scale is read just like any other, but you must determine the values of the divisions in the region where the reading is being taken, since all the divisions do not have the same value. For example, from 1 to 2, there is one in-between mark, which must be 1 1/2. From 10 to 20, there are ten divisions, so each must equal 1. From 20 to 50 there are six divisions, so each equals 5. Thus, you have to determine the division values for the section of the scale you are reading.

The scale in Fig. 4A is for a “series-type” ohmmeter. Its scale has zero (0) at the extreme right, the reverse of the usual voltage or current scale. This scale is marked 0-300, but on a typical multimeter using this scale the ranges actually are 300,000 ohms, 3,000,000 ohms, and 30,000,000 ohms, so when reading the meter, you must add to your reading the correct number of zeros for the range being used. For the 300,000-ohm range you add three zeros (000), the 3,000,000-ohm range calls for the addition of four zeros (0,000) to your reading, and five zeros (0,000) are required when you use the 30,000,000-ohm range. (The 30,000,000 ohm value is 30 megohms, so you can read the 0-300 scale in megohms by dropping a zero. Thus, 300 at the extreme left is 30 megohms; 10 is 1 megohm; 5 is .5 megohm, etc.)

On some ohmmeters the “low-ohm” range is provided by a shunt-type ohmmeter. A scale of this kind is shown in Fig. 4B. Notice that zero on this scale is at the left hand end. However, the modern trend is to use a series type ohmmeter even for the low-ohm range; in this type of ohmmeter, all ranges have zero at the right-hand end of the scale.

A Typical Multimeter Scale. Fig. 5 shows a typical multimeter scale. The scale at the bottom is a dual scale for d.c. voltage and current. The next dual scale is for a.c. voltage (this scale is usually colored red on the standard dial to help make it stand out from the others). The top scales are for the resistance ranges.

Although there appear to be a number of markings on a dial of this kind, you don’t have to worry about any of the markings except the ones on the scale that you happen to be reading at the moment. With practice you will soon learn to disregard all other scales.

In Fig. 5, notice that the voltage scales are 0-12 an 0-30. On an actual meter of this kind, the voltage range may be 0-3, read on the 0-30 scale by moving the decimal point one place to the left; 0-12 volts, read directly.

![Fig. 4](image-url)
on the meter; 0-30 volts, read directly on the meter; 0-500 volts, read by adding a zero to the 0-30 scale; and 0-1200 volts, read on the 0-12 scale by adding two zeros to each reading.

Once you build the tester for the experiments (or obtain a test instrument) and practice a little, you will find it surprisingly easy to read meter scales. ALWAYS READ THE SCALE FOR THE PURPOSE AND THE RANGE YOU ARE USING.

MULTIMETER CONNECTIONS

Now that you have learned something about reading meter scales, let us see how to connect a multimeter in order to get readings. Fig. 5 shows a typical modern multimeter.

A pair of test leads, one red and one black, are used to make connections between the tester and the circuit or part under test. One end of each lead is fitted with a pin connector which is plugged into the jacks mounted on the multimeter. The other end of each lead has a large insulated probe, which is used to make connections in the circuit.

At the right of the meter, there are four connecting jacks, colored (in order from top to bottom) red, black, red, and black. The bottom (black) jack, labeled "COM" as the "common" jack that is used for one connection in all uses of the multimeter. The black-colored test lead is always plugged into this jack. Then, the red test lead is plugged into one of the other three jacks, depending on what is to be checked.

For most purposes, the red lead is plugged into the red jack next to the bottom—the one labeled "V-O-MA." This is the jack for making voltage, resistance, and current measurements. The other two jacks are for special purposes that will be explained elsewhere in your course.

Suppose you want to use this multimeter as an ohmmeter. First, plug the black test lead into the "COM" jack and plug the red test lead into the "V-O-MA" jack. Next, turn the selector knob to the desired ohmmeter range. There are four ohms ranges—2000 ohms, 20,000 ohms, 2 megohms, and 100 megohms. The names of these ranges come from the highest values that can be read on each. Thus, you should always choose a range that is higher than the resistance you want to read.

To calibrate the ohmmeter properly, touch the tips
the test probes together, and turn the lower left-hand knob (labeled “OHMS ADJ.”) until the meter reads zero at the right-hand end of the ohms scale. The instrument is now ready to be used as an ohmmeter.

**Continuity Testing With an Ohmmeter.** When we say that a circuit or a part has continuity, we mean that there is a continuous metallic or conductive path for the flow of direct current through the circuit or part. A circuit does not have continuity when an “open” (a break) occurs, for then the metallic path is not complete.

Elsewhere in your Course, you learned that the series ohmmeter consists of a voltage source (either a battery or a power pack) and a meter in series. If the ohmmeter test probes are held together, the voltage sends current through the test probes and through the meter, causing the meter pointer to move to a full-scale position, indicating that there is no resistance between the test probes. Thus, on the series-type ohmmeter, zero resistance between the test probes causes a “full-scale” reading. When the probes are separated or “open,” there is no deflection—the pointer remains at the left of the scale.

Now, when the test probes are held on the terminals of a part having continuity, the battery causes current to flow through both this part and the meter. Because of the resistance of the part being tested, the current flow is less than when the test probes are held together, so the meter pointer deflects to some position other than that for zero resistance. The higher the resistance of the part, the less the current that will flow and the less the meter pointer will deflect from its “open circuit” position. If the part has no continuity (is open) the pointer does not move from the “open” position because no current can flow through the break.

Notice that the ohmmeter has two uses: 1, it indicates whether the part (or the circuit) has continuity; and 2, if the scale is calibrated properly, it shows the resistance of the part.

- When you are testing for continuity, you should use one of the higher ohmmeter ranges; you will then get a deflection regardless of the part resistance if the part has continuity. For example, when a resistor is being tested as in Fig. 7A, the ohmmeter test probes are touched to the resistor terminals. The ohmmeter battery sends current through the resistor, and the meter needle deflects to some position on the scale, the exact position depending on the resistance value.

When the ohmmeter probes are placed on the terminals of a defective resistor as in Fig. 7B, no current can flow because the circuit is open (the resistor is broken), and there is no deflection of the meter needle.

In Fig. 7B, the break in the resistor is visible. In practice, however, it is rare to find a part that is visibly defective. Furthermore, in a radio receiver, parts are frequently concealed by shield cans or other parts. In such cases, you could not possibly see a break, so you could check only by means of test instruments. **This is one of the most important reasons for using an ohmmeter.**

- Much of your continuity checking inside a radio set will be between what radio men call “reference points”—tube sockets, the chassis, and the high voltage terminal of the power supply, for example. A number of parts may be checked at one time by making ohmmeter readings between these points. If continuity is found, all the parts being checked are at least temporarily cleared of suspicion. You will learn much more about the use of reference points later in your Course.

**Resistance Measurements.** Continuity tests are
made merely to find out if a complete circuit exists. You don’t try to read the meter—you just look to see if the pointer moves. Often, however, you will want to determine the exact resistance of a part or of a whole circuit. For example, you may find continuity through a short-circuited part, but the resistance of the circuit in which the part is used will be lower than normal.

You can measure resistance, as well as check continuity, with the ohmmeter section of your multimeter. However, there are certain precautions you must take. In later Booklets, we will show you how to test individual parts and circuits; you will learn the “do’s and don’ts” of resistance measurements. If, right now, you have a fair idea of what continuity testing means, you are making real progress.

**D.C. Voltage Measurements.** A serviceman uses a d.c. voltmeter almost as much as he does his ohmmeter. Multimeters have a number of voltage ranges so that they can be used to measure voltages of widely different values. The meter shown in Fig. 6, for example, has five d.c. voltage ranges. To use the instrument as a d.c. voltmeter, you should put the test probes in the same jacks as for the ohmmeter, with the black lead in the “COM” jack; then turn the center selector switch to the desired range.

Since some defect of the circuit you are checking may create an unexpectedly high voltage, always start with the highest voltage range first, then shift to lower ranges when the readings indicate it is safe to do so. Memorize this rule to safeguard your meter. Make a habit of turning the range selector switch to the highest d.c. voltage position (or the “OFF” position) immediately after completing the measurements; this will prevent your accidentally making another test later with the selector set to a low range.

You have learned from your Lessons in Fundamental Radio Principles that voltage exists between two points. In other words, you can’t connect just one voltmeter probe to a single terminal and obtain a reading; both probes must be used, and they must be connected to points of different potential.

For example, each radio tube has a plate and a cathode, and there is a voltage between these elements that radio men call the plate voltage. To measure this plate voltage, you connect the negative (black) voltmeter probe to the cathode socket terminal, and the positive (red) probe to the plate socket terminal. (Take care that the black and red probes are in the proper test jacks, or the meter pointer will swing the wrong way.) With the proper test probe connections and the proper range, you will read the plate-to-cathode voltage on the meter.

Later you will receive an RSM Booklet on voltage measurements and will learn just how to make measurements, what to expect, and how to use the results of your tests.

**A.C. Measurements.** To measure a.c. voltage with a meter like that in Fig. 6, set the center selector switch on the multimeter to the proper a.c. range, and touch the test probes to the points between which the voltage is supposed to exist.

Servicemen frequently measure the a.c. output signal voltage. However, in a receiver operated from the power line, the only other a.c. voltages are those used to heat the filaments of the tubes, and the high a.c. volt-
FIG. 8. The wrong and right ways to measure current. You can damage the pointer of a milliammeter, or even burn out its coil, if you connect the meter across some part in the circuit. Even if the meter is not harmed, you will not get a true reading of the circuit current. Always connect a current meter in series with the circuit, and be sure to start with the instrument set for its highest range.

age in the power pack (which is changed to the high d.c. voltage required to operate the other tubes).

Filament voltages are rarely measured, except in a.c.-d.c. universal receivers, for little can go wrong with the usual filament circuit. The high a.c. voltage applied to the rectifier is seldom measured because it must be all right if any tube has the correct d.c. voltage. Sometimes, however, you'll want to measure the line voltage. This is done by inserting the test probes into the wall socket holes.

As in making other voltage measurements, always use the highest range of your meter first, switching to a lower range if necessary.

Current Measurements. It takes time and some unsoldering to make current measurements, for, as you learned from your regular Course, the meter with which the current measurement is to be made must always be inserted in series with the circuit so that the circuit current will flow through the meter. Failure to observe this precaution may result in a burned out meter, or at least in a bent meter pointer. Fig. 8 shows the right and wrong ways to measure current.

You should always use the highest current range first, switching to a lower range if necessary. Even so it is a good idea to find out if the current is abnormal; high before making any current measurement. Test with an ohmmeter and with a voltmeteter will disclose such a condition and will show you if current measurements can be safely made. However, these same ohmmeter and voltmeter tests usually eliminate all need for current measurements in practical service work, as you will learn in other RSM Booklets.

Don't Burn Out Your Meter. The meters used in tes instruments are fairly rugged, but in most multimeter too much voltage or current will burn out the meter coil or bend the pointer by making it hit the stop too hard. You can avoid such an experience by always remembering the following:

1. Using the ohmmeter.
   a. The ohmmeter cannot be damaged unless the circuit under test is alive. Disconnect the set from it power source by pulling the plug out of the wall outlet before making measurements. (If it is a battery set, disconnect the batteries—don't just turn off the set.) A charged condenser can furnish enough current to damage an ohmmeter, so wait a few moments after disconnecting the set from it power source to let the charge leak off the condensers.

2. Using d.c. and a.c. voltmeters.
   a. Always use the highest range of your meter first switch to a lower range if it is safe to do so.
   b. Know what you want to measure and where to place your probes.
   c. If the meter pointer comes up with a rush and looks as if it will go off-scale, take one or both test probes off the circuit quickly—use a higher range since the one in use is too low.
d. Don't try to measure a.c. voltage with the selector set at a d.c. position. The meter will not read, but, if the a.c. voltage is higher than the meter range employed, the meter coil will burn out.
e. When you are through using some range of a multimeter, always reset it to the highest d.c. voltage value, or to the OFF position if there is one. If you don't form this habit, you may leave the multimeter set to an ohmmeter range and try to measure voltage. This will ruin the meter.
f. If the meter starts to read down-scale, reverse the test probes.

3. Using the milliammeter for current measurements.
   a. Don't make current measurements when voltage and ohmmeter measurements will do.
   b. Before connecting the milliammeter, satisfy yourself that the circuit has no defects that will cause excess current to flow.
   c. Break the circuit so the meter can be placed in series with it. Never connect a current meter across a radio part or across a voltage source. (Don't try to measure the "current" of a battery or of a power line.)
   d. Always start with the highest range of the meter, being ready at an instant's notice to remove one or both of the probes if the meter needle shows signs of going off-scale. If the first range is too high for you to read easily, move the meter switch to a lower range.
   e. If the meter starts to read down-scale, turn off the circuit, reverse the test probes, and turn on the circuit again. As you know, all circuits in which current measurements are made have a source of voltage. The meter must be placed in the circuit so that electrons will enter its negative terminal, hence the positive meter probe goes to the positive side of the voltage source.

THE SIGNAL GENERATOR

As its name suggests, the signal generator supplies or generates a radio signal. It is a miniature broadcasting station but does not produce (radio men say "is not modulated by") words or music. Instead, it has a steady low-pitched tone that will be heard when its signal is tuned in on a receiver.

A signal generator has a dial similar to that of a receiver and can be tuned like a receiver. You are probably familiar with all-wave receivers that pick up short wave as well as broadcast-band stations. Such receivers have a band switch to change from one wave band to another—signal generators are similarly equipped.

