SECTION 3

SPECIALIZED BROADCAST RADIO ENGINEERING

LOUDSPEAKERS

304 E

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LOUDSPEAKERS

SCOPE OF ASSIGNMENT

This assignment will deal with loudspeakers; how they work and how they are used. First the nature of sound will be investigated as a preamble to the study of the loudspeaker itself. Then the characteristics of the ear will be discussed so as to appreciate the loudspeaker requirements, and then the different types of units will be studied.

First will be taken up the balanced-armature type of motor unit, and then the more universally used dynamic loudspeaker type. The various components, such as the cone, voice coil, and field magnet, will be studied in detail, so as to obtain a better idea as to how satisfactory sound reproduction is obtained.

Then various kinds of speaker enclosures will be investigated, including the exponential horn type, and speaker systems including high- and low-frequency units will be studied so as to see what is involved in a high-fidelity system. The assignment will close with a discussion of stage speaker placement and multiple speaker installations.

PRELIMINARY CONSIDERATIONS

SOUND.—In order intelligently to study loudspeakers (more properly called "reproducers") it is necessary to know something of the nature of "Sound" and its effect on the human brain. At this point no attempt will be made to go into detailed study of sound and the human ear but a brief outline will facilitate the understanding of the problems involved in the design or selection of a reproducer.

Sound is the sensation produced in the brain by *sound waves*. A sound wave is a form of vibration set up in some elastic medium such as air, water, steel—in fact in practically any material substance. Sound waves consist of compressions and rarefactions set up in a fluid transmitting medium by a vibrating body and traveling outward in radiating concentric spheres. The frequency of vibration in the medium is the same as that of the vibrating body.

The *velocity* of the transmission of sound waves is a function of the density/elasticity ratio of the transmitting medium. Some velocities of sound through common mediums are:

Air =	1130 feet per second at 20°, increasing 2′ for each degree C.
Water =	4728 feet per second. (Usually taken as 4800 for normal temperatures).
Hard Wood =	10,900 feet per second.
Brick =	11,980 feet per second.
Steel =	16,360 feet per second.

It will be observed that the velocity of propagation of a sound wave through any medium is *much* less than that of the electromagnetic radio wave. Also, unlike radio, the velocity is not even approximately constant for different media and conditions.

The intensity of sound at a given distance from the vibrating body varies directly with the amplitude of vibration at the source and inversely as the square of the distance from the source. Thus for a given source the sound intensity I and I_1 at two points P and P_1 at different distances d_1 and d_2 from the source will be expressed by the ratio,

$$\frac{I}{I_1} = \frac{d_1^2}{d^2}$$

As the cone of a speaker vibrates, it moves first in one direction, then in the other. As the cone moves forward it compresses the air in front of it and rarefies the air immediately behind. As it moves back to normal the air molecules on both sides assume normal positions. As the cone then moves backward the air behind it is compressed and the air in front rarefied. When the cone returns to the normal position the air again assumes normal pressure on the two sides of the cone. Since air is an elastic medium these compressions and rarefactions travel outward from the cone at an approximate velocity of 1130 feet per second. Thus one complete sound wave consists of,

1. Area of compression.
2. Area of normal pressure.
3. Area of rarefaction.
4. Area of normal pressure.

The air molecules move only very short distances, each transferring its motion to the next adjacent molecules. The number of complete waves per second equals the pitch or frequency, and is usually designated as the number of waves or cycles per second. Since sound waves affect the auditory sense, the frequency of such waves is spoken of as an "audio frequency."

As with all wave motion, the sound wave has a "wavelength" (λ). λ = V/F where V = velocity of propagation per second (1130 feet per second in air), and F = frequency in cycles per second. Thus a sound frequency of 500 cycles per second will have wavelength of,

In Air, $\lambda = \frac{V}{F} = \frac{1130}{500} = 2.26$ feet.

In Water, $\lambda = \frac{V}{F} = \frac{4800}{500} = 9.6$ feet.

The sound wave may be illustrated as in Fig. 1 in which the pressures are plotted against distance. (+) pressure represents a compression, (-) pressure represents a rarefaction, and (0) is normal pressure.

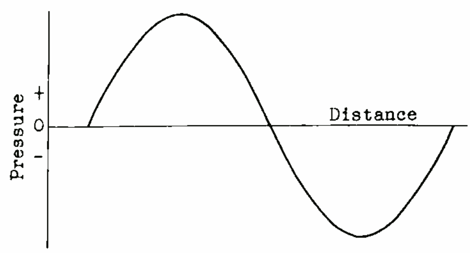
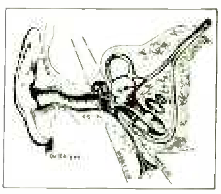


Fig. 1.—Sound or acoustic pressure plotted against distance.

The vibrations of a body at audio frequencies are too rapid to be directly visible, but they may be made visible and the exact motion studied by means of a stroboscope.

THE EAR.—The human ear is a highly sensitive and delicate mechanism which responds to extremely minute variations in air pressure. Although very sensitive the ear has certain peculiarities of response that must be understood if the engineer is intelligently to grasp the problems of reproducer design.

The ear as shown in Fig. 2 may be divided into three major sections, the outer, the middle and the inner ears. The outer ear consists of the auricle and external auditory canal. The auricle is a flap of cartilage which serves the purpose of collecting sound waves over a large projected area and to direct the waves into the auditory canal. The canal while offering free passage to the sound waves offers a certain amount of mechanical and thermal protection to the more delicate structures of the middle and inner ears. The inner end of the auditory canal is closed by an oval shaped membrane or diaphragm known as the tympanic membrane or, more commonly, as the ear drum. The ear drum vibrates in accordance with the sound waves striking it.



Outer Ear P - Auricle
C - Auditory canal
D - Ear drum

Middle Ear H - Auditory ossicles
E - Eustachian tube

Inner Ear S - Basilar membrane
N - Auditory nerve trunk

Fig. 2.—The anatomy of the ear.

The middle ear is enclosed in the tympanic cavity which is located in the temporal base of the skull. In this cavity are three small bones, the malleus, the incus, and the stapes, known as the auditory ossicles. The bones are articulated or jointed and form a mechanical impedance network between the ear drum and the oval window membrane of the inner ear. The motion imparted to the inner ear membrane by the ossicles is about 1/60 of the motion of the ear drum. If the correct impedance match is to be maintained between the inner and outer ears the middle ear must be kept at atmospheric pressure. Equalization of pressure is maintained by a slow exchange of air through the Eustachian Tube leading from the inner ear cavity to the pharynx in the throat. Any stoppage in this tube, such as might be caused by a head cold, may seriously interfere with normal hearing.

The inner ear is a snail shaped cavity in the skull, filled with liquid and divided into two sections by a flexible membrane with a bypass for allowing some interflow. Sound waves are set up in the liquid of the inner ear by vibrations of the oval window membrane which is mechanically driven by the auditory ossicles in the middle ear. Into the liquid project about 24,000 fibers carrying the nerve terminals (3500 nerves per octave). Each has a different length, thickness and mass, thus each responds to a different frequency. These nerves carry the impulses to the brain where they are interpreted as sound. The high frequencies affect the nerves at the entrance end; the low frequencies affect the nerves at the inner end.

The ear is delicate and differs in persons owing to heredity, disease, training, etc. Thus a sound level that is satisfactory to one person may be unsatisfactory to another. The maximum high frequency response of the ear is usually between 15,000 and 20,000 cycles but in some persons the response is limited to frequencies below 10,000 cycles. The response of the ear to different sound levels is approximately logarithmic, therefore to appear twice as loud, a sound level must be increased by ten times.

In the reproduction of sound and in the design of the apparatus—amplifiers, speakers, etc.—the characteristics of the ear must be taken into consideration. In the determination of the frequency range it is necessary to cover, the characteristics of the ear and the nature of sound waves become of major importance.

In exhaustive tests it has been found that a very great improvement is observed by the average person when the band of frequencies reproduced for speech and music is increased at the high frequency end from 5000 to 8000 cycles. To the average person for most sounds, little difference is observed when the range is increased from 8000 to 15,000 cycles. But for some sounds, such as the jingling of keys and the rustle of paper, the reproduced sound is much more "natural" with a sound system capable of reproducing frequencies up to 15,000 cycles. Most speech and music does not absolutely require frequency response below 100 cycles, but for certain instruments flat response to 30 cycles is necessary for *true* reproduction, and the music lacks "body" if the low frequencies are attenuated unduly.

HEARING RESPONSE.—Fig. 3 shows a typical hearing response curve in which sound pressure variation (or level in db) is plotted as a function of frequency. Sound wave pressure variation (or level in db) is plotted as a function of frequency. Sound wave pressures for any value of frequency which lie below the "Threshold of audibility" curve do not produce the sensation of "Sound" in the brain. Sound wave pressures which lie above the "Threshold of feeling" curve will produce a definite sensation of feeling or pain, in addition to the sensation of hearing and become very uncomfortable to the listener. The ordinate between the two curves shows the range of sound pressures over which the ear gives a usable range of hearing. It will be noted that the ear has its maximum latitude of response in the range of from 1000 to 2000 cycles. This latitude of hearing response becomes much less for the low and high frequencies. The broken portions of the curves indicate the range over which it is difficult to accurately determine the distinction between the sensations of hearing and those of feeling.

The "Threshold of audibility" curve may also be interpreted as indicating the sensitivity of the normal ear to sound waves over the audio spectrum. It will be noted that the region of maximum sensitivity is around 3000 cycles. For frequencies below 2000 cycles the sensitivity falls off in a fairly uniform manner down to the lowest frequency to which the ear will respond. For frequencies above 4000 cycles the sensitivity falls off rapidly. For example, the sound pressure of a 16,000 cycle note must be approximately 250 times that of a 2000 cycle note to produce a barely audible sound. Threshold of audibility curves may be drawn for any individual person from data obtained

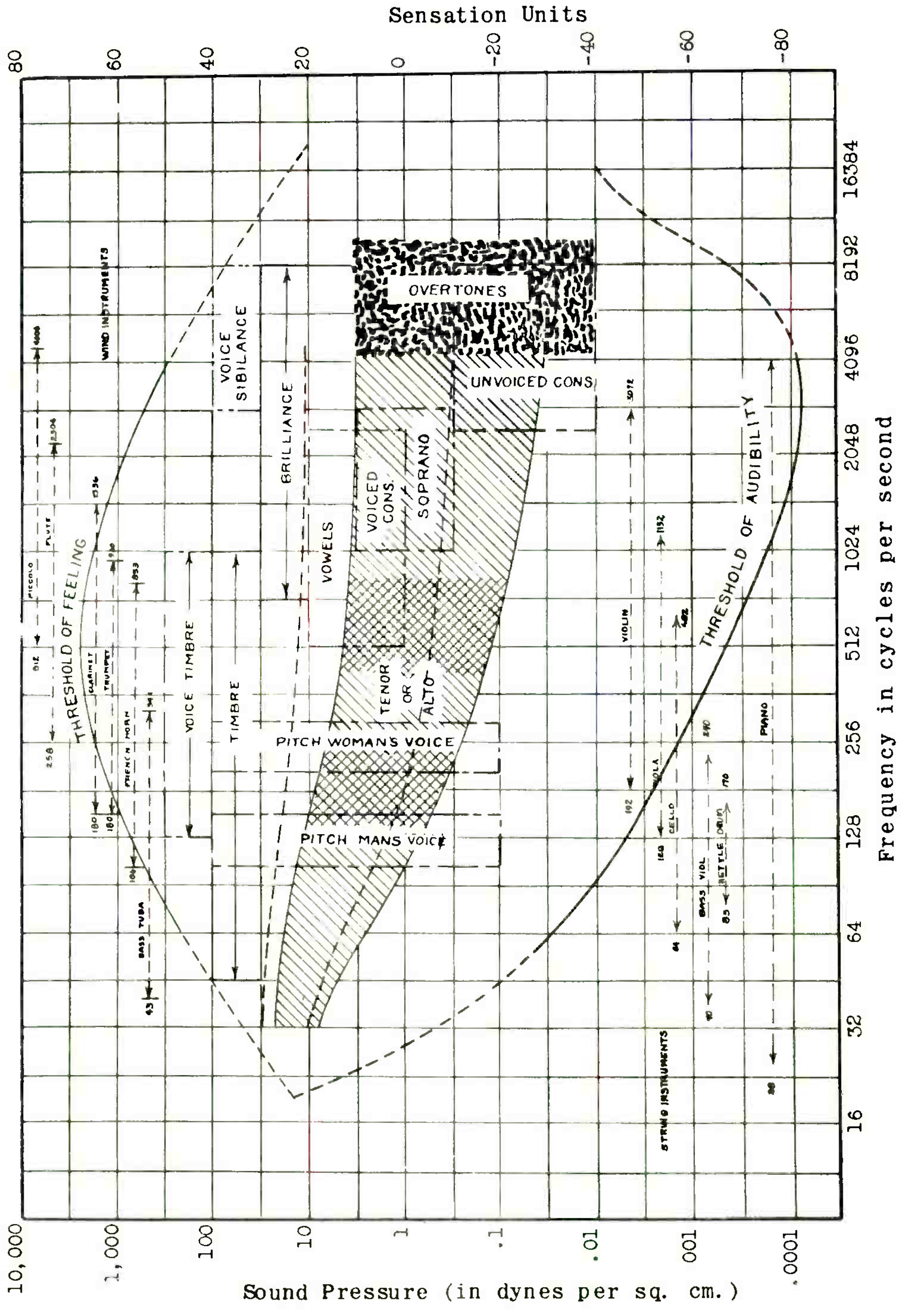


Fig. 3. — Characteristics of the ear.

by a calibrated audiometer and the results compared to the normal, or average curve, to determine the hearing loss for the individual.

Superimposed on the audibility curves of Fig. 3 are additional curves indicating the relative frequency and amplitude range of typical musical instruments and for the human singing voice. It will be noted that the normal ear has a frequency and amplitude range considerably in excess of average requirements. Note that the average ear will respond to a frequency more than 3 times greater than the highest piano frequency hence the ear is capable of hearing the third harmonic of the highest fundamental frequency of the piano.

The audibility curves of Fig. 3 show that the normal ear is enormously more sensitive to frequencies around 3000 cycles than for frequencies in the lower or higher register. This characteristic of the ear should be well understood as it has an important bearing on the design and operating characteristics of audio equipment.

Fig. 4 shows a family of curves known as equal loudness contours and indicates the sound intensity level required at different frequencies to produce the same loudness response. (These curves have been presented in a previous assignment on audio amplifiers). It will be observed that the relative ratio of ear sensitivity to low frequencies compared to the sensitivity at 1000 cycles depends largely on the existing average sound intensity at 1000 cycles. For example at a level of 0 db the ear sensitivity is *down* more than 50 db at 50 cycles from what it is at 1000 cycles. At a level of 50 db the 50-cycle response is down approximately 25 db from the 1000-cycle response and at a level of 100 db the sensitivity at 50 cycles is essentially the same as for 1000 cycles. These curves clearly indicate the reason why the low-frequency response of an amplifier or receiver is poor at low operating levels although the overall response may be in correct balance for somewhat higher operating levels. To compensate for the poor low-frequency response at low volume levels a bass compensated volume control may be used. As mentioned in a previous assignment, this volume control automatically increases the relative ratio of low-frequency to intermediate-frequency output as the control is adjusted to reduce the output level.

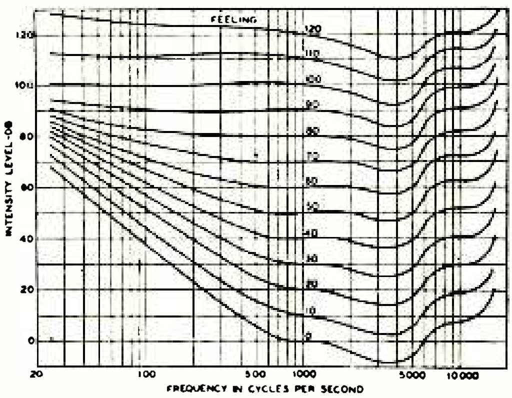


Fig. 4.—Equal loudness contours.

LOUDSPEAKER REQUIREMENTS.—The requirements for reproducers for various purposes differ widely with the uses to which the reproducers are to be put. For a broadcast receiver it is only necessary that the sound output be adequate to fill the average room to a comfortable level. For public address work the reproducer may have to deliver sufficient sound for a large audience and/or to fill a large auditorium. For this purpose it usually is desirable that the reproducer be directional; for the broadcast receiver the reproducer should be nondirectional, radiating most of its sound in the form of a semicircle.

The low-frequency response of a speaker is determined largely by its size and the area of the baffle used with it, the larger the area the lower the frequencies the speaker can reproduce. If the speaker is to be used with an amplifier having a sharp cut-off below 200 cycles and above 3000 cycles as in some inexpensive broadcast receivers, it is obviously a waste of money to install a speaker capable of reproducing from 30 to 10,000 cycles. On the other hand, for auditorium use where the particular amplifying equipment is capable of handling frequencies from 30 to 12,000 cycles, a reproducer capable of handling frequencies between 100 and 6000 cycles is entirely inadequate and will nullify the expensive efforts taken toward high fidelity in the rest of the system.

For high fidelity reproduction the speaker must be free from wave distortion; that is, the sound power produced must correspond to the electrical power. It must also respond uniformly to all audio frequencies and should not introduce frequency components not present in the amplifier output.

The efficiency of loudspeakers is, as a rule, very low. (By efficiency is meant the factor by which the electrical energy driving the speaker is converted into sound energy.) For radio speakers, the average efficiency is usually between 1 and 5 per cent. Speakers well designed for public address work have efficiencies between 25 and 50 per cent. Only the very best speakers will approach 50 per cent efficiency.

The efficiency of a typical 10" speaker of the type commonly employed in modern broadcast receivers is shown in Table 1. The maximum efficiency occurs at the frequency where the moving system of the speaker is mechanically resonant, about 100 cps in this case. In the design of a speaker to be used with a receiver operating from a 60-cycle power line particular care must be exercised to avoid mechanical resonance near 120 cps since that is the predominant hum frequency at the rectifier output.

