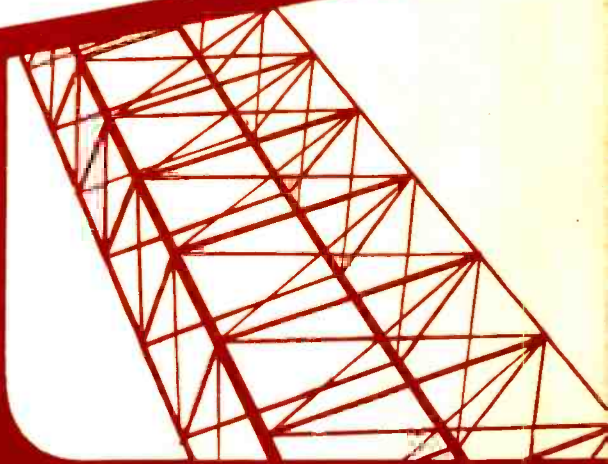
P-A SYSTEMS

Lesson RRT-5



DE FOREST'S TRAINING, INC.

2533 N. Ashland Ave., Chicago 14, Illinois

RRT-5

Radio Reception
and Transmission

LESSON RRT-5
P-A SYSTEMS

CHRONOLOGICAL HISTORY OF RADIO AND TELEVISION DEVELOPMENTS

1865—Mahlon Loomis sent wireless signals between two mountain tops 14 miles apart confirming his vision of a "static sea" within which electric disturbances could be set up to transmit messages through space. The first concept of electromagnetic wave propagation.

1872—Mahlon Loomis received the first wireless telegraph patent issued in the United States, covering the use of an elevated aerial to radiate or receive electric pulsations through space.

1873—The first ideas on television—an Irish telegraph operator noticed that his instruments acted erratically when the sun shone on his selenium resistors. This gave several inventors ideas on the transmission of pictures.

1873—Maxwell published his book on Electricity and Magnetism in which he advanced the concept of electromagnetic waves and laid the ground for wireless communication.

DE FOREST'S TRAINING, INC.
2533 N. ASHLAND AVE., CHICAGO 14, ILLINOIS

RADIO RECEPTION AND TRANSMISSION

LESSON RRT-5

P-A SYSTEMS

INDEX

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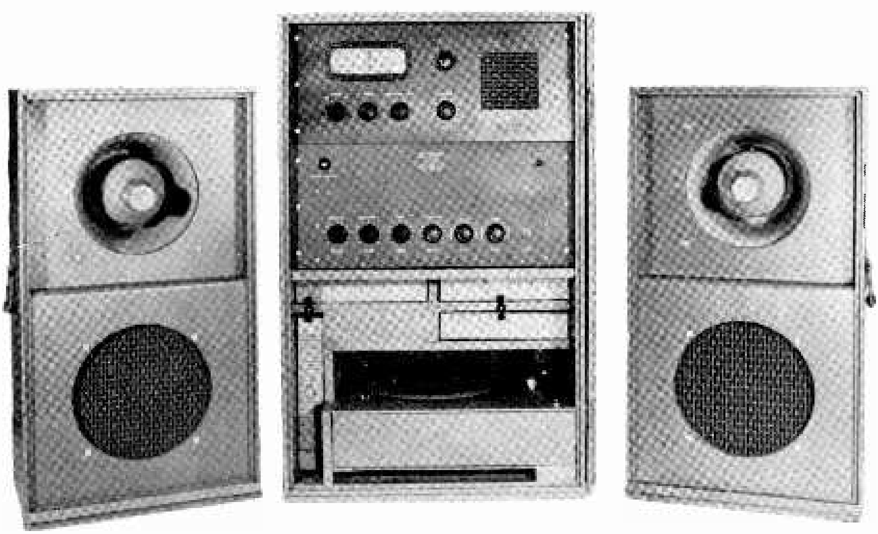
The quitter never wins and the winner never quits.
Alexander Mitchell

P-A SYSTEMS

PUBLIC ADDRESS SYSTEMS

The earliest models of microphones and headphones were designed for the original telephone systems which had a comparatively short operating range over wired circuits. Many devices were developed to extend this range and shortly after the invention of the triode tube, the telephone companies built the first "electronic repeaters" which now are known generally as audio amplifiers. It was the efficient operation of these amplifiers that made long distance telephony possible and practical.

With the advent of Broadcast Radio, telephone types of microphones were used for transmission and headphones for reception to make up what can be thought of as a Wireless telephone system, but, the requirements of radio differ quite widely from those of a telephone. At the transmitter, the microphone must be more sensitive to pick up sounds over a larger area and must respond to a much wider band of



Portable sound system for indoor and outdoor use. It is equipped with radio tuner, microphone, and phono input. A monitor speaker is mounted behind the grill in the upper right-hand corner of the center unit.

Courtesy David Bogen Company, Inc.

frequencies than required for speech only. At the receiver, the signals must be reproduced at a volume level high enough to be heard easily and distinctly without holding a headphone to the ear.

With the rapid growth of radio, the original telephone units were modified to meet the new requirements and naturally, new and improved types were developed. Then, it became evident that if the audio amplifier of a radio receiver could raise the level of the weak signals of the detector output it could do the same for the output of a microphone connected directly to it. Thus, by the use of a microphone, audio amplifier and one or more speakers, the voice of a person can be reinforced and increased to almost any desired volume level. Used mainly at large gatherings, arrangements of this type are known usually as Public Address Systems.

In general, a public address system includes one or more microphones and other pickup devices, one or more audio amplifiers and one or more speakers each capable of handling considerable power. There are many

A 50-watt amplifier shown removed from cabinet.

Courtesy Langevin Company, Inc.

specific applications for equipment of this type which today will be found in practically all auditoriums, churches, clubs, ballrooms, baseball parks and other similar places which accommodate audiences large or small.

SELECTION OF EQUIPMENT

In the selection of equipment for any p-a system, the size of the installation, the use to which it is to be put, the quality desired, and the average noise level at the location must all be considered. The type of system used for a large outdoor audience at a race track or baseball park, where there is a large amount of audience noise, is quite different from that used for providing background dinner music in a restaurant, or for sound reinforcement in a concert hall.

These factors determine the power requirements, fidelity, amplifier gain, controls, input sources, loudspeakers, and transmission lines. For outdoor gatherings such as fairs and picnics, where the size of the group may range anywhere from about 100 to 25,000 people, and where the noise level is high, a portable system having sufficient power for adequate coverage and intelligibility is needed. Here, fidelity is of minor importance. A similar though permanent installation is required for a baseball park where the crowd may number up to 50,000, or at an airport where the normal noise level is extremely high.

For an open air concert, where the audience may number from 500 to 5000 people, the noise level is not high, and a medium power system capable of high quality reproduction is needed. For a restaurant, where 50 to 300 people may be seated, the noise level is medium and a fairly low power system having a mellow quality is desirable. For a dance hall the system should have a good bass response.

The desired characteristics of a system are obtained by the proper selection of parts which can be divided into the three main divisions of (1) Input devices, such as microphones, phono pickups, and radio tuners, (2) The amplifier and (3) The number and types of speakers. Amplifiers, loud speakers, microphones and some types of radio tuners have been described in previous assignments while phonograph pickups and other types of radio tuners will be taken up in later lessons. However, as the basic p-a systems include only microphones, amplifiers and speakers the following explanations of this Lesson will emphasize these units.

MICROPHONES

With respect to their selection for use in p-a systems, the following review of the characteristics of the main types of microphones will be of benefit.

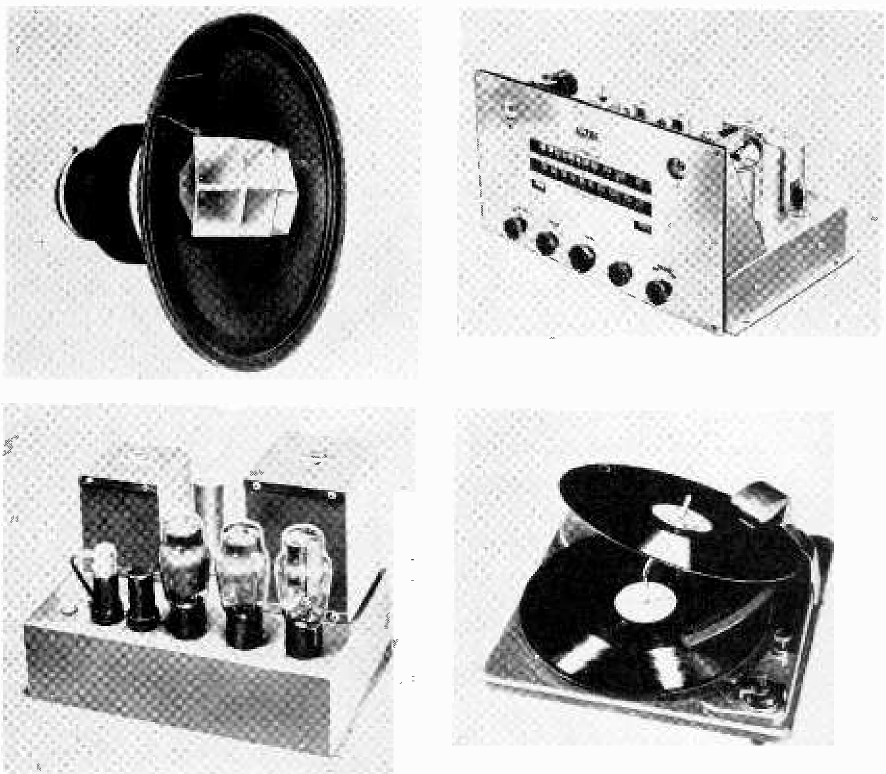
Although the output of the carbon microphone is relatively high, its use results in serious attenuation of the higher audio frequencies, and therefore it is not employed generally for the pickup of music or other programs which require high fidelity reproduction. However, it does find application where speech frequencies only are involved, such as in police, amateur, and aircraft radio telephone systems. Other disadvantages of the carbon microphone are its high noise level due to the great number of contacts presented by the large number of carbon granules, a tendency of the granules to pack together, and the effect of vibration, positioning, handling, etc. on its sensitivity. Its advantages are its comparative ruggedness and low cost.

The condenser microphone has a better high frequency response characteristic with a much lower noise level than the carbon microphone, and is not affected appreciably by temperature, humidity, or ordinary handling. However, it is affected by cavity, diaphragm and air-pocket resonances, and its sensitivity is much lower than that of the carbon type. Because of this low sensitivity, a preamplifier is usually mounted in the same case or housing along with the microphone itself.

From the viewpoint of the sound engineer, this arrangement constitutes a disadvantage because preamplifier noises and failures are often encountered while the microphone is in use. It then becomes necessary to disconnect the microphone from its preamplifier and connect it to a spare while a program is in progress. These interruptions are highly undesirable and the condenser microphone is seldom encountered today except in special applications such as in laboratory test equipment.

The dynamic microphone is popular for p-a systems and remote pickups as well as studio work in radio because of its adaptability. Due to its low impedance, it may be used with shielded cable over considerable distances from its preamplifier, and has the advantage of mechanical ruggedness with a higher sensitivity than the ribbon type. Most types of dynamic microphones are capable of operating with uniform frequency response up to about seven or eight thousand cps, and their noise output is comparatively low.