Signal generators are used to adjust (or align) receivers so that stations will come in at the proper point on the receiver dials, and to adjust receivers so the weak, far-away stations can be heard with good volume. Sometimes signals from broadcast stations can be used for this purpose, but in many cases they will not do so plan to get a signal generator eventually.

b. Signal generators have another important use. If you have a dead receiver, you already know that the troub
s caused by a defective part that kills the action in one stage. The rest of the stages may be all right, and if you can find the bad stage, the job is almost half done. The signal generator helps in this. Just inject its signal into the various stages one at a time, working back a stage at a time from the output stage towards the antenna. As long as the signal tone is heard in the loudspeaker, all stages from the point of signal injection to the loudspeaker are in working order. When you pass through the dead stage, the speaker will be silent.

From your Lessons and the RSM Booklets, you will earn how to identify stages, how to tune your signal generator, and how to connect a signal generator to the different stages in a receiver.

**TUBE TESTERS**

There is probably more difference between tube testers than between other pieces of service equipment. Some are rather simple in the tests they perform, while others will make a more complete test of the tubes. In general, the more elaborate the tests that can be made, the more expensive the tester. However, elaborateness of tests is not always desirable—manufacturers are constantly bringing out new tubes. Sooner or later, tubes are developed that the tester cannot test without redesign. When enough tubes like this have been brought out, the serviceman is forced to junk his tester and buy a new one.

For this reason, the alert serviceman chooses a simple, inexpensive tube tester that will check tubes for shorts or undesired resistance (leakage, servicemen say) and for emission (the ability of a tube cathode to give off, or emit, electrons). Modern testers of this type are made with controls and circuits that are adaptable to a wide variety of conditions. The better ones have individual selector switches for the various tube elements so that practically any arrangement of elements can be handled. With testers of this kind, only tubes with radically different sockets or with remarkably different characteristics require new instructions or changes in design.

Even so, it is advisable to put off the purchase of a tube tester just as long as possible, then to purchase the very latest style. (It is never advisable to waste money purchasing some out-of-date, second-hand tester.) Remember, the multimeter and the signal generator are the most necessary of the basic instruments. At the beginning, you can get radio dealers or parts distributor to test tubes for you, or you can substitute good tubes for defective ones. Then, once you are in the service business, get your tube tester, and pay for it from your service earnings.

**SIGNAL TRACERS**

The multimeter, the signal generator, and the tube tester are the basic instruments required for professional servicing. However, there are one or two additional pieces of equipment that you will want to own eventually, if you plan to build up your service business to a high volume.

The most important of these instruments is the signa
tracer. The signal tracer is important because it speeds up service work. At the beginning, speed may not be absolutely necessary, but later on you will be faced with the fact that the more radios you can service in a given time, the greater your income will be. Anything you can do to speed up this service will be definitely worth while. At that time, you will find that a signal tracer, which helps you to localize the trouble in a quick, logical, and definite manner, will be very desirable. We won't go into the theory of operation of signal tracers here—this will be covered thoroughly in future Lessons and Booklets. Just keep this instrument in mind as something you will probably purchase after you have established your service business.

**R-C TESTERS**

Another supplementary instrument, found in some service shops, is the R-C Tester. This device is most commonly used to test condensers to determine their capacity and leakage values. Although this tester is not an absolute necessity, it can speed up your service work by making a direct test, when otherwise you might have to make an indirect and more time-consuming test to find the same answer.

In addition, an R-C Tester measures resistance. The ohmmeter of your multimeter is satisfactory for most purposes, but the R-C Tester can be made much more accurate in its measurements. For certain purposes, this greater accuracy may prove desirable.
No. 1 How Radio Receivers Are Serviced

RADIO SERVICING METHODS
FOREWORD

As you open this Booklet, you may well ask: "What is the purpose of these Radio Servicing Methods Booklets?" In a nutshell, we can answer: "The purpose of these Booklets is to give you what amounts to apprenticeship training—the kind of knowledge that most people believe can be obtained only by working for months or years at a trade!" And, this training is planned to "fit in" with your regular Course, so that your theoretical and practical training will go along hand-in-hand.

Your Lessons in Fundamental Radio Principles and the Lessons for Specializing will give you all the necessary radio theory—the how-and-why-it-works knowledge. In these Lessons, you will learn what breakdowns may occur and how they affect the operation of the radio; you need this knowledge to find troubles when they occur.

Then, you will get the actual "feel" of working with radio parts in the Experimental Kits. The experiments you carry out will supplement your technical knowledge and give you practice in constructing and testing radio circuits.

Finally, these Radio Servicing Methods Booklets will show you how to use the methods actually followed by professional servicemen. You will learn how to remove the set from its cabinet, make tests, replace defective parts, and make adjustments on the complete radio. Yes, these Booklets are your shop training. In them, you will learn as an apprentice does—by following the tested, proven methods used by professional servicemen. You will learn what to do, and even more important, you will learn why to do it that way!

To get the most from these Booklets, plan to read one after each of your regular Lessons. Study a Lesson, then read the Radio Servicing Methods Booklet having the same number. Then, from time to time, review the Booklets to keep fresh in your mind all the details of how to service receivers.

J. E. SMITH.

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RADIO servicing is an ideal field—an interesting profitable occupation that will give you real professional standing in your community. Almost everyo- owns a radio, but relatively few people know how to fix one. The man who has this knowledge is sure of comfortable living and the respect of his fellows.

► When you are an expert radio serviceman, what will your life be like? Well, you'll certainly find it to be varied! In a typical day, you may repair a half-dozen radios—ranging in size from a tiny midget to a large phono-radio combination—with each of them having entirely different defects. You may make service calls to several homes, meeting all sorts of people. Each day will be different from the one before it—new work to do, new people to meet—a vast improvement over the humdrum, monotonous days of which the average job consists.

Your working conditions and hours, too, will be better than those most jobs offer. You will find that rad servicing is an enjoyable combination of brain work and mechanical work-with-your-hands. The physical work almost always will be light and non-fatiguing—a pleasant relaxation, in fact, from your thinking—ar generally will be carried out in a clean, well-lighted place. Your hours you will set for yourself if you have your own full-time radio business, or work part-time to get additional income. Of course, if you prefer regular hours, you may work for someone else.

All in all, being a radio serviceman is a wholly sati
fying and *profitable* way of earning your living.

What do you need to know to get into this field? How does the successful serviceman fix radios? Why are some radio men more successful than others?

These RSM (Radio Servicing Methods) Booklets are going to answer these questions by giving you the *practical*, step-by-step procedures followed by expert radio servicemen. You may be familiar with some of these procedures—you may have tinkered with radio at some time in the past, or you may have had experience as a serviceman. However, here at NRI we do not take anything for granted. We are going to build on a rock-bottom foundation of radio knowledge, leaving nothing to your imagination. Let’s start now, and see how radio troubles develop and how radio servicemen find and correct these troubles.

**WHAT A SERVICEMAN DOES**

To the eye of the average person, a radio chassis is a jumble of strange-looking objects. However, the radio man doesn’t worry about how the radio *looks*. Whether the set is a large phono-radio combination, a television set, or a tiny midget—whether it is an a.m. (amplitude modulation) or an f.m. (frequency modulation) type—the serviceman knows that it contains only a few general types of parts. He knows that these parts are connected in certain ways to produce certain desirable operations.

In other words, the serviceman recognizes a radio receiver as an electrical device that operates according to well-known electrical rules. To him, there is nothing mysterious about a radio that operates improperly, or goes dead altogether. He knows that some part or connection has become defective, and that he is to find and repair the fault.

Repairing a defect is simply a mechanical procedure of mending a poor connection or substituting a good part for a bad one. Almost any handyman can repair a radio, once he is shown the trouble. But it takes real knowledge of radio to locate the defect with reasonable speed. It is this *specialized knowledge* that sets the expert serviceman apart from ordinary “fixers”—and is for this knowledge that he is paid.

The quicker a serviceman can find the trouble, the more receivers he can service and the more profit he can make. Hence, you should have two goals as you start on your radio career: first, to learn *how to service*; second, to learn how to service *more quickly*.

This *second* goal is the one that makes the difference in the earnings of servicemen. Many are “stuck in a rut,” having learned just enough to get by, and having stopped their radio education before reaching the point where it would really pay dividends. Remember, it’s knowledge for which you are paid!

Let us begin now to build up your knowledge of radio servicing by giving you some facts about the parts of the radio receiver and how they become defective.

**HOW RADIO PARTS BREAK DOWN**

The basic parts of a radio receiver are tubes, condensers, and resistors. Any one of these parts may become defective. Two of your first steps on the road to becoming a serviceman will be to learn to recog

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*Can you imagine yourself working at a modern, fully-equip service bench like this? Here you see an excellent example of a good, well-designed two-man service bench, built for an NRI graduate. This man started in business using a small desk-like bench and a limited stock (a picture of this is shown later in the booklet). In just two or three years his business grew to such an extent that this larger bench was necessary.*
these various parts and to understand what may go wrong with each of them. Let’s take a few examples, and preview some of the parts you will study in detail in your Lessons in Fundamental Radio Principles.

Radio Coils. Several typical coils are shown in Fig. 1. As the name implies, each consists of a coil of copper wire wound around a form. It may be wound in a single layer, or it may be wound in layers, like thread on a spool. The coil may be wound on a bakelite or fiber cylinder, or it may be wound on a cardboard bobbin. In some cases, thin sheets of iron may be inserted inside the bobbin, so that the coil is around the iron. Frequently two or more coils are wound on a single core of iron, bakelite, or fiber. Such a combination of coils is called a “transformer.” It is used to transfer power from one circuit to another.

The wire is made of copper because copper is a good “conductor” of electricity. (Little electrical power is lost in the wire.) Silver would be slightly better, but its high cost rules it out except for special applications.

Open Circuits. Regardless of its appearance, a radio coil is a continuous piece of wire through which an electrical current can flow. If this coil wire breaks, the current no longer can flow through the coil. Then we say that the coil has opened, because the circuit is broken and no longer has “continuity.”

A break of this kind may occur sometimes because the coil wire (which usually has a very small diameter) was pulled too tightly when it was first fastened to a terminal lug connector, and stretching or expansion of the coil form has snapped the wire. A more likely reason for a break in the wire is electrolysis—a kind of corrosion that attacks and eventually eats through wire that is carrying a current. Also, if too much current flows through a wire, it will overheat and melt.

Whatever the cause of the break, the circuit is opened and the radio no longer can perform normally, or it may be dead altogether. We may be able to see this break if it has occurred at a terminal. However, it is quite likely that the break is underneath several layers of wire, in some position where we cannot possibly see it. If so, to find the trouble, we either have to substitute parts until the radio comes back to normal or have to make electrical tests to determine just which part is defective. (Servicemen make these tests by using indicating devices that show just what is happening electrically within the circuit. In your regular study Lessons, future RSM Booklets, and the Experimental Kita, you will learn all about the different methods of testing radio parts and circuits.)

Short Circuits. An open circuit is not the only trouble that can occur in a coil. The turns of wire in a coil may be wound close together, or there may be many layers of turns. It is important that the wire-turns be electrically separated so that there is no copper-to-copper contact between wires or layers. To prevent contact the wire is covered by “insulation.” This insulation may be a varnish, or it may be a silk or cotton “sleeve,” either of which has the property of blocking the flow of current, thus forcing the current to stay within the wire.

Should this insulating material become defective, it will be possible for electric current to flow through the break to some adjacent wire, or to an adjacent layer.
without having to follow the turns of wire. Any such path through a break in the insulation is called a short circuit, so named because the current is following an undesirable (and usually shorter) path. Thus, it is not flowing at its full intensity through a portion of some electrical device. Again we have a defect that probably will not be visible.

Part Value Changes. When you study coils in your Lessons in Radio Fundamentals, you will learn that certain electrical properties of the coil depend on the spacing between the turns of wire. Should this spacing change ever so slightly, the coil will not have exactly the same properties. Also, you will find that if moisture is absorbed by the form on which the coil is wound, another electrical value of the coil will be affected. Consequently, it is possible for the electrical characteristics of the coil to change, and for the operation of the receiver to be affected thereby, without any visible alteration in the appearance of the coil.

Radio Condensers. As you can see, coils are subject to a variety of possible defects. This is true also of other radio parts. Let us see how some of the condenser types, shown in Fig. 2, can become defective.

One kind of condenser is made of two metal plates separated by an electrical insulator, as shown in Fig. 3. The plates are strips of tin foil, separated by an insulator made of waxed paper. The condenser is rolled up in the form of a cylinder, and a wire is pressed against each foil plate. The condenser then is dipped in wax. On hardening, this wax holds the wires against their plates. These wires are used to connect the condenser to other parts of the circuit in which it is installed.

Should one of these wires pull away from its plate, the connection between other parts and that particular plate of the condenser will be broken, which opens the circuit. Since the contacts between the wires and the plates are sealed within the condenser housing, you can't see whether a wire has pulled away or not.

If the insulation between the plates breaks down, an electrical circuit will be completed between them. This short circuit will ruin most types of condensers. The excess current flow may also ruin other parts as well, so more than one part may have to be replaced.

► Another class of condenser has plates with variable spacing. One plate is made of a spring material and can be moved either closer to or farther from the other plate by a controlling screw. These condensers are used to adjust circuits so that they are exactly in step with each other. Once they are adjusted properly, the radio performance will be at its best. However, the spring tension of such a condenser may change with age so that the spacing between the plates alters. When this occurs, the electrical value will change also. The actual amount of space variation may be so small that it cannot be de-

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**FIG. 2.** The name by which a condenser is called sometimes indicates what it is made of, sometimes what it is used for. Two kinds of electrolytic condensers are shown at A, a paper condenser at B, a mica condenser at C; these names come from the materials used in making them. The trimmer condenser (D) and the tuning condenser (E) get their names from their uses.