TABLE I

Frequency Cps	Efficiency per cent
50	.5
75	4.5
100	12.
200	10.
500	5.3
1000	2.3
1500	.95
2000	.7

The acoustical reproducer consists of two essential units: the driving unit or motor and the loading unit, usually a horn or cone. In the driving unit it is desired to obtain the maximum amplitude of motion for a minimum current variation in the winding. There are several different types of driving units most of which will fall under one of the general classifications of moving-iron or moving-coil types.

MOVING-IRON DRIVING UNITS

IRON-DIAPHRAGM TYPE.—There are two general types of moving-iron units, the iron diaphragm and the balanced armature. The iron diaphragm type is the one so commonly used in the ordinary telephone receiver and radio headset. It consists of a stiff iron diaphragm rigidly mounted above an electromagnet, the windings of which carry the signal current. For high sensitivity the diaphragm must be mounted quite close to the poles of the magnet. This limits the distance through which the diaphragm may move and hence the amount of power the unit can handle. Because the diaphragm is stiff it has bad resonant peaks, usually around 1000 cycles, which makes this type of unit unsatisfactory for high quality reproduction.

BALANCED-ARMATURE TYPE.—The balanced-armature type of driving unit is shown in Fig.5. It consists of a soft iron armature balanced between the poles of an electromagnet. The signal current is used to energize the magnet. To one end of the armature is connected a driving rod which in turn drives a diaphragm which may be either metal or paper composition. This type, if made sensitive by reducing the air gap, has a tendency to rattle against the poles of the magnet on strong signals. If the length of the air gap is increased the unit will handle more power but the sensitivity is decreased. Placing more turns on the magnet coil increases the sensitivity but reduces the high frequency response because of increased inductance.

The balanced armature unit represents a mechanical pushpull system which reduces the tendency toward production of second-harmonic distortion, a fault that is particularly bad in the iron-diaphragm unit. However, the mechanical stiffness of the moving system must be kept high in order to provide the necessary restoring force to keep the armature magnetically centered. Since stiffness is a factor which limits the motion and output at the lower frequencies the low frequency output of the balanced armature unit is ordinarily quite poor.

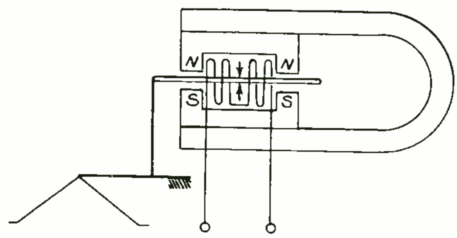


Fig.5.—Balanced-armature type of loudspeaker motor.

DYNAMIC LOUDSPEAKERS

MOVING-COIL UNIT.—This driving unit is built in two types, the "electrodynamic" in which a strong fixed magnetic field is produced by an electromagnet, and the "dynamic" or PM type which utilizes a permanent magnet to produce the fixed field. The electrodynamic type has been very popular for several years. The field for the electromagnet is obtained either from a battery, a rectifier of the tube or copper oxide type or by using the plate current of the radio receiver or amplifier. In the last mentioned case the field winding is usually a part of the receiver rectifier filter circuit. In recent years the development of high quality permanent magnets has resulted in considerable increase in the use of PM type reproducers.

ELECTRODYNAMIC SPEAKER.—In this type of reproducer a small coil, called the voice coil, carries the signal current which causes the voice coil to move back and forth within a strong fixed magnetic field set up by the field current flowing in the field winding. The voice coil is mounted on a very light form which is attached to a corrugated paper cone.

As shown in Fig. 6, the cone is so mounted that the voice coil is centered in the air gap formed by the construction of the iron core, and is free to move back and forth in this gap, the extent of the movement being determined, for a given magnetic field density and for other constant factors, by the amplitude of the signal current in the voice coil. Since the voice coil is rigidly attached to the cone, the cone is moved back and forth by the movement of the voice coil.

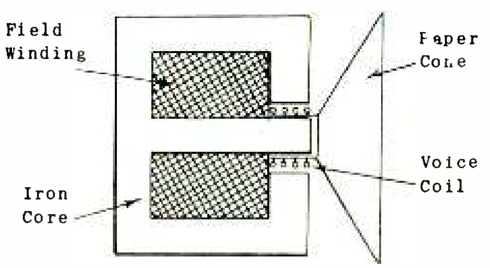


Fig. 6.—Electrodynamic cone type of loudspeaker.

Now refer to Fig. 7. Assume the direction of current flow through the field winding on the central field pole as shown, which will produce a magnetic flux in the annular gap *entering* the central pole. If the instantaneous direction of current flow through the voice coil is into the paper in the top conductors and out of the paper in the bottom conductors, the voice coil and the attached cone will be driven to the left in accordance with the *left hand* rule for motor action. As the current flow through the voice coil reverses during the next half-cycle the direction of mechanical force is reversed and the cone moves to the right by an amount proportional to the amplitude of the current delivered to the voice coil by the amplifier.

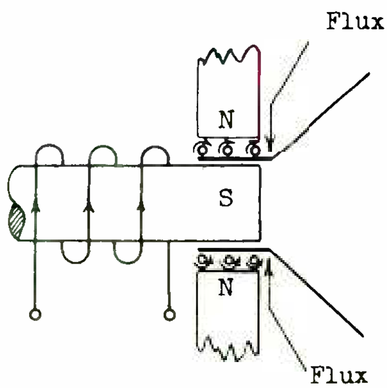


Fig. 7.—Current and motion of a cone diaphragm.

Since the required magnetizing force, for a given volume and permeability of iron in the core, and for a given density of magnetic field, is determined largely by the length of the air gap in which the voice coil is mounted, this gap should be made as small as practical. However the voice coil must be able to move freely within the gap and manufacturing tolerance must not be too small, so the gap must be great enough to allow easy centering of the voice coil and free movement at all signal amplitudes. The back and forth movement of the cone with a strong signal may be as much as 1/8" or more at the low frequencies of speech or music.

FIELD CONSTRUCTION.—The field core may be either cast or stamped. When cast iron or cast steel is used the core and shell are considerably heavier due to the lower permeability of this material. Stamped cores usually are made of ingot iron or Swedish steel and, due to the higher permeability of the material, such cores may be made considerably smaller, lighter and at a lower cost.

The efficiency of the speaker depends on the density of the field flux produced in the air gap and the number of ampere turns of the voice coil relative to its resistance. The density of the magnetic field, for a given length of air gap, is limited by the cost of construction and the current required to establish the magnetizing force. The magnetizing force may be anywhere from 500 to 2000 ampere turns, but a magnetizing force of from 1000 to 2000 ampere turns is most common. Three examples of commercial speaker field windings are given.

While all three fields require very nearly the same magnetizing force, there is considerable difference in the types of winding and in the current and voltage requirements. No. 1 is designed to operate from d-c power supply of 110 volts or a rectified a-c power supply of the same voltage. The resistance is moderately high in order to keep the current requirements within reasonable limits. No. 2 is designed to operate from a tube rectifier and, to keep the current as low as practical, employs a high resistance winding.

No. 3 is designed to operate from a 6-volt storage battery with a moderate current which, however, is much higher than the magnetizing currents of the other speaker fields due to the smaller number of turns of larger wire. With the same kind and amount of iron in the core, the same air gap, and the same voice coil, No. 1 and No. 2 will have about equal efficiency while No. 3 will operate at somewhat higher efficiency due to the greater number of ampere turns in the field winding and consequent higher flux density. The flux density in a good loudspeaker is in the neighborhood of 12000 gauss (lines per sq. cm.); this value can also be obtained with a permanent magnet field.

Fig. 8 shows several methods of energizing the speaker field from the power supply of a radio receiver.

	Turns	Wire Size B and S	Resistance	Voltage	Current	Ampere Turns
1.	22,000	34	2500 ohms	110	.044 Amp.	968
2.	40,000	36	75000 ohms	180	.024 Amp.	960
3.	1,600	20	8.5 ohms	6	.7 Amp.	1120

(A) and (B) are for fields of more than 6500 ohms resistance. Both place an additional load on the rectifier. (b) is less desirable as it increases the current taken through the entire filter with a corresponding increase in the drop through the chokes, nearer approach to core saturation, and less complete filtering with given capacitors. However if adequate filter capacity is provided the A.C. component of speaker field flux should be reduced to a minimum with a corresponding low hum level.

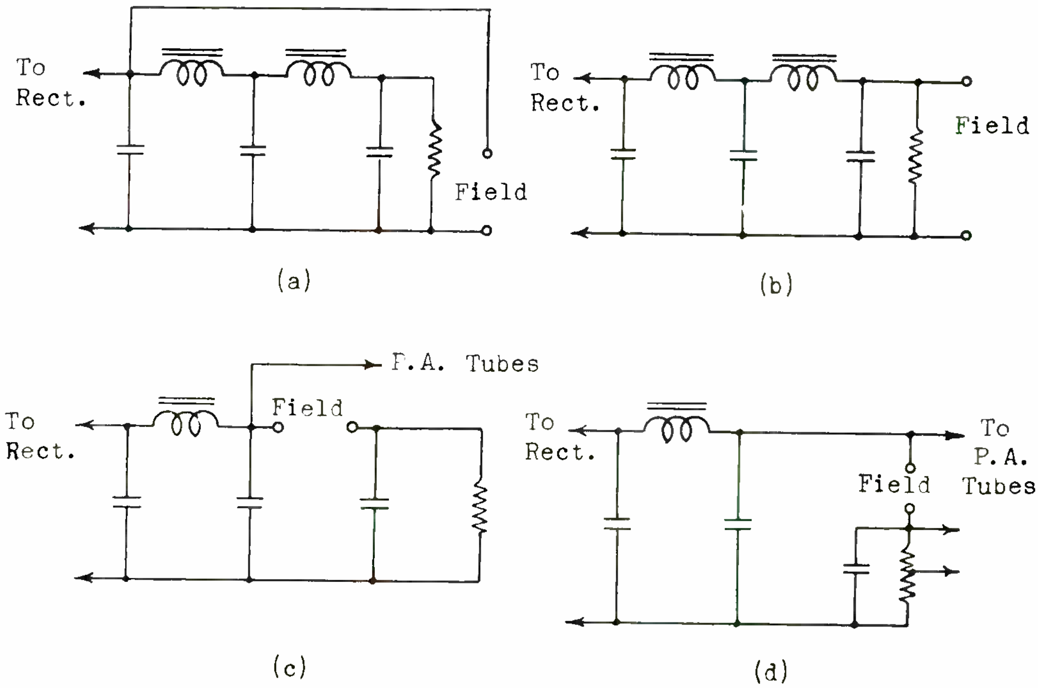


Fig. 8.—Various methods of energizing the field coil of an electrodynamic loudspeaker.

The circuit shown in Fig. 8(c) is used with field winding of 500 to 1000 ohms. Here the speaker field acts as the second filter choke of the power supply and thus is an aid toward economical receiver construction. All the plate current of the receiver except for the power amplifier stage is used to energize the field winding. This method is most universally employed in commercial receivers.

The circuit of Fig. 8(d) is used with field windings of from 2000 to 4000 ohms. Here the field winding serves both as a filter choke and as a voltage dropping resistor. All of the receiver plate current except that of the power amplifier tubes flows through the speaker field, the voltage for the power amplifier being taken off above the field winding.

HUM BUCKING COIL.—When the field coil current is supplied from the output of a full wave 60 cycle rectifier (tube or copper oxide type) an appreciable 120 cycle component of field flux is usually present which may result in objectionable hum from the speaker.

This condition may be remedied by the use of a hum bucking coil in series with the voice coil as shown in Fig. 9. as a very low resistance short-circuited secondary turn and shield, and thus effectively prevents any hum from being induced in the voice coil.

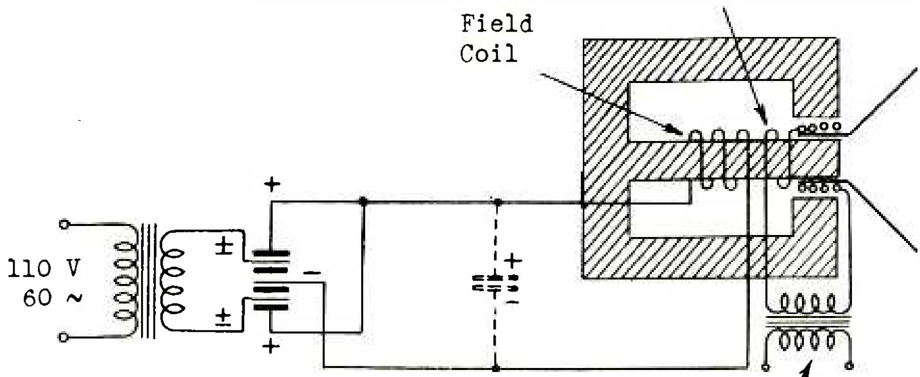


Fig. 9.—Dry-disc rectifier field supply.

The hum bucking coil is wound on the field core, usually directly behind the voice coil. The location and number of turns used should be such that voltages induced by any component of A.C. field flux will be equal and opposite in the H-B and voice coils. Complete cancellation is difficult to obtain due to the fact that the hum voltages induced in the two coils are not exactly 180° out of phase.

However, with the H-B coil wound with the proper number of turns and properly located with respect to the voice coil the hum can usually be reduced to a satisfactory level. If the H-B coil is not properly connected the hum will be increased, a condition that can be remedied by reversing the connections of the coil.

Some speakers use a form of shading ring to reduce hum. This consists of a heavy copper disc secured to the field core so that it encircles the center core. This acts

Fig. 9 also illustrates the manner in which the field may be supplied from a dry-disc rectifier. Note polarity of connections as used for standard assemblies. A power transformer reduces the 100 volts from the source to about 7.5 volts to supply the rectifier unit. Satisfactory filtering is obtained by the use of a high capacity low voltage electrolytic condenser. Usual capacity values range from 1500 to 2000 microfarads.

VOICE COIL.—The voice coil may consist of anywhere from a few to 100 turns of from No. 12 to No. 32 wire wound on a very accurately fitted form attached to the cone. The voice coil is "floated" in the gap by means of a "spider" which should be rigid enough to maintain the same clearance between coil and magnetic pole and yet sufficiently flexible not to interfere seriously with the motion of the coil. It is essential that the voice coil be properly centered in the gap, otherwise the coil may rub against the field pole producing a rasping sound and eventually destroying the insulation between turns. The length of the air gap is made as small as practical because a few one-thousandths of an inch in gap length make a considerable difference in the magnetizing force required to maintain a given strength of field. The length of gap required, allowing adequate clearance, is determined largely by the amount of wire on the voice coil.

It is important, from the viewpoint of electrical efficiency, that as much of the magnetic gap as possible be filled with wire. Four layers of No. 36 wire will occupy practically the same space as two layers of No. 30. The former will have four times as many turns but, since the resistance of No. 36 is 4 times as great as that of No. 30, the former coil will have 16 times as much resistance which would result, for the same voltage, in a greatly decreased voice coil current and actually fewer ampere turns. However since the voice coil is fed from the power amplifier tubes through a step-down impedance matching transformer, by proper design of the transformer either winding can be made to give essentially similar results. Single layer windings consisting of copper tape wound edgewise is preferred, because although more expensive, it permits the coil to be self-supporting and minimizes the amount of air gap space occupied by insulation.

In present day dynamic speakers, the voice coil impedances range from about 4 ohms or less to about 30 ohms. Commonly encountered impedances are 4 ohms, 8 ohms and 16 ohms. It must be remembered that the impedance of a voice coil is more than its d-c resistance, the inductive component of the winding usually having an appreciable effect, particularly at the higher frequencies. The inductive effect is about the same relative to the resistance of the coil whether it consists of few or many turns.

The manner in which the voice coil impedance may be expected to vary with frequency is shown by the following values obtained from an 8-inch speaker in a standard broadcast receiver. The voice coil was 1 inch in diameter, consisting of 105 turns of No. 32 wire having a D.C. resistance of 4.41 ohms.

Frequency	Impedance
100	4.6 ohms
200	4.7 "
500	4.9 "
1000	5.3 "
2000	6.6 "
3000	7.5 "
4000	8.4 "
5000	9.5 "
6000	10.4 "
7000	11.3 "

In practice one must elect to match the impedance of the speaker at some particular value of frequency (usually 400 cycles) at which frequency will be obtained the most efficient power transfer. For frequencies above or below that value the impedance match can be only approximate and maximum power transfer is not obtained over the entire audio range. The output transformer which raises the speaker voice coil impedance to the value required to match the power amplifier tubes is frequently mounted directly on the speaker chassis as this facilitates mounting and wiring in most cases.

THE CONE.—The cone is made of special type of heavy paper or fiber and is sometimes corrugated. The edge of the cone must be free so that the entire cone can be driven by the voice coil; at the same time it must be held in place with the voice coil accurately centered in the air gap around the field pole and the only place from which to suspend it is the outer edge. An arrangement such as that shown in Fig. 10 is used. Around the outer edge of the cone is glued a ring of *very flexible* membrane; such as leather. The outer edge of this flexible membrane is clamped to a steel ring and the steel ring is supported by steel arms from the outer case of the field winding, the arms and steel ring making up a frame which hold the cone in position accurately. Thus while suspended from the outer edge of the cone, the cone edge is essentially free and can move under the influence of the voice coil along the axis of the central pole.

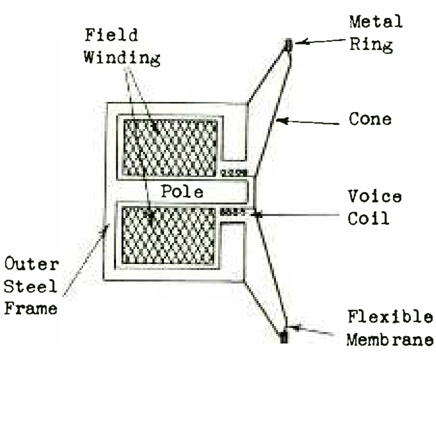


Fig. 10.—Cross section of an electrodynamic loudspeaker unit.

In recent years the trend has been to form or mould the cone into its finished shape as a one piece seamless unit as shown in Fig. 11. Corrugations pressed into the outer edge provide the necessary flexibility for the to-and-fro motion.

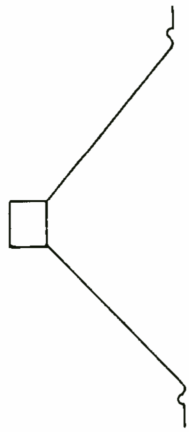


Fig. 11.—One piece type of cone.