A uniform frequency response up to 15,000 cps can be obtained by use of a crystal microphone which has low noise output, a high impedance and is relatively inexpensive. The advantage of its high impedance is that the microphone output can be fed directly to the grid of an amplifier tube without the necessity of an impedance matching transformer. However, the sensitivity of the crystal type is comparatively low, and therefore it should not be located at too great a distance from its preamplifier.



Complete home music system, consisting of radio tuner, phono turntable, amplifier and speaker.

Courtesy Altec Lansing Corporation

Its frequency characteristic is affected by humidity and temperature changes, therefore it should be employed only in recording studios or on musical stages, etc. where temperature and humidity conditions can be controlled rigidly. The crystal microphone is non-directional, that is, it picks up sound equally well from all directions, and the desirability of this characteristic should be considered in the selection of a microphone for a particular application.

Next to the crystal in its quality of frequency response characteristic is the ribbon, or velocity, microphone. Its sensitivity is low, but its impedance is low also, and therefore it may be connected to its preamplifier by means of a comparatively long length of shielded low-impedance transmission line. That is, the preamplifier need not be as close to the ribbon microphone as with the condenser and crystal types. When used for speech pickup, this type often produces a middle frequency hum, which is not apparent for music. However, it may be overloaded easily, and sudden sounds such as gun shots, explosions, etc., may blow the ribbon entirely out of the air gap. For this reason a different type, such as a dynamic, is usually used for sound effects in radio studio work.

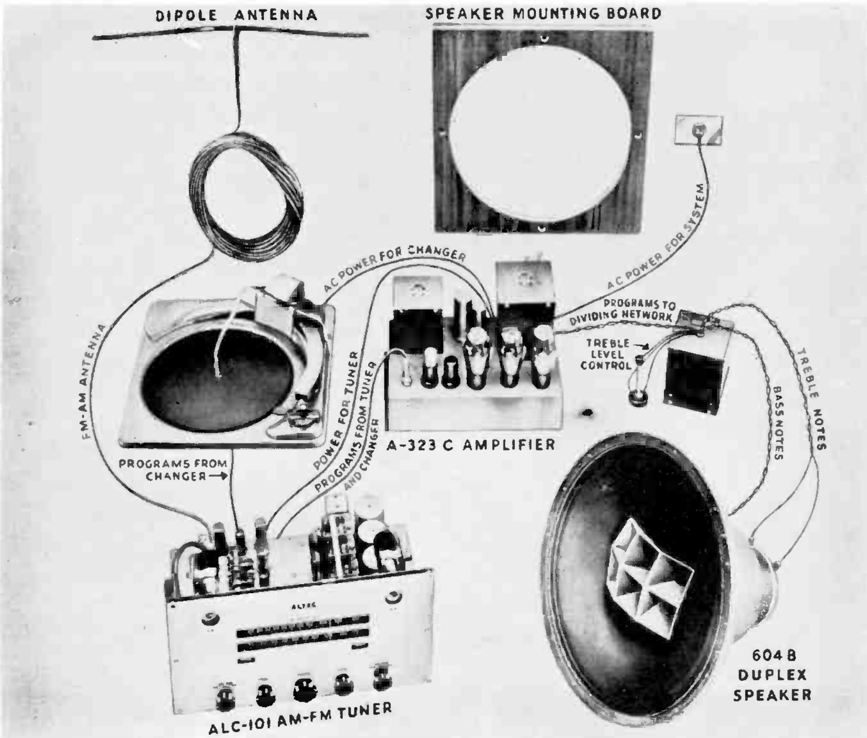
The ribbon microphone is bidirectional in that it is most responsive to sounds originating at its front and rear, but least responsive to sounds originating at its sides. This characteristic is of considerable value in the solution of some of the difficulties usually encountered in reverberant locations, because it reduces the effect of undesired sound reflections. Also, it increases the possibilities of obtaining better balance, clarity, naturalness and selectivity in sound pickup. When used for public address purposes, the directional characteristic of the ribbon microphone permits the reduction of feedback effects between the microphone and loudspeakers.

MICROPHONE PLACEMENT

Some of the possible ribbon microphone placements, illustrating the use of its directional characteristics, are given in the diagrams of Figure 1. In any particular direction the sensitivity is directly proportional to the length of a line drawn from the microphone to the point where it intersects one of the circles.

Figure 1A shows the general arrangement for a soloist and piano. The actual distance must be determined by the strength of his or her voice, but under no conditions should it be less than 2 feet. As shown by the dotted-line piano, interesting effects may be obtained by changing the angle of the microphone with respect to the piano, thus changing the ratio of reverberation to direct pickup.

For stage plays, the movements of the actors seeking advantageous positions may be reduced by utilizing the directional characteristics of a ribbon microphone as shown in the arrangement of Figure 1B. If the microphone is used by a person seated at a table or desk, it should be placed so that it picks up sound directly from him rather than that reflected from the surface of the table.



Picture diagram of complete home music system shown in preceding illustration.

Courtesy Altec Lansing Corporation

A studio arrangement of a ribbon microphone, for soloist and orchestra is given in Figure 1C. Because of the fact that artists and announcers cannot work closer than two (and preferably three) feet from the microphone, difficulty is sometimes experienced in obtaining the proper balance between the artist or announcer and the orchestra. This difficulty may be overcome by using two microphones, one for the orchestra and one for the artist or announcer. When this is done, the artist's microphone should be oriented so that its dead zone is toward the orchestra.

In public address work, usually the microphone can be placed within three or four feet of the artist, but to prevent acoustic feedback, it should be arranged so that its direction of minimum pickup is toward the loudspeaker system. In certain cases it may be necessary to have a microphone installed at each side of the stage to give the artist the required freedom of movement.

The cardioid microphone, the response characteristic of which is shown in Figure 2, is useful where it is necessary to discriminate against sounds coming from the rear and which would be picked up by the pressure-difference type. For this reason the cardioid type is also very popular for p-a work.

PHONO PICKUPS

In a complete p-a system, it is desirable and often necessary to make provisions for the reproduction of phonograph records, either to provide musical programs or supplement the voice and music of live performers. For this service, it is necessary to install a phonograph turntable and pickup, the output of which is similar in frequency but of higher amplitude than that of most microphones.

As will be explained in a following lesson, many types of phono pickups have been developed, but where cost is an important factor, the best performance will be obtained from the crystal-type like those generally found in the better home type phonograph equipment designed for 10 inch and 12 inch records. Good magnetic and dynamic pickups are more expensive, but generally have wider frequency range with lower distortion and usually are used with equipment designed for the professional type of 16 inch transcriptions.

The frequency response of all types is affected to some extent by the mechanical resonance of the pickup itself and of the pickup and arm as a whole, distortion due to non-linear relationship between the movement of the needle and the output voltage, non-uniform output over the frequency range, wear of the record due to the weight of the pickup and arm, poor tracking, mechanical damping, etc.

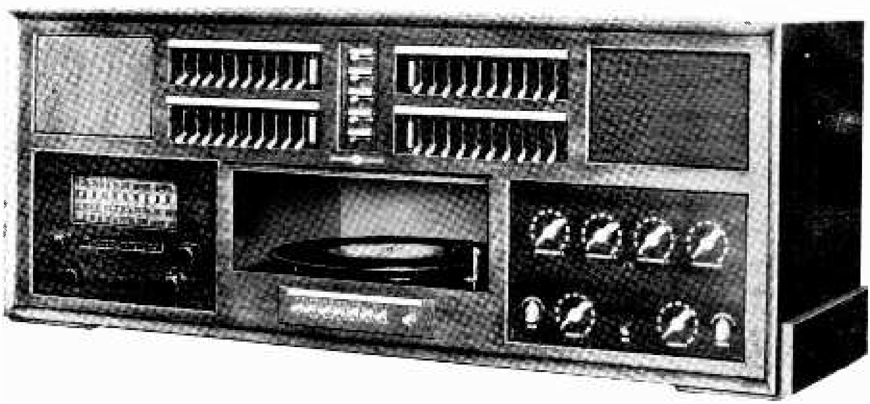
In order to minimize hum, the line between the pickup and amplifier should be as short as possible and placed in a grounded shield. Also, the pickup arm, the turntable, and the turntable motor should be grounded. Even with these precautions, hum is often experienced. Some types of motors induce hum into certain pickups, and this can be discovered by moving the pickup to and from (or across) the record. With certain types, hum may be reduced by reversing the connections to the motor.

In many cases it is desirable that arrangements be made so that the p-a system can reproduce radio programs as they are broadcast. For this purpose there are special types of radio receivers, called radio tuners, which include all the high frequency tuned circuits and terminate at the detector. As far as the p-a amplifier is concerned, this is merely another type of input and as the amplitude of its output compares quite closely to that of a phono pickup, usually it is connected to the phono input circuits of the amplifier. Some commercial p-a amplifiers provide a connection specifically designated as the radio input.

Thus, a complete p-a amplifier may make provision for three types of input signals, (1) Microphone, (2) Phono and (3) Radio. These are classed according to the source of the signals because the frequencies of all of them are in the audio band. The amplifiers of the larger p-a systems may include input circuits for one, two or more of each type of input with the necessary controls for each.

P-A AMPLIFIERS

The explanations of the earlier lessons described the various separate circuits of the main types of audio amplifiers and controls but, for a brief review, we show them all combined in Figure 3 and will trace the paths of the signals from the input circuits to the output transformer.



School sound system with radio, microphone, and phono input. By means of selector switches the sound can be distributed to the various rooms in a building.
Courtesy Stromberg-Carlson Company

Starting with input circuit of Mic. #2, the output of the microphone develops a voltage across the grid load R_2 and thus applies the signal to the control grid of V_2. The bias voltage for this grid is obtained from the drop across R_3 which carries the cathode currents of V_1 and V_2. The signal voltage on the control grid of V_2 causes variations of plate current which, in turn, vary the voltage across the load resistor R_{10}. These voltage changes are carried over to the left hand grid of V_3 through coupling condenser C_3 and potentiometer R_{12} which acts as the grid resistor and also as a volume control for this channel.

The plate voltage for V_2 is obtained from a tap on the voltage divider made up of resistors R_{11}, R_5 and R_4. The screen-grid voltage is also obtained from this divider, but is tapped off between R_5 and R_4, and thus the screen operates at a lower potential than the plate. The bias on the left hand grid of V_3 is obtained from the voltage drop across R_{13}.

With the signal applied to the left hand grid of V_3, there will be a variation of current in the load resistor R_{15}, and the voltage drop across it is carried over to the control grid of V4 through the coupling condenser and grid load R_{18}. This coupling condenser is made up of C_8 and C_9, so that its total impedance can be controlled to a great extent by varying the position of the contact of R_{17}. The screen grid and plate of the pentode V_4 are connected together so that the tube operates as a triode.

The plate circuit of tube V_4 is coupled to the grid circuits of the output tubes, V_5 and V_6, by means of transformer T_1. Notice here that instead of the usual center tap for push pull operation, the secondary consists of two separate windings to permit inverse feedback action by means of the voltage drops across resistors R_{22} and R_{23}. The plate circuits of the output tubes are coupled to the speakers by output transformer T_2 which has a center tapped primary and two secondaries, each of which has several taps to provide various turns ratios for matching different combinations of speakers.