**FIG. 3.** This is how a paper condenser is made. Sheets of metal foil and waxed paper are stacked (A) and rolled up (B). Then leads are pressed against the ends of the foil sheets (C).
ected by the eye, but it will affect the operation of the circuit.

Another kind of condenser contains a chemical solution that causes a film to form on an aluminum plate. The film acts as an insulator between this plate and the solution. As the condenser ages, evaporation of water from the solution will reduce the amount of liquid in contact with the insulating film, which will change the electrical value of the condenser. Also, the liquid may become a poorer conductor of electricity, which will reduce the usefulness of the condenser. These changes will not be visible, although sometimes the escaping moisture will deposit some of the chemical in a whitish crust around the vent holes of the condenser housing.

Resistors. Lengths of special wire or carbon material that have the property of opposing the flow of current, but not of stopping it altogether, are known as resistors. Their electrical effects are in between those of conductors, which carry current with ease, and insulators, which prevent current flow.

Several types of resistors are shown in Fig. 4. Various defects may appear in them. For example, the wires used to connect them to the rest of a circuit may pull away from the resistance material inside the resistor, or the resistance material may break; either of these defects will open the circuit. Short circuits may occur within the resistor housing. And, as you will learn, heat may alter the characteristics of the resistance material so that it changes in electrical value.

Radio Tubes. A tube is perhaps the most familiar of all the parts in the radio. As shown in Fig. 5, it consists essentially of a glass or metal bulb, inside of which are a number of pieces of metal and wire. Heat is necessary for the operation of radio tubes, so one of the pieces of wire is arranged in the form of a filament, somewhat similar to the filament in a lamp bulb or light globe. An electric current flowing through this filament produces heat, and this heat makes it possible for the tube to operate. If too much current flows through the filament, the heat may become so great that the filament wire will melt. This breaks or opens the filament circuit, and the tube will no longer work.

The various pieces of wire and metal within the tube must not touch each other except as intended—otherwise, a short circuit will exist within the tube. Should the position of some of these parts change, the tube characteristics will be altered.

Connections. Radio parts must be connected to one another by pieces of wire. Each connection is soldered to produce a good electrical contact. Although a connection is not a radio “part,” a broken connection can open a circuit just as well as a defective part. Excess solder may drop from a connection and cause a short circuit to the set chassis or to another terminal or part. If dirty wires are connected, or if improper soldering lets chemical actions occur at a connection, then the resulting poor contact will oppose the flow of current like an unwanted resistor.
The manufacturer of the radio is careful to see that the proper soldering techniques are followed to avoid these troubles. However, servicemen frequently either do not know how to solder properly, or grow careless. Thus they may make defective connections which can cause much trouble later.

When you receive the Experimental Kit giving instructions and practice in soldering, be sure to learn all you can about this important service step. You will have to unsolder connections either to test parts or get them out for replacement; you will have to resolder the connections to the new part, so you will constantly be using a soldering iron in your radio work.

Summary. We have barely touched upon some of the types of radio parts with which you are going to become familiar. However, you can see a pattern repeating itself over and over—regardless of the part, you know that you always look for a mechanical defect that has opened a circuit, caused a short circuit, or resulted in a change in the electrical characteristics of some part.

It is rare to find a part that looks bad, although occasionally one will be found that has been overloaded so severely that it is actually burned or is otherwise visibly defective. Generally, the mechanical trouble will be inside some sealed container or will be of such a nature that it cannot be seen. The only way we then have of finding the trouble is to observe the electrical effects produced by that particular trouble. From a thorough knowledge of radio theory and of service procedures, it is possible to localize troubles by reasoning. However, in most cases, the test procedures soon reach a point where test equipment is needed.

As you can see, radio servicing is basically easy—you already know just what the serviceman looks for!

TEST EQUIPMENT

To do service work, you will need only three pieces of test equipment—known as a multimeter, a tube tester, and a signal generator. Later RSM Booklets will describe these devices in detail and show you just how to use them. For now, let's see briefly what their uses are.

Multimeter. This device is a combination instrument that can be used to measure resistance, voltage, or current. It is the most useful test instrument any serviceman has, for with it, he can locate open or short circuits in any part or connection. In addition, he can sometimes use it to determine whether a part has changed its electrical characteristics.

Tube Tester. As the name implies, this device is used to test tubes. It is a very handy instrument, since faulty tubes are one of the most frequent causes of serv complaints.

Signal Generator. The circuits of a set must be justed from time to time to produce maximum performance. The signal generator is used to supply an electrical signal that allows the receiver to be adjusted properly. The instrument also proves very useful in tracking down certain kinds of defects.

These three items are the basic pieces of test equipment that all servicemen must have and use. It is possible to carry on a large servicing business with no oth
equipment. However, a serviceman who has a large volume of business usually acquires additional kinds of specialized testers that help him to service faster. This supplementary equipment will be described in later booklets and in your Course.

CLASSES OF RADIO SERVICEMEN

We cannot classify servicemen as beginners or experts solely from the length of time that they have been in the service business, because an absolute beginner may use advanced techniques that are unknown to one serviceman who has been in business for years. Instead, it is better to classify servicemen according to the methods that they use in servicing. We might split them into three classes—the radio mechanic, the semi-professional serviceman, and the professional serviceman. Let's see which methods each class of serviceman uses to locate the mechanical troubles we have just described.

The Radio Mechanic. The most elementary way of servicing is to test each and every radio part, in turn, until the defective part is located. The only requirements for servicing in this manner are: 1, a knowledge of the appearance and characteristics of radio parts; 2, the three basic pieces of test equipment; and 3, a lot of patience. The procedure is so mechanical that we apply the name "Radio Mechanic" to such a serviceman. He uses only his hands and his test equipment, and does not yet have the radio knowledge to "use his head."

While this is not the only way to service radio receivers—or the best or quickest way—it does require the least radio knowledge and allows one to start servicing the soonest. It is the way many servicemen start out, and it was once the way of even the expert.

In the early days of radio, even the large receivers contained no more than twenty or thirty parts. When something went wrong, it was practical to test each and every part and thus localize the trouble. Of course, if the radio man was unlucky, the defective part might be the last one tested, but eventually the trouble would be found.

It is still possible to test radio receivers this way—but now you have seventy or eighty parts in the average seven-tube receiver, so the problem of testing all parts or trying others in their places is too time-consuming to be profitable. Obviously, if one man can service six receivers in the time it takes another man service one, then the first man will have the greater claim. This need for greater speed in service led to the modern, professional servicing techniques that take full advantage of a thorough knowledge of radio parts and the circuits in which they are used.

Before we describe the procedures of the real expert, however, let us discuss those of the intermediate servicemen—those men between the mechanic and the expert. We might call these the semi-professional servicemen.

The Semi-Professional Serviceman. The next step up the ladder to becoming a true professional serviceman is to learn the purpose of radio parts—why they are used in certain combinations, and what they are supposed to do. With this knowledge, it is possible to see why certain troubles are common to certain receivers, and why certain troubles produce certain de
A good example of how most men get started. A sturdy table
for a bench, a few tools neatly held on a board, a multimeter, a
signal generator, and a tube tester are all this man needs now.
These basic test instruments are all many servicemen ever get for
spare-time or small one-man businesses. However, it is a good
deal to set aside some of your earnings to purchase additional
equipment when such equipment can speed up your work.

The semi-professional still must use the three pi
of basic test equipment, and must know how to test ti
parts. In addition, he has to know what purpose is set
by the parts, and should pay careful attention to
way in which the receiver operates. This last often g
clues that can guide him directly to the defect.

► Parts break down more frequently in certain secti
of a receiver than in others. The coil in one circui
the radio may be subject to frequent breakdowns, w
the coil in another circuit in the same kind of radio
never give trouble. There is, of course, a reason for
—radio parts in some circuits are required to ha
more power than in others. The fact that certain trou
occur frequently in radios has led to the developmen
service charts that describe the different ways
which receivers may act and list a number of poss
causes for each kind of abnormal behavior.

The difficulty with using these charts is that you n
either memorize them or refer to them constantly.
soon as you develop a complete knowledge of the fi
tion of radio parts, you'll know what can cause trou
without having to memorize a list of trout

The Professional Serviceman. The ability of
semi-professional serviceman to locate defects rap
is limited. As soon as he is out of ideas on what ;
be wrong with a receiver, he must revert to the " "
everything" methods of the radio mechanic.

The true professional serviceman, however, ha
thorough knowledge of radio parts and circuits. F
this knowledge, he first tries to reason out the n
probable faults, much in the manner of the semi-p
fessional. However, if this step fails, he does not hav
test all seventy or eighty parts in the radio recei
because he knows methods of isolating the trouble; 
small group of parts. These processes of isolation al
him to concentrate on just the three or four items t
could be causing that particular trouble; this limits
amount of testing he has to do. This excludes the m
ment of luck, and makes it possible for him to ser
receivers in a minimum of time.

► The professional knows that radio parts are grou
FIG. 6. This diagram gives the basis for the quick professional methods used in the isolation of troubles. Notice that the radio stages are grouped into sections. A single test (or a simple series of tests) will show in which section the trouble exists. Then, other tests will further localize the trouble to the defective stage. There are a number of systems of localization, and you will study them all. Some are better for one kind of trouble than for another; you would follow a different procedure when working on a "dead" receiver than you would when working on one with distortion, for example. All the systems require that you know how the radio should function—in fact, the more you know (and the better you know it) the faster you can service receivers.
how they fail, and their weaknesses; it also requires knowledge of the testing methods that will permit you to determine when a part is defective.

- The second method requires the same knowledge as the first and, in addition, makes use of some knowledge of the functions of radio parts.
- The professional method requires the same knowledge as the other methods, plus a thorough knowledge of how radio stages and sections function. If you ever watch an expert work, you will find that he touches or pulls out a tube over here, or makes a simple test over there and, from two or three such tests, locates the source of the trouble. Remember, the tests themselves are very simple—the knowledge required is used in properly interpreting the results of these tests.

**A GUIDE TO THE FUTURE**

As you may know, part-time servicing has allowed many an NRI student to earn more than the cost of his course before graduation. One of the chief purposes of these RSM Booklets is to get you started in spare-time servicing very quickly. They are designed to give you both the training and the experience you need to be a successful serviceman.

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Another graduate, and his simply constructed bench. There is no need for an elaborate bench while you are getting yourself established. Later, you can construct as elaborate and as decorative a bench as you desire. You will soon receive an RSM Booklet giving detailed plans for simple workbenches that you can build.

Students sometimes believe that the only way to get experience is just to start servicing, accepting any job they can get. Well, you can get experience this way—but it will certainly be a long time before you get the well-rounded experience on all sorts of defects that you need to be a professional serviceman. Furthermore right at the start, you will undoubtedly suffer the embarrassing experience of accepting jobs that prove to be too tough for you to handle.

How, then, are you to get the experience you need in a reasonable time? You will find the answer later on in whole series of your RSM Booklets will contain sections devoted to giving you practical training in locating and repairing specific defects in an actual receiver. The NRI Practical Training Plan will give you as much all around servicing experience as you might get from fixing hundreds of sets—and give it to you in a matter of weeks, instead of the months or even years the other way would take.

Yes, your NRI training is going to be both thorough and fast. Your RSM Booklets will concentrate first on giving you the knowledge you need to be a radio mechanic. Once you have learned this, your Booklets wil
lead you by easy stages to semi-professional and professional servicing methods. In addition to strictly service instruction, the Booklets will give you other information of a related nature—such as how to build a workbench, how to set up a shop, how to get business, and so forth. Throughout, the emphasis will be on practical, how-to-do-it information. At the same time, your Lessons in Radio Fundamentals will be teaching you the theory of radio circuits and stages. The Course and Booklets are so planned that you will always have the theoretical knowledge you need to understand the servicing methods you learn. Our experience in teaching thousands of Radiotricians has been that this system of instruction, tying theory and practice closely together, is by far the most satisfactory way to give you professional training.

Your step-by-step training begins in the next RSM Booklet, in which we describe the few simple tools you will need to start servicing. You may be surprised to learn that common tools, such as you probably already have around the house, will do for most of your service work.

THE N. R. L. COURSE PREPARES YOU TO BECOME A RADIOTRICIAN & TELETRICIAN
No. 4  How To Remove and Overhaul the Chassis

RADIO SERVICING METHODS
NRI TRAINING

Dear Mr. Smith:

I made money before completing your Course by doing spare time Radio work. In May I opened a Radio repair store, and in about two months time there were about 150 radios ahead of me. In August I hired a Radio technician to help me. During the past year my profits were approximately $3600. All I had was the will to get ahead — NRI furnished the rest. This Course includes everything to make a man tops in Radio.

R.B.F., Michigan

HOW TO Remove AND Overhaul THE Chassis

ALMOST every time you repair a radio, you will have to remove the chassis from the cabinet to locate and replace the defective part. In some receivers (particularly small portables), you must even do so to remove the tubes for testing. Now removing a chassis may seem to be a simple job—yet, even when you know how, “pulling” the chassis can easily take longer than the actual repair!

In this Booklet, we shall give you practical hints on how to remove the chassis rapidly. In addition, you will learn how to overhaul a chassis so that it will be in first-class condition when you have made the repair and put back the chassis in the cabinet. Study this information carefully—it will help you to service faster and better.

First, let’s learn how to remove a chassis.