The spider which maintains the proper alignment of the voice coil in the air gap may be of the internal or external type as shown in Fig. 12(a) and (b). The external type, shown in (b), due to the longer arms, provides a greater range of motion and most usually will be employed in the high capacity cone speakers. The internal spider is less expensive to build and provides a satisfactory range of motion for small speakers. The spider arms may be made of metal or fiber. Excessive overloading, particularly around the point of mechanical resonance, is a common cause of spider failure.

The actual vibrations of a cone are *very* complex. If the entire cone moves as a piston proportional to the signal current at the high frequencies, the high frequencies will be accentuated. If the cone flexes at the high frequencies, the high-frequency response is reduced but additional frequencies may be introduced, particularly second harmonic distortion.

of the moving system and the mass reactance of the air load.

The compliance represents the reciprocal of the stiffness introduced by the spider at the center of the cone plus the stiffness of the support at the free edge of the cone.

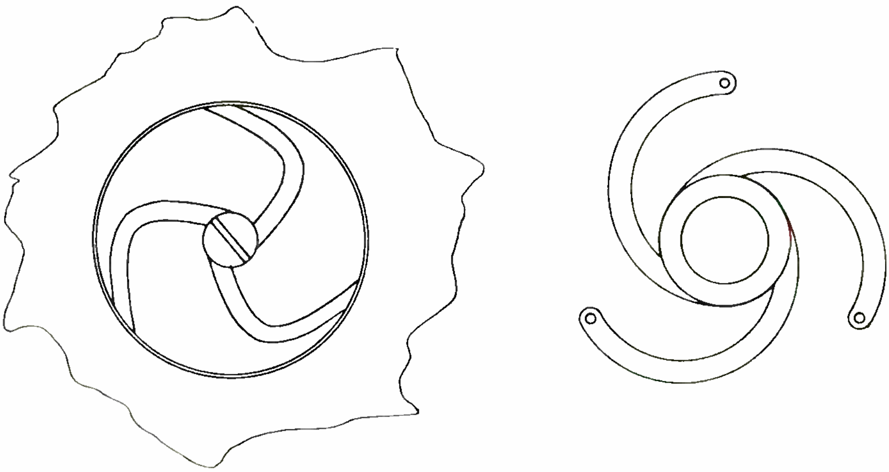


Fig. 12.—Internal and external spiders used to center the cone.

The free-edge cone need not be large to reproduce the low frequencies *if the proper baffle is used.* A diameter of from 6" to 12" is sufficient depending on the amount of power to be radiated, particularly if a horn is employed.

WIDE-RANGE SPEAKERS

BASIC DESIGN CONSIDERATIONS.—The vibrating system of a speaker may be considered as a mechanical network consisting of mass reactance, stiffness (compliance) and mechanical resistance which are equivalent to the circuit constants of X_L, X_c and R in an electrical network.

The mass reactance represents the sum of the mechanical reactance of the moving system and the mass reactance of the air load.

The mechanical resistance R represents the sum of the friction losses in the cone material and the spider and outer support, plus the mechanical resistance due to the air load. The latter value, call it R_r, represents the radiation resistance and is the useful load to which energy is delivered.

The velocity of the moving cone is proportional to the force developed by the driving motor and inversely proportional to the opposition or impedance of the moving system.

$$\text{Velocity} = \frac{BIL}{\sqrt{R^2 + (X_m - X_{cm})^2}}$$

B = Flux density in gap in Gauss.
I = Current in voice coil in amperes.
L = Length of voice coil winding in cm.
R = Mechanical resistance component of the mechanical impedance.
X_m = Mass reactance component.
X_{cm} = Compliance component.

At the frequency at which $X_m = X_{cm}$ the total mechanical impedance is a minimum and is represented by R, and the moving system becomes mechanically resonant. For a given driving force the velocity therefore becomes a maximum at the frequency which produces mechanical resonance. At frequencies below resonance X_{cm} (stiffness) becomes the controlling factor while at frequencies above resonance X_m becomes the controlling factor.

Consider frequencies above this resonant point, say up to 1000 c.p.s. or so. In this range, the cone moves as a piston, and its inertia may be considered as controlling its motion. This means that its acceleration is proportional to the applied force. Where the latter is sinusoidal because it is produced by a sinusoidal current in the voice coil, it turns out that the velocity V of the cone varies *inversely* as the frequency of the voice coil current.

Now as to the acoustic properties of the cone. At the lower frequencies the diameter of the cone is small compared to the wavelength, and the cone may be regarded as a point source of sound. Such a source has the characteristic that its radiation resistance R_r increases as the *square* of the frequency.

The acoustic or sound output of a speaker can be expressed as V^2R_r, which is similar to the expression for electrical power $P = I^2R$. Hence, if we double the frequency, R_r becomes $2^2 = 4$ times as great, but V becomes half as great, and V^2 becomes ONE-QUARTER as great. As a result, V^2 decreases as rapidly as R_r increases, and the product V^2R_r remains therefore constant.

Hence we can summarize the behavior of the loudspeaker by saying that in the lower end of the audio spectrum, where the cone's motion is controlled by its inertia or mass, and where furthermore it is essentially a point source of sound, its output is essentially independent of frequency; i.e., it has substantially a flat frequency response.

At the higher frequencies it is practically impossible to make the entire cone move as a rigid unit or piston when it is driven by a voice coil located essentially at its center. Moreover, it would cease acting as a point source and confine its acoustic radiation in the form of a narrow beam.

A large cone moving as a piston therefore becomes highly directional in its high-frequency output and much less directional in its low-frequency output. A large cone is necessary to handle a large low-frequency output but due to its large mechanical mass it will be inefficient for reproducing high frequencies if the cone is rigid and moves as a piston.

It may therefore seem desirable to have a number of cones of graduated sizes for most effectively reproducing the entire audio band: i.e., a small cone of low X_m for the high frequencies, a larger cone for the intermediate frequencies and a still larger one for the lowest frequencies. Such arrangements of graduated speakers have frequently been employed as will be shown later in this assignment.

For economic reasons it is desirable to combine the basic operating advantages of the graduated multiple speaker combination into a single cone structure which is less expensive to build and requires less space for mounting. The curvilinear cone as shown in Fig. 13 has been developed and introduced on the market in an attempt to accomplish this result.

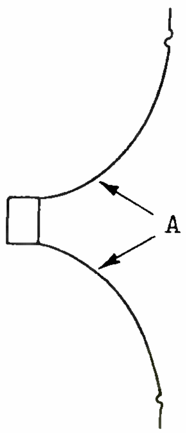


Fig. 13.—Curvilinear cone designed to vary amount of material vibrating with frequency.

The graduated curvature of the moulded cone structure results in sectional vibration. At low frequencies the entire cone moves as a unit with piston action and provides maximum radiation surface. At some higher frequency, for example 600 cycles, only the small portion has motion imparted to it by the voice coil. To explain the action one may assume the cone flexes at point A, allowing the central portion to vibrate through the small range necessary to reproduce the high frequencies, while the larger outer portion of the cone moves very little or not at all at these high frequencies due to the high value of X_m.

Another type of cone designed to accomplish the same result in a similar manner is the Polyfibrous cone which is moulded in such a manner as to give a continously graduated thickness from center to outer edge. This allows for flexing of the cone material and allows the central portion to move more or less independently of the main area for reproduction of the high frequencies. The compliance of the cone varies from extreme stiffness at the apex to considerable suppleness at the rim. High frequency reflection from the rim is reduced to a minimum.

Still another manner of accomplishing the same result is to mould or stamp corrugated rings into the cone structure as shown in Fig. 14.

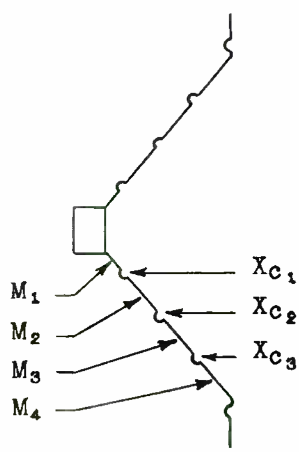


Fig. 14.—Corrugations break up cone structure into small masses coupled to one another by compliances.

The corrugations function as mechanical compliances to couple the several sections of the cone representing mechanical masses, as M_1, M_2, M_3, M_4, but which permit the smaller cone sections to move more or less independently of the large sections at high frequencies.

Fig. 15 shows a graphic comparison of the operating constants and characteristics for three similar types of speakers having cone diameters of 1", 4" and 16". The speaker designs were so chosen that the computed efficiency (μ) for each speaker remains essentially constant at 4 per cent over its useful range up to cut-off. Note that the small 1" speaker may have an efficiency as high as the larger speaker to the point of ultimate resistance where the air load, r_{MA}, becomes constant, but its low frequency power handling capacity is much less. The top chart shows a field flux density of 10,000 gauss for each speaker, which is 64,500 lines per square inch and is probably somewhat higher than the value used in the average receiver speaker. In the 1" speaker designed for high frequency reproduction the mass of the cone is only .015 grams and a light weight aluminum voice coil of the same mass is employed so as to keep the total moving mass down to a minimum.

The second row of charts in Fig. 15 show values of mechanical impedance plotted as a function of frequency for each of the three speakers. The mechanical reactance X_{MC} varies directly as the product of the mass of the moving system and the frequency. The mechanical impedance of the large cone becomes large at high frequencies and the resulting motion at the cone becomes very small. This graphically shows why a large rigid cone is unsuitable for high frequency reproduction. By comparing the X_{MC} values it may be seen that for any value of frequency the mechanical reactance is directly proportional to the total mass of the moving system (mass of cone plus mass of voice coil).

The mechanical reactance of the air load X_{MA} varies in the same manner as the mechanical reactance of the moving system up to the frequency corresponding to ultimate resistance, after which it drops off in the manner shown.

The mechanical resistance of the air load r_{MA} represents the useful radiation resistance into which the cone delivers useful acoustic energy. It should be noted that r_{MA} is proportional to the square of the frequency up to the frequency which produces ultimate resistance, beyond which it remains substantially constant. The point of ultimate resistance will be reached when the frequency rises to a value at which the corresponding half-wavelength is approximately equal to the cone diameter.

For example: The 16" cone (1.33 ft.) represents in dimension the half-wavelength of a 2.66' sound wave. The frequency corresponding to this wavelength is 1130'/2.66 = 425 cycles beyond which point the value of r_{MA} remains substantially constant. Above this frequency a rigid 16" cone will be inefficient. Proceeding in a similar manner for the 4" cone it is found that the r_{MA} value approaches a constant value at about 1720 cycles.

If a cone is to deliver a constant acoustic output over the frequency range above which r_{MA} becomes constant the impedance of the moving system must be made independent of frequency by proper processing of the cone. This condition may be realized by embossing suitable corrugations in the cone material which progressively reduce the mass reactance to the higher frequencies.

The curves in Fig. 16 show the necessity of using a large-size cone to handle a large acoustic output in the low frequency range. Such large cones must be of heavy construction to avoid introduction of harmonics due to flexing of the cone material. The curves show the amplitude of motion required at a given frequency to radiate one *acoustic* watt from each side of the cone.

SYSTEM	A	B	C
DIAMETER INCHES	16	4	1
MASS OF CONE GRAMS	40	1	.015
MASS OF VOICE COIL GRAMS	4	.35	.015
VOICE COIL MATERIAL	COPPER	COPPER	ALUMINUM
AIR GAP FLUX GAUSS	10000	10000	10000

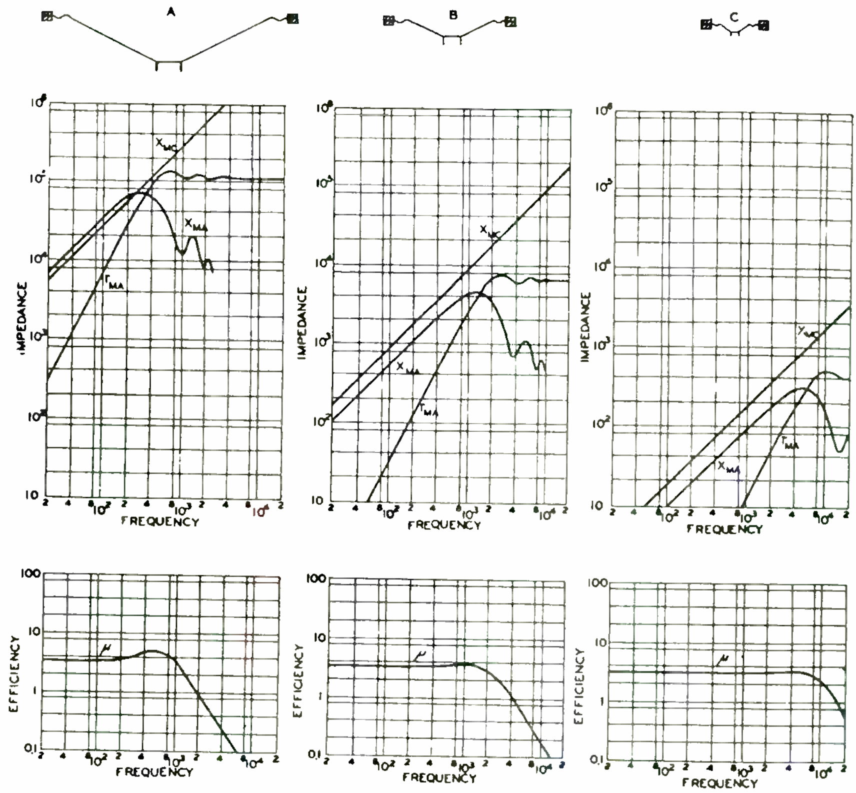


Fig. 15.—The impedance frequency characteristics of three cone loudspeakers having 1", 4" and 16" diameter cones and the efficiency frequency characteristics of the three loudspeakers. X_{MC} = mechanical reactance of the cone and coil, X_{MA} = mechanical reactance due to air load, r_{MA} = mechanical resistance due to the air load and μ = efficiency.

For example; at 100 cycles the 16" cone must move through a peak amplitude of .045"; the 4" cone through a peak amplitude of .72" and, the 1" cone through a peak amplitude of 11.5". It is obviously impractical for the small cone to radiate appreciable output at such low frequencies. *The required peak amplitude of motion, for a fixed acoustic output, varies inversely as the square of the cone diameter.* At this point it should be clearly understood that a small moving mass is necessary for high frequency reproduction and a large coupling surface is necessary if large outputs are to be handled at low frequencies.

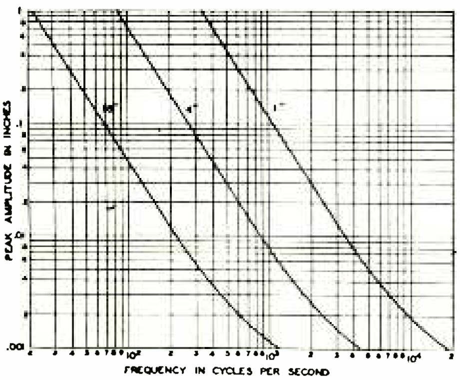


Fig. 16.—The amplitude frequency characteristics of vibrating pistons, of various diameters mounted in an infinite wall, for one watt output on one side.

DOUBLE VOICE COIL, SINGLE CONE SPEAKER.—Fig. 17 shows in cross-section a double-voice coil single cone speaker developed to extend the high frequency range beyond that of the usual low-price speaker. The speaker has a voice coil form on which are wound two voice coils: one of many turns of copper wire; the other, a few turns of aluminum wire. The latter coil is closest to the cone; the copper coil is farthest from the cone. Between the two voice coils is a crimp or corrugation in the voice coil form; this crimp acts as a compliance or spring member between the two coils.

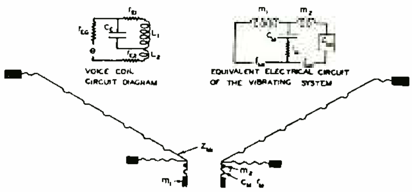


Fig. 17.—Cross-sectional view of the double coil, single cone loudspeaker with the voice coil circuit diagram and equivalent electrical circuit of the mechanical system. r_{EG} internal resistance of generator, e internal voltage of the generator, r_{B1} and L_1 resistance and inductance of the large coil and r_{B2} and L_2 resistance and inductance of the small coil and C_E capacitance of the condenser shunting the large coil, m_1 mass of the large coil, m_2 mass of the small coil, Z_{M1} mechanical impedance of the cone at the voice coil, C_m and r_m compliance and mechanical resistance of the corrugation, f_{m1} force generated in the large coil and f_{m2} force generated in the small coil.

The copper coil is the heavier, and vibrates only at the lower audio frequencies, in conjunction with the aluminum voice coil and the attached cone. The three parts vibrate as a single unit; the crimp between the two coils does not "give" or yield at the lower frequencies.

At the higher audio frequencies the inertia of the copper voice coil becomes so great that it tends to remain stationary, and instead only the aluminum voice coil and cone vibrate. The crimp in between flexes and thereby permits the copper voice coil to remain stationary while the aluminum voice coil vibrates.

The cone has a series of crimps or corrugations, too. These act to isolate the outer and heavier portions of the cone from the lighter inner sections closer to the aluminum voice coil, so that a smaller and smaller portion of the cone vibrates as the frequency is raised.

The result is that at the higher audio frequencies the vibrating mass is reduced to that of the light aluminum voice coil and the portion of the cone closest to it. This means a better high-frequency response because a smaller vibrating mass is involved.

Acoustically, the reduction in the vibrating portion of the cone means that a point source radiator is involved, with a better angular spread of the higher frequencies. Thus, the "highs" are radiated in as wide a beam as the "lows", so that the quality of the sound remains about the same off the axis of the speaker as on the axis.

Since the copper voice coil is inactive at the higher audio frequencies, there is no point in passing these components through it. Instead, a shunting capacitor bypasses this voice coil so that the "highs" flow only through the aluminum voice coil. This has another beneficial effect besides elimination of the I^2R losses in the copper voice coil; the impedance of the voice coil system does not rise materially with increase in frequency owing to the inductance of the coil system because essentially the inductance to the "highs" is reduced.