When the output voltage of a microphone is applied to the Mic. #1 input channel, it will appear across R_1 and be applied to the grid of V_1. Like V_2, this tube obtains plate and screen grid voltages from a divider arrangement consisting of R_{11}, R_7 and R_6 and its bias from the common cathode resistor R_3, with its bypass condenser C. Variations of voltage on the control grid cause changes of current in the load resistor R_8.

The resulting variations in voltage drop are applied to the right hand grid of V_3 through the coupling condenser C_1, and the tapped variable resistor R_9. This, of course, assumes that the movable arm of R_9 is in its present position or somewhere between grounded center tap and the condenser C_1 end. By changing the position of the arm of potentiometer R_9, between the center tap and the coupling condenser C_1 end, the amount of signal voltage from this channel can be controlled. Because the plates of V_3 are connected directly to each other, a signal on the right hand grid of V_3, is carried on through the succeeding stages the same as explained for the Mic. #2 channel.

If the output of a phonograph pickup is applied across the phono input connections, there will be a corresponding voltage drop between the upper end of R_9 and ground. Therefore, to apply this signal to the right hand grid of V_3, it will be necessary to set the movable contact of R_9 somewhere between its upper end and the center tap. The phono pickup feeds directly into this second stage of V_3 because its output voltage is considerably higher than that of a low-level microphone, and therefore requires less amplification.

With the movable arm of R_9 in the position shown, the input signal to the right grid of V_3 will be from the Mic. #1 channel, and when the arm is moved into the upper position, the input signal will be from the phono input. Thus, R_9 not only controls the volume of these two channels, but it acts also as a fader so that the signal from either channel can be applied to the right-hand grid of V_3. In changing from one channel to the other there is no sudden cut-off, but when moving the arm from one end to the other, after the volume of one is reduced to zero, the volume of the other starts to increase. It is because of this gradual change from one signal source to another, that a control like R_9 is commonly referred to as a Fader.

Let us assume that a pickup is connected across the phono input and the fader is in a position to apply the signal to the right-hand grid of V_3. Also, we will assume that a microphone is connected to the Mic. #2 channel, so that its output will be applied to the left grid of V_3 through V_2 and the coupling network.

Under these conditions, there will be two separate and distinct signal sources applied to the respective grids of V_3. However, because the plates of V_3 are tied together, there can be but one value of signal current which will have a waveform dependent on the values of the two signals. In other words, the signals will mix, and because this has been accomplished electronically, a circuit of this kind is known as an Electronic Mixer. Notice that either the phono or Mic. #1 can be mixed with Mic. #2.

Thus control R_9 serves as a fader for the phono and Mic. #1 channels as well as a volume control for each, while control R_{12} serves only as a volume control for the Mic. #2 channel. Operated together, these two controls serve as a mixer for signals in Mic. #2 channel and those in either the phono or Mic. #1 channels.

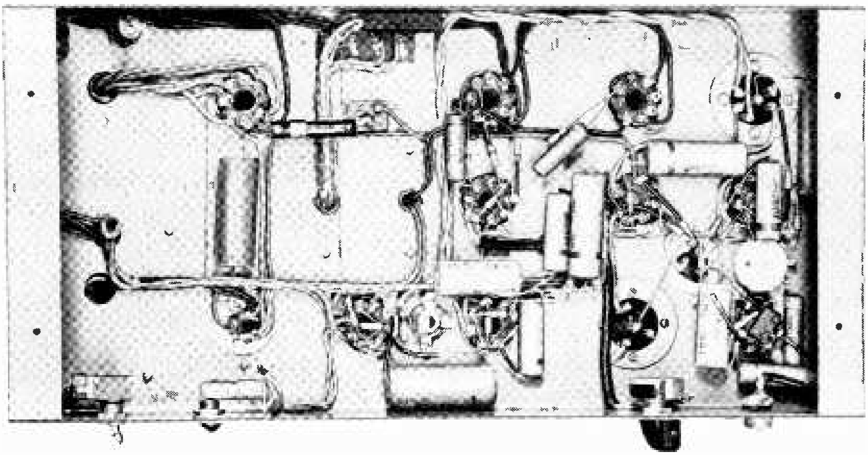
Possibly you are wondering about the advantage of such an arrangement, and a good example is a piano accompaniment for a vocalist. The usual procedure is to place one microphone close to the piano and the other in a position so that the vocalist can be more or less at a distance from the accompanying music. By properly manipulating the controls, the most desired sound levels of each can be obtained. Without a mixer, it would be necessary to use only one microphone; and it would be impossible to regulate the two signals separately in order to obtain the desired relative levels.

The heart of a p-a system is the amplifier.
Courtesy Thordarson Electric Manufacturing Division

P-A AMPLIFIER OUTPUT CIRCUITS

As with other electronic equipment, most audio amplifiers are of the highest quality design consistent with the total allowable cost of the various tubes, transformers, condensers, etc. employed in the circuit. This accounts for the great variations in design, as well as price, of the various units commercially available, the power output or fidelity of which depends upon the requirements of the application for which they are intended.

The use of inverse feedback to improve the tone quality by reducing distortion has been explained in an earlier lesson, and as mentioned previously, an example of its application to a push-pull output circuit is given in the p-a amplifier diagram of Figure 3. The use of resistor-condenser feedback coupling networks is the most common method employed, and is suitable when the range of frequencies to be amplified is such that the reactance of condensers, such as C_{12} and C_{13} in Figure 3, is negligible.



Layout of underchassis parts and wiring of amplifier shown in previous illustration.
Courtesy Thordarson Electric Manufacturing Division

However, in the case of large commercial public-address installations, where cost of equipment is of lesser importance, and where real high-fidelity amplification is required, an output circuit like that of Figure 4 may be used. Here the secondary of input transformer T_1 consists of two windings, the inner end of each being connected to a tertiary winding in the special output transformer T_2. The grid bias of tubes V_1 and V_2 is supplied through this tertiary winding, the center tap of which connects to C—.

Besides providing inverse feedback without frequency discrimination; the arrangement of Figure 4 has the further advantage that, with negligible grid circuit resistance, more power output can be obtained by operating the tubes in classes of amplification which require grid current.

For p-a installations requiring large power output, two or more tubes may be operated in parallel as shown in Figure 5. Here, the respective tube elements are connected together, grid to grid, plate to plate, etc. The advantage of this arrangement over a push-pull circuit is that, with an input voltage equal to that required for only one tube, the output power available is equal approximately to that of a single tube multipled by the number of tubes used. However, the parallel system does not provide the cancellation of even order hamonics, etc. as does push-pull operation.

Since the effective plate resistance of two tubes in parallel is equal to one-half that of one tube, the required plate load for maximum undistorted output is only one half that for a single tube. This means that the turns ratio of the output transformer, T_2 in Figure 5, must be such that the total load impedance reflected from the speaker voice coils, is half of that required when only one tube is employed.

The parallel tubes are operated at the same bias as for single tube operation, and they can be biased separately or by a common bias resistor as shown. In the calculation of R in Figure 5, it must be remembered that although the required voltage drop is the same, the current in the resistor is twice that for one tube. Sometimes the parallel tubes are biased separately in order to compensate for slight differences in their plate current characteristics.

Although the same power output could be obtained by substituting a single tube of double the rating of each parallel tube, these larger tubes usually require higher plate voltages. The advantage of multi-tube operation for the same power output is the reduced size and cost of the power supply.

the features of both the circuits of Figures 4 and 5 is the push-pull parallel output stage shown in the schematic diagram of Figure 6. Here, output tube V_3 is connected in parallel with V_5, while V_4 is in parallel with V_6, and the two parallel groups are operated in push-pull. This system permits at least four times the output of one tube, with the attendant hum and distortion cancelling advantages of push-pull operation. In fact, by employing a push-pull power amplifier driving stage, V_1 and V_2, sufficient driving signal power is available so that four tubes such as beam power type 6L6 can be operated in class AB_2 and provide about 100 watts output power with reasonably low distortion.

The output transformer T_3 is of the special inverse feedback type described for Figure 4, and contains a terminal board to which connections can be made to various taps on the secondary winding. This permits the necessary impedance match to be made to whatever combination of loudspeakers may be employed. As shown, a choice of several different output impedance values is available. To permit better regulation of the C− and screen-grid voltages, a separate power supply, $B+_2$, is used for the output tube plates only.

LOUDSPEAKERS

The problems concerning p-a loudspeakers are very important and deal with the type and number required, their individual power requirements, directivity, and placement. Of equal importance is the manner in which the various speakers are connected to the output of the p-a amplifier and to each other.

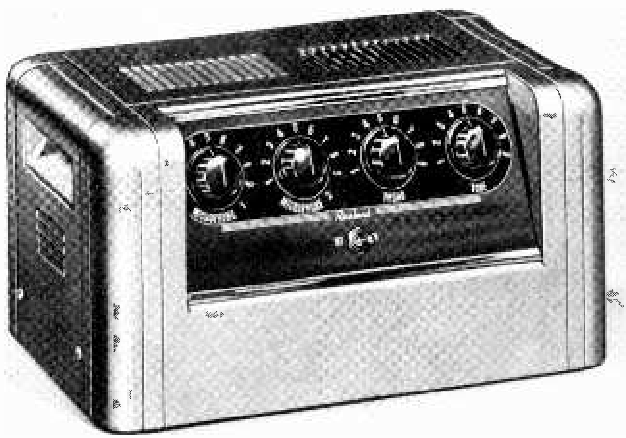
Speakers generally are classified according to their power handling capabilities, their size, shape, and their frequency response. The power handling capability of a speaker is the measure of the electric audio frequency power which can be impressed on its voice coil without rasping, rattling, burn-out, or destruction of the cone, diaphragm, or voice coil structure. The size of a speaker is taken as the diameter of the cone, or, in the case of horns, by the length of the horn and the width of the bell. The shape is self-explanatory, such as round or oval cones, square horns, etc., while the response concerns the frequency range over which the speaker output is essentially constant.

Besides this, a speaker is described technically by the impedance, resonance point, power capacity, and size of its voice coil; by the d-c resistance or the inductance of its field coil, by the weight of its permanent magnet; and also by its exact overall dimensions or those of a particular part. Depending upon the use to which the speaker is to be put, any or all of these technical facts must be known.

PLACEMENT OF SPEAKERS

Whenever sound is heard from two or more sources, such as a performer and one or more loudspeakers which are at different distances from the listener, the difference in the time it takes the sound to reach the listener's ears causes an echo effect, and the intelligibility is reduced. Therefore, the most desirable location for all the loudspeakers of a p-a system is at the exact source of the sound. For a number of reasons, this ideal arrangement can not always be duplicated in actual practice. No one speaker should be employed which is capable of the high sound level output required for a large outdoor gathering, or a large noisy factory. If one speaker were used to cover an area of this size, its sound output would be so high as to be very uncomfortable for the listeners located near it.

Where the area to be served is small enough to be covered by one speaker, or by a single group of speakers, the units should be placed above the center of the stage or bandstand and oriented so that there is a minimum of overlap of their coverage areas. In this case, the selection of the type of speaker as regards size, power capabilities and coverage angles, depends upon where they are to be located in a given room or hall.