GENERAL PROCEDURES FOR REMOVING THE CHASSIS

There are three types of cabinets: the midget, which includes the portable; the table model, a receiver of medium size that is set on a table or bench; and the console, a big cabinet that stands on the floor and may also include a phonograph. We will take up each type in turn. But, before we do, let’s learn some general procedures that apply to all three.

↑ Before you attempt to remove or reinstall a chassis, always unplug the receiver power cord from the power outlet to prevent the possibility of shock. Then take a few minutes to look over the receiver and study its ar-
rangement. Usually the fastenings will be simple.

As your first step, remove the control knobs. (These knobs fit on the ends of the control shafts which come from the chassis through the front panel of the cabinet.) The exact method of removal depends on the fastening. Some knobs have set screws; others are held on by friction springs. Just pull off the latter type. Further examination will show whether you must remove push buttons, dial pointers, etc.; instructions for removing these are given later.

Next, turn the cabinet around and look for speaker cables, power wires, antennas, etc., that have to be disconnected. Sometimes you will find it more convenient to disconnect these after the chassis has been wholly or partly removed from the cabinet. Some receivers have back covers or loop antennas that must be removed.

Next, locate the chassis fastenings. There should be screws or bolts holding the chassis in the cabinet, and sometimes shipping bolts or clamps used to protect the receiver in transport from the factory may still be in place. (On the other hand, another serviceman may have left off the securing bolts, so be sure never to tilt a cabinet until you are certain that the chassis is securely fastened in it.) Most sets have a straightforward arrangement of two to four screws or bolts, but some have “hidden” screws. Sets with inclined tuning dials, or those in period style console cabinets, sometimes have extra screws holding the tuning mechanism in place.

Once you have located the fastenings, remove them and lift the chassis out of the cabinet. If it sticks, do not force it. Pull and lift cautiously to learn where it is being held. Many sets stand on rubber “feet” or blocks, which are used to reduce vibration. These blocks may stick, particularly when placed in counter sunk holes, and will have to be pulled or pried loose.

▶ Let’s sum up the steps in removing a chassis:
1. Unplug power cord.
2. Remove control knobs.
3. Remove push-button knobs if necessary.
4. Remove dial pointer, dial scale, or dial cord connection to pointer if necessary.

5. Remove back cover or loop antenna if in the way.
6. Disconnect speaker cable, antenna and ground leads, phonograph cables, etc., when necessary.
7. Remove chassis bolts and take out chassis.

▶ Now, we’ll see how these steps apply to specific models. Remember that the methods we discuss will apply to any similar case, whether the cabinet is a midget, table, or console type. Thus, we may describe the removal of a certain type of knob under “midget sets,” but this would apply to similar knobs on any set. Also remember that we are describing only representative types—we could not possibly cover them all.

**REMOVING A MIDGET CHASSIS**

Fig. 1 shows a midget receiver in a plastic cabinet. This universal a.c.-d.c. set has a manual tuning knob or wheel on the right-hand side. The knob on the front is the volume control and ON-OFF switch. Push buttons for automatic tuning are above this knob.

▶ The first step in removing the chassis is to take off the knobs. This will allow the shafts to slip out of their cabinet holes when the chassis is removed.

There are three methods of fastening control knobs to shafts. There may be a set screw in the side of the knob which, when tightened, bites into the shaft; the knob may be held on simply by friction; or, as is the
case with the tuning knob in Fig. 1, a screw may pass through the knob and into the end of the shaft.

To remove a knob held by a screw, simply loosen or remove the screw and pull off the knob. If no screw is used, then the knob is held on by friction. TO REMOVE ANY FRICTION TYPE KNOB, JUST GRASP IT FIRMLY AND PULL IT OFF. Hold the cabinet with the other hand, as illustrated in Fig. 1.

Now remove the main tuning knob which projects from the side of the cabinet in the set shown in Fig. 1. A screwdriver is the only tool required. On many sets of this kind, releasing this knob also releases the locking mechanism for the push buttons. The push buttons will then probably get out of adjustment and will have to be reset. Details on resetting push buttons will be given in a later RSM Booklet.

As you take each part off the cabinet, put it in a small box so that it will not be lost, and you won't have to waste time looking for it when you reassemble the receiver.

► Next, remove the back cover, if one is used. In most cases four screws will hold the back in place on the cabinet. If a wooden cabinet and wooden back are used, a number of wood screws may pass through the back into the edges of the cabinet. However, in our example, snap fasteners are used instead of screws. They pass through holes in the back and snap into holes provided in the cabinet. Fig. 2 shows how you can pry out these snap fasteners with a screwdriver blade.

After the back cover is loose, tilt it back to see if a loop is mounted on it. If one is, it will be wired to the chassis, and you must be careful not to break the connecting leads. Sometimes you can leave the loop attached to the chassis, sometimes you'll have to disconnect it—but you should never rip it loose.

► Examine the back of the chassis to see how it is held in the cabinet. In our example the mounting bolts pass through the bottom of the cabinet into the chassis (see Fig. 3). Here, the cabinet has been turned on its back so the mounting bolts can be removed with a screwdriver. (Sometimes hex-head bolts are used; remove this type with a socket wrench.) Take out the lower bolts first, allowing the chassis to hang from the upper ones. If you take out the upper bolts first, the chassis will tend to twist in the cabinet and may jam.

► Notice, on the bottom of the cabinet, the chart giving the positions of the tubes and the make and model number of the receiver. In some cases, you will find a complete wiring diagram on the bottom of the cabinet.

As you take out the last mounting bolt, put your hand under the back of the cabinet. Hold the chassis up to prevent it from dropping out of the cabinet. Next, place
the cabinet right side up with the back toward you. Hold the cabinet with one hand while you pull the chassis out with the other. Pull on a coil can or some other rigidly mounted part that won't be crushed or damaged.

Fig. 4 shows the chassis out of the cabinet and ready to be serviced. The numbers stamped on the front chassis wall are inspection numbers which do not identify the chassis. The model number of this receiver is on the label on the bottom of the cabinet.

The set shown in Fig. 4 has the speaker mounted right on the chassis, so it does not remain in the cabinet. This is generally true in midgets; however, in larger receivers, the chassis and the speaker are mechanically separate and are connected electrically with a cable.

In Fig. 4 the hardware (knobs, snap fasteners, and screws) have been grouped together on the workbench so that you can see them. On an actual job, you'd put each part in a container as you took it off.

The push buttons on the set pictured did not have to be removed, but in some receivers, like the one shown in Fig. 5A, it is necessary to remove the push buttons before the chassis can be taken from the cabinet. The push buttons shown in Fig. 4 could be pulled off if necessary, but those in Fig. 5A have to be unscrewed in order to remove them, as shown in Fig. 5B. In each case a preliminary examination will show you what must be done and how to do it.

The midget shown in Fig. 5B is somewhat different from the one in Figs. 1 to 4. Instead of screws through the bottom of the cabinet, the chassis is held in place by the bolt marked C in Fig. 5B and by another one on the opposite end of the chassis. Also, it does not have a back cover but it does have a loop. This loop can be removed to make it possible to remove tubes for testing without pulling out the chassis. The loop is held on by two screws. If the chassis is to be taken out for servicing, the loop can remain in place.

In still another type of midget, the tuning dial pointer slips over a shaft that protrudes through the front of the cabinet. The pointer is held in place by friction and must be pulled off before you take the chassis out of the cabinet. You will find many variations like this—watch for them.

**REMOVING A TABLE-MODEL CHASSIS**

Table-model receivers are larger than midgets, but are in cabinets designed to be placed on tables or shelves. Remove the control knobs first. If you find a set screw holding a knob, rotate the knob to make sure there aren't two screws—some early receivers used them.
Next, turn the cabinet on its side and loosen the four mounting bolts as shown in Fig. 6. Don’t take them out completely—if you do, the heavy chassis may fall over inside the cabinet, and some parts may be damaged.

After the mounting bolts are loosened, set the cabinet upright on the workbench with one edge of the cabinet sticking out over the edge of the bench, as shown in Fig. 7. Remove the mounting bolt thus exposed, then repeat the process to remove the other mounting bolts.

Next, grasp the power transformer (or some other large, firmly mounted part), and pull the chassis out of the cabinet (Fig. 8). If you look closely at this figure, you can see the cable that connects the loudspeaker and the chassis. If the cable is long enough, you may not have to remove the speaker from the cabinet. If you wish, you can unplug the speaker cable, but be sure you NEVER turn on the set with its speaker disconnected. (As you will learn later, this may damage the set.) On this set, the connecting plug is mounted on the speaker; sometimes you will find it on the back, the side, or the top of the chassis. Some sets have no plug in the speaker cable.

When the cable is not long enough to let you take the chassis out of the cabinet and work on it readily, the loudspeaker must be removed. To remove the loudspeaker, loosen the nuts around the back of the speaker rim with a socket wrench. Hold the speaker in place with your hand while taking off the last nut. Then, grasping the magnet frame, pull it straight back, and place it face down on the workbench on a clean piece of paper. Be careful not to puncture the speaker cone by putting it on some sharp object.

When you reinstall the speaker, be very careful to line up the holes in the speaker rim with the proper mounting bolts. Carelessness in reinstalling the speaker will allow the mounting bolts to punch holes in the speaker cone. If the cable is short, turn the speaker so that the cable will be long enough to reach from set to speaker when both are reinstalled. If the output transformer is mounted on the speaker, be sure it is placed so that it will clear all chassis parts when the chassis is slipped in place.

**REMOVING A CONSOLE CHASSIS**

Fig. 9 shows part of the front of a console receiver. All knobs must be removed. These knobs are usually of the friction type. In this particular case, the knob is stuck and will not come off easily, so the serviceman has folded a handkerchief and slipped it under the knob. By pulling on the ends of the handkerchief, he can remove the knob without trouble. This is a trick to remember—you’ll find it handy time and time again.

On this model, the dial pointer must be removed. Its operating mechanism protrudes through a slot cut in the cabinet. Fig. 10 shows how the dial scale is removed so that the dial pointer may be disconnected from the
FIG. 11. Removing the dial scale pointer. This is not necessary in all sets, but must be done in this one so that the pointer will not be bent when the chassis is removed.

chassis tuning mechanism. The scale is held in place by four wood screws. (Watch for Phillips screws here; these require the use of a small Phillips screwdriver.) When these screws have been removed, slip off the dial scale. Then, as shown in Fig. 11, remove the dial pointer by loosening the screw that holds it at its bottom. The dial pointer will be bent or broken if you pull out the chassis without removing the pointer first.

Fig. 12 shows a back view of the chassis. The speaker is in a compartment below the chassis. So is the loop antenna, which is inside the cardboard form that surrounds the speaker. Cable leads pass from the speaker and the loop through holes in the chassis mounting board and plug into jacks on the left-hand side of the chassis. **Never cut a cable that passes through a hole in the cabinet.** You will always find a plug at one end of the cable. Disconnect such plugs, making careful note of where each goes.

Now take out the bolts holding the chassis to the wooden shelf. You'll probably have to lie on the floor so that you can see the bolts and get a large screwdriver into them, as shown in Fig. 13. Before the last bolt is completely removed, put your hand on the back of the chassis so that it can't slide down the inclined shelf and fall on you or the floor. Fig. 14 shows the chassis being slid out of the cabinet.

In a few receivers, you will find that the dial is mounted on the chassis and also is screwed to the front of the cabinet to give it greater rigidity. If, when you start to pull the chassis out of the cabinet, you find that it is being held, don't just pull harder. Stop and see what's holding it. If the dial is screwed to the cabinet, remove the screws.

> Notice the tools and parts left on top of the cabinet in Fig. 14. This is something that should never be done, for the top of the cabinet may be scratched, and no housewife is going to like the idea even if the cabinet isn't damaged. It is best to replace all tools in the toolbox as soon as you are finished with them, and to place knobs and screws in a container (a small jar, can, or box carried in your tool kit).

Installing a chassis of this kind is much easier than taking it out, because generally you don't have to lie down on the floor. You can tell from the old dust marks or press marks just how the chassis was placed originally in the cabinet, and when you have it exactly in place, you can easily start the chassis mounting bolts with your hand and finish up with a screwdriver. By feeling with your fingers, you can slip the screwdriver blade
into the screw head slot without much trouble.

In some receivers, the control knobs stick up through the top of the cabinet. In this instance, the chassis may be mounted on a baseboard, as shown in Fig. 15. This baseboard will be screwed to the front of the cabinet. Loosen the screws A-B-C-D-E-F one at a time, taking out first those that are hardest to reach. These are usually the ones in the top. Hold the chassis with one hand or put books or blocks under it to hold it in place while you take out the last screws. Lift the chassis out of the cabinet. You can then lay the chassis on its back or on its side and take out the bolts that fasten the mounting board to the bottom of the chassis.

Fig. 16 shows a side view of a chassis designed for many uses. It can be used as a small public address system by plugging a microphone into the jack provided for this purpose, or it can be used to amplify the output of an electrically-operated phonograph. Note the number of jacks and sockets for the various attachments. You will not always find a label pasted on the chassis indicating the use of each jack. It's a good idea to look first, and, if there is no label, to mark with a pencil on the side of the chassis the position occupied by each plug you remove. Then you will have no trouble in getting the right plug back in the right jack.

Of course, you can look at the side of the chassis when you are ready to put back the plugs, and you won't have much trouble, since the plug pins generally are arranged in such a way that they will fit only into the proper jacks. If you can't see the jacks, however, you may do considerable fumbling around before you get the right plugs into the right jacks, and a sketch of some sort will save you time.