Thus, the inductance involved is that only of the aluminum voice coil of relatively few turns instead of both voice coils. Of course the force produced is decreased if the number of turns is reduced, but this is in part compensated for by the lowered voice-coil impedance which allows more audio current to flow. A further benefit is that the more nearly constant impedance of the speaker with frequency makes it better suited to a pentode power output stage, since it will be recalled that a pentode tube tends to generate excessive distortion if the load impedance presented to it is too high.

Fig. 17 shows both the actual electrical circuit of the voice coil system, and the electrical ANALOGUE of the mechanical vibrating characteristics of the system. Thus, for example, r_{E1} is the ohmic resistance of the copper voice coil, and L_1 its inductance, whereas m_1 is the electrical inductance analogue representing the actual mass of the copper voice coil, etc. The diagram is self-explanatory, and of course corroberates the statements made above.

To summarize the action: at the higher audio frequencies the ordinary cone refuses to vibrate as a single unit, and normally only the portion nearest the voice coil vibrates. This action is encouraged here by the corrugations in the cone. When this occurs, the voice coil mass becomes excessive. To reduce this, the voice coil is made in two parts, with the lightest (aluminum) part nearest the cone, and with a crimp or corrugation between it and the copper voice coil. Also electrically the copper voice coil is bypassed by a capacitor.

This permits the aluminum voice coil to vibrate independently of the copper voice coil, which remains stationary because of its mass and also because no appreciable amount of high-frequency current is flowing through it. We therefore have at the higher frequencies the effect of a small light cone driven by a light aluminum voice coil, whereas at the lower frequencies we have a larger heavier cone driven by a voice coil of more turns and also incidentally of greater mass. Acoustically the speaker acts essentially as a point source throughout most of the frequency range, and therefore provides a wide distribution in space of the radiated sound.

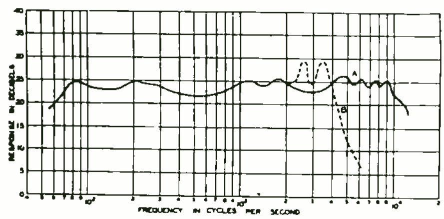


Fig. 18.—A. The response frequency characteristic of a double coil, single cone loudspeaker. B. The response frequency characteristic of the same cone as in A driven by a single coil.

Fig. 18 shows response frequency characteristics of a cone when driven by a double voice coil system. Note that considerable extension of the characteristic is obtained in the high register.

MULTIPLE CONE LOUDSPEAKER.— Instead of using multiple voice coils, an alternative design good up to about 8,000 c.p.s. is one employing a plurality of cones, actually three in number. This design is somewhat cheaper than the double voice coil, but not quite as good at the higher frequencies.

Fig. 19 shows a single voice coil multiple cone speaker. Three cones of graduated sizes are employed to obtain better response over a wide frequency band. Increased frequency range is obtained by coupling decreasing sizes of cones to the voice coil for the high frequency reproduction. The cone sections Z_{M1}, Z_{M2} and Z_{M3} are coupled to each other by the compliances C_{M1} and C_{M2}.

At low frequencies the compliance reactances of C_{M1} and C_{M2} are large compared to the mechanical impedances of Z_{M1} and Z_{M2} respectively. There is therefore no flexing of the coupling compliances C_{M1} and C_{M2} and the three cone sections are essentially rigidly coupled together and move together as a single large cone element.

In the upper mid-frequency range the compliance reactance of C_{M1} becomes small compared to the cone impedance Z_{M1} and the large cone section Z_{M1} is therefore essentially decoupled from the voice coil m and remains stationary. The voice coil now delivers its force f_m to the two smaller cone sections Z_{M2} and Z_{M3} which, taken together, have much less mass than the large cone section Z_{M1} and permit more uniform reproduction of the frequencies in this range.

At high frequencies the compliance reactance of C_{M2} becomes small compared to the mechanical impedance of cone section Z_{M2} and therefore the intermediate cone section Z_{M2} also becomes decoupled from the voice coil. It is thus observed that the voice coil now delivers its force f_m to the small cone section Z_{M3} which is best adapted to high frequency response due to its small mechanical mass.

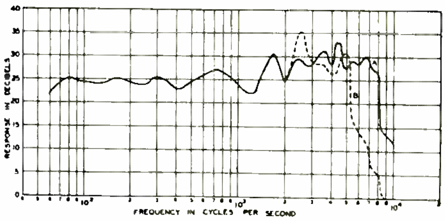


Fig. 20. — A. The response frequency characteristic of a single coil, multiple cone loudspeaker. B. The response frequency characteristic of the same large cone driven by the same coil as in A.

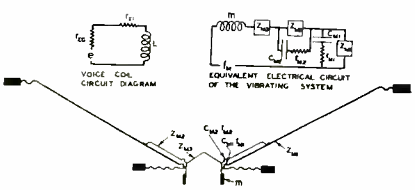


Fig. 19.—Cross-sectional view of the multiple cone, single coil loudspeaker with the voice coil circuit diagram and the equivalent electrical circuit of the mechanical system, r_{EG} internal resistance of the generator, e internal voltage of the generator, r_{E1} and L resistance and inductance of the voice coil, m_1 mass of the coil, Z_{M1}, Z_{M2} and Z_{M3} mechanical impedance of the large, intermediate and small diameter cones, C_{M1} and r_{M1} compliance and mechanical resistance of the corrugation in the large cone and C_{M2} and r_{M2} compliance and mechanical resistance of the corrugation in the intermediate cone. f_M force generated in the voice coil.

Fig. 20 shows the response characteristic for a single coil driving a single large cone and when driving a multiple section cone as described in the preceding paragraph. The multiple cone extends the upper frequency from 5100 cycles to 8000 cycles, an increase in frequency range of more than one-half octave.

MULTIPLE CONES AND VOICE COILS.—Fig. 21 shows a speaker consisting of a double voice coil (as in Fig. 17) driving a double cone (similar to Fig. 19), thus combining the inherent advantage of reduced voice coil mass *and* reduced cone mass at the higher frequencies. The equivalent voice coil and impedance circuits are also shown in detail. The large voice coil section m_1 is rigidly coupled to the large cone section Z_{M1}, and also coupled through the mechanical compliance C_M to the small voice coil section m_2. The small voice coil m_2 is rigidly coupled to the small cone section Z_{M2}. The large voice coil section m_1 is represented electrically as $L_1 r_{E1}$ and is shunted by a capacitor C_E. Improved high frequency performance is obtained by a reduction in mass of voice coil and cone.

At low frequencies the shunting reactance of C_E is large compared to the impedance of the large voice coil section $L_1 r_{E1}$ and essentially the entire current from source e flows through both coils in series. The compliance reactance C_M is large compared to the mass of the large voice coil m_1 and the entire system functions as a single voice coil-single cone speaker.

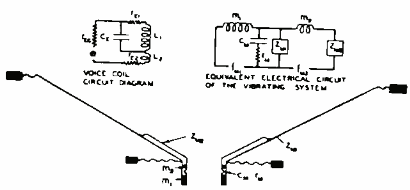


Fig. 21.—Cross-sectional view of the double coil, double cone loudspeaker with the voice coil circuit diagram and the equivalent electrical circuit of the mechanical system, r_{EG} internal resistance of the generator, e internal voltage of the generator, r_{E1} and L_1 resistance and inductance of the large coil and r_{E2} and L_1 resistance and inductance of the small coil and C_E capacitance of the capacitor shunting the large cone, m_1 mass of the large coil, m_2 mass of the small coil. Z_{M1} mechanical impedance of the large cone, Z_{M2} mechanical impedance of the small cone, C_{M1} and r_M compliance and mechanical resistance of the corrugation, f_{M1} force generated in the large coil and f_{M2} force generated in the small coil.

At high frequencies the electrical impedance of C_E is small compared to the electrical impedance of either voice coil, $L_1 r_{E1}$ or $L_2 r_{E2}$ and therefore the source voltage e will establish maximum current through the small voice coil $L_2 r_{E2}$. The compliance reactance of C_M will be small compared to the mechanical impedance of m_1 and Z_{M1} and the large voice coil section and large cone are effectively decoupled from the small voice coil section and small cone section. Therefore at high frequency the large coil and cone sections remain essentially stationary and the speaker functions as a light weight cone and coil system.

Fig. 22 shows that good response is obtained up to about 14,000 cycles with a single speaker unit and without using cross-over filters as may be required with multiple speakers. It is also interesting to note the small drop in high frequency response for points 30° off the speaker axis.

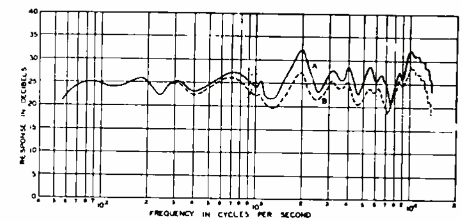
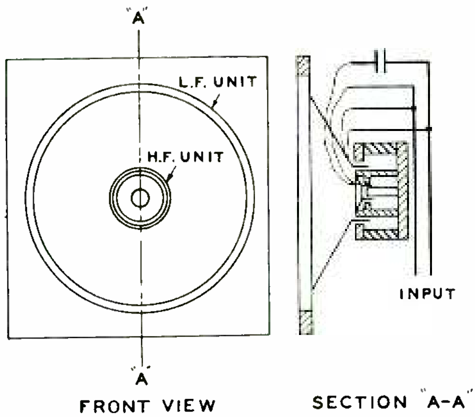


Fig. 22.—A. The response frequency characteristic of a double coil, double cone loudspeaker on the axis. B. The response frequency characteristic on a line making 30° with the axis.

THE TYPE LC-1A LOUDSPEAKER.—A more recent design similar to the one just described is the Type LC-1A loudspeaker manufactured by RCA for broadcast studio work. This speaker also has two cones and separate voice coils mounted on SEPARATE voice-coil forms. Indeed, the high frequency cone is suspended WITHIN the voice coil for the low-frequency unit.

It is illustrated in Fig. 23. The larger cone is the low-frequency or "WOOFER" unit,* and the small cone within the voice coil of the woofer is the high-frequency unit or "TWEETER". The latter has its own, smaller voice coil.



(*Courtesy RCA*)

Fig. 23.—A congruent coaxial combination of low-frequency and high-frequency direct-radiator loudspeaker units.

It will be observed that the permanent-magnet field structure has a central core of permanent magnet (alnico) material, which feeds flux through the inner air gap in which the tweeter cone's voice coil is immersed. An outer ring of alnico produces a separate field in the outer air gap in which the woofer voice coil is immersed. Thus the outer annular iron ring for the tweeter acts as the inner pole for the woofer.

The advantage of this arrangement is that the apparent sources for the low and the high frequencies are two concentric cones located at the same point; i.e., sounds all seem to come from one point in the room. As a result, there are no interference effects between the two cones in the frequency range where they overlap; indeed, they vibrate in phase and hence like a single cone in this region.

This can be compared with the behavior of two antennas operating on a common frequency, as in an array. If they are at the same place in space (within a fraction of a wave length, and are driven in phase, they operate essentially as a single antenna. If, on the other hand, they are separated from one another by an appreciable portion of a wavelength, then in certain directions their radiations meet in phase and reinforce each other, whereas in other directions their radiations meet out of phase (owing to the difference in path length) and tend to cancel. The result is a directional pattern having in general several lobes and nulls.

Exactly the same sort of effect can be obtained with two loudspeakers separated by an appreciable portion of a wavelength, such as a quarter wavelength. It is true that in the case of sound, the walls of the enclosure develop standing-wave effects, so that your ear is rather used to nulls and peaks; nevertheless, it was felt that the loudspeaker itself should not be guilty of such variations in response.

*It has also been called the "growler" unit.

There is another advantage of this design. In the usual case where the tweeter and woofer units are not mounted coaxially, it is desirable to make their overlap region in the frequency spectrum as narrow as possible so as to confine any nulls and peaks to a small portion of the response range. This in turn makes the electrical "cross-over" network more difficult and critical to design and adjust, whereas in this "duo-cone" coaxial design, considerable frequency overlap is possible.

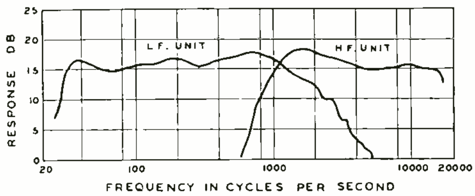
The large cone is 15 inches in diameter, so that it can radiate a reasonable amount of power down to 30 c.p.s. or so without requiring to vibrate through an excessive amplitude. It is made in the form of a shallow rather than a deep cone so as to radiate the upper frequencies in its range through a relatively wide angle.

The high-frequency cone is 2 inches in diameter and hence of very low mass. It behaves essentially as a piston radiator up to 10,000 c.p.s. Above this frequency its response would fall off rapidly; to prevent this a compliance is provided between the voice coil and the cone to separate the two masses and obtain a progressive wave-motion effect that permits the response to be extended to 15,000 c.p.s. The shallow nature of the cone enables the sound to be radiated through a wide angle in the room.

The cross-over network that confines the low frequencies to the 15-inch cone and the high frequencies to the 2-inch cone is a very simple affair. A capacitor is connected in series with the high-frequency voice coil, and the two then connected in parallel with the low-frequency voice coil to the output terminals of the amplifier.

The relatively high inherent inductance of the low-frequency voice coil prevents the higher frequencies from flowing through it; instead; they flow through the capacitor and the high-frequency voice coil. Thus no high-frequency power is wasted in the low-frequency coil. Similarly, the high reactance of the capacitor keeps the low-frequency power out of the high-frequency coil.

Fig. 24 shows the frequency response of the low-and high-frequency units mounted on a large flat baffle. Where the two outputs are exactly equal (slightly above 1000 c.p.s.) the combined power is double either one, or the level rises, 3DB. This must be borne in mind in interpreting Fig. 24; it means that the overall response is practically flat.



(Courtesy RCA)

Fig. 24. — Frequency response of the low-and high-frequency units mounted in a large flat baffle.

The speaker may be mounted in a flat baffle or an enclosure, such as that illustrated in Fig. 25. The action of the latter will be discussed farther on in this assignment, but it will be noted here that it has an opening or port below the loudspeaker unit, and this port enables the very low frequency radiation to be increased without increasing the low-frequency cone excursion. This in turn means more low-frequency output with less distortion.

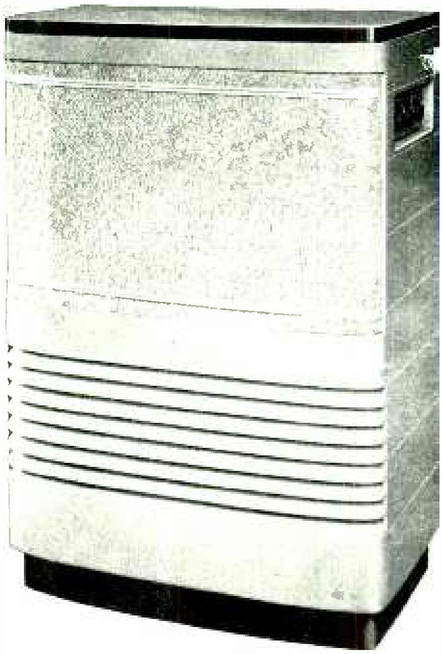


Fig. 25.—LC-1A Speaker mounted in a phase inverter cabinet.

TRI-AXIAL LOUDSPEAKER.—Fig. 26 shows another design for a high fidelity system, the Jensen Tri-axial Loudspeaker Model G-610. It consists of three units: a low-frequency direct-radiator cone unit, a small horn intermediate-frequency unit, and a small horn high-frequency unit. The latter is mounted out front so as not to have its sound beam obstructed and modified by the obstructing effect of the other two units.

In order to obtain a piston-like behavior on the part of the low-frequency cone, the cross-over between it and the intermediate-frequency unit is set around 600 c.p.s. The next cross-over between the intermediate and high-frequency units is set at 4000 c.p.s.

An interesting feature of this system is the use of phenolic materials for the diaphragms. This material has high internal damping which minimizes modes of vibration differing from a piston-like action. It is therefore more free of sharp resonances and "birdies" characteristic of metallic diaphragms.

The low-frequency unit has a 15-inch cone shaped in a certain curve known as a Hypex curve. This curve is a variation on the exponential curve and a horn of this shape has a somewhat better impedance characteristic at the low-frequency cut-off point.* The cone not only acts as a low-frequency direct radiator, but also as a continuation of the hypex horn of the intermediate-frequency radiator so as to afford sufficient bell area for a cut-off somewhat below 600 c.p.s., as will be explained farther on.

A three-inch voice coil drives this cone, and it is immersed in the magnetic field of a 6 1/2-pound Alnico V magnet. This magnet struc-

*This and other horn characteristics are discussed farther on in this assignment.

ture is separated from that of the intermediate-frequency magnet by a non-magnetic material shown in heavy cross hatching. An unusually large spider together with a rigid cone provides good linearity and low distortion even at high operating levels.

voice coil. The sound is taken off the diaphragm around its circumference; a so-called high-frequency plug covers up the center of the diaphragm. This improves the frequency response near the upper 4000-cycle cross-over, and will be explained in greater detail farther on.

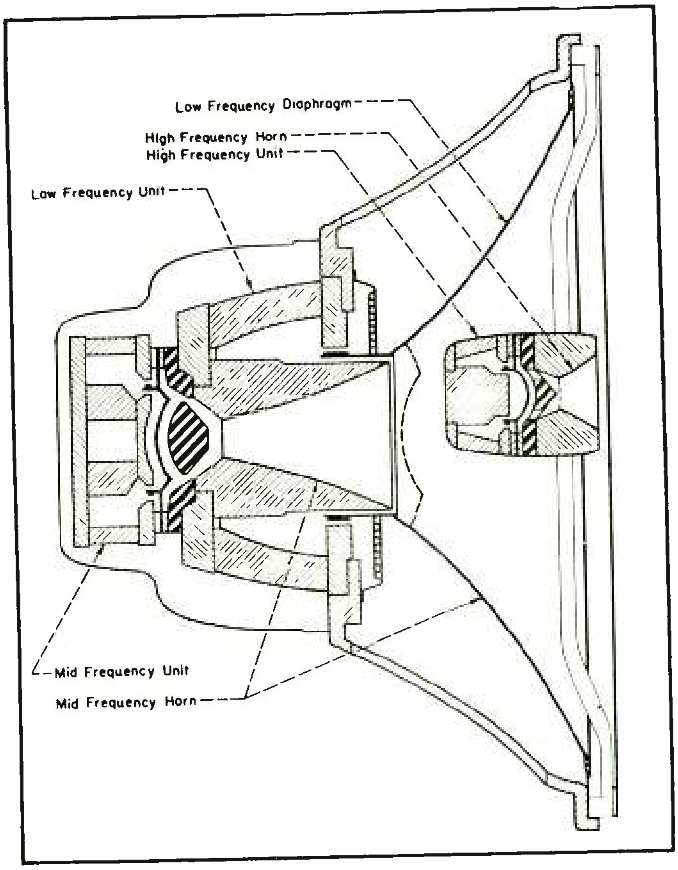


Fig. 26.—The Jensen Tri-axial Loudspeaker system in cross section, showing the three units mounted on a common axis.