P-A amplifier showing various operating controls.
Courtesy The Rauland Corporation

For example, in the arrangement of Figure 7A the sound system is located at the middle of one side of a rectangular room and the area is covered best by means of three baffle mounted cone type speakers having wide coverage angles. In Figure 7B, with the microphone at the end of the room or auditorium, best coverage is obtained with the narrower angle projector type speakers mounted above the stage in the position shown. Although, there is some overlap of the coverage areas in the region at x, this causes but little trouble because most listeners in this area are about equidistant from both speakers.

Where a large area is to be covered, speakers may be placed at intervals along the walls as shown in Figure 7C. This represents one possible arrangement for a factory or similar installation. If the noise level is very high, projector or trumpet type speakers may be needed, since they concentrate their output into a small angle so that it can override the machinery noise, etc. Since the narrower the coverage angles employed, the larger the number of speakers needed, in this case the speaker placement depends upon the types of units used.

A fourth arrangement is shown in Figure 7D. Here, three rooms are supplied with sound by the same p-a system. Whereas a single speaker might be sufficient to cover the total area of all rooms, the partitions (inside walls) require the use of a separate speaker for each. An advantage of the particular placement shown is that speakers A and B are approximately equidistant from the doorway at X as are speakers A and C from the doorway at Y. Thus, the delay in sound waves arriving from either pair of sources is reduced to a minimum in the doorways, the only points at which two speakers would be heard at the same time.

ACOUSTIC FEEDBACK

So far, the placement of the speakers has been considered only with respect to the area of sound coverage but when located in the same room as the microphone, their relative positions are important. Under no conditions should there be a direct return path for sound from the speakers to the microphone. This plan is followed in the arrangements of Figure 7A and 7B where the microphones are placed at the rear of the speakers. However, under these conditions, some sound reflected by the outer walls will reach the microphones. Like the original, this reflected sound will be picked up by the microphone, amplified and reproduced by the speakers.

Because of its travel from the speakers to the outer walls and back, the reflected sound will reach the microphone a short time later than the original and, if it is of low amplitude, will produce an echo. As a simple illustration, assume the original sound is a sustained note and the reflected sound reaching the microphone has the same level as the original. This condition will double the microphone input, and approximately double the speaker output which, in turn will double the level of the reflected sound. This building up of microphone input and speaker output will continue until the amplifier is overloaded and produces only a sustained howl.

This type of feedback can be recognized readily by its gradual build-up, the time of which depends upon the travel distance of the reflected sound. If low frequencies are reflected at a higher level than others, the acoustic feedback will cause a rumble or speaker rattle. If high frequencies are reflected at a higher level, the acoustic feedback will cause a whistle.

To avoid this action, the common procedure is to locate the speakers and microphones in the most desirable positions, as previously explained and then, with the system in operation, to turn up the amplifier gain until acoustic feedback occurs. The gain is then reduced to a point of stable operation and, if the resulting sound level is satisfactory, nothing more need be done except to remember the point of maximum allowable gain. However, if the maximum allowable gain does not provide sufficient sound levels, adjustments by either acoustic or electric methods must be made to reduce the reflected sound.

Acoustically, the speakers or microphones can be moved or the microphone can be shielded with sound absorbent material. Electrically, tone controls in the amplifier can be adjusted to attentuate the particular frequencies, which are reflected at the highest levels.

SOUND LEVELS

In the selection of loudspeakers for a particular p-a installation, the question of power requirements must be answered at the same time that the problems of coverage angle and speaker placement are being considered. It has been mentioned that the required power depends upon the size of the area to be covered as well as its ambient or normal noise level. Therefore, the noise level must be determined first to permit the selection of a speaker having the proper wattage rating.

For large installations with a number of speakers, the power requirements of each are determined separately and then added to find the total power which must be provided by the amplifier.

The basic purpose of all p-a equipment is to reproduce sound and, in the speakers, the electrical power output of the amplifier is converted into sound energy or power. The electrical power can be measured accurately in terms of watts or volts and amperes but there is no absolute unit for the measurement of sound. Instead, measurements are made by comparing the intensity of one sound with that of another. Experiments have shown that for any intensity, sound power must be increased by approximately 25% before the difference is noticeable to the average human ear. This ratio, used as the unit of sound measure, is called a "decibel" abbreviated "db"

For an increase of 25%, the power ratio is,

$$\frac{P_2}{P_1} = \frac{1.25}{1.00} = \frac{5}{4}$$

and the resulting change of sound is one db. For a second 25% increase of power, the ratio is.

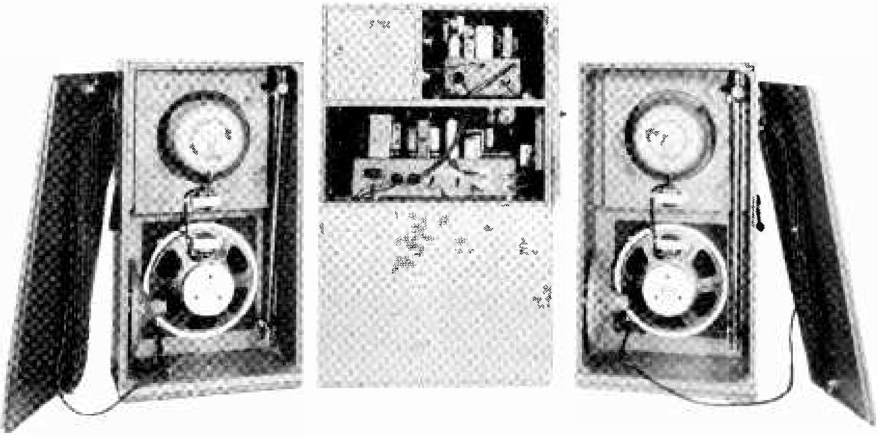
$$\frac{P_2}{P_1} = \frac{5}{4} \times \frac{5}{4} = \frac{5^2}{4} = \frac{25}{16} = 1.56$$

and the resulting increase of sound is two db. Following the same plan, for a third 25% increase, equivalent to three db, the power ratio is,

$$\frac{P_2}{P_1} = \frac{5}{4} \times \frac{5}{4} \times \frac{5}{4} = \frac{5^3}{4^3} = \frac{125}{64}$$

$$= 1.95 = 2 \text{ approx.}$$

This method becomes somewhat involved for the common



Portable p-a system suitable for indoor and outdoor use.

Courtesy David Bogen Company, Inc.

problems of calculating the number of db equivalent to a given power ratio. Therefore, as the relationship is logarithmic, usually it is stated as,

$$db = 10 \log \frac{P_2}{P_1}$$

Applying this equation to the last example given above, from a table of common logarithms we find, log 2 = .3010 and substituting this value for the power ratio,

$$db = 10 \log \frac{P_2}{P_1} = 10 (\log 2)$$

$$= 10 \times .3010 = 3.01$$

Assuming the measurements are made with resistances of equal value, voltage and current ratios may be converted to equivalent db values by means of the following equations.

$$db = 20 \log \frac{E_2}{E_1}$$

$$db = 20 \log \frac{I_2}{I_1}$$

By means of these equations, power, voltage and current ratios can be converted to equivalent values of decibel gain or loss. For example, a three stage amplifier with a voltage gain of 20 per stage has an overall gain of 20 x 20 x 20 = 8000. Expressed in decibels, for each stage

$$db = 20 \log \frac{E_2}{E_1} = 20 (\log 20)$$

$$= 20 \times 1.301 = 26.02$$

For the overall gain,

$$db = 20 \log \frac{E_2}{E_1} = 20 \log 8000$$

$$= 20 \times 3.901 = 78.06$$

Expressed in db, the total gain is equal to the sum of the stage gains.

Because the decibel represents a ratio between two values, it is convenient to establish an arbitrary reference level which can be considered as O db. For p-a work, one common reference level for power is .006 watt which is 6 milliwatts. For voltage and current ratios the common reference values for resistance are 600 ohms or 500 ohms.

On this basis, for an amplifier which, with an input of 6 milliwatts provides an output of 24 watts,

$$\text{Power gain} = \frac{P_2}{P_1} = \frac{24}{.006} = 4000$$

$$db = 10 \log \frac{P_2}{P_1} = 10 \log 4000$$

$$= 10 \times 3.602 = 36.02$$

The 6 milliwatt reference level was chosen because, in telephone systems, it was the minimum electrical power required to produce audible sound. However, sound consists of waves of varying frequency and pressure and can be produced by other than electrical methods. Disregarding its source, the minimum amount of sound energy, necessary to produce audible sound has been standardized as .000,000,000,-000,000,1 watt per square centimeter. This is called the "Threshold of Audibility" and is the 0 db reference of the sound level scale. For a person with good hearing, sound will be barely audible at 0 db but will be so loud as to be painful at 120 db.

The following table shows a few representative sound levels, measured in decibels, above the threshold of audibility:

TABLE I

Source	Level in db
Threshold of Pain	120
Hammer blows on steel plate—2 ft.	114
Pneumatic riveter—25 ft.	100
Factory, heavy manufacturing	80
Very heavy street traffic—25 ft.	80
Factory, light manufacturing	70
Large office or store	65
Restaurant or residential street	60
Ordinary conversation—5 ft.	60
Garage	55
Hotel or apartment (city)	42
House (country)	30
Average whisper—5 ft.	18
Quiet whisper—5 ft.	10
Threshold of audibility	0

As listed, ordinary conversation level at 5 feet is 60 db, and to make oneself heard in a factory this level must be increased to 80 or 85 db to prevent the factory noise masking the speech sounds. This may be done by talking more loudly, by decreasing the distance between talker and listener, or both. Thus it is seen that the sound level at the listener's ear depends upon the distance from the source as well as the level at the source.

For installation work, sound levels are measured with a "sound level meter" which consists of a microphone, amplifier, and an indicating meter calibrated in decibels above the threshold of audibility. Where such a meter is not available, a rough estimate can be made by comparison with the levels of the sources listed in the above table. However, this method can provide only approximate values because of the wide variations which exist in individual cases.