In a few receivers, you will find that the leads from the phonograph motor are soldered inside the receiver chassis instead of being plugged into a socket (look for a plug at the motor). When there is no plug, these leads must be cut before you can take the chassis out of the cabinet. Be sure that the receiver is disconnected from the power line before you try to cut these leads with your side cutters. It's best to stagger the cuts on the
two wires, as shown in Fig. 17A, instead of cutting them both at the same place. When you reassemble the receiver, strip the insulation off these four wires, reconnect them with twist connections (Fig. 17B), solder the joints, and cover them with tape (Figs. 17C and 17D).

![Image of four steps in making a splice.]

**Overhauling Radio Receivers**

Regardless of the complaint for which the receiver is being serviced, certain general overhauling steps should be taken while the receiver is out of the cabinet. These steps are simple ones—removing dust and dirt, resoldering poor connections, replacing pilot lamps, etc.—but they all have a bearing on the appearance and operation of the receiver. Let us see just what you should do. We will assume that the chassis and the speaker have already been removed from the cabinet (in the manner given earlier in this Booklet).

**Cleaning the Chassis.** A thorough cleaning is part of every service job. There is always an accumulation of dust on a radio, for its heat of operation sets up air currents that carry dust to it. Furthermore, the average housewife is afraid to clean the inside of a radio for fear of possible damage to the radio (and, perhaps, shocks). A set owner may be embarrassed by the dirty appearance of his radio, and he is certain to notice and appreciate its return in a clean and shining condition.

There are technical reasons for cleaning, too. Moisture is the greatest enemy of a radio receiver, and dust tends to collect and hold moisture. When dust gets between the plates of a tuning condenser, it causes noise. Dust is somewhat conductive, so dust can provide leakage paths between circuits.

For these reasons, and to be able to work under the cleanest conditions possible, remove all dust as one of your first service steps when the receiver is in your shop. (In the home of the customer, don’t raise a dust storm in the living room. It is preferable to carry the dusty receiver out with you; avoid cleaning inside the home if possible.)

To keep your shop and yourself clean, wear a shop apron, and do your cleaning outdoors if possible. Don’t inhale dust. Provide plenty of ventilation if you must clean up indoors.

Dust may be removed by wiping, by blowing, or by vacuuming. Outdoors, you could use a small bellows, bicycle pump or vacuum cleaner hose attachment to blow away most of the dust. Indoors, it is best to wipe the chassis with a clean cloth or with a small clean paint brush.

Next, remove the tubes, one at a time, wiping off all dust with the cloth. If the tube base prongs appear corroded, clean them with fine sandpaper. Wipe the top of the tube socket and the chassis in its vicinity, then replace the tube. Repeat the process on each other tube in turn. Never take out more than one tube at a time, or you may get them mixed and replace them incorrectly.

Quite often a tube will be surrounded with a metal can known as a tube shield. One type of tube shield is removed by pulling upward (if there is a connection to a top cap on the tube, remove this first). Another type, made in two parts and known as a form-fitting shield, is held together by a spring-steel ring. Pull out the tube and shield together, then push this ring out of its groove with a screwdriver; the shield will then fall apart and can be removed easily for cleaning.

It is most important that all dirt be removed from the variable condensers. Rotate the gang tuning condenser to its open or minimum-capacity position and run a pipe cleaner (the type obtained from tobacco stores) between each pair of plates to loosen the dirt,
then blow out any dust that remains. Fig. 18 shows the method. (The condenser was removed from the chassis for clarity in this illustration, but, of course, you should not remove the condenser to clean it.)

Any grease or dirt remaining on the chassis after the dusting procedure should be removed with a cloth dipped in a cleaning fluid such as carbon tetrachloride (Carbona). NEVER USE WATER ON A RADIO CHASSIS! Clean the tuning dial with a soft dry cloth

_Tightening Screws._ It is rare indeed to find any “loose” screws on a radio receiver except trimmer adjusters. Many beginners find these, tighten them, and then find that the receiver no longer functions. An expert must be called in to realign the receiver—in practically all cases a signal generator must be used. Therefore this WARNING: Do not blindly tighten every screw head and nut you see on a chassis; a great many of these screws and hex-head bolts or nuts are for tune-up or alignment purposes; they control the settings of trimmer condensers or adjustable coils. You can recognize these screws by the simple fact that they are mounted ON a part and do not serve to fasten that part to the chassis. A few typical alignment screws are shown in Fig. 19. Some parts, even, are not supposed to be secured tightly to the chassis; tuning condensers, for example, are frequently mounted on rubber “feet.” Therefore, there is rarely any necessity for tightening mountings unless you have loosened them for some testing purpose.

_Soldering._ Be sure to notice the condition of the soldered joints. Manufacturers use good soldering techniques, but you can _never_ tell what some other serviceman has done. Resolder any joints that appear to be defective—joints that have a green and corroded appearance, joints covered with excessive rosin, and joints over which the solder did not flow smoothly. You can test joints for looseness by pulling on the wires with a pair of long-nosed pliers.

_Insulation._ Be on the lookout for frayed or cut insulation, particularly on wires that go through the chassis. If you find a lead with defective insulation, replace it with a new lead; or wrap insulating tape over it; or unsolder one end and slip a piece of varnished cloth tubing (called “spaghetti” by radio men) over the wire so it will cover the break in the insulation, then resolder the lead.

Inspect the power cord for defective insulation, particularly at the point where it enters the chassis and at the wall plug. The rubber insulation on power cords will dry out and crumble with age. Replace any cord in poor condition, because such a cord is a fire hazard.

Examine the leads on top of the chassis going to the tube top caps. Polish the grid lead clips with sandpaper or scrape them with a knife, squeeze, and adjust them so they make firm contact with the top caps of the tubes. Be sure that each tube shield makes good contact with the chassis, and that the grid leads do not touch the shield.

Fig. 19. Typical locations of alignment screws on: tuning condensers; i.f. transformers; and trimmer condensers. Never tighten or loosen these screws; they are used to adjust tuned circuits, not to hold the parts to the chassis.
REPLACING PILOT LAMPS

Nearly all radio receivers use pilot lamps to illuminate the tuning dial and to indicate that the receiver is "on." Like any lamps, these burn out with either age or overloading.

The two types of pilot lamps most widely used are shown in Fig. 20. The miniature screw-base lamp shown at A is like a flashlight bulb. The miniature bayonet-base lamp at B has a smooth metal base with two side studs, like an automobile dash lamp. Typical sockets for these pilot lamps are shown with them.

Removing a Pilot Lamp. Pilot lamps can usually be removed by reaching in from the rear of the radio cabinet. In some cases, however, (particularly in midget receivers), it is necessary to remove the chassis from the cabinet to get at the pilot lamp. (In the receiver shown in Fig. 11 you can see that the dial scale may be removed to replace the pilot lamps.)

Always make sure that a pilot lamp fits tightly in its socket before declaring it burned out, for the lamps often become loose. If the socket seems oversized, squeeze it a little with your fingers or pliers to get a tighter fit.

You can remove screw-base lamps by twisting them in a counter-clockwise direction; bayonet-base lamps by pushing them down slightly and twisting them a small amount counter-clockwise, just as automobile lamps are removed. Some sockets are held on brackets by spring clips. When you find a socket of this type, just pull the entire socket off its mount if that will make removal of the lamp easier.

Pilot Lamp Ratings. Pilot lamps are rated for certain supply voltages, and also for the amount of current needed for proper illumination.

Typical operating voltage ratings are 2, 2.5, 3.2, and 6.3 volts, and a few new lamps with these ratings are usually carried by servicemen in their tool boxes. The voltage rating is usually marked somewhere on the lamp; a burned-out lamp must, naturally, be replaced with a new lamp having the same voltage rating. (The 6.3-volt lamps are marked 6-8 volts, which indicates that the lamps will work on any voltage between 6 and 8 volts.)

Different current ratings are also available; typical values are .06, .15, .2, and .25 ampere. To distinguish lamps according to current ratings, the small glass beads (see Fig. 20) supporting the filaments are colored. There are three common values of 6.3-volt bulbs; a brown bead is used to indicate .15 ampere, a white bead for .2 ampere, and a blue bead for .25 ampere.

Battery sets rarely use pilot lamps. However, a few of the 2-volt types do use the 2-volt, pink bead, .06-ampere bulb. An extra switch is used to turn on these bulbs when they are needed for tuning; otherwise they are turned off to save the batteries. Don't presume these bulbs are burned out if they are not lit, at least until you have operated the lighting switch.

On a.c. sets with power transformers, the most important rating for the pilot lamp is its voltage. (You will learn how to identify power transformers later.)

If the markings on a defective lamp are not clear or are missing entirely, you could measure the voltage at the lamp socket terminal with the a.c. voltmeter in your multimeter. Choose a lamp having approximately the same voltage. However, a higher-voltage lamp is quite satisfactory if it gives sufficient light, and will have considerably longer life than a lamp rated lower than the measured voltage. Thus, the 3.2-volt lamp was developed to operate on 2.5 volts, giving adequate illumination and longer life.

In a.c. receivers using power transformers, the filament voltage ratings of the amplifying tubes are a di-
rect guide to the pilot-lamp voltage rating, because in most cases the pilot lamp is operated from a filament winding. This means that the pilot lamp will have the same voltage rating as these tube filaments. However, since there are only two standard a.c. voltages in receivers using power transformers (2.5 volts and 6.3 volts), you could use an elimination procedure. First try a 6.3-volt lamp. If it lights very dimly, then try a 3.2- or 2.5-volt type. In practically all cases, any current rating (any color of glass bead) will do.

- Universal receivers, which can operate from either a.c. or d.c. lines, require special consideration. Don't try to measure the voltage across a pilot lamp socket in one of these receivers, because the voltage will be far higher than normal until the pilot lamp is installed. You will learn why this is so in your Lessons in Fundamental Radio Principles. The pilot bulbs used are rated at 6.3 volts, but the current rating is quite important because of the special circuit used.

If you can be sure no one has previously installed the wrong lamp, you can put in a replacement having a glass bead of the same color. If there is any doubt, however, the proper size must be determined from the service information on the receiver.

**Finishing the Job.** The foregoing overhauling procedure may be carried out before or after the repair of the defect, as circumstances may require. (Future RSM Booklets will discuss repairs in great detail.) However, check the receiver operation to be sure everything is normal after these procedures. Then, replace the speaker and the chassis in the cabinet by reversing the steps of removal. After doing so, polish the outside of the cabinet carefully with a good grade of furniture polish.

The receiver can now be returned to its owner with both its appearance and its operation improved. Connect it to its antenna and ground, plug in the power cord and make a final check of its operation. If it performs properly, then you have completed a service job—one of which you can well be proud!
No. 5  How To Restring Dial Cords and Set Push Buttons

RADIO SERVICING METHODS
HOW TO RESTRING

Dial Cords
AND SET
Push
Buttons

MOST radio repairs call for considerable technical knowledge, but, if you have a little knack for mechanics, you'll be ready to take on two of the most common repair jobs as soon as you've finished this Booklet. We're going to show you how to repair dial drives, and how to set push buttons, which are two servicing jobs you'll be meeting all the time.

DIAL-DRIVE MECHANISMS

When you turn a knob to tune a radio, your action changes the settings of condensers or coils within the set, and also operates a mechanism that indicates the frequency to which the radio is tuned (usually by moving a pointer over a dial or a dial past a pointer). When we speak of the dial-drive mechanism, we mean the mechanical system that causes these actions when you turn the tuning knob.

Belt and cord drives are the two types in most common use today. Other kinds of drives have been used—particularly direct drives, in which the tuning knob is attached to the tuning condenser shaft, and friction drives, in which a rubber roller, secured to the tuning knob, bears against a dial secured to the tuning condenser shaft—but these systems are so simple that you can repair them without instructions.

Belt Drives. A typical belt drive is shown in Fig. 1. As you can see, there are two pulleys, one mounted on
the tuning shaft, the other on the condenser shaft, over which an endless belt passes. Usually there is some way of controlling the belt tension; in the illustration, this is done by the idler pulley $P$, which is held against the belt by the spring $S$ with enough force to create the desired belt tension.

Although the dial has been shown as transparent here so you could see how the system works, it is actually made of metal. The condenser shaft projects through a hole in the middle of the dial, and the pointer is fastened to the end of the shaft by a machine screw. The condenser pulley is mounted behind the dial.

**Cord Drives.** Cord drives are usually considerably more complicated than belt drives. Fig. 2 shows one of the simpler forms. Notice that the basic difference between this and the belt-drive system is that the dial cord (usually strong fishline or similar material) is securely fastened to the condenser pulley, or drum, instead of merely running around it as a belt does. In fact, the dial cord is brought down inside the condenser drum (through a slit in the drum rim) and is hooked to a spring that keeps it taut.

Besides connecting the tuning shaft and the condenser drum, the cord also passes over two small pulleys. A pointer is clamped to the cord in the length between these pulleys, and, as the arrows show, this pointer slides along a supporting edge from left to right when the tuning shaft is rotated clockwise. Thus, this system gives us a horizontal movement of the pointer instead of the rotating movement produced by a belt drive, and so permits use of the rectangular “slide-rule” type of dial that has become so popular.

Now let’s see how to repair belt and cord drives when they become defective.

**REPAIRING BELT DRIVES**

The usual defect of a belt drive is that the belt slips, either because it has stretched or frayed, or because the idler pulley does not hold it under tension. There is usually some way to increase the tension on the belt. In the system shown in Fig. 1, for example, the belt can be made tighter by shortening the spring that holds the
idler pulley against the belt. Sometimes the tuning shaft is in a slot, in which case the belt can be tightened by sliding the shaft in the slot. If the belt is stretched or frayed, however, it must be replaced.