The intermediate-frequency unit has a rigid re-entrant type phenolic diaphragm (the inner and outer portions are of reversed curvature) and is driven around the edge rather than near the center by a two-inch

The moving system has a mass of two grams, and the voice coil is immersed in a magnetic field of a density of 17,500 gauss, which is extremely high, and makes for high efficiency of operation. Note that the initial section of the Hypex horn is fashioned from the center core of the low-frequency magnet structure.

The tweeter also has a light stiff phenolic diaphragm driven by a one-inch aluminum voice coil, with a moving system dynamic mass of but 130 milligrams. The flux density is here 15,500 gauss, which is also very high. The clearance between the diaphragm and the horn plug, while adequate, is so small that the die casting unit has to be machined after it is cast to obtain the accuracy required.

The result is a response that is down only 3 db at 18,000 c.p.s. compared to that in the 60 to 400 c.p.s. range. The Hypex horn employed has a bell diameter of but 1 1/2 inches, but this is adequate for a 4000-cycle cutoff. At the same time, the outer surface of the tweeter unit is streamlined so as to present a minimum obstacle to the output of the other two units.

Fig. 27 shows in block diagram form the cross-over and control system for the loudspeaker, as well as the individual and combined frequency responses. Air-core inductances in conjunction with capacitors are used for the cross-over networks, together with three-step level adjustments for controlling the middle-and high-frequency response of the speaker. These are all contained in a separate chassis that is connected to the speaker assembly by means of a cable.

In addition an over-all L-pad volume adjustment supplies individual loudspeaker control for use in multiple installations, and a four-position high-frequency cutoff control permits attenuation of the output of the tweeter unit where the program material requires such reduction in frequency response.

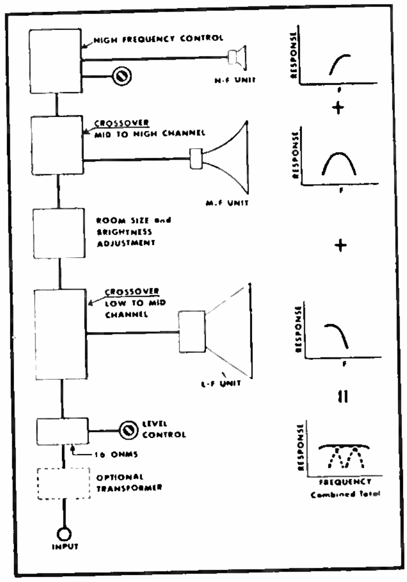


Fig. 27. — Schematic of three-way cross-over and control system of Model G-610 Tri-Axial Loudspeaker.

THE 820A CORNER SPEAKER SYSTEM.—Fig. 28 shows the external appearance and the internal components of the 820A Corner Speaker system built by the Altec Lansing Corporation, who furnish speaker systems for motion picture theatres, etc. This speaker employs the same professional units that are installed in motion picture theatres.

The woofers are two 803A units feeding directly through a straight (not folded) horn into the room. The horn cabinet is sturdily braced and arranged to be placed in a corner of a room, where the walls and floor can act as continuations of the horn.

The tweeter is an 802B unit feeding an H-808 multicellular horn. The latter is designed to radiate the high-frequency components over a wide angle (as will be explained farther on) and provides an efficient unit capable of a smooth response from 800 c.p.s. (the cross-over frequency) to 15,000 c.p.s. An aluminum edge-wise wound voice coil is employed in conjunction with a 2-mil thick aluminum diaphragm, and a large Alnico 5 permanent magnet is employed to provide the magnetic field.

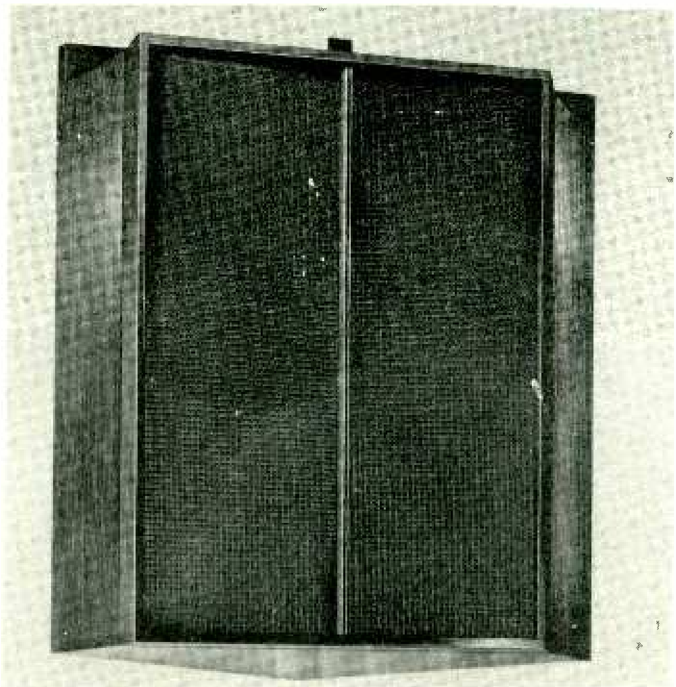


Fig. 28.--External and internal views of the 820A Corner Speaker System manufactured by the Altec Lansing Corporation.

A cross-over network in a separate chassis is provided. It also enables the high-frequency output to be attenuated in four 1-db steps so as to coordinate the performance of the speaker to the surroundings. A power input of 30 watts is possible, and the required amplifier output impedance is 6 to 12 ohms. The vertical distribution of the sound is 40 degrees, and the horizontal distribution is 90 degrees, so that when placed in the corner of a room, it covers the entire room.

SPEAKER BAFFLES AND ENCLOSURES

BAFFLES.—A baffle is required with a loudspeaker because both sides of the cone produce sound waves. The baffle is used to prevent the sound waves on the two sides from neutralizing each other; i.e., the rarefaction from one side neutralizing the simultaneous condensation from the other side. In other words, as the cone moves back and forth the air flows around the cone and tends to neutralize the air pressures.

At any given frequency the cone must move in *each* direction a number of times corresponding to the frequency. Thus at 100 cycles the cone must move in each direction 100 times. If complete neutralization of the air pressures takes place there will be no sound wave. With partial neutralization there will be less sound, particularly at the low frequencies, and thus frequency distortion. The neutralization of air pressure can be prevented by making the path between the front and back long enough so that the air cannot flow from one side to the other in time to equalize the pressures.

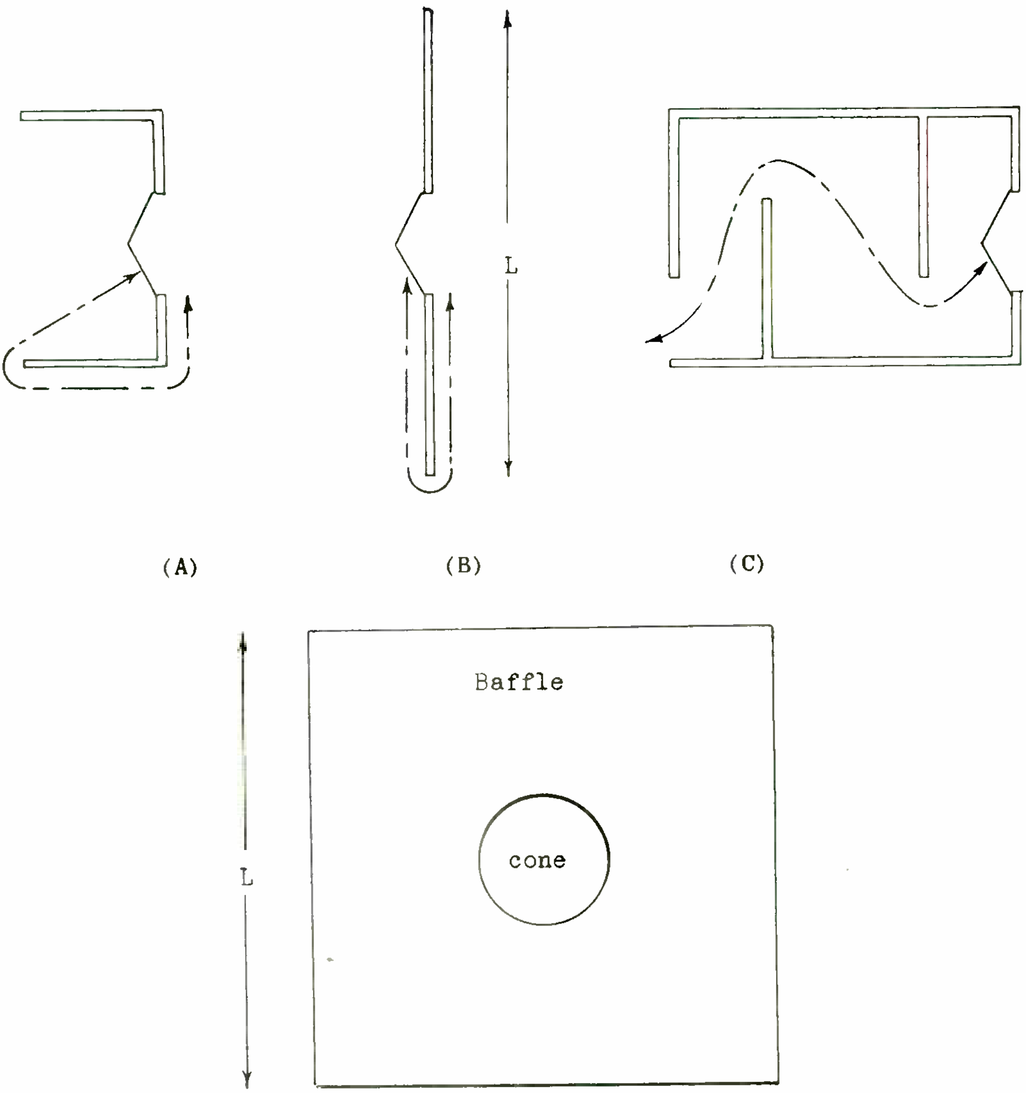


Fig. 29.—Various types of baffles.

Methods of doing this are shown in Fig. 29. The extensions beyond the edges of the cone form the baffle. This baffle may be flat, in the form of a box, or may be a horn. Fig. 29(A) shows an ordinary form of baffle which may consist of the front and sides of the radio cabinet in which the reproducer is mounted. 29(B) shows the type of flat baffle often used with permanent installations of public address systems where space is of secondary consideration, such as in a larger auditorium. 29(C) illustrates the manner in which the length of the air path may be increased where the space available is quite limited by using a series of baffles. 29(D) is a front view of 29(B).

Fundamentally, the front and back waves must be separated for a distance and corresponding amount of time such that each wave has a chance to spread into space before it meets the other wave. In this way the energy from each side of the cone has a chance to spread and dilute itself to such a degree that the portions of the two waves that do meet and neutralize each other constitute but a small fraction of the total energy radiated.

The rule as to baffle size is that the path length from one side of the cone to the other side should be at least half a wave length. This is illustrated in Fig. 30. Roughly, the distance from the front to the rear of the cone, as measured along the surface of the baffle, should be one-half wavelength. Then, if a compression wave starts out from the front side, by the time it gets to the back of the cone, the cone is moving to the left and generating a compression wave from that side. Hence no substantial amount of neutralization occurs.

This is of course a rough rule. At the edge of the baffle the back and front waves meet in phase opposition because they travel equal distances from the center to the edge, and it is in this region that cancellation will occur. But this is only a local effect, and in the case of a flat baffle, nobody is sitting in line with the edge of the baffle. To the front and rear, neutralization has been prevented in the manner described above.

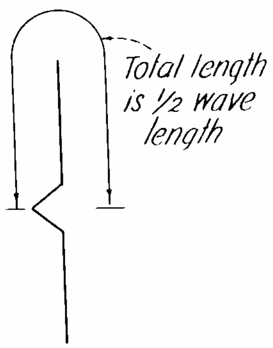


Fig. 30. — The baffle should provide a path length from the front to the rear of the cone equal to half a wavelength.

Hence the rule is that the baffle should present a path length from front to rear of half a wavelength ($\lambda/2$). If the baffle is flat and circular in shape, then its radius should be $\lambda/4$, and its diameter therefore $\lambda/2$. Since

$$\lambda = V/f \qquad (1)$$

where V is the velocity of sound, (= 1130 ft./sec.) and f its frequency, it follows that the diameter D of such a baffle should be

$$D = \lambda/2 = V/2f = 1130/2f$$

For a low frequency response down to 100 c.p.s.,

$$D = 1130/2 \times 100 = 5.65 \text{ ft.}$$

For 60 c.p.s.

$$D = 1130/2 \times 60 = 9.417 \text{ ft.},$$

and for 40 c.p.s.,

$$D = 1130/2 \times 40 = 14.12 \text{ ft.}$$

If the baffle is square instead of circular, the length of a side, L, can be used for D. This represents the square which circumscribes (surrounds) the circle of diameter D. At sufficiently high frequencies, the cone itself can act as its own baffle, so that the baffle just mentioned is solely to increase the low-frequency response.

LOW-FREQUENCY RESONANCE.— It may therefore be asked why the baffle is not made in the form of a closed box surrounding the cone, so as to act the same as an infinite baffle. In this case the response should extend down to very low frequencies.

The answer is that a closed box is like a dashpot or acoustic spring, and acts as a stiffness component to the motion of the cone. In other words, the cone in attempting to vibrate back and forth has to compress and expand the enclosed air, just as if a spring were fastened at one end to the cone and at the other end to the enclosure or speaker frame.

Such a spring effect tends to raise the so-called low-frequency resonant point. This refers ordinarily to the interaction of the mass of the cone and the stiffness of the spider and rim suspensions; the two cause the cone to vibrate through large and even excessive amplitudes at the frequency at which they balance (resonance). This occurs at 40 c.p.s. for large heavy cones having large compliance suspensions (low stiffness) to perhaps 150 c.p.s. or more for a small light cone and stiff spider or rim suspension.

The resonance causes excessive output at this frequency; i.e., a peak in the response curve. Above and below this resonant frequency the amplitude of vibration decreases rapidly. Fortunately at higher frequencies much less amplitude is required to radiate adequate amounts of sound power, so that the amount of power radiated above the resonant frequency is not very much less than at the resonant frequency.

Below this frequency, however, *more* rather than *less* amplitude is needed for appreciable acoustic radiation so that if the amplitude decreases below (as well as above) the resonant frequency, the power output drops rapidly. From this it follows that a loudspeaker radiates appreciable amounts of power down to its low-frequency resonance point; below this its output drops off very rapidly.

Consequently, if an enclosed baffle adds stiffness to its motion and hence raises its resonant frequency, it thereby reduces its frequency range (even though the totally enclosed baffle prevents low-frequency attenuation by preventing interaction between the front and rear waves).

This is illustrated in Fig. 31. Normally the low-frequency resonant point is f_1, and the frequency range is f_1 to f_3. But if the speaker is placed in a small box, its resonant frequency is raised to f_2, and the range correspondingly reduced to from f_2 to f_3.

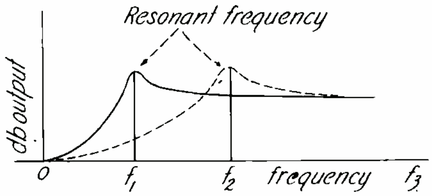


Fig. 31.—A small enclosure raises the resonant frequency of a loudspeaker and thereby reduces its frequency range.

SPEAKER ENCLOSURES.—It is therefore necessary to use a speaker enclosure or box that is fairly large,—several cubic feet or so in volume, in order that its stiffness be low. In that case the low-frequency resonance is raised but a small amount, and the inherent frequency range of the speaker is maintained. At the same time the actual low-frequency radiation is permitted to occur without interference between the front and rear waves of the cone.

The only flaw in all this is that one ends up with a fairly large baffle, which was the very thing it was hoped to avoid with a total enclosure. However, the box still is not nearly as large in wall area as the flat baffle, and besides, in the ordinary console radio, a box is required anyway to house the amplifier and other electronic gear.

Although many different principles have been applied to speaker cabinets or enclosures, such as resonant pipes, half-wavelength tubes, and the like, present day practice is to employ either a totally enclosed box, or more often a vented baffle, also known as a reflex baffle. This is illustrated in Fig. 32. where it will be observed that an opening has been provided below the cone speaker. This opening can be just an opening, or a short tube extending back into the baffle.

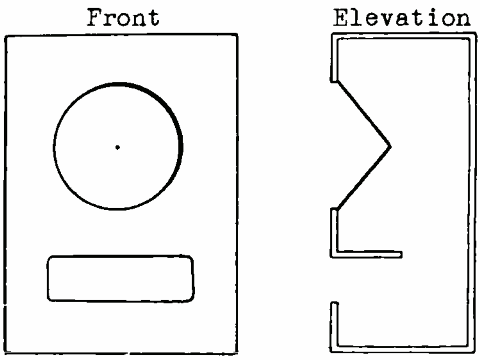


Fig. 32.—Use of a speaker vent in the cabinet to augment the low-frequency response.

The principle of operation is based on the behavior of two coupled circuits resonant to the same frequency, such as a double-tuned i-f transformer. As illustrated in Fig. 33, either winding will resonate with the associated tuning capacitor to the same frequency, but when the two are coupled together and adequately damped, two broad resonant peaks are obtained, one on either side of the original resonant frequency. In other words, the band width has been increased, and the response flattened.

The loudspeaker action is similar. The compliance of the enclosure is made approximately resonant in conjunction with the mass of the air in the port of the same low-frequency as that of the cone unit, by proper choice of volume and port area. The volume or cabinet dimensions, and the port opening are not critical; a volume of 6 to 8 cu. ft. is generally ample, and the port area is adjusted accordingly.