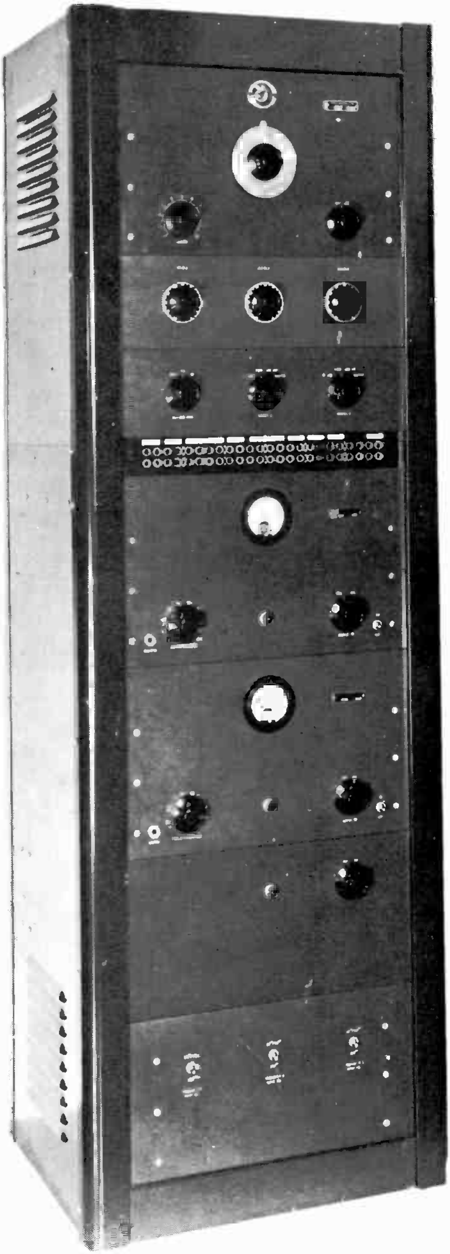
POWER REQUIREMENTS

The following Table II lists the coverage angle, the sound level at 10 feet with 1 watt input, and the recommended power input for four representative commercial loudspeakers:

TABLE II

Speaker	Coverage Angle degrees	Sound Level at 10 ft. 1 w input decibels	Power Input watts Ave.	Power Input watts Max.
5″ Dia. cone type	140°	79	3	5
Cone driver (5″ dia.) in small reflex horn (10″ bell)	80°	82	3.5	6
12″ Dia. cone type	140°	89	—	10
Metal diaphragm unit with reflex horn (20″ dia. bell)	54°	105	15	25

The listed ratings will vary somewhat with different makes of speakers, and therefore, like Table I, the data is only an approximation of the performance of any particular units which may be employed.



P-A rack and panel assembly mounted in steel cabinet.
Courtesy Presto Recording Corporation

The sound level values of Table II are based on a distance of 10 ft. and an input power of 1 watt but in actual installations, both of these will vary. Therefore, the chart of Figure 8 indicates the relationship between sound level and distance. The curve crosses the 0 db line of the left hand vertical scale on the 10 ft. ordinate of the lower horizontal scale to coincide with the table. Reading the chart, if the distance is reduced to 1 ft. the sound level is + 20 db while, if the distance is increased to 100 ft. the sound level is reduced to − 20 db.

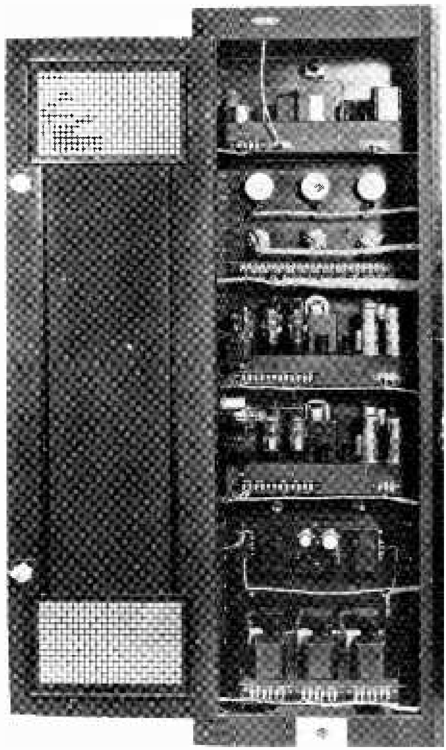
Following the same general plan, the chart of Figure 9 indicates the relationship between speaker input power and sound levels. Here, the curve crosses the 0 line of the left hand vertical scale at the 1 watt ordinate to coincide with the table. Reading the chart, if the power is reduced to .1 watt the sound level drops to —10, while if the power is increased to 10 watts, the sound level rises to + 10. When the ambient noise level is known, by combining the data of Table II with the variations listed by the charts of Figures 8 and 9, the speaker power requirements for any particular p-a installation may be determined.

As an example, suppose it is desired to install a p-a system to provide background dinner music in a restaurant, and to give the effect of sound emanating from a single source, a grouping of three speakers similar to the arrangement of Figure 7A has been decided upon. The greatest distance that the sound must be projected is from the speakers to the far corners of the room, and a rough measurement shows this to be about 40 feet.

A sound level test during the busiest hours proves the ambient noise level to be about equal in all parts of the room, and approximately the value given in Table I, 60 db.

Experience has shown it is desirable that the sound level produced by the loudspeakers be from 5 to 20 db above the noise level, depending upon the use to which the p-a system is to be put. Since in this example its main use is for background music, the speaker output need not be very high as far as volume is concerned. However, in order that all of the high frequencies be heard, and since they largely determine the quality of the music, a sound-over-noise margin of 10 db is decided upon. Thus, the sound level desired is 60 db (noise level) plus 10 db (margin) or 70 db, at 40 feet from the speakers.

Table II shows that the level at 10 feet from a 12″ cone type speaker is about 89 db with 1 watt of input power. From Figure 8 it is found that at 40 feet from the speaker, the level is 12 db less than the value at 10 feet, or $89 - 12 = 77$ db. These values refer to the level on an axis perpendicular to the face of the speaker, and as seen in Figure



Cabinet door open showing internal arrangement of the p-a equipment presented in the previous illustration.

Courtesy Presto Recording Corporation

7A, a line drawn from one of the far corners of the room to the nearest speaker would be at an angle to the speaker axis. Since the sound level decreases as this angle increases, for the arrangement shown in Figure 7A, it is conservatively estimated that the level in the far corners of the room will be about 80% of the above value, or 77 x .8 = 61.6 db.

To increase this to the desired level of 70 db, an increase of 70 − 61.6 = 8.4 db is necessary. Figure 9 shows that to increase the sound level to 8.4 db above the level with 1 watt input, the speaker input power must be raised to a value of about 7 watts. This can be done since this value falls well under the recommended maximum power input for the 12″ cone type speakers, as given in Table II. Note that if it were attempted to employ the 5″ cone type speakers, for example, the required input power would be much higher than the maximum value allowable.

Since three such speakers are to be used, Figure 7A, the total power needed is 21 watts, requiring an audio amplifier having an output of from 20 to 25 watts. To provide a reserve which may be needed when the normal noise level is higher than usual, a 30 watt amplifier would be more desirable.

IMPEDANCE MATCHING

In the Lesson on Microphones, the purpose of impedance matching transformers was described and in the Lesson on Audio Amplification, the action by which the output transformer matches the plate load of the power amplifier tube to the speaker voice coil was explained. With most p-a systems, it is necessary that a number of speakers, often with different voice coil impedances, be properly matched to the output of the audio amplifier; and it is highly important that the installing technician be thoroughly acquainted with the problems involved. Therefore, before taking up the procedures employed in connecting speaker circuits, the necessity of providing the correct impedance match will be explained.

As has been stated, the maximum transfer of energy from a source to its load takes place when their respective impedances are equal. This can be shown by means of the simplified circuit of Figure 10 where Z_G represents the internal impedance of generator G which produces the voltage E_G, while W_o is the useful output power appearing in the load Z_L. Letting $E_G = 100$ volts and $Z_G = 2500$ ohms, for the first case suppose Z_L is equal to Z_G, and, for simplicity, that they are both resistive. Then the total circuit impedance $Z_T = Z_G + Z_L = 2500 + 2500 = 5000$ ohms. The circuit current, found from Ohm's Law is:

$$I = \frac{E_G}{Z_T} = \frac{100}{5000} = .02 \text{ ampere,}$$

and the output power in Z_L is:

$$W_o = I^2Z_L = .0004 \times 2500 = 1 \text{ watt}$$

For the second case, let the load impedance be less than that of the source. If $Z_L = 500$ ohms, with all the other values the same as above, then:

$$Z_T = Z_G + Z_L = 2500 + 500$$
$$= 3000 \text{ ohms},$$

$$I = \frac{E_G}{Z_T} = \frac{100}{3000}$$
$$= .033 \text{ ampere (approx.)}$$

and

$$W_o = I^2Z_L = .001 \times 500$$
$$= 0.5 \text{ watt (approx.)}$$

For the remaining case, let $Z_L = 7500$ ohms, which is greater than the value of Z_G. Then:

$$Z_T = 2500 + 7500 = 10{,}000 \text{ ohms},$$

$$I = \frac{E_G}{Z_T} = \frac{100}{10{,}000} = .01 \text{ ampere},$$

and

$$W_o = I^2Z_L = .0001 \times 7500 = 0.75 \text{ watt}.$$

As seen, in both the second and third cases, in which the value of the load Z_L is different from that of the source Z_G, the output power W_o is less than in the first case where Z_L is equal to Z_G. It is for this reason that in electrical circuits involving the transfer of power, the necessity of obtaining the desired impedance relationships is very important.

Direct-coupled amplifier housed in steel cabinet.
Courtesy Amplifier Corporation of America

The impedance values of speaker voice coils are much lower than the required load impedance of an output tube plate circuit. However, as explained in the Lesson on Audio Amplification, by virtue of its primary-secondary turns ratio, the output transformer changes the respective voltage and current values in such a way that the speaker voice coil impedance can be made to reflect, or appear to be, the same value as the required plate load of the tube.

Since audio output transformers usually have a secondary with several taps as shown in Figures 3 and 6, the currect turns ratio may be employed automatically simply by selecting the tap, the rated value of which corresponds to the total impedance of the circuit containing all the speakers which are to be supplied with power by the p-a amplifier.

CONNECTING SPEAKERS

In common commercial practice, the impedance of loudspeaker voice coils is measured at 400 cps, though there are cases of special coils which are measured at 1000 cps. The values range from 3 ohms to about 45 ohms, and in connecting a speaker, this ohmic value is of greater importance than the frequency at which the voice coil was measured.

The output transformers on most commercially built amplifiers are tapped for 4, 8, 15, 250 and 500 ohms, as shown in Figure 6. However, some of the larger units are equipped with various additional taps having values such as 2, 14, 16, 30, 50, 60, 125, 200 and 350 ohms.

For the simplest case, a speaker with a 4 ohm voice coil would be connected to the 4 ohm tap and common terminal C on the output transformer of Figure 6, while the 8 ohm tap and C would be used if the speaker were an 8 ohm unit. However, if it should be necessary to connect a 15 ohm speaker to a transformer which has only 16 ohm and 14 ohm taps, the 16 ohm tap should be used. This will reflect a somewhat lower value of plate load so that it is possible to obtain slightly more power from the amplifier, although the distortion will be somewhat greater at the peak output. If the speaker were connected to the 14 ohm tap, the reflected plate load would be somewhat higher than the recommended value and tend to cause overloading at a lower value of power output.

When two 16 ohm speakers are to be used with an amplifier, they can be connected in parallel so that their total impedance is equal to 8 ohms, and they may then be connected to the 8 ohm transformer tap. Likewise, two 4 ohm speakers can be connected in series for a total impedance of 8 ohms, and then connected across the transformer 8 ohm and C terminals. This series arrangement, however, should be made only if it is absolutely necessary because of the following situation.

The impedance of a speaker voice coil as an electrical unit will change with the frequency of the signal applied to it because of the variation of the air load. At the speaker resonant frequency, its voice coil impedance may rise to four or six times its value at the measured frequency (400 or 1000 cps). Normally, for frequencies other than at resonance, the speaker impedance does not depart greatly from the measured value and its reproduction of sound is not greatly affected.