The important thing to watch in replacing a belt is that you have a belt of the right size. There are more than a hundred different sizes in use, and generally the wrong size will not work; either it will be too tight, in which case it will break very soon and make the set hard to tune in the meantime, or it will be too loose, and the tuning system will not work at all. The best way to get the right size is to order an exact duplicate belt for that particular receiver from either your supply house or the set distributor or manufacturer. The make and model number of the set are all you need to know to get the right belt from one of these sources.

If an exact duplicate belt is not available, you will have to know the precise size of belt you want. One way to find this out is to cut the old belt and measure it carefully. Sometimes, though, the old belt will be missing, or will have stretched so much that a measurement won’t give you accurate information. In this case, the best thing you can do is to run a silk cord (which will not stretch) over the pulleys to find the right length.

Installation of endless belts is easy. Usually you will have to remove the dial to put one on, and sometimes you must unscrew the bracket holding the tuning shaft so that its pulley can be moved closer to the condenser pulley; then the belt can readily be slipped over the pulleys.

REPAIRING CORD-DRIVE SYSTEMS

There are so many variations of cord-drive systems that a much larger book than this could not cover them all. However, each manufacturer usually issues diagrams showing how to repair and restrung his sets; these diagrams, and your own mechanical ability, will let you repair almost any system. We’ve included a number of samples of manufacturers’ diagrams in this Booklet to show you what they’re like. It would be a good idea for you to build up a file of such information;
you can always get the instruction leaflets from the manufacturer, and usually from his distributor.

We’re not going to attempt to cover specific drive systems here. Instead, we are going to give you a series of service hints that apply to any system.

➔ The first is—be sure you know what the drive is supposed to do. If you have the manufacturer’s diagram, or the old cord is still on the set, trace what happens when you turn the tuning knob. Before you remove the cord, if you don’t have the manufacturer’s diagram, make a sketch to show where the cord is supposed to go, with arrows to show the direction the cord and the pointer move when the tuning knob is turned. (Generally, but not always, the pointer moves across the dial from left to right, and the condenser gang opens, when the knob is turned clockwise.)

If the cord is not on the set, or has broken and been pulled off the pulleys, you may have to study the set carefully to figure out just what the system is supposed to do. Once you have decided how to make the repair, draw a diagram to show just what you intend doing. This will serve a double purpose: it will keep you reminded of how you are going to make the repair, and, if you find you are wrong, it will show you what not to do the next time.

The cord is usually wound around the tuning shaft at least twice, often more, and you must be careful to wind the correct number of turns on the shaft when you install a new cord. If you put on too few turns, the cord will probably slip; too many turns, on the other hand, will tend to bunch up and may jam the system. If the cord is gone, so you can’t tell how many turns there should be, try using two or three.

What Cord to Use. Ordinary string or thin cotton fishline is not satisfactory, because it will stretch. If a thin cord is needed, silk or nylon fish cord is best. A cord with a Fibreglass core and a synthetic braid cover will also work well. Cords of medium thickness (diameter approximately .04 inch) are made of nylon, linen, or cotton; cotton cords of this diameter are satisfactory because they will not stretch at the tensions normally
used. Any fishline you use should have a breaking strength of at least 18 pounds.

Phosphor-bronze wire cords are also available. These are particularly useful in sets where the cord must move a heavy mechanical system. Heavy linen cords are also used in such installations.

Cords usually come in 10- and 25-foot lengths, wound on spools. You can get them from any radio supply house.

**Common Defects.** Several things may happen to cord-drive systems. The cord may lose tension, either because it stretches or because the tension spring does; the cord may slip; the pointer may stick; or the cord may jump off its pulleys, or fray, or break. Let’s see what to do in each case.

**Loss of Tension.** If the cord is too loose, it will simply slip around the tuning shaft instead of turning with it. Usually this defect can be remedied by shortening the cord. One way is to knot it again at the point where it is attached to the tension spring inside the condenser shaft pulley. Always use a square knot (shown in Fig. 5), which will not slip. You can put a drop of speaker cement, fingernail polish, or shellac on the knot as an added precaution against slipping.

Sometimes the cord is loose because the tension spring has stretched too much. If the spring allows the knot in the cord to come almost out of the slit in the condenser shaft drum, tighten the spring rather than shorten the cord. Inspect the end of the spring that is not hooked onto the cord. This end is anchored inside the drum, usually either to a bent-up metal ear or in a hole. There may be other ears or holes, farther from the slit, in which the spring can be anchored; if so, try one of them and see if the cord tension is sufficiently increased.

If not, or if no other anchor points are provided, you can either shorten the spring or install a new one. To shorten the spring, cut off a few turns from the anchor end with a pair of cutting pliers, and bend the cut end to form a new hook.

If a shortened spring is still too weak, install a new one. The exact size is not important, but be sure it is strong enough so that it will not be stretched out of shape when it is installed—otherwise it will quickly lose its tension.

When you install a new spring, or shorten an old one, be careful not to let the cord slip off the pulley system—if it does, you may have to restring the whole drive. You can hold the cord in place by pressing your thumb firmly over the slit where the two parts of the cord emerge from the condenser shaft drum. If you need both hands for the spring, put a piece of scotch tape over the slit instead.

Be sure you seat the end of the spring firmly in its anchor hole or around its anchor post. Usually it’s easiest to do this by grasping the end of the spring with a pair of needle-nose pliers, stretching it slightly past the anchor point, then allowing it to relax and guiding it into or around its anchor as it does so.

**Slipping Cord.** If the cord seems tight, but slips on the tuning shaft, probably grease or oil has gotten on it. You can remedy this condition by working powdered rosin into the cord. A commercial non-slip compound, having a rosin base, is available in stick and liquid form. This compound has the advantage that it shrinks the cord slightly in drying, thus giving increased tension as well as eliminating the effects of oil or grease.

**Sticking Pointer.** As we said earlier, cord drives are always used with slide-rule dials, in which a pointer moves horizontally over a long, rectangular dial that resembles a slide rule. If the pointer sticks or binds, the cord will get taut on one side of the pointer and loose on the other, and may jump off its pulleys; if the cord does not jump off, the tuning knob will at least be difficult to turn.

The pointer of such a dial usually slides along a metal...
track on the edge of the dial, or on the dial plate edge itself. If the pointer sticks, inspect the track for burrs that may cause increased friction. Remove them with fine sandpaper, and spread a light film of vaseline over the track. (Don’t get oil or grease on the cord.) Make sure the dial lights do not interfere with the pointer movement; if they do, bend their brackets slightly.

**Cord Jumps Off Pulleys.** Provided the cord is tight enough so that it should normally stay on its pulleys, the usual reason it jumps off is that it is caught somewhere in the system. Turning the tuning knob then tightens part of the cord and loosens part of it until the loose part finally slips off its pulleys altogether. The usual cause is a sticking pointer, as we just said. Whatever the cause, remedy it, then put the cord on again. Make sure that it is tight enough to stay.

**Frayed or Broken Cord.** Either of these must be replaced. The broken cord is usually harder to replace, because the drive system will probably be completely unstrung, and you will have to figure out how the system works. Replacement is not difficult if you have the manufacturer’s sketch of the stringing arrangement. If you do not, be sure to make a diagram of the system. If the cord is frayed but still in place, make the diagram before you remove it.

Turn the condenser gang either fully closed or fully open (maximum or minimum capacity) before you start restringing, then string the drive in a direction such that any tension you put on the cord will tend to keep the condenser in position. This will let you pull the cord taut during the operation without fear of its slipping. Manufacturer’s instructions usually specify whether the gang is in or out for the direction of stringing shown. (Be sure you cut enough cord off the spool to do the job. It’s better to waste a few inches than to have to waste the whole piece because it is short by half an inch.)

Finally, connect the pointer to the cord in a temporary manner. (Most pointers clip on the cord, but in some systems the cord is wrapped around a stud on the pointer slider.) See that it is at the high-frequency end of the dial when the condenser gang is full out, and goes toward the low-frequency end when the gang is turned in. You may have the system strung backwards, in which case all you can do is try to smile and do the job over.

Turn on the set and check the accuracy of the pointer setting. If necessary, adjust the pointer position until both the frequency of the station being received and the frequency indicated by the pointer are exactly the same. Then, place a drop of speaker cement or collodion on the pointer clip to bind it to the cord. It’s worth while to take a little trouble with this, because your customer may find it irritating to have the pointer indication even a little off what it should be. In fact, you can demonstrate to him how accurately the pointer is set when you’re finished.

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*Diagram of GE-US51 Receiver. Notice use of flat springs to hold pulleys.*
Setting Up Push Buttons In Radio Receivers

Automatic tuning, in which a set is tuned to a desired station simply by pressing a push button, has become an almost universal feature. Adjusting the push buttons is an easy job, but one a set owner seldom attempts. If the button doesn't bring the station in "on the nose," or if he wants a different station, he will usually call in a serviceman.

- There are three main types of automatic tuning systems—electrical, mechanical, and electro-mechanical. In an electrical system, pushing in the button switches a preadjusted set of trimmer condensers or adjustable inductances (coils) into the tuning circuits of the radio, and, at the same time, releases any button previously pushed.

- In a mechanical system, the motion produced by pushing in the button actuates a lever system that turns the regular gang tuning condenser of the set to the desired station.

- In an electro-mechanical system, pushing in the button (or, in some systems, turning a telephone-type dial) starts a small electric motor that turns the gang tuning condenser to the desired station. This is the only system that is particularly complicated. We're going to cover this system and its variations further along in your regular course. It would be best for you not to attempt to adjust an electro-mechanical system until you've studied the lesson on automatic tuning systems—unless you have the manufacturer's adjustment instructions for the set in question. These instructions are usually complete enough for you to make the adjustment without difficulty, if you follow them closely.

However, electro-mechanical tuning is used in but few modern sets. In this booklet, we're going to explain how to adjust electrical and mechanical systems—the kind servicemen are called upon to adjust almost every day. We will give you the general procedure for all sets instead of instructions for just a few specific radios. You can easily take care of any small variations you find in a particular set. Our instructions will be for setting up all the buttons in a set; if you're interested in
adjusting just one button, follow the procedure only until that button is set up.

**ADJUSTING ELECTRICAL SYSTEMS**

If you have the manufacturer's instructions for the set you're working on, read them carefully. They may contain some short-cut methods that apply only to that set (which, of course, we will not give here). If you don't have them, the method described in the following paragraphs will work perfectly well.

First, turn on the set and let it warm up for at least twenty minutes. This will prevent drifting of the adjustments after the set reaches operating temperature.

While you're waiting for the set to warm up, count the number of buttons available for station selecting. Ignore any that are used for other purposes (for turning the set on and off, for manual tuning, for phonograph operation, etc.). Have the customer select a corresponding number of stations. These should be local or medium distant stations that can be received well. (Naturally, if you are merely readjusting buttons that have drifted off their settings, you will reset them to the originally selected stations unless the owner wants a change.) Look up the frequency of each station in a log book or the radio listings in the newspaper, then list each station in order of increasing frequency (lowest frequency first on the list).

Next, locate the trimmer adjustments on or above the chassis (or directly behind the push buttons). There will be two of these for each button, and they will be marked to indicate the frequency range to which they can be tuned. (For an example, see Fig. 4.) Assign a station to each button on the basis of these frequency ranges, then make a rough sketch to show which station has been assigned to each button. If the customer has selected a station for which no tuning trimmers are available, (or two stations that need the same set of trimmers), have him pick some other station (or one of the two) and tell him he'll have to use manual tuning for the one you cannot set up.

You can usually reach these trimmer adjustments from the back of the cabinet, sometimes by removing a back cover. On a few sets, you can reach them from the front of the cabinet by removing either a cover plate or the escutcheon plate through which the push buttons protrude.

When the set has warmed up sufficiently, you're ready to adjust the first button. Tune the set manually to the station selected for the button, then press in the button. Now turn the oscillator trimmer (sometimes called the tuning or station-selecting trimmer) for that button until you get the same program. (You can identify the oscillator trimmer by the fact that you will hear several stations as you turn the adjusting screw.)

Next, turn the other trimmer (usually called the antenna trimmer) for that button until maximum volume is produced. This is not a critical adjustment. Readjust the oscillator trimmer carefully until the station is perfectly tuned. Return the set to manual tuning and make sure the station is heard at least as well on automatic tuning as it is on manual. (Usually the automatic tuning will be slightly better, since the trimmer condensers can be adjusted more accurately than the gang tuning condenser.)

This completes the set-up for one button; adjust each other button in exactly the same way. Since there's always a chance that you have tuned a button to another station carrying the same program as the one you want, check your work by pressing one button after another during the station announcement period. If the station you want is a local, you can readily check by comparing the background noise heard on automatic and manual tuning; if the background is much noisier on automatic tuning, you probably have tuned to some other station on the network.

**FIG. 4. Typical trimmer adjustments.**

The oscillator trimmers (marked A) are at the left, the antenna trimmers (marked B) at the right. In some radios, one set of trimmers is above the other.
Precautions and Hints. Never turn a trimmer condenser screw more than a few turns out (counter-clockwise) or it will fall out. Never apply force to a screw.

If you can’t bring a station in with the oscillator trimmer, loosen the antenna trimmer a turn or two and try again.

You may sometimes meet coaxial (one inside the other) adjustments like that shown in Fig. 5. Special wrenches are made for these, like the one shown.

Adjustable inductances called permeability-tuned coils are often used in place of trimmer condensers. A typical one is shown in Fig. 6. To produce the same change in frequency, the adjusting screw of a permeability-tuned coil must be turned many more times than that of a trimmer condenser.