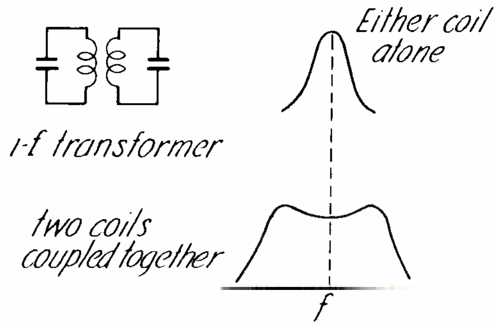


Fig. 33. —Coupling two resonant circuits together provides a broader response consisting of two resonant peaks.

This cabinet is acoustically coupled to the back of the cone unit by its proximity. The result is a response similar to Fig. 33, and shown in Fig. 34. Without the enclosure, a single resonant peak is obtained at f_1. With the enclosure, the response—shown by dotted lines—has two peaks, one BELOW f_1 at f_2, and the other above at f_3.

Usually the latter is masked by the improved radiation-resistance damping at this higher frequency, but several advantages are obtained:

1. The low frequency response is extended from f_1 to f_2, a worth-while gain.
2. The single high peak is replaced by two smaller peaks that are usually unobjectionable; i.e., the response has been *flattened* as well as extended.
3. The greater damping at f_1 means that the cone excursions have been reduced. This means that the harmonic distortion resulting from the voice coil "popping out" of the air gap at high levels and low frequencies has been reduced.
4. The vent, port, or opening in the cabinet acts as an additional acoustic source or radiator at the lower frequencies, in addition to its acting as a mass that resonates with the compliance of the enclosure. Its area can be on the order of that of a 15- or 20-inch cone loudspeaker, even though actually only an 8-inch cone may be employed. The effect is as if at the lower frequencies a much larger cone were employed, that had a higher radiation resistance and hence operated at a higher efficiency in this frequency range.

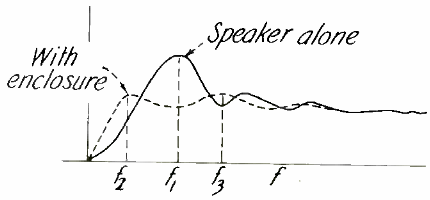


Fig. 34. —Effect of a vented baffle on the response of a cone loudspeaker unit.

The reader may question the utility of such a port, particularly since it is preferably mounted quite close to the loudspeaker unit. The basis for such doubt is the fact that the port provides sound coming from the back of the cone and therefore presumably in phase opposition to the sound coming directly from the front of the cone. It would appear that cancellation of the sound energy should occur, the very thing the baffle is supposed to prevent.

The answer is that in this region of resonance, large phase shifts occur in the wave, with the result that the wave emerging from the port is substantially in phase with the wave coming from the front of the cone, and therefore produces an additive effect.

At somewhat higher audio frequencies, the difference in path length begins to shift the phase of wave coming from the rear of the cone via the port into the room, so that again an approximately additive effect can be obtained.

However, at frequencies above the low-frequency end, the front of the cone can obtain sufficient "bite" on the air in front of it because of the increasing radiation resistance and therefore does not require the support of the radiation from the port. In other words, the cabinet had done its job at the low-frequency end of the spectrum; it is not required at the higher audio frequencies.

It is therefore found desirable to line the interior of the box with acoustic absorbing material, such as cotton batting. A thickness of 2 to 3 inches is not sufficient to present appreciable damping to frequencies at 100 c.p.s. and below, particularly around 30 c.p.s., so that this acoustic material does not damp the double resonance peak unduly and absorb and thus waste energy in this region. Instead, it is the improved radiation resistance that acts as damping, and this sort of damping is desirable, as it means power output.

It is at the higher frequencies, where the output from the rear of the cone is not required, that the acoustic material performs its function of absorbing this output, and preventing standing wave phenomena with resulting mass and compliance variations in the loudspeaker unit. Cabinets without vents or ports are also lined with acoustic material to prevent such standing-wave phenomena in them, too.

The size of the port can be varied in some cases so as to vary the response characteristic. As more of the port is uncovered, more very low-frequency radiation is obtained, up to a certain limit, and closing up the port shifts the peak to a higher frequency, as well as increases its amplitude. By properly adjusting the port to the loudspeaker unit employed, the frequency response can be made to have the shape desired.

BAFFLE RESONANCES.—The baffle must be non-resonant. It should be made of plywood 3/4" or more thick or of heavy celotex. For theatre reproduction the baffle may be as large as 9' square, and 1 1/2" thick. For a home receiver a baffle 3' to 4' square is ordinarily considered sufficient.

Cabinet resonance is very common with radio receivers employing large dynamic speakers. Resonance of cabinet panels and confined air causes distortion by reinforcing some frequencies more than others. Since it is usually not practical to remove resonant panels they may be made highly absorbent and the open condition approximated by lining with some absorbent material such as celotex, as mentioned previously, and/or a system of strong internal bracing may be used. The metallic sound of metal cabinets may be eliminated by lining with 1/2" celotex. The chassis and speaker may be mounted on this material to reduce microphonic effects. A good speaker cabinet should be of heavy solid material, well braced and solidly glued to minimize cabinet resonance.

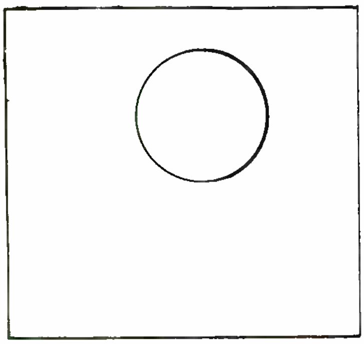


Fig. 35.—By mounting the speaker unit offset in the baffle, equalization of the frequency response may be obtained.

At frequencies for which the distance from front to back of the cone is one wavelength the sound waves will be more or less cancelled as they are in phase opposition if the baffle is symmetrical as, for example, a circular baffle. Speakers mounted on an approximately symmetrical baffle show a severe dip in the response characteristic at the frequency which corresponds to a wavelength which is equal to the distance from front to back of cone. This effect may be minimized by mounting the speaker off center in an unsymmetrical baffle as indicated in Fig. 35. There will now be an infinite number of paths of different lengths from one side of the cone to the other which avoids severe cancellation at any one frequency.

HORNS

HORN LOUDSPEAKER.—The horn is an acoustic output transformer; it transforms the low radiation resistance of the room per unit area at its mouth or bell into a larger radiation resistance at its throat and presents this higher radiation resistance to the diaphragm located at that point.

As a result of this better coupling, the efficiency is very much improved over that of a direct-radiator loudspeaker such as the cone unit described previously, and the amplitude of vibration of the diaphragm is reduced to a value that obviates the excessive distortion that would occur were the small diaphragm coupled directly to the air.

Fig. 36 is a simple line drawing of a horn and loudspeaker unit. Such a unit generally employs an aluminum or duralumin diaphragm, driven by a voice coil similar to that employed in a cone unit. The shape shown in Fig. 36 is that of a disc; as will be explained later, the actual shape may be different in order to obtain a stiffer diaphragm.

The voice coil and diaphragm have a certain amount of mass, and the clamped edge presents a certain amount of series compliance, as is indicated by M_d and C_d in Fig. 37. The air chamber presents a certain amount of compliance represented by C_a, and F represents the a-c force developed in the voice coil by the flow of audio currents in it.

The air pressing against the diaphragm exerts a force equal to the area of the diaphragm multiplied by the pressure developed in the enclosed air when the diaphragm squeezes (or expands) this air. The same *pressure* is developed in the throat (by Pascal's principle) but over a smaller area, so that the *force* in this region is less than on the diaphragm. transformed into a high velocity of air in the throat, just as water flows slowly in the wide parts of a river and much faster in the narrow parts or *rapids*.

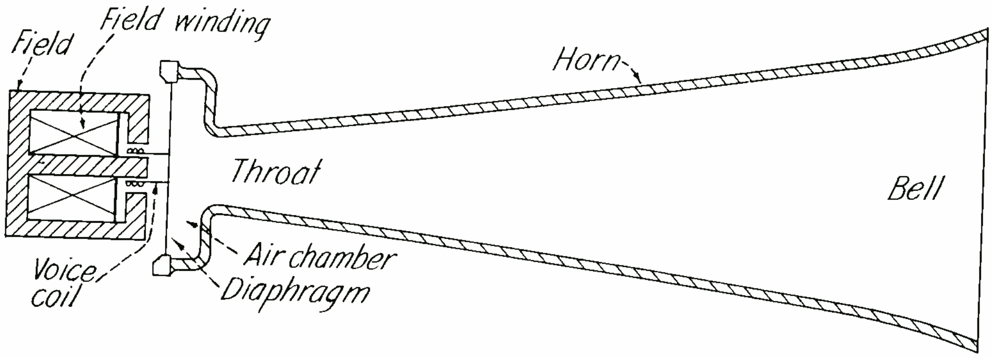


Fig. 36. — Representative form of horn loudspeaker.

This means that the force on the diaphragm is to the force on the pellet of air in the throat of the horn as the area of the diaphragm, call it A_d, is to the area of the throat, call it A_t. In other words, the force (mechanical voltage) is *stepped up* going from the throat to the diaphragm, or *stepped down* when viewed in the opposite direction. At the same time a low velocity of the air in front of the diaphragm is

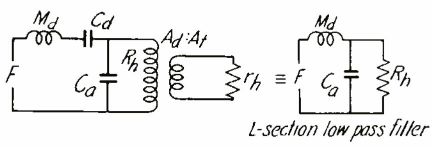


Fig. 37. — Equivalent electrical circuit for a horn type loudspeaker.

Since *velocity*, of the air or of the diaphragm, etc., corresponds to current in an electrical circuit, the diaphragm and throat areas act like an electrical transformer of turns ratio A_d/A_t, in that the force is stepped down from the diaphragm to the throat in the ratio of A_d/A_t, and the velocity is correspondingly stepped up. Moreover, such escape of velocity through the throat relieves the pressure built up in the air chamber or compliance, so that the transformer in Fig. 37 is shown *shunting* C_a, and thus relieving the pressure in the air chamber.

Finally, the horn appears as a (mechanical) resistive load r_h at the throat end when the bell or mouth end feeds acoustic energy into the room or auditorium. Thus r_h is shown connected to the secondary terminals of the transformer; it reflects to the primary terminals a higher resistance $R_h = r_h(A_d/A_t)^2$ in parallel with C_a.

The horn itself is a kind of impedance-matching network of the transmission-line type; i.e., one having distributed constants. It matches the low impedance of the air in the room to the higher impedance of the air in the throat, but only down to a certain low frequency known as its cutoff frequency. Thus it acts both as a high-pass filter and an impedance-matching device; above the cutoff frequency it presents an acoustic resistive impedance r_h to the loudspeaker unit connected to its throat.

An examination of Fig. 37 shows that the horn loudspeaker looks like a low-pass filter unit if the diaphragm edge compliance C_d is made very large (diaphragm edge clamping made very flexible) so that its reactance is negligible in the normal frequency range. By proper design of the unit, its cutoff frequency can be extended to 5000 or even 15000 c.p.s.; by proper design of the horn its cut-off frequency can be brought down to 30 c.p.s. or so. In that way the audio band can be covered with one or two such units, as will be shown.

THE EXPONENTIAL HORN.—Horns can be made in a variety of shapes: straight-sided or conical, and flaring in various ways. Of the latter type, the one that flares in an exponential fashion is the one most generally employed.

An exponential horn is one whose PERCENTAGE INCREASE IN AREA PER UNIT OF LENGTH IS CONSTANT. For example, if we start with a throat area A_t, and one foot toward the bell, the area has increased by 20% or to $1.2A_t$, (Fig. 38) and one foot from this area the area has increased by another 20%, or $1.2(1.2A_t) = 1.44A_t$, and so on, then the horn's area is said to increase as $\epsilon^{\beta l}$ or *exponentially* with distance, where β is the flare constant and l is the length from the starting point, namely the throat.

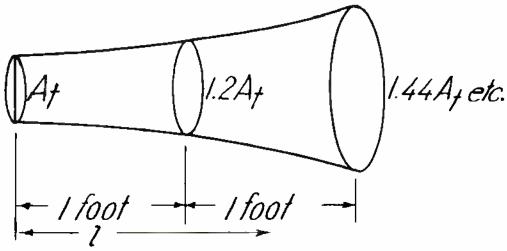


Fig. 38.—The percentage increase in cross-sectional area per unit length is constant in an exponential horn.

The larger β is, the more rapidly does the horn flare. The initial area A_t is determined by the design of the driver unit and how much damping this unit requires for flat frequency response.

The rate of flare and final bell area are determined by THE LOWEST FREQUENCY IT IS DESIRED TO TRANSMIT THROUGH THE HORN. The lower the frequency desired to transmit, the smaller β must be and hence the more gradually must the horn flare, and at the same time the larger must be its final bell or mouth area.

The reason for the latter requirement is that the mouth must act like a large source (similar to a large cone) rather than as a point source, in order that a condensation of the air, for example, coming out be not suddenly expanded as it enters the room. A sudden expansion would cause the condensation to become a rarefaction at this point, which would then travel BACK INTO THE HORN, AND CONSTITUTE A REFLECTION OF THE WAVE MOTION BACK INTO THE HORN. Similar considerations apply to a rarefaction leaving the mouth of the horn; a condensation would return into it.

The phenomenon is exactly analogous to that occurring in a transmission line if not matched properly at its far end; if the terminating impedance is less than the characteristic impedance of the line, the reflected wave is opposite in phase to the oncoming wave.

The upshot of all this is that a horn required to pass very low frequencies, such as 50 or perhaps 30 c.p.s., must flare very slowly, have a very large bell or mouth, and therefore be very LONG. This is one of the most serious objections to an exponential horn.

HORN FORMULAS.—The flare constant β is given by

$$\beta = \frac{4\pi f_c}{a} = \frac{4\pi f_c}{1130} = 0.01112 f_c \qquad (2)$$

where f_c is the cut-off frequency, and a is the velocity of sound = 1130 ft./sec. For example, suppose it is desired to design a horn that will transmit down to 30 c.p.s. Then the flare constant should be

$$\beta = 0.01112 \times 30 = 0.3336$$

The diameter of the mouth or bell (assumed circular in cross section) is

$$D = \frac{4}{\beta} = \frac{\lambda}{\pi} \qquad (3)$$

Waves emerging from a mouth of this diameter will not be appreciably reflected back into the horn; the loudspeaker unit will be matched in impedance to free space (or the room). For a 30-cycle cutoff frequency,

$$D = 4/.3336 = 12 \text{ feet}$$

This is truly a large diameter, and shows how large the horn mouth must be to transmit 30-cycle notes into the room without reflection.

The area at any point in the horn, distant l feet from the throat, is given by

$$A = A_t \epsilon^{\beta l} \qquad (4)$$

where A_t is the throat area (in any units desired, such as sq. cm., or sq. in.), and ϵ is the natural base of logarithms (= 2.7183....). If we require a final diameter D, the final area is

$$A = \pi D^2 \qquad (5)$$

For D = 12 feet = 144 inches,

$$A = \pi(144)^2 = 65,100 \text{ sq. in.}$$

If the horn is to have a mouth other than circular, its *area* should be the same as the circular one, namely, 65,100 sq. in. Suppose the initial or throat diameter (determined by the loudspeaker unit, is 1 inch. Then

$$A_t = \pi(1)^2 = 3.14 \text{ sq. in.}$$

Eq. (4) can be solved for l in terms of A, A_t, and β. Thus

$$l = \frac{2.3}{\beta} \log (A/A_t) \qquad (6)$$

Suppose A is taken as the final mouth area of 65,100 sq. in., and A_t = 3.14 sq. in. Then

$$l = \frac{2.3}{0.3336} \log (65,100/3.14)$$

$$= \frac{2.3}{.3336} \log 20,700$$

$$= \frac{2.3}{.3336} (4.3160) = 29.8 \text{ feet!}$$

Thus a horn practically 30 feet long

and with a final diameter of 12 feet is required to transmit 30-cycle tones from an initial throat diameter of one inch to free space or an auditorium (Fig. 39). This is truly a huge horn, and is required because such low frequencies as 30 c.p.s. are desired to be transmitted.

FOLDED HORNS.—In order to conserve useful space, the horn is usually folded in one way or another. Fig. 40 shows a rather obvious way to fold the horn; the total space is more nearly a cube and more practical a shape than that shown in Fig. 39.

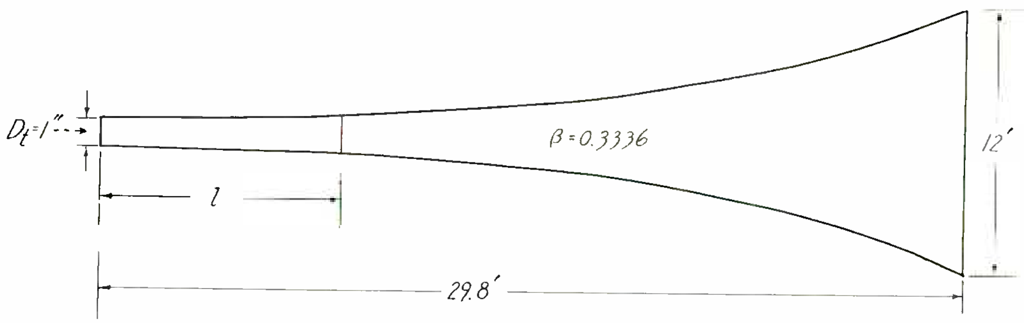


Fig. 39.—Design characteristics of a horn whose cut-off frequency is 30 c.p.s.

The horn area at any distance can be readily computed from Eq. (4). However, the following simple rule will short-cut much of the calculations otherwise required:

1. If the area doubles every 2 feet, the low-frequency cut-off = 32 c.p.s.

2. If the area doubles every 1 foot, the low-frequency cut-off = 64 c.p.s.

3. If the area doubles every 1/2 foot, the low-frequency cut-off = 128 c.p.s., and so on.

If we assume 32 c.p.s. is sufficiently close to 30 c.p.s., then the horn of Fig. 39 will increase from 3.14 sq. in. at the throat to 6.28 sq. in. two feet from the throat; 12.56 sq. in. at four feet from the throat, and so on.