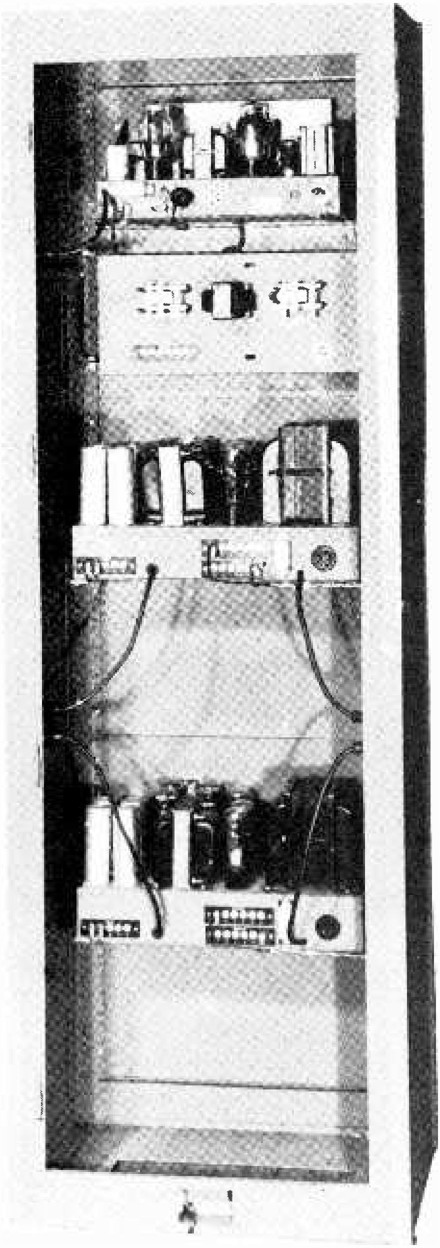
Since no two speakers have exactly the same resonance point, a circuit consisting of two or more speakers in series will have several resonance points. Normally, each of several series connected speakers should take its proper share of the total amplifier output in accordance with the values of their respective voice coil impedances.

Thus, each of two equal impedances would take half of the total power supplied by the amplifier. However, when the signal to be reproduced matches the resonant frequency of one of the speakers, the impedance of this speaker becomes high, and it consumes much more than its share of the output power. Also, it permits less current in the circuit, and as a result, the other speaker is forced to accept the decreased current and its output is reduced.

Likewise, similar actions occur at the resonance point of each of several series-connected speakers, causing undesired reductions in the volume of those units not at resonance. The total result is a highly distorted and unpleasant reproduction of the sound output by the entire system. For this reason, whenever several speakers are to be connected to the same amplifier, it is best that they be connected in parallel. As in all parallel circuits, the voltage across the various branches remains equal, and thus only the speaker which is at resonance will draw less current because of its high resonant impedance.

TRANSMISSION LINES

The speaker connections explained so far apply to cases where the speakers are located fairly close to the amplifier, such as distances of about 15 feet or less. For greater distances than this, one factor which limits the length of the transmission line is the power loss due to the resistance of the wires. In general, the higher the impedance values of the units which the line connects, the lower will be the loss of power due to the impedance of the line. However, as explained in the Lesson on Microphones, the high-frequency losses in a transmission line, due to the capacitance be-



Rear view of industrial p-a system assembled in steel cabinet.
Courtesy David Bogen Company, Inc.

tween the conductors, are greater in a high-impedance line than in a low impedance line. Also, a high-impedance line has greater ability to pick up a-c hum and other undesired interference, but as there is no amplification following a speaker line, this characteristic is seldom important. For these reasons, a 500 ohm line is usually considered to provide an acceptable compromise between the power losses and the undesirable qualities of a high-impedance line.

The term 500 ohm line, as used with reference to audio frequencies, means that a two conductor line, twisted or otherwise, is connected between a source having an output impedance of 500 ohms and a load having an input impedance of 500 ohms. If the respective output and input impedances of these units are 50 ohms each, then this connecting line is said to be a 50 ohm line, etc. This designation should not be confused with what is called the characteristic, or surge, impedance of a transmission line. Transmission line surge impedance depends upon the physical structure of the line itself, and is important at radio frequencies.

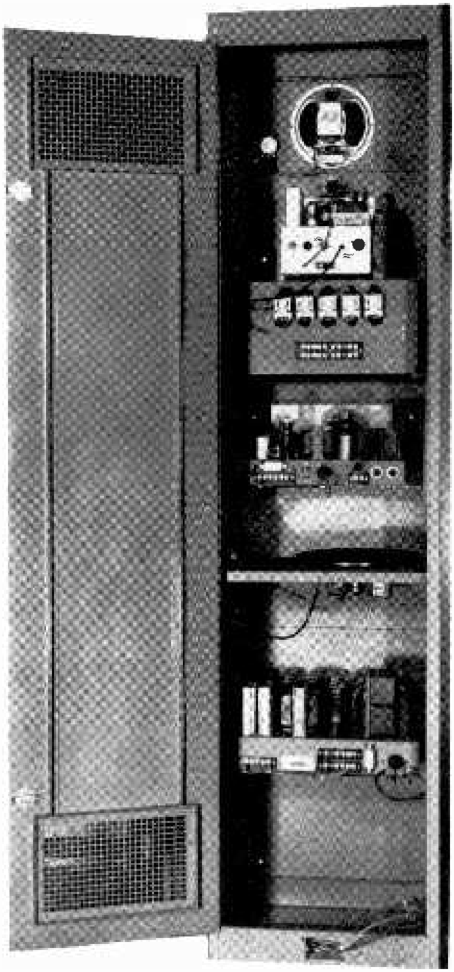
A second factor which limits the length of the transmission line is the impedance mismatch introduced by connecting the wires in series with the speaker load. For example, suppose two

4 ohm speaker voice coils are connected in parallel to the 2 ohm tap on the output transformer of the amplifier in Figure 3 by means of a transmission line 38′ 4″ long. Also, assume that this line is made of No. 18, B and S Gauge wire which, according to the wire table, has a resistance of about 6.5 ohms per thousand feet.

This amounts to 0.0065 ohm per foot, and multiplying this value by the length of the lines gives 0.25 ohm (approx.). Since the transmission line has two wires, the total resistance of the line is 2 × 0.25 = 0.5 ohm. Thus a total impedance of 2 ohms (that of the two speakers in parallel) plus 0.5 ohm (resistance of the line), or 2.5 ohms is connected to the 2 ohm transformer tap. This results in an impedance mismatch of 25% and although experience has shown that a mismatch of 15% is audibly acceptable, 25% is about the maximum permissible in present day sound equipment.

Therefore it is common practice for p-a manufacturers to supply only 35 feet of No. 18 stranded speaker wire. Rather than an economy measure on the part of the manufacturer, this is the longest length that can be used and still remain within the bounds of permissible distortion due to mismatching.

Of greater importance, however, is the power loss due to the resistance in a low impedance line. For example, suppose the amplifier of Figure 3 supplies a total of 30 watts to the speaker and the line load in the above ex-



Complete p-a assembly showing monitor speaker at top, below which is the radio tuner, amplifier, phono turntable, and power supply.

Courtesy David Bogen Company, Inc.

ample. Since the total load impedance is 2.5 ohms, the current in the transmission line is:

$$I = \sqrt{\frac{W}{R}} = \sqrt{\frac{30}{2.5}} = 3.46 \text{ amperes.}$$

The resistance of the line is 0.5 ohm, and therefore the power lost in the line is:

$$W = I^2R = 3.46^2 \times 0.5 = 6 \text{ watts,}$$

which is 20% of the total power supplied by the amplifier.

If the same length of speaker line were connected to the 500 ohm primary of a matching transformer at the speaker end of the line, then with properly matched relations existing at the amplifier output terminals, the total load impedance would become 500 ohms (primary of the speaker transformer) plus 0.5 ohm (resistance of the line), or 500.5 ohms. The current in the line would then be,

$$I = \sqrt{\frac{W}{R}} = \sqrt{\frac{30}{500.5}} = 0.245 \text{ ampere,}$$

and the power lost in the line only:

$$W = I^2R = 0.245^2 \times 0.5 = .03 \text{ watts.}$$

Thus, the use of the higher impedance line reduces the power loss to 0.1% of the total supplied by the amplifier. Furthermore, the impedance mismatch is now reduced to a value of 0.5/500, or 0.1% also.

For installations where the speakers are located at considerable distances from the amplifier, the employment of 500 or 250 ohm lines is necessary, since for such length the power loss in low-impedance lines would be prohibitive. In practice, the lengths of low-impedance transmission lines are limited to values which allow a maximum line resistance equal to 15% of the voice coil impedance. For high-impedance lines, the total resistance of the line should not be more than 5% of the terminating impedance. Based on these percentage figures, the following table gives the maximum length in feet which transmission lines of various wire sizes and impedances may have.

TABLE III

Wire Size (B and S Gauge)	Maximum Length of Line in Feet				
	2 ohms	5 ohms	8 ohms	250 ohms	500 ohms
No. 12	95 ft.	190 ft.	380 ft.	4000 ft.	8000 ft.
No. 14	60	120	240	2500	5000
No. 16	37	75	150	1500	3100
No. 18	28	47	95	1000	2000
No. 20	15	30	60	600	1200
No. 22	9.5	19	38	390	780

LINE-TO-VOICE COIL TRANSFORMERS

The employment of line-to-voice coil or speaker transformers, was mentioned above in connection with the use of high-impedance transmission lines. In a case where several speakers are all to produce the same output volume and in addition are to be connected in parallel across a transmission line, the matching problem may be solved as follows:

Standard line-to-voice coil transformers have primaries which usually are tapped for impedance values of 500, 1000, 1500, and 2000 ohms. Some have additional taps which give impedance choices of 375, 600, 2500, 3000 and 4000 ohms. The secondary impedance of various makes range from 0.05 to 500 ohms. though they usually contain only a few taps, such as 2, 4, 6 and 8 ohms.

First, a tentative value of line impedance is decided upon, and this value multiplied by the number of speakers to be used gives the value of primary impedance which the speaker transformers should have. For instance, suppose six speakers are to be connected to the output transformer T_o, of a p-a amplifier as shown in Figure 11. It is desired to use a 500 ohm line, therefore speaker transformers, T_1, T_2, T_3, T_4, T_5, and T_6 must have primary impedances of $500 \times 6 = 3000$ ohms each. In other words, the total impedance of the six 3000 ohm primaries in parallel is 500 ohms, thus matching the 500 ohm tap of the output transformer T_o to which the line is connected. Of course, each speaker transformer must be equipped with secondary taps which allow it to be matched to its respective speaker.

If in the above example, the speakers are equipped with line-to-voice coil transformers all of which do not have 3000 ohm primary taps, then a 250 ohm line might be used by connecting the leads to T_0 as indicated by the dotted line in Figure 11. In this case each speaker transformer primary must have $250 \times 6 = 1500$ ohms impedance, their secondary connections remaining as above.