ADJUSTING MECHANICAL SYSTEMS

There are two types of mechanical automatic-tuning systems. In one, the rocker-bar type, each button has its own locking adjustment. In the other, the cam-and-

lever type, one locking adjustment takes care of all the buttons.

Rocker-Bar Mechanisms. A typical rocker-bar mechanism is shown in Fig. 7. The rocker bar is a flat pivoted metal piece that is connected to the gang tuning condenser through a gear system in such a way that the angle to which the rocker bar is rotated determines the condenser setting. Each button is on a plunger that goes through a slit in the rocker bar. On each plunger is a metal finger that can be set to any desired angle by adjusting a screw. When a button is pushed, the finger on its plunger bears on the rocker bar and turns the bar to the same angle as the finger; this changes the setting of the gang tuning condenser, and so tunes the radio.

To set up the buttons of a rocker-bar system, take the same initial steps as you do with an electrical sys-
tem: turn on the set and let it warm up, have the stations selected, then list them and assign them to buttons. You don't have to consider tuning ranges, because any button can tune any station. It is usually best, but not actually necessary, to assign stations to the buttons from left to right in the order of increasing or decreasing values of frequency.

While you wait for the set to warm up, locate the adjusting screws. These are always accessible from the front of the receiver, but you may have to remove the push buttons, remove the station tabs from the buttons, remove the station tabs from the escutcheon (the ornamental plate around the buttons on the panel), or remove the escutcheon to get at them. In some sets, the push buttons themselves serve as the adjusting screws. If you have the manufacturer's instructions, they will tell you where the screws can be found.

Set up each button as follows: Back off its adjusting screw. Press the button in and hold it in. Carefully tune in the station desired for that button with the manual control. Run in the adjusting screw as far as it will go. Release the button.

That's all you need to do to set up one button; set up each of the others the same way. Check your work when you're through by pressing each button in turn.

Cam-and-Lever Mechanisms. A typical cam-and-lever mechanism is shown in Fig. 8. The cams—heart-shaped metal discs—are secured to an extension of the gang tuning condenser shaft by friction washers. A single locking adjustment, when tightened, locks all cams to the shaft simultaneously. The push buttons are mounted on the ends of pivoted levers. When a button is depressed, a roller on the other end of its lever is forced against one of the cams; this turns the cam, and the shaft to which it is locked, to the point where the roller reaches the bottom of the V of the cam.

To set up one of these systems, take the same initial steps you would for a rocker-bar mechanism: First, warm up the set and assign stations to buttons. Then locate the locking adjustment. This may be a screw in the center of the manual tuning knob, as it is in the mechanism in Fig. 8, or a knurled screw on the side of the receiver, a wing nut on the side of the dial assembly, a screw accessible from the back of the receiver, a screw exposed by removing the push-button escutcheon or removing a snap-in button on the escutcheon, or a screw reached through a hole located below the tuning unit. Sometimes the tuning knob itself must be pushed in or pulled out, then turned, to unlock the cams. The manufacturer's instructions, if you have them, will show you where the adjustment is.

Once you have found the adjustment, loosen it. Firmly push down the first button to be set and carefully tune the set to the desired station with the manual control. Be sure to hold the button depressed until you are through tuning. Then release the button, but do not
touch the locking adjustment. Repeat the process until all the buttons have been set up; then, and only then, tighten the locking adjustment.

FINISHING UP (ALL SYSTEMS)

No matter what kind of automatic tuning system you adjust, be sure to give it a final check by comparing push-button reception with manual-tuning reception for each station. If manual tuning produces better reception on any station, readjust the automatic tuning for that station.

A sheet of tabs on which are printed the call letters of all U. S. stations is usually supplied with automatic tuning sets. After setting up the buttons, secure the appropriate tab for each on the button or in the escutcheon surrounding them. When call-letter tabs are not furnished, get the sheets from the distributor of the set or from your radio parts supplier.

Whenever possible, demonstrate the set to the customer in his own home. Press each button to show that it works as it should, and make sure he knows how to operate the automatic tuning system. You may think that anyone can operate automatic tuning—but a surprisingly large number of people don’t realize that the buttons must be pressed in all the way in a mechanically-tuned set, and some even forget that an electrically-tuned receiver must be switched from manual to automatic and vice versa. You’ll build good-will, and save unnecessary call-backs, by giving a short demonstration of every set you adjust.
NRI TRAINING Pays...

Dear Mr. Smith:

I cannot praise NRI and its staff too highly. I started to earn a little about the twelfth lesson by the time the course was completed, I had earned enough to pay for the course and buy a tube tester and other servicing equipment. At present I am doing part-time service work - making on an average of $75 per month, and getting a larger amount of business every day.

I value my training more every day.

O.E.H., Penna.

ONE of the many good features of radio servicing as a profession is that you need not invest much money in tools to get started. The familiar, every-day tools shown in Fig. 1, along with the soldering iron that we will send you, will be all you'll need for the experiments of your Practical Demonstration Course and for your first servicing jobs. Although you will need a few more as you advance, tools will never be a major expense.

This does not mean that tools are unimportant in servicing. Quite the contrary—you will use them constantly, and, to make fast, neat, professional repairs, you must know how to use them properly. This Booklet will teach you what tools to use and how to care for them. Details on their use will be in other RSM Booklets.

We will describe three groups of tools. First are those you will need for all radio jobs, both in the customer's home and at your bench. Next are tools that you will need occasionally at your bench. Last are those that you will want when you develop so large a business that it is profitable for you to use time-saving equipment in spite of its initial expense.

GENERAL FACTS ABOUT TOOLS

Before discussing individual tools, let's get some practical pointers on tools in general.

First—NEVER BUY A TOOL YOU DON'T REALLY NEED. To tie up your capital in tools and equipment that you use very seldom is poor business—and
you must be a good business man as well as a good repair man to succeed in radio servicing.

Since the tools in Fig. 1 are so common, you may already have them all. If you must buy one or two, buy the best you can afford. You can always do better, faster work with high-quality tools. If you must get your first ones at the "five and ten" or the bargain counter of a hardware store, buy better ones as soon as your servicing profits give you the money to do so.

Care of Tools. Start from the beginning to use a tool only as it is intended to be used. Never, for instance, use a good screwdriver as a chisel or a screwdriver handle as a hammer. Have a place close at hand for your tools so that you can pick them up and lay them down without tossing them across the workbench. Good tools will last a long time if not abused.

Above all, be neat—it will save you time. Have a place at your bench for each tool, and teach yourself to put back the tool in its proper place when you're through with it—or, if you intend to use it again soon, lay it in a nearby work tray. Nothing is more annoying or kills more time than having to hunt for a tool lost on a cluttered bench.

► For the same reason, as you take screws, nuts, dial knobs, and other small parts out of a set, put them in a small box, a tin can, or a jar. Then, if you have to put

FIG. 1. The soldering iron we send you, plus these few tools that you supply yourself, are all the tools that you need to start servicing.

Now where did I put that nut?

A disorderly, cluttered workbench can cost you money—you may waste many hours of valuable time looking for misplaced tools and parts.

Keep your tools in a rack—keep small parts and tools you're using in a work tray—and you'll find you can work much faster.

the receiver aside for a while, these parts won't be lost.

Keeping things in order will help you develop a logical, orderly work-procedure which will save the maximum amount of time.

Keep your tools and your tool box thoroughly clean. Use clean dustcloths to wipe off dust and moisture. To protect the tools, oil them regularly. Dip them into a flat pan containing light oil, then wipe them thoroughly dry. Just enough oil will remain on the surface to prevent rust. Don't leave excess oil on the tools—it will make your hands dirty, leave spots everywhere you lay the tools, and collect dust and dirt.

► A customer will be impressed either favorably or unfavorably by the condition of your tools, not by their number. Rusty, broken tools are certain to make him wonder if you should work on his radio.

We'll now discuss the various tools you'll need—indicating in each case whether you should have the tool soon or can wait for some time before you get it.

PLIERS

Fig. 2 shows typical styles of the most useful types of pliers. One of each type will eventually find its way into your kit of tools.

Long-Nose Pliers. There are several styles of long-
nose (sometimes called thin-nose or needle-nose) pliers. The types in Figs. 1 and 2 are most commonly used, but suit your own preferences. In any case, be sure to get a pair with jaws that meet squarely and have no side play.

Fig. 3 shows two major uses of long-nose pliers. Don't try to loosen a nut with them in the manner shown in Fig. 4. You will spring the jaws so that they no longer meet squarely. These pliers are intended only for light-duty holding or pulling, not for heavy twisting.

Automobile Pliers. Both straight- and bent-nose slip-joint or automobile pliers are useful for removing large nuts. The bent-nose type is preferable because it helps you to remove nuts close to the chassis or the chas-

FIG. 3. Common uses of long-nose pliers: A, holding a wire for soldering; B, bending a loop in a wire.

sis wall. Buy a pair made from thin stock, as it will be easier to use them in crowded places.

Ignition Pliers. The ignition pliers shown in Fig. are lighter and less bulky than automobile pliers, but are surprisingly powerful. They are useful on smsguts, on nuts in hard-to-reach places, and in adjusting certain loudspeakers.

CUTTING TOOLS

Side-cutting Pliers. Side-cutting pliers are used to snip the wires off defective parts, to cut pieces of hool-up wire from a roll of wire, to cut connecting leads to the proper lengths, and to cut off soldering lugs. The side-cutter shown in Fig. 5 is used because it is easy to get under wires and other parts than are other cutters. You will need side-cutters right away; get a pair about 5 or 6 inches long.

Cheap side-cutters are a waste of money. Before buying a pair, close the jaws (but don't squeeze the handle tightly), and hold the pliers against the light. The cutting edges should fit together perfectly (see Fig. 6). If you can see light between the cutting edges, the pliers will not cut insulation cleanly; don't buy them.

Never use side-cutters for cutting nails, bolts, or heavy wire; you will ruin them.

Courtesy India Drop Forge and Tool Corp.

FIG. 5. This is the type of side-cutting pliers you will find most useful.
**Jackknife.** Right from the start you will need a jackknife with a rugged blade. A typical knife is shown in use in Fig. 7, trimming insulation from a wire. Always be careful not to cut into the wire you are cleaning, as a deep nick will weaken it. Since scraping wires will full even the best knife blade, some servicemen use an old knife or just one blade (or only part of one blade) for this purpose.

Never use a jackknife as a screwdriver. Don't cut wires with it either; you are certain to nick the blade.

**SCREWDRIVERS**

Screws of all sizes are used in radios—from big woC or machine screws used to mount large radios in the cabinets, down to tiny set screws used to hold contr knobs on their shafts. Some screws are easy to reach others are difficult. You will need both short- and long bladed screwdrivers.

It is very important that the screwdriver tip fit the screw slot snugly so that the screw will be relatively eas to turn. Tips that are too wide, narrow, thin, or thick fe the slot tend to twist off the head, chew up the slot, or make the screw hard to turn. You will need sever screwdrivers to fit the various screws found in radio

Fig. 8 shows a good beginning assortment of screw drivers. There are six conventional types: two with small tips (these can both be the same length—you need two mostly because they are easily mislaid); three with medium tips, in the shaft lengths and weights shown and one heavy-duty type with a medium-length shaf You can add other medium-tip sizes later if you fin you need them.

The sixth screwdriver shown in Fig. 8 is intende
or use with the Phillips screw, now used in many radios.

As Fig. 9 shows, this screw has a recess instead of a
lot in the head and requires a special screwdriver.
because of the taper of the recess in the screw head,
e Phillips screwdriver will take screw sizes up to
No. 4, and another will take sizes 5 to 9. These two
Phillips screwdrivers will be the only ones you will need
in radio work. You will need both almost as soon as you
tart servicing.

* Screwdrivers with the hold-tight feature shown in
Fig. 10 are handy for starting screws in hard-to-reach
daces. Get a medium and a small one of this type after
a while.

* When you buy a screwdriver, be sure that the blade
will not loosen under strain and turn at the handle.
While some wooden-handle screwdrivers are satisfac-
tory, the best for radio work are those with shanks
soldered into handles of transparent, insulating plastic.
One point—keep these handles away from heat and
lame; they will burn.

**SOLDERING IRON**

The most used of all your tools will be the soldering
iron. Whenever you replace a part or disconnect a lead,
ou will unsolder and resolder one or more connections.

*Fig. 10. The clamp on the end of the screwdriver is handy for
holding the screw until it is well started in the work.
Courtesy Yako Products Company*
detailed instructions on how to solder, and lots of practice.

However, you can't get too much practice—soldering is an art. An astounding number of receivers are serviced only because of poor soldering. Learn the rules in Table 1, and practice them constantly in your work. If you do make a poor joint, don't fail to do it over. Remember—good soldering is essential.

**Hints on Unsoldering.** Unsoldering, the reverse of soldering, is considerably easier. To unsolder, simply apply the iron tip to the joint until the solder melts. (If a coating of oxide or grease keeps it from melting, apply a very little flux to the solder with the end of a toothpick and bring the iron to the joint.) You can then pull the wire gently with a pair of pliers and break the joint. Never jerk or pull too hard; you may break leads, soldering lugs, or the parts themselves.

You can readily pick up excess solder from a joint by holding the hot tip under it. The solder that runs down onto the iron can be shaken off onto the shop floor. Repeat this until only a little solder is left on the joint, then, as it cools, wiggle the leads back and forth so the remaining solder cannot set properly. In many cases, this makes it easy to untwist and remove the leads.

If it proves too difficult to remove leads this way, just cut one wire with a pair of cutters. You can then remove the small end of the wire if it is in the way of a new connection.