Another form is that shown in Fig. 41, and is employed for the low-frequency or woofer unit in theatre and other high-power installations.

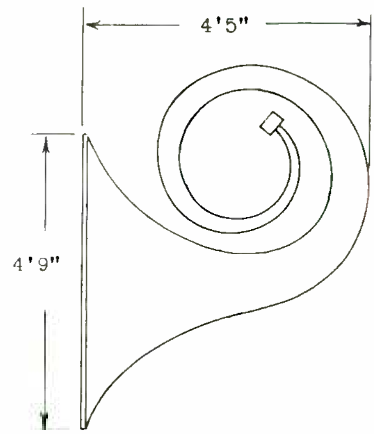
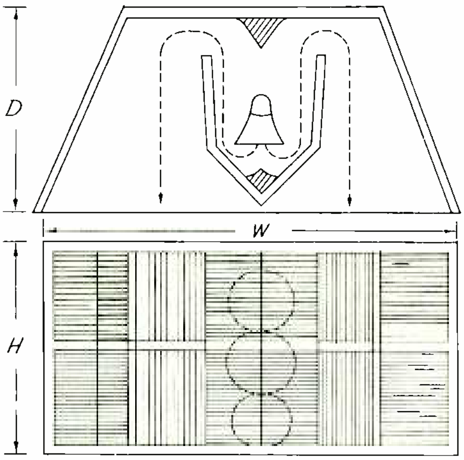


Fig. 40.—Coiling the narrow end of an exponential horn conserves space.

As is clear from the top view, the cone speaker unit is mounted facing the front, but is nevertheless enclosed by an angled partition that reflects the sound back to the rear partition. There it is redirected forward in two parallel paths, and emerges from the front or mouth of this reentrant horn (a horn folded back on itself.)



(Courtesy Jensen)

Fig. 41.—Folded horn or baffle for use with low-frequency cone loudspeaker unit.

The blocks shown in cross hatching in the top view can be omitted in low-frequency work. It will be observed that the length of the horn is either dotted-line path, and that in effect there are two horns in parallel. The cross-sectional area A at any distance l from the cone is given in terms of the initial area by Eq. (4).

As is suggested in the lower, front view by the circular dotted lines, three cones can be mounted, one above the other, and connected so as to all move in phase and therefore act essentially as one large piston. This means that the initial throat area A is very large, and no step-down transformer action is utilized in this particular horn design. The main feature of this design is that the overall length of the horn assembly is a distance D which is much less than the actual horn length, although the bell or mouth dimensions H and W are not reduced in this design.

A horn folded as shown suffers from one defect, and that is that the path length along the outside of the curve is considerably different from that along the inside of the curve, particularly when measured in wavelengths at the higher frequencies. This is brought out in Fig. 42, and illustrates the fact that sound proceeding along the shorter path may reach point A at the mouth in the form of a condensation, whereas sound reaching point B along the longer path may arrive there simultaneously as a rarefaction.

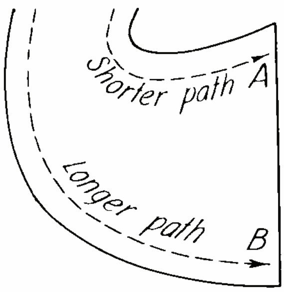


Fig. 42.—The difference in path length along an abrupt bend may lead to destructive interference effetts.

The result will be cross currents and cancellation of the two opposite pressures.

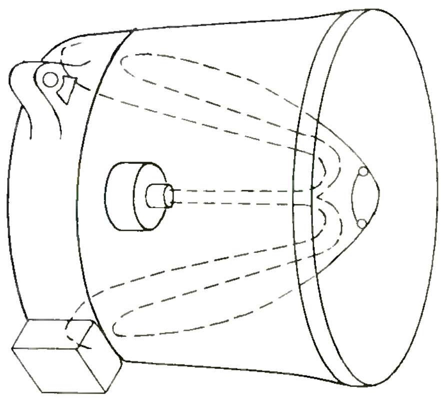
This effect is more pronounced, the sharper the bend; the larger the cross section at which it occurs, and the shorter the wavelength of the sound and hence higher the frequency. For that reason, horns that reverse themselves like a stocking half turned inside out, and called for that reason reentrant horns, are suitable mainly for the lower frequencies, where the wave length is large relative to the difference in path length. But this is precisely the region where a horn is required.

At the higher frequencies even a small cone can act as a direct radiator with fair efficiency, or a tweeter with a high-frequency unit and straight horn can be employed with very little additional room required. The latter unit will be discussed farther on.

It is interesting to note, however, that the reflection and resistance losses in a re-entrant horn cause about a 2db attenuation as compared to a straight horn or trumpet. A re-entrant horn, nevertheless, is much more efficient as a impedance-matching device for a loudspeaker unit than a cone operating in a flat or even reflex baffle, so that even though it may be 2db "down" on a straight horn, it is still very worth while using.

Fig. 43 shows another type of horn unit used ordinarily for high-level speech public address work. This particular loudspeaker is designed to be used aboard ship, and the driver unit is completely enclosed so as to assure troublefree watertight performance under adverse conditions of temperature and humidity. The horn is an aluminum casting suitably lacquered.

Although it is true that the horn has considerable curvature, it will be observed from the dotted line cross section that the most abrupt curvature is at the end of the first section, and that at the wider cross sections the curvature is more gentle. Furthermore, this unit is intended for speech rather than high-fidelity music, and as such functions very satisfactorily even if there should be some irregularities in the frequency response.

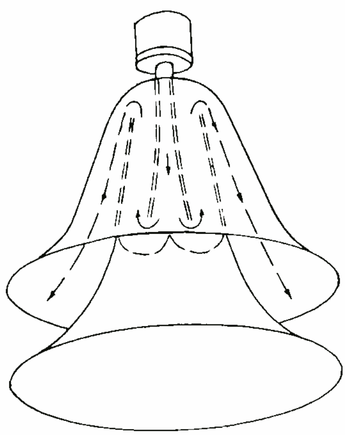


(Courtesy Racon Electric Co.)

Fig. 43. — Double Re-entrant Marine Loudspeaker.

Another arrangement is shown in Fig. 44. Here the sound is radiated in a relatively narrow beam (as measured in a vertical direction) over 360° azimuth (horizontal angle). At the same time the high loading of a horn on the loudspeaker unit is obtained, with consequent efficiency of operation.

KLIPSCH HORN. — A re-entrant type of horn developed by Paul W. Klipsch has an added important feature that it is capable of utilizing the walls of the room as extensions of its sides. In this way a cabinet of modest proportions is capable of radiating sounds down to as low as 30 c.p.s. with fair efficiency.



(Courtesy Racon Electric Co.)
Fig. 44. — Radial re-entrant trumpet loudspeaker transmits sound uniformly over a complete circumference of 360°.

Fig. 45 shows a cut-away oblique view of the new horn speaker arranged to take advantage of the high acoustical impedance of the corner of a room. The speaker is in a triangular enclosure or box, with a square opening in the front of the box acting as a throat to the horn portion. The latter consists of a vertical passage in front of the cone, with half of the sound energy traveling upward and the other half downward until the waves reach passageways at the top and bottom, whence they proceed toward each other to the rear.

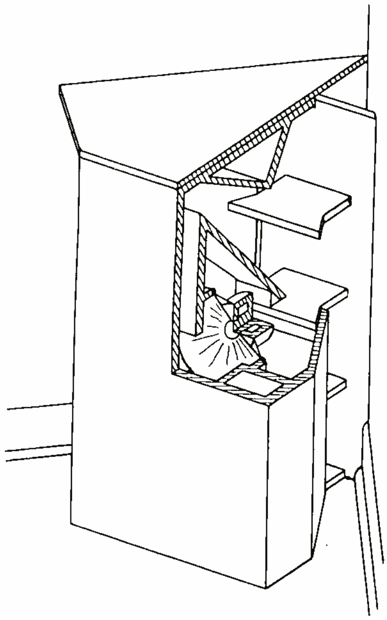


Fig. 45. — Oblique cut-away of the Klipsch horn.

The later path is more clearly shown in Fig. 46. At the rear the sound waves from the top and bottom passages meet at the center, only to split in a vertical plane and pass around the two sloping outsides. This path is best shown in the top view portrayed in Fig. 47.

The two sides come together but terminate before they would otherwise meet in the corner of the room. The sound wave can then pass out laterally and proceed along either passage formed by the wall and the adjacent side. At the front there are in effect two mouths, one on either side of the front panel.

The sound wave then proceeds into the room, guided by either wall and the floor, with the result that the resultant or effective mouth area is *four* times the actual mouth area. The latter is in this particular example 570 sq. inches and the horn length is approximately 40 inches, as compared to a mouth area of 4500 sq. inches for a horn in free space, or 2300 sq. inches if the horn is operated close to a floor or wall, with a length of 80 inches. Thus, a saving of about 75 per cent in volume is obtained.

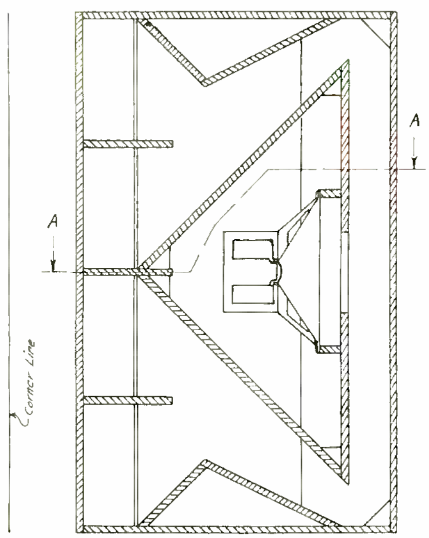


Fig. 46. — Sectional view from the side.

For a 10.5-inch diameter diaphragm weighing between 14 and 18 grams, a throat area of about 50 sq. inches is required at a frequency of 400 c.p.s. for good loading. At lower frequencies, however, this throat area is too small, and the step-up transformer effect is consequently excessive, that is, the loading becomes too great at the lower frequencies.

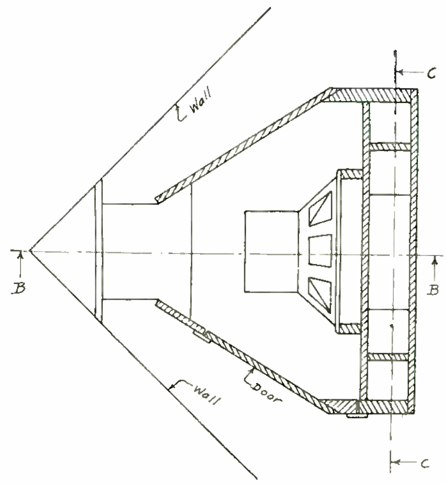


Fig. 47. — Sectional view from the top of the new horn.

To obviate this effect, the first part of the throat (the two portions in front of the cone) are conical rather than exponential in shape, and expand at a more rapid rate. This is illustrated in Fig. 48. The result is a "rubber throat", that presents a larger throat area in effect at lower frequencies. Thus, at 400 c.p.s., the effective throat area is 50 sq. inches, but at frequencies below 100 c.p.s. it is about 100 sq. inches.

This "rubber throat" has a taper such that its area doubles every 8 inches, which corresponds to a cut-off of 100 c.p.s. The remainder of the horn flares at such a rate that the area doubles every 16 inches, which corresponds to a cut-off of 42 c.p.s.

The multiple taper introduces an inertia effect or mechanical inductance, which is balanced by a compliance in the form of an air chamber of about 250 cubic inches between the diaphragm and the throat. The air chamber is shorter than 1/8 wavelength at 400 c.p.s.

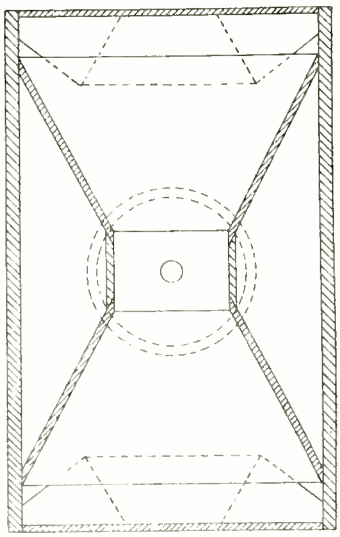


Fig. 48. — Sectional view from the front.

At low frequencies the throat does not act as a pure resistance, but has a certain amount of inertia effect or reactance. This is balanced by an air-chamber compliance in back of the cone. About 3000 to 3500 cubic inches is found satisfactory.

As a result of all these design factors, excellent efficiency and performance is had down to 40 c.p.s. for this particular design, and yet the cabinet is hardly any larger than that of an ordinary radio cabinet. The horn loading can produce an efficiency of perhaps 45% at 45 c.p.s., and 35% up to 200 c.p.s. or so. This is to be compared with 1 or 2% for a flat baffle.

HORN UNITS

ANALYSIS OF HORN UNIT ACTION. — The horn unit was described previously, and the equivalent electrical circuit was derived and presented in Fig. 34. This figure is repeated here in Fig. 49(A). The action of C_d is of importance at the lowest frequencies only, and produces in conjunction with the diaphragm and voice-coil mass M_d a resonance at some low-frequency which can be designed to be below the cut-off frequency of the connected horn.

Above this resonant frequency the circuit is essentially that shown in (B). This is recognized as a half-or L-section constant-K type of low-pass filter, terminated in a matching resistance R_h which is the reflected resistance of the actual throat resistance r_h. Its frequency response is portrayed in (C). Up to a so-called cut-off frequency f_c its response is relatively flat if R_h, C_a, and M_d are properly coordinated. Above this frequency the response drops off rapidly.

The cut-off frequency is given by

$$f_c = \frac{1}{\pi/M_d C_a} \qquad (7)$$

and for flat response

$$R_h = M_d/C_a \qquad (8)$$

Eq. (7) indicates that the smaller M_d or C_a are, the higher is f_c. This means that if the mass of the diaphragm is reduced, and the volume of the air chamber is reduced (thereby decreasing C_2), f_c can be increased. Unfortunately, the mass of the diaphragm cannot be reduced too much or, for a given area, it will be too flimsy, even when corrugated to stiffen and strengthen it. Furthermore, if the volume of the air chamber is reduced by decreasing its thickness, the diaphragm will strike the wall of the air chamber on loud low-frequency notes, so the compliance cannot be reduced too much by this means.

into the horn mouth, even though the thickness of the duralumin is only .002". The tangential striations or pleats act as a very flexible compliance, so that compliance C_d in Fig. 49(A) is very large and its reactance is very low. This means the series resonant frequency of the diaphragm can be below the low-frequency cut-off of the horn and hence of no consequence.

Finally, the voice coil consists of flat aluminum tape wound on edge as shown. This means that the only insulation is that required between turns; none is required on the edges. As a result the maximum amount of conductive material is

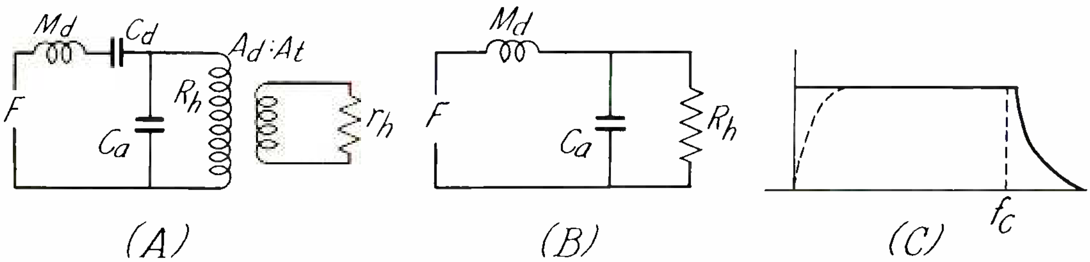


Fig. 49.—Horn loudspeaker unit: The actual equivalent circuit; the simplified equivalent, and the filter frequency response curve.

Nevertheless, by careful design, f_c can be raised to 5000 c.p.s., and even up to 12,000 or 15,000 c.p.s. for reduced power outputs and smaller air chamber volume. One factor is reduction in the mass M_d of the voice coil and diaphragm. Fig. 50 illustrates the design factors involved. Although this is an older Western Electric Type 595AW loudspeaker diaphragm, it is representative of modern design.

The reversed dome shape makes the center rigid and act like a piston in pumping air back and forth

available for a given amount of mass, or conversely, for a given force, a minimum of voice-coil mass is required.

This means that M_d can be made less by such construction. A further advantage is that the heat generated in the voice coil by the audio currents can be transferred to the iron sides of the air gap more readily than if the flow has to be partly through insulation, which is generally a poor heat as well as electrical conductor. For high-power speakers, particularly if the average power is high, (as contrasted to the peak power) burn out of the voice coil is a serious limitation to the loudspeaker performance.

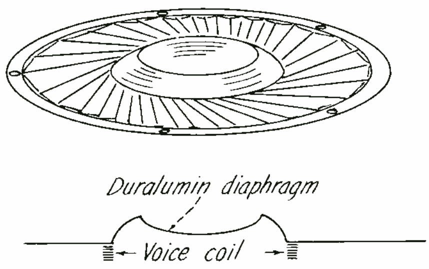


Fig. 50.—Duralumin diaphragm and voice coil.

HIGH-FREQUENCY CHANNELS.—One factor in horn units is that at the higher audio frequencies the path length from the edges of the diaphragm to the horn throat are longer than that from the center of the diaphragm by an amount comparable to the wavelength of the sound. Thus, in Fig. 51(A), path length AC may exceed path length BC by half a wavelength at say, 5000 c.p.s., since the wavelength at this frequency is $1130 \times 12/5000 = 2.71$ inches, and $\lambda/2 = 1.36''$.

In such a case, the waves will meet 180° out of phase and tend to cancel, and the high-frequency response will be greatly attenuated, particularly at those particular wavelengths. Of course, at 10,000 c.p.s., if the cutoff frequency extended to this point, the waves would meet in phase once more, and no attenuation would occur.

To obviate attenuation anywhere in the range of transmission, the path lengths must be equalized, and Fig. 51(B) shows how this is done. The passageway from the air chamber to the horn throat is not at the center, as at (A), but is an *annular* connection about halfway out along the radius of the diaphragm. As a result of the high-frequency channels produced, path lengths such as CED and AB are about equal, and do not exceed the shortest path length ED by as much as half a wavelength.