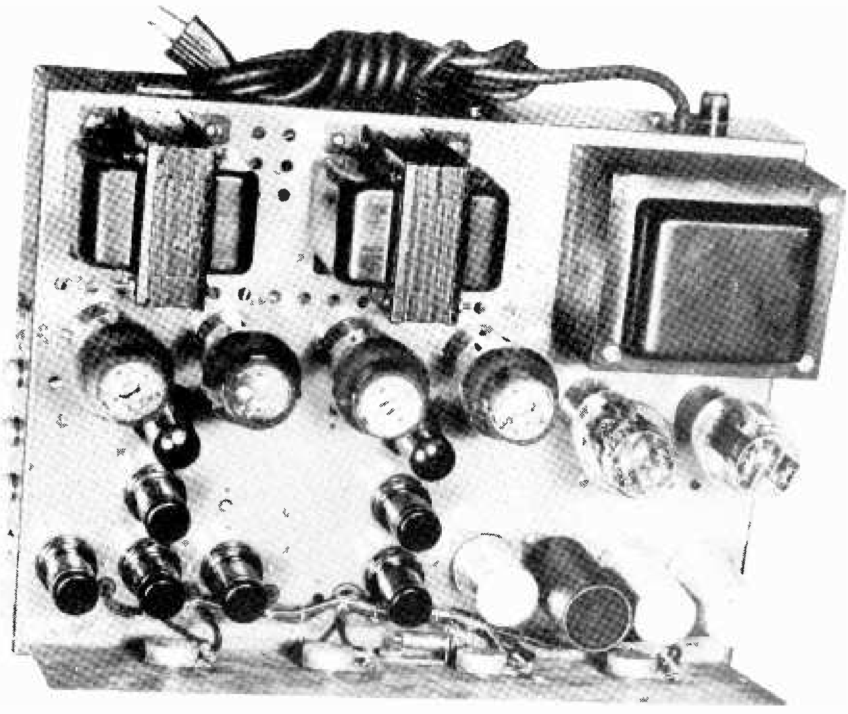
With a 500 ohm line, and but four speakers, the line-to-voice coil transformer primaries should have $500 \times 4 = 2000$ ohm impedances. With a 250 ohm line, their primary impedances should be 250×4 or 1000 ohms.

The total wattage required to operate the group of equal output speakers is equal to the number of speakers times the wattage rating of one. Thus, with six 3 watt speakers, the total power needed is 6×3 or 18 watts, which when allowing for a little reserve, can be handled adequately by a 20 or 25 watt amplifier.

In a case where the group of speakers have transformers with primaries of a single impedance, the line impedance and output transformer tap must be selected in accordance with this value. The total speaker load is found by dividing the value of the matching transformer primary impedance by the number of speakers and then connecting the line to that tap on the amplifier output transformer which most nearly approximates the total speaker load. For example, to use four speakers with fixed 500 ohm transformer primaries, the line should be connected to the output tap which provides a value of 500 ÷ 4 or 125 ohms.

equal output but, with many p-a systems it is necessary that one or a group of speakers receive more power than the others, and that several be of different sizes. When this is the case, the matching procedure is a little more complicated and, briefly, consists of the following:

After the total power require-



Top view of powerful p-a amplifier chassis.
Courtesy Stromberg-Carlson Company

SPEAKERS OF UNEQUAL POWER

So far, these explanations have applied only to speakers having

ment has been determined, a chart like that of Figure 12 is prepared to determine which output and speaker transformer taps are to be used. As such a chart is seldom available, its construction will be explained.

Suppose a p-a speaker system is to consist of three speakers, one of which is to operate at 18 watts, the second at 8 watts, and the third at 4 watts, as determined by the sound level requirements and area to be covered by each unit. Each has a voice coil impedance of 8 ohms and is equipped with a line-to-voice coil transformer having an 8 ohm secondary and a primary with taps at 500, 1000, 1500 and 2000 ohms.

The total power required is $18 + 8 + 4 = 30$ watts, and a 35 watt amplifier is selected with an output transformer that has taps at 4, 8, 15, 30, 250 and 500 ohms. A chart like that of Figure 12 is ruled on a piece of paper, listing the amplifier output impedance along the top and the speaker line-to-voice coil transformer primary impedances along the left-hand side.

The speaker power values are then computed for each combination by means of the following formula:

$$W_s = \frac{Z_o W_A}{Z_s}$$

when:

W_s = Power in watts to be applied to speaker,
W_A = Rated amplifier power in watts,
Z_o = Impedance in ohms at output transformer tap,
Z_s = Impedance in ohms at speaker transformer tap,

For example, the first value:

$$W_s = \frac{4 \times 35}{500} = 0.28 \text{ watt,}$$

is entered in the 500 ohm line and 4 ohm column of the chart, as shown. Similarly, the second value in the 4 ohm vertical column is:

$$W_s = \frac{4 \times 35}{1000} = 0.14 \text{ watt,}$$

which is entered directly below the first value. These calculations are continued until all of the combinations of taps have been covered, and the chart is complete as shown. Note that these values apply to a 35 watt amplifier only, and if, for example, a 50 watt amplifier is to be used, the value of W_A becomes 50 instead of 35 in each of the computations.

Next, by inspection of the chart, select the amplifier output transformer tap which will provide as nearly as possible, the three values of power required by the respective speakers. In the chart of Figure 12, the 250 ohm amplifier output tap lists a value of 17.5 watts when the speaker transformer primary is 500 ohms; a value of 8.76 watts when the speaker transformer primary is 1000 ohms; and 4.38 watts when the primary is 2000 ohms. Thus, the transmission line may be connected from the 250 ohm output transformer tap; to the 500 ohm tap on the primary of the speaker transformer which is to receive 18 watts of audio power; to the 1000 ohm tap of the 8 watt speaker transformer; and to the 2000 ohm tap on the primary of the transformer connected to the 4 watt speaker.

The impedance match may now be checked by calculating the total impedance of the three speaker transformer primaries in parallel. The formula:

$$\frac{1}{Z_T} = \frac{1}{Z_1} + \frac{1}{Z_2} + \frac{1}{Z_3} \text{ etc.}$$

is used, and upon substituting the above values,

$$\frac{1}{Z_T} = \frac{1}{500} + \frac{1}{1000} + \frac{1}{2000}.$$

This step may be speeded by reference to the following table of reciprocals.

TABLE IV

Speaker Transformer Tap	Reciprocal	Speaker Transformer Tap	Reciprocal
375	.002666	3000	.000333
500	.002000	4000	.000250
750	.001333	5000	.000200
1000	.001000	6000	.000166
1500	.000666	7500	.000133
2000	.000500	8000	.000125
2500	.000400	10000	.000100

Adding the reciprocals of the above impedance values gives:

$$\frac{1}{Z_T} = .002 + .001 + .0005 = .0035$$

and the total load $Z_T = 286$ ohms, which, as mentioned, has been connected to the 250 ohm amplifier output tap, therefore the mismatch is only 12.6%.

Sometimes it is possible to achieve a somewhat closer match by using two of the intermediate taps on the amplifier output transformer. The impedance Z between any two of these taps can be found from:

$$Z = (\sqrt{Z_H} - \sqrt{Z_L})^2$$

when:

Z_H = Impedance of the higher tap

Z_L = Impedance of the lower tap

For example, if the output transformer of the above example had a 30 ohm tap, this together with the 500 ohm tap would provide an output impedance of:

$$Z = (\sqrt{500} - \sqrt{30})^2$$
$$Z = (22.36 - 5.48)^2$$
$$= 284.93 \text{ ohms}$$

which almost exactly matches the value of load impedance calculated above. By using these connections, the speaker circuit of the above example is as shown in Figure 13.

To eliminate the repetition of these calculations for every installation, the following Table V lists the values of output transformer impedance which can be obtained by connecting the speaker line to various combinations of the more common taps.

TABLE V

Z	Z_H	Z_L	Z	Z_H	Z_L
* 0.64	8	4	85.0	125	4
* 1.00	15	8	106.0	200	30
* 3.00	15	4	150.0	250	15
* 4.80	60	30	170.0	250	8
*12.10	125	60	190.0	250	4
*21.00	250	125	*215.0	500	60
24.00	60	8	285.0	500	30
32.50	60	4	350.0	500	15
*43.66	500	250	385.0	500	8
71.00	125	8	415.0	500	4

The combinations marked with an asterisk should be used only when the power involved is fairly low to avoid burning out the transformer winding. unless a check of the transformer specifications show it to be capable of carrying a higher current. This is necessary because many transformers are made with smaller wire between the high impedance taps than between the low impedance taps. Ordinarily, no trouble will be encountered if the lower of the two taps used has an impedance of not more than 30 ohms.

PHASING SPEAKERS

When more than one speaker is used in an installation, it is important to operate all the voice coils "in-phase". That is, all the diaphragms should move in the same direction at the same instant. If they are not in-phase, the total output will be materially reduced because the sound from one unit will tend to cancel that of the others. The simplest method of checking the phase of speakers is first to connect and excite the fields of all the speakers, not of the p-m type, and then temporarily short out any hum-bucking coils. Next take a dry cell (1½ volts) and touch its positive terminal to one voice coil lead and its negative terminal to the other voice coil lead. The speaker cone will jump either in or out at the instant the cell is connected. Reversing the cell connections will reverse the direction in which the cone jumps. Test all the speakers in the same manner, and on each mark the voice coil lead which, when connected to the positive terminal of the cell causes the cone to jump outward.

For parallel operation, connect all the marked leads to one side of the line and all the remaining leads to the other side of the line, and the speakers will be correctly phased. If series operation is necessary, connect the marked lead at one voice coil to the unmarked lead of the next one in the usual manner of connecting series units.

If it is desired to phase several speakers, each having its own transformer attached, the same procedure can be followed except that a battery of about 22½ volts is connected across the primary leads of the transformer. Voice coil leads must be attached to transformer secondary leads, and bucking coils shorted out temporarily. Any color coding, terminal identification, or wire position is then recorded so that the desired operating conditions can be identified.

IMPORTANT WORDS USED IN THIS LESSON

ACOUSTIC FEEDBACK—The action of certain sound waves returning from a loudspeaker to the microphone in such a way as to reinforce the input signal to a degree that causes a howl.

AMBIENT—Refers to the conditions surrounding a given point or region.

DECIBEL—A unit for measuring differences in sound intensity.

FIDELITY—As applied to sound systems, it refers to the degree of accuracy with which original sound signals are amplified and reproduced.

FREQUENCY DISCRIMINATION—The characteristics of a device or circuit of operating upon or influencing some frequencies more than others.

MATCHING TRANSFORMERS—An audio coupling transformer that also matches the impedance of the two associated circuits.

MECHANICAL DAMPING—The act of retarding the motion or vibration of an object by introducing some form of friction that dissipates part of the kinetic energy of the moving element.

NOISE LEVEL—The intensity of the audible noise at a given location. Usually expressed in decibels.

NONLINEAR—Not directly proportional, that is, the variation of one quantity with respect to another is not a straight line when a graphic illustration of the changes is made.

PHASING SPEAKERS—The process of connecting a group of speakers in a p-a system so that the diaphragms of all move in the same direction at the same instant.

PUBLIC ADDRESS SYSTEM—Equipment for reproducing speech and music with sufficient volume for a large public gathering. It consists of an input device (microphone or phono pickup), an audio amplifier, and a reproducer or loudspeaker system.

SOUND LEVEL METER—A portable device used to measure sound levels, consisting usually of a microphone, amplifier and output indicator calibrated in decibels.