**SOCKET WRENCHES**

Socket wrenches are better than pliers for removing nuts from bolts. A socket wrench fits over the nut and does not slip off. You do not have to move it to get another grip; just “spin” the nut right off.

Socket wrenches come in sets like that shown in Fig. 12, or may be bought individually. They come in sizes to fit nuts of: 3/16" — 7/32" — 1/4" — 9/32" — 5/16" — 11/32" — 3/8" — 7/16" — 1/2". A set, one of which you should get soon after you start servicing, contains the most used sizes. Eventually, you will want a complete collection of these handy tools. Special types (1/2" and 9/16") with extra-deep hollow shafts are available for removing volume control nuts.

Good socket wrenches are expensive, so wait until your service earnings will pay for them—cheap types are a waste of money. A good one has a socket that is...
free from burrs and is of sufficient depth to fit over two nuts. Also, a good wrench has a shaft that is hollowed out for a considerable distance, so that the wrench can fit down over long bolts to engage the nut; some cheaper wrenches do not have this feature. Furthermore, the better types have thin, strong sockets which can get into small spaces.

The better sets have stands, so you can keep them together and in order. Recently, plastic-handled types have been made with different colored handles; the size can be recognized by the color—this is a real time-saver.

MISCELLANEOUS ITEMS

Alignment Tools. With age, receivers get out of adjustment, losing their pep or their ability to separate signals from different stations. To restore the set to its original characteristics, it is “aligned” by resetting a number of variable adjusters.

It is possible to align many radio receivers with ordinary screwdrivers and socket wrenches. However, for greatest ease and most accurate work, alignment tools should be used. Typical examples of these tools are given in Fig. 13.

These tools are small screwdrivers and wrenches made from plastic materials instead of metal. This eliminates the upsetting effect of bringing metal near certain adjusting screws and nuts, thus making it easier to align accurately. These tools can be obtained from almost any radio supply house—individually or in complete kits. Naturally, you won’t need them until you learn how to align a receiver.

Tool Box. You will need a tool box right from the start to carry your tools with you on service calls. A sectional or tackle box like that shown in Fig. 14 is best: you can see all the tools when you open the box, and you can keep your tools separated according to types so that you can find them easily.

Have a place in your box for each tool, and put it there as soon as you are through using it. Otherwise, you’ll often leave tools in your customers’ homes. Don’t toss the tools into the box either; the clatter is annoying, and you might miss and scar some furniture. Remember—you will be working right in the living room, perhaps surrounded by interested onlookers. The condition of your tools, and the way you treat them, will have much to do with the impression you make on the customer.

Empty your tool box regularly in your shop and clean both it and the tools carefully. Many servicemen place small bits of wire, excess solder, and other odds and ends in the tool box when in the home of the customer. It is perfectly all right to clean up this way—in fact, it makes a good impression on the customer—but be sure to remove this junk when you get back to your shop. Don’t carry useless tools, either.

Hardware. A certain amount of hardware will be necessary as soon as you start your service business. You’ll need a roll of hook-up wire, a roll of friction or tire tape, and an assortment of screws and nuts.

The most used screws for mounting radio parts are 6-32 and 8-32 machine screws. (By the way, say “six thirty-two” and “eight thirty-two,” not “six thirty-second” or “eight thirty-second.”) The six in 6-32 refers to the diameter of the wire from which the screw is made, and the thirty-two tells the number of threads
to the inch. The lengths you will need in radio service work are ¼-inch and ½-inch. One gross each of ¼-inch and ½-inch 6-32 and 8-32 (round or binder head) machine screws, with nuts to fit, makes a good stock. Get screws that are cadmium- or nickel-plated (to prevent rust), and keep them separated according to sizes in small glass jars.

Eventually, you will probably get a box of assorted wood screws, and perhaps another box of assorted self-tapping screws. The latter do not require nuts; they cut a thread in metal and thus hold themselves.

WORKBENCH TOOLS

The tools we’ve mentioned so far are those you will use both at the workbench and on service calls. There are others that you will eventually find desirable for use at your workbench.

Hack saw. You’ll need a hacksaw fairly soon for sawing the shafts off volume or tone controls, to make them the correct length. (General-purpose replacement controls come with extra long shafts so that they may be fitted to different receivers.) A typical hacksaw is shown in use in Fig. 15. Here are some pointers on using a hacksaw:

1. Place the blade on the frame holding-pins, with the blade teeth pointing away from the handle.
2. Adjust the blade between the frame holding-pins so that there will be no twist, and keep it stressed tightly.
3. Bear down sufficiently hard on the forward stroke to make the blade cut into the work. If the teeth merely slide over the work, their cutting edges will be dulled.
4. Lift the saw slightly on the return stroke to disengage the teeth—otherwise they will be dulled.
5. Don’t saw too fast; you’ll overheat the blade and cause it to lose its hardness. About forty strokes per minute is satisfactory for general work.

Hacksaw blades come in 8-, 10-, and 12-inch lengths. Be sure to get blades the right length for your holder (unless it is adjustable for various lengths). Blades are made with 14, 18, 24, or 32 teeth to the inch, depending on the cutting job they must do: those with fewer teeth are for soft metals like aluminum alloys, zinc, and copper, while those with closely set teeth are for iron, steel, and other hard metals. Blades with 24 teeth to the inch are most useful for radio work, but eventually you’ll want others also. For instance, a fine-toothed blade is necessary for cutting thin metal (such as a chassis).

Vise. Eventually, you will need a husky vise like that shown in Fig. 16 for such jobs as holding a volume-control shaft firm while you cut it and for holding the soldering iron steady while you clean the tip with a file.

FIG. 15. Cutting off a volume control shaft with a hacksaw. This is a common radio servicing job.

FIG. 16. A vise like this is the best kind for service work. The jaws should be at least 3 or 4 inches wide.
Get a strong vise that has 3- or 4-inch jaws and can be locked in any desired position. Bolt it firmly to your bench.

**Files.** One flat file and one round or rat-tail file will prove useful to you almost at once. The rougher the work or the softer the metal to be smoothed, the coarser the file should be. Get a medium-sized double-cut bastard flat file at first. A double-cut file has teeth in two sets of rows, the sets crossing each other at an angle. The term bastard refers to the rather coarse size of the teeth or degree of cut. Other cuts are known as second-cut, smooth, and dead-smooth, each of which is finer-toothed than the one mentioned before it. If too fine a file is used, the teeth quickly clog with “pins” (particles of metal).

When filing, hold the lower end of the file between the thumb and first finger of your left hand and the handle in your right hand. Move the file back and forth across the work in a direction parallel with the length of the file. Use pressure only on the forward or downward stroke, and release the pressure on the backward stroke so that you will not break the file teeth.

▶ When file teeth clog they must be cleaned. A special wire brush is available for this purpose, but you can easily make a satisfactory cleaning tool. Hammer the point of a large nail until it becomes spade-shaped. File this flat end square until the nail looks like the one in Fig. 17. Now hold the nail at an angle and work it back and forth across the file, in the same direction as the rows of teeth. This will cut teeth in the end of the nail that will fit down into the ridges on the file and scrape out all the metal particles.

Don’t abuse your files. They are very hard and brittle. If you throw one on top of another in the tool box, the teeth or the whole file may break. Don’t use a file as a prying tool—it will be sure to break.

**Hand Drill.** A hand drill is very useful for removing rivets holding parts to a chassis or for making mounting holes in a chassis. You’ll need one soon after you start servicing. An ordinary hand drill, with three-jawed chuck capable of taking round-shank drill up to ¼-inch size, is all that is necessary for radio work.

Fig. 18 shows a typical drill in use.
▶ You will, of course, need some drill points or drill bits. There are two kinds—carbon and high-speed. The carbon type is cheap and entirely satisfactory for use in hand drill. Only high-speed drills should be used in electric drills, because they run so fast that they will overheat and destroy a carbon drill-point.

Fig. 19 shows a typical drill point and a table giving the numbers for the drill sizes that you will be most likely to use.

Even the best drills become dull with use. Carbide drills cost so little you may as well discard them, but pays to have high-speed drills sharpened by a machinist. Don’t try to sharpen them yourself unless you’ve experienced in doing so.

Always turn a drill slowly enough and use enough pressure so that the drill point will not slide.

After drilling a hole, particularly through thin metal, you may find a rough burr on the side where the drill point came through. You can easily remove this burr with a counter-sink or with a drill about twice the size of the one you used to make the hole. Wrap the Shank end of the drill in a piece of cloth, grasp it firmly with your thumb and index finger, and make a few turns to remove the burr.

**Drill Bits and Gears**

<table>
<thead>
<tr>
<th>Drill Diam. Gears</th>
<th>No. (in.)</th>
<th>Screw</th>
</tr>
</thead>
<tbody>
<tr>
<td>1/16</td>
<td>2090</td>
<td>12-20</td>
</tr>
<tr>
<td>1/8</td>
<td>1935</td>
<td>10-32</td>
</tr>
<tr>
<td>3/16</td>
<td>1685</td>
<td>8-32</td>
</tr>
<tr>
<td>1/4</td>
<td>1405</td>
<td>6-32</td>
</tr>
<tr>
<td>5/32</td>
<td>1130</td>
<td>4-40</td>
</tr>
</tbody>
</table>

Fig. 19. These five drills will take care of most service work. Order them by number.
your fingers, hold the cutting point to the burr side of the hole, and turn the drill clockwise. This will trim off the burr and leave a smooth edge.

When you have to make a hole much larger than the drills you have, either drill a series of small holes and smooth the hole out with a file, or drill as large a hole as possible and use a reamer.

**Hammer, Punch, and Cold Chisel.** You’ll probably need a hammer and a punch fairly soon. Any ordinary household hammer will do for radio work. A small center punch is helpful when you have to drill holes in a chassis. (Unless there is a starting indentation, a drill point will tend to skid about rather than enter the desired point.) To make a mark with a punch, center the point of the punch at the spot where you wish to drill and strike the top of the punch sharply with a hammer.

Before using a punch, take all the tubes out of the chassis so they won’t be jarred by the blow, and, if possible, support the chassis at the point you plan to punch. A block of hardwood is just right for this. If it is impossible to put a support under it, don’t strike hard enough to bend the chassis.

You will occasionally find a cold chisel useful for making a large, square hole in a chassis or for knocking off rivet heads.

**Marking Tools.** You will find occasional use for a wood or metal ruler. A pair of dividers will be helpful at times in determining the spacing between holes, though usually the ruler will be sufficient.

You can mark on most receivers with a pencil, though some are finished so that a sharp-pointed scriber must be used. In a pinch, an ordinary straight pin can be used to scratch a mark on a chassis.

**TIME-SAVING TOOLS**

We’ll mention now a few tools you won’t need unless you get a large shop. Before then, the time you can save with these tools won’t be worth their cost.

**Chassis Cradle.** Often, a receiver must be turned upside down so that you can work underneath it. In some cases there will be large parts, such as transformers and electrolytic condensers, so placed on top of the chassis that they will make steady supports while the chassis is turned over; but frequently you’ll find a delicate part, such as the tuning mechanism, so placed that it will be damaged if the weight of the chassis rests on it.

Most servicemen use boxes, wooden blocks, or radio parts to support the chassis. However, a chassis cradle like that shown in Fig. 20 is better. With the chassis fastened in such a cradle, it may be placed any desired working position without danger of damaging any parts in the receiver.

**Wire Strippers.** Much of the wire used by servicemen has push-back insulation. You simply push back the insulation with your thumb and forefinger to uncover an end for soldering. However, some wires are covered by braided, plastic, or rubber insulation, which must be cut off the wire. Although you can cut this with a sharp knife if you are careful to avoid nicking the wire, it is a rather awkward procedure in tight places. In such cases, the wire-stripping tool shown in Fig. 21 is quite a time-saver. This tool strips off insulation neatly and cleanly, without nicking the wire.

**Ratchet Screwdrivers.** The ratchet screwdriver shown in Fig. 22 is convenient for removing and replacing long screws. Only the best grade ratchets last very long and have the necessary mechanical strength to tighten large screws firmly.
**Auto Radio Tools.** If you specialize in installing automobile receivers, you will find that both an adjustable end-wrench and an electric drill are necessary jobs. Fig. 21 illustrates both.

**CONSTRUCTIONAL TOOLS**

The following tools are rarely necessary in a radio service shop, but will prove useful if you ever specialize building or remodeling radio equipment. Naturally, you won't even consider entering these fields until you've had plenty of servicing experience, so you won't need these tools for some time:

a. **Reamer and brace.** These are used to enlarge holes illed in metal.

b. **Tap wrench and tap.** These are used to cut screw reads in drilled holes.

c. **Socket punch.** This is used to cut a hole in a chassis for a tube socket or the socket of an electrolytic condenser.

d. **Power tools.** A drill press and other power tools will be very useful if you do much constructional wor.

**A Look Ahead.** You will need some place on which to work. At the beginning, all you need is a reasonab solid flat surface. Later, you will find you need a regul workbench; that will be time enough to consider makir one. A future RSM Booklet will give you plans and constructional data for several types of workbenches.

Your next RSM Booklet, “Equipment Used by Servicemen,” will tell you the basic features of testir equipment that servicemen use. Before you tackle RS3, however, study Lesson No. 3 of the regular Course so you will be able to understand thoroughly everythin in this practical Booklet.

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**G. 21.** The electric drill (top) and the adjustable end wrench (bottom) are needed for auto-radio work. The wire stripper (bottom left) and the ratchet screwdriver (bottom right) are speed-up tools that come in handy in the well-established shop.