STUDENT NOTES

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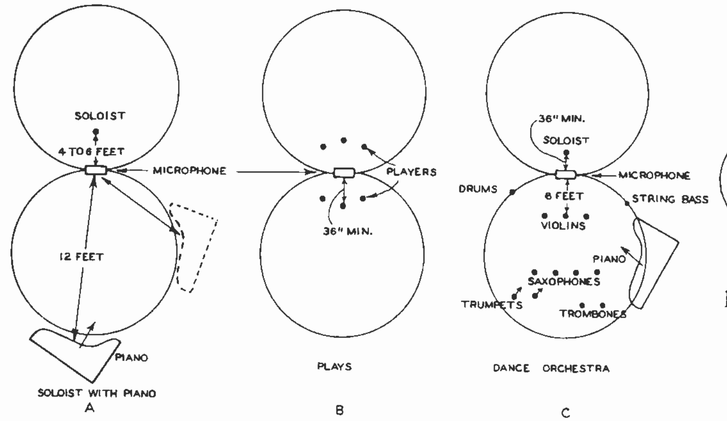

FIGURE 1

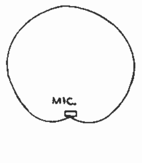

FIGURE 2

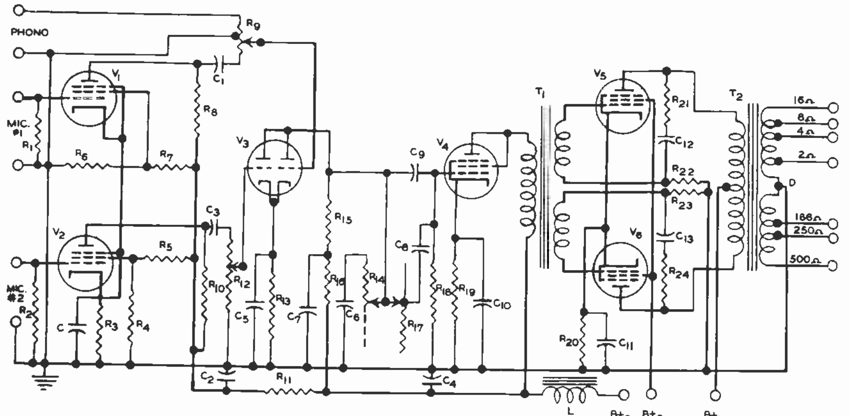

FIGURE 3

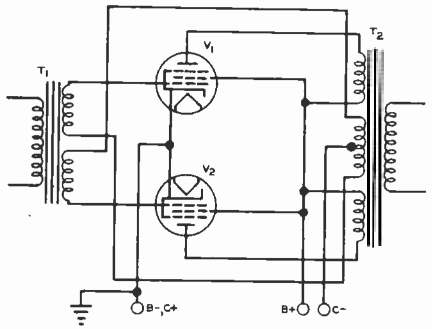

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FIGURE 4

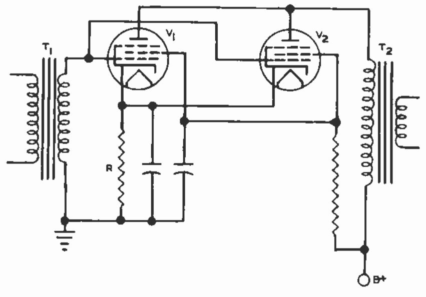

FIGURE 5

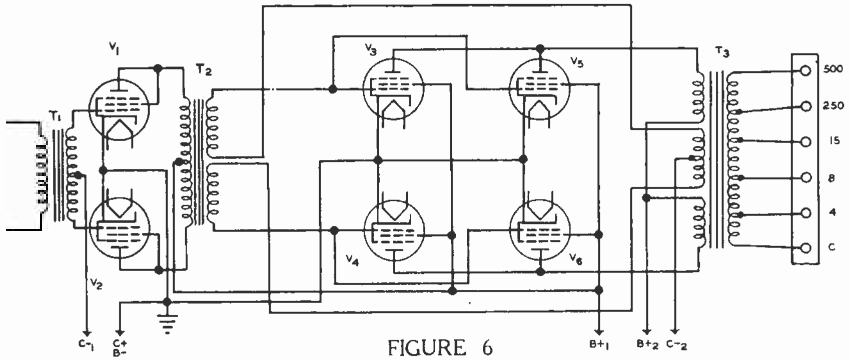

FIGURE 6

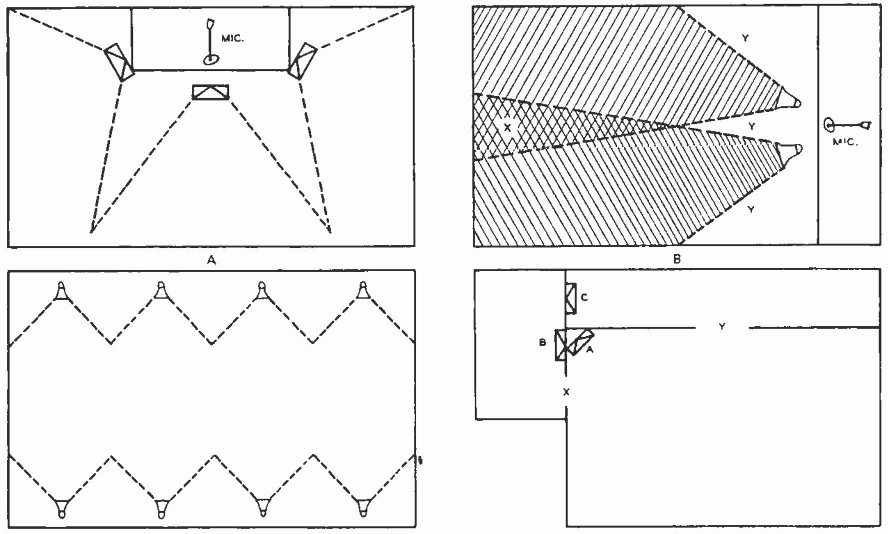

FIGURE 7

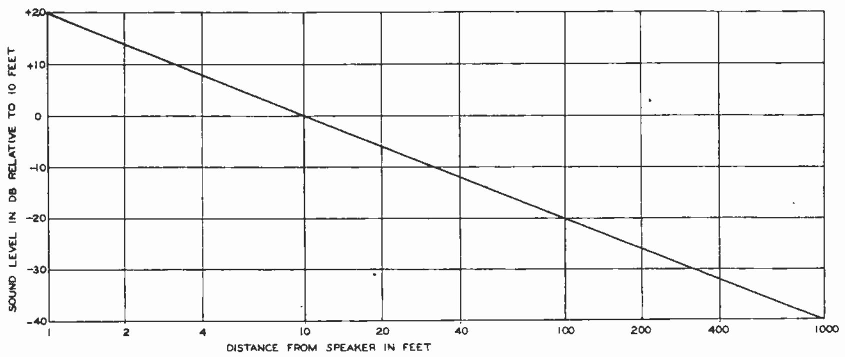

FIGURE 8

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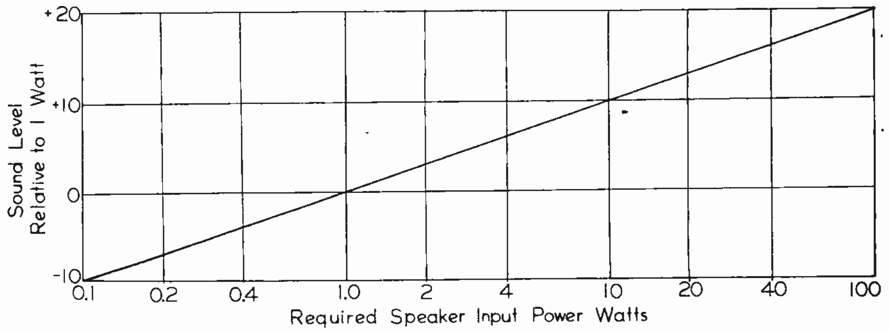

FIGURE 9

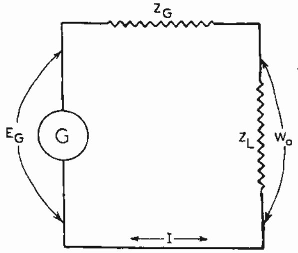

FIGURE 10

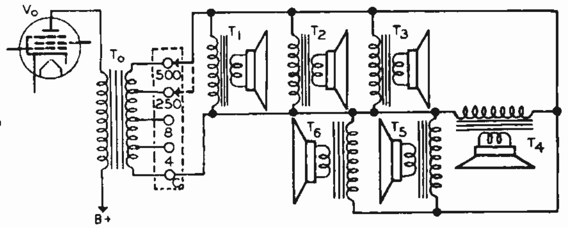

FIGURE 11

SPEAKER POWER IN WATTS WITH 35 WATT AMPLIFIER					
Speaker Transformer Primary Taps	Amplifier Output Transformer Taps				
	4	8	15	250	500
500	0.28	0.56	1.05	17.50	35.00
1000	0.14	0.28	0.53	8.76	17.50
1500	0.09	0.19	0.35	5.84	11.67
2000	0.07	0.14	0.27	4.38	8.75

FIGURE 12

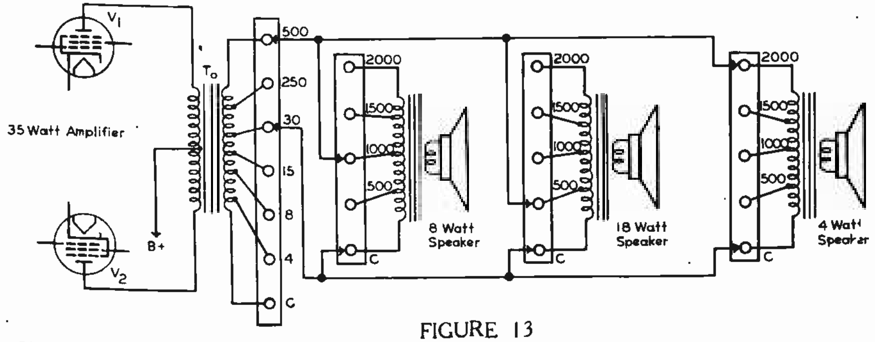

FIGURE 13

RRT-5

FROM OUR *President's* NOTEBOOK

SUCCESS COMMANDMENTS

1. **WORK HARD.** Hard work is the best investment a man can make.
2. **STUDY HARD.** Knowledge enables a man to work more intelligently and effectively.
3. **HAVE INITIATIVE.** Ruts often deepen into graves.
4. **LOVE YOUR WORK.** Then you will find pleasure in mastering it.
5. **BE EXACT.** Slipshod methods bring only slipshod results.
6. **HAVE THE AMERICAN SPIRIT OF CONQUEST.** Then you can successfully battle with and overcome difficulties.
7. **CULTIVATE PERSONALITY.** Personality is to a man what perfume is to a flower.
8. **HELP AND SHARE WITH OTHERS.** The real test of business greatness lies in giving opportunity to others.
9. **BE DEMOCRATIC.** Unless you FEEL right toward your fellow men, you can never be a successful leader of men.
10. **IN ALL THINGS DO YOUR BEST.** The man who has done his best has done everything. The man who has done less than his best has done nothing.

—Charles M. Schwab

Yours for success,

E.B. DeVry

PRESIDENT