The conical plug in the center is supported from the walls of the

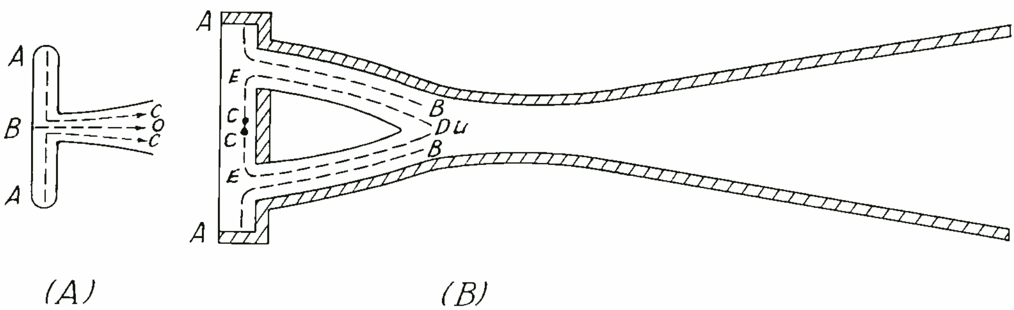


Fig. 51.—Method of equalizing path lengths from various parts of the diaphragm to the horn throat.

horn by thin webs that do not appreciably obstruct the passage of the sound waves. In the case of the Western Electric 595AW loudspeaker unit, the construction is slightly modified, as indicated in Fig. 52, so as to conform to the shape of the stiffened diaphragm, but the principle is exactly the same.

HIGH-FREQUENCY UNITS.—High-frequency units or "tweeters" are built on this principle. Furthermore, there is no attempt made to have such a unit cover a range below 250 c.p.s., or whatever the so-called "cross-over frequency" between woofer and tweeter is.

Restriction of the tweeter's range to the high-frequency end means that the amplitude of excursion of the diaphragm can be relatively small even for several watts of power output, and this in turn means that a shallow air chamber can be used. Also a smaller diaphragm is possible because of the more limited power requirements at the higher frequencies. The result is a smaller mass M_d and a smaller compliance C_a; from Eq. (7) it is seen that f_c, the cutoff frequency is correspondingly raised. Thus f_c can be raised to 12,000 and even 15,000 c.p.s.

One difficulty experienced in the case of a tweeter, is the narrow beam effect obtained at the higher frequencies. At low frequencies the sound tends to spread out in a wide beam, but at high frequencies the reverse is true. As a result, a person located on the axis of the speakers picks up an excess of the high frequencies (particularly the very high frequency components of noise or hiss), whereas a person located off the axis picks up an excess of the low-frequencies.

In the RCA LC-1A duo-cone loudspeaker, this is minimized by having a small cone for the tweeter, which acts as a point source and thereby radiates the sound through a large space angle, as well as by using shallow cones, all parts of which are within a half wavelengths distance to the front of the baffle.

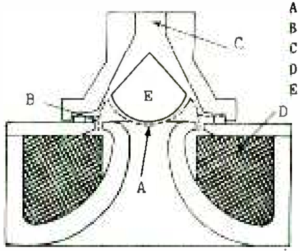


Fig. 52.—High-frequency channels produced by the horn plug in the W. E. 595AW loudspeaker unit.

In a horn type loudspeaker, however, the mouth or bell of the horn tends to act as a large rather than as a point source, and hence tends to radiate in a narrow beam. It is this factor that makes the horn so much more efficient a radiator than a cone loudspeaker.

Fortunately, this directional effect can be minimized to some extent because a tweeter does not radiate down to such low-frequencies, and hence can use an exponential horn of larger flare constant β. The more rapid flare of the horn resulting from this gives a beam of sound of wider spread.

However, the spread from a single horn is not sufficient. The Bell Laboratories have therefore developed a cellular horn structure which in principle looks like Fig. 53 (A), and in practice is shaped as in Fig. 53(B).

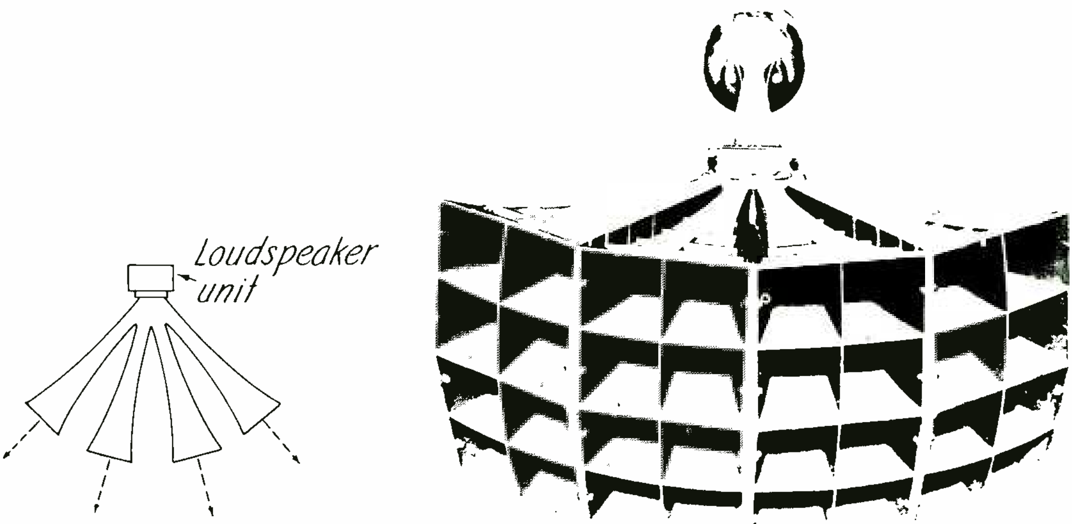
Each horn may have a narrow beam, but since each beam is aimed in a different direction, as indicated in (A), the total effect is as if a radiator were radiating sound through a large solid angle. As shown in (B), the individual horn mouths touch each other, so that some overlapping of the beams occurs, but the resultant beam has a fairly uniform sound intensity across its cross section, and is of the desired width.

The construction is such that the horn looks like a honeycomb or cellular structure when viewed from the front. The sides can be of metal, but should be of sufficient thickness, or else be sufficiently damped by some asphalt or similar material so as not to vibrate appreciably when the sound waves pass through it.

The throat can in itself be split up into one or more branches so as to accomodate more than one unit. The advantage of using two or more units instead of a single larger one in order to handle more power is that the cut-off frequency of the plurality of units is just as high as that of each individual unit, whereas one single large unit would have a lower cutoff frequency.

LOUDSPEAKER CIRCUITS

CROSS-OVER NETWORKS.—When separate loudspeakers are employed to



(A) (B)

Fig. 53.—Use of multiple horn construction to produce a wider beam at the higher audio frequencies.

handle the high- and low-frequencies ends of the audio range, such as woofers and tweeters, means must be had to divide the frequency bands properly between the units. Such means are known as cross-over networks.

Some elementary examples of this were mentioned previously, such as the double-voice-coil loudspeaker, in which the low-frequency voice coil was shunted by a capacitor. In the case of the duo-cone loudspeaker, the inherent reactance of the low-frequency voice coil tended to exclude the higher frequencies which, however, readily passed through the capacitor in series with the high-frequency voice coil. At the same time, the reactance of this capacitor at the lower frequencies prevented these components from flowing through the high-frequency voice coil.

Such an elementary cross-over network is broad in its action. As indicated in Fig. 54 by the solid-line curves, the audio energy fed into the woofer and into the tweeter each tapers off gradually in the frequency range. At the so-called cross-over frequency f_{co}, each response is down 3 db, or to one-half the power, so that the sum of the two loudspeaker outputs is 100%, the same as in those parts of the spectrum where either is essentially functioning by itself.

The reason such gradual attenuation is feasible is that the speaker voice coils or cones are coaxial (on a common axis) and hence located practically at the same point. In this case, as mentioned previously, no "array" effect occurs, with resultant nulls in certain directions, if the speakers overlap in their radiation, and hence the attenuation of the cross-over network can be gradual.

Where, however, separate woofer and tweeter combinations are employed, in which the units are not on a common axis, nulls in the response in certain directions can be expected. To minimize this, the frequency overlap between the two speakers should be made as small as possible, and this means rapid attenuation of either speaker's network. This is indicated by the dotted lines in Fig. 54.

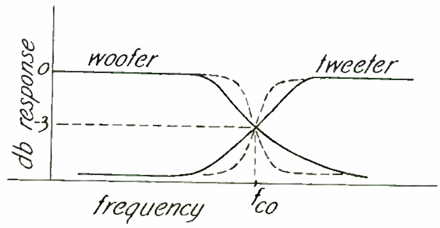


Fig. 54.—**Narrow and broad cross-over network frequency characteristics.**

The attenuation should be at least 12db one octave from the cross-over frequency; i.e., at twice the cross-over frequency. The attenuation should not exceed 18db per octave because the filter arrangement becomes more complicated and costly and tends to have excessive losses in the transmission range.*

Actually there are two types of networks: one based on filter theory and the other on constant resistance networks. The former is

*See, for example, "Loudspeaker Dividing Networks," by John K. Hilliard, Electronics, January 1941.

somewhat more flexible in application, and will be considered first. It, in turn can be designed in a parallel or in a series form, as is illustrated in Fig. 55.

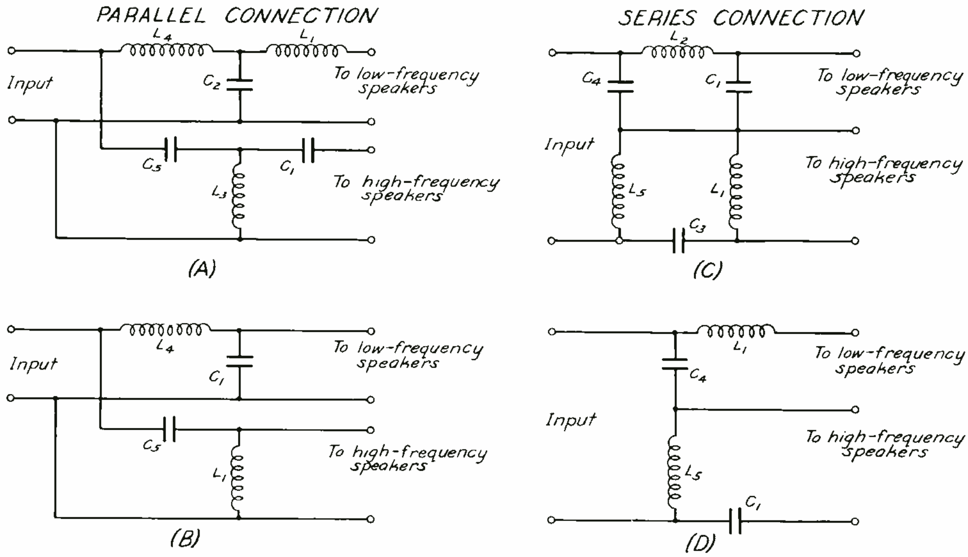


Fig. 55.—Filter sections used in parallel and series connection for speaker divider networks. (Top), full-section filter; (bottom), half-section filter.

In the parallel form, (A) and (B) the inputs of the two filters are connected in parallel, and the outputs connect independently to the low- and high-frequency speakers. In the series connection, (C) and (D), both inputs and outputs are in series. The operation is the same in either case.

In (A) filter elements of the Tee type are employed; and in (C), of the π type. In either case two full or complete filter sections are employed. The top section in either case is of the low-pass type, and "cuts off" at the cross-over frequency; i.e., it transmits flat nearly up to the cut-off frequency, and then begins to attenuate, as is indicated by the "woofer" curve in Fig. 54. This section therefore connects to the woofer or low-frequency speaker.

The lower section in either case is of the high-pass filter type; it does not begin to transmit until just a little below the crossover frequency (which is its cutoff frequency. Hence the tweeter or high-frequency speaker is connected to its output terminals. At the crossover frequency, each network attenuates by 3db, so that the power output of either speaker is 50% of its output in the pass band. This makes the output of *both* speakers 100% at the cross-over frequency.

The attenuation for a full section is 18db per octave. Where only 12 db per octave is desired, half instead of full sections can be used. These are shown in (B) and (D).

The values for the various circuit elements can be obtained from the following formulas:

Let R_o be the speaker voice-coil impedance; and $\omega_c/2\pi$ the cross-over frequency. This, as just mentioned, is also the common cut-off frequency of the low-pass and high-pass filter elements. Then

Then from Eq. (9),

$$L_1 = \frac{10}{2\pi \times 800} = 0.001991 \text{ henries}$$
$$= 1.991 \text{ millihenries}$$

$$L_4 = (1+0.6)\frac{10}{2\pi \times 800} = 0.00319 \text{ henries}$$
$$= 3.19 \text{ millihenries}$$

$$C_1 = \frac{1}{2\pi\ 800 \times 10} = 19.91 \times 10^{-6} \text{ farads}$$
$$= 19.91\ \mu\text{f.}$$

$$C_5 = \left(\frac{1}{1+0.6}\right)\frac{1}{2\pi \times 800 \times 10}$$

$$= 12.44 \times 10^{-6} \text{ farads} = 12.44\ \mu\text{f.}$$

$$L_1 = \frac{R_o}{\omega_c} \text{ henries} \qquad C_1 = \frac{1}{\omega_c R_o} \text{ farads}$$

$$L_2 = \frac{2R_o}{\omega_c} \text{ henries} \qquad C_2 = \frac{2}{\omega_c R_o} \text{ farads}$$

$$L_3 = \frac{R_o}{2\,\omega_c} \text{ henries} \qquad C_3 = \frac{1}{2\,\omega_c R_o} \text{ farads}$$

$$L_4 = (1+m)\frac{R_o}{\omega_c} \text{ henries} \qquad C_4 = (1+m)\frac{1}{\omega_c R_o} \text{ farads}$$

$$L_5 = \frac{1}{1+m}\frac{R_o}{\omega_c} \text{ henries} \qquad C_5 = \frac{1}{1+m}\frac{1}{\omega_c R_o} \text{ farads}$$

where m is a filter design factor usually taken as equal to 0.6.

As an example of the use of these formulas, suppose it is desired to cross over at a frequency of 800 c.p.s., and the woofer and tweeter voice coil impedances are both 10 ohms. Furthermore, a 12-db-per-octave attenuation is sufficient, and the parallel type network is desired. This means that the network shown in Fig. 55(B) will be satisfactory.

In particular, it is to be noted that the resistance of either coil should be low compared to $R_o = 10$ ohms. For about 0.5 db I^2R loss in either coil, the resistance should be 10% of R_o, or 1 ohm.

The other type of divider network is based on a rather curious network condition illustrated in Fig. 56. In either (A) or (B), if $R_o = \sqrt{L/C}$, then the input impedance is a pure resistance, also of magnitude R_o. In other words, either circuit is in a sense resonant at all frequencies!

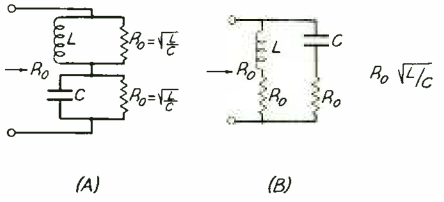


Fig. 56.—Two forms of constant resistance networks.

This property is utilized by considering the woofer and tweeter voice coil as each of resistance R_o, and connecting them in series with the L's and C's. The cross-over frequency is then chosen as desired. The actual circuit is shown in Fig. 57.

Network (A) here corresponds to (B) of Fig. 57. Either one provides an attenuation of 6db per octave. For 12db per octave, networks (C) or (D) can be used.

The formulas are as follows:

$$L_o = \frac{R_o}{\omega_c} \qquad C_o = 1/\omega_c R_o$$

$$L_1 = L_o/\sqrt{2} \qquad C_1 = \sqrt{2}\, C_o \qquad (10)$$

$$L_2 = \sqrt{2}\, L_o \qquad C_2 = C_o/\sqrt{2}$$

If R_o is again chosen as 10 ohms, and $\omega_c/2\pi$ as 800 c.p.s., then, for network (D), we must first solve for L_o and C_o from Eq. (10). Thus:

$$L_o = 10/2\pi \times 800 = 1.991 \text{ mh.}$$

and

$$L_2 = \sqrt{2} \times 1.991 = 2.82 \text{ mh.}$$

$$C_o = 1/2\pi\, 800 \times 10 = 19.91\ \mu\text{f.}$$

and

$$C_2 = 19.91/\sqrt{2} = 1.408\ \mu\text{f.}$$

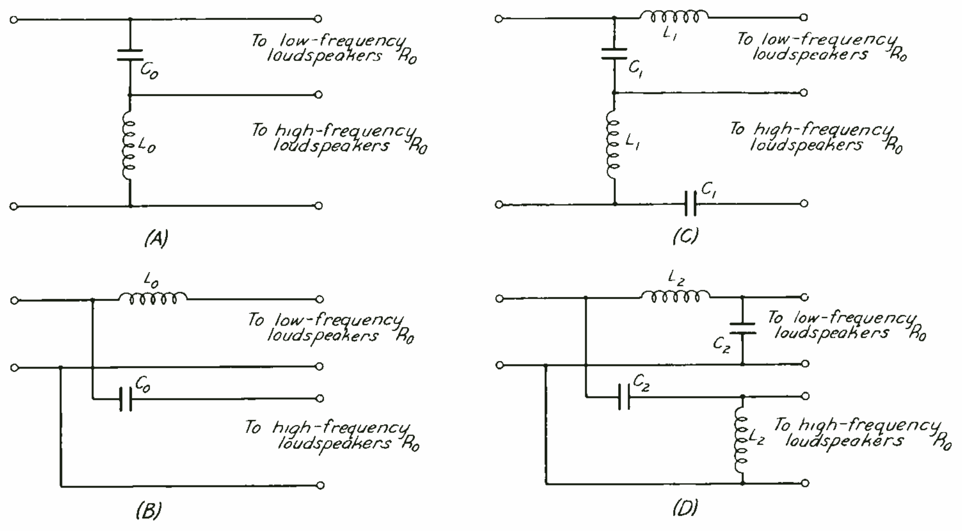


Fig. 57.—Constant-resistance speaker divider networks. (A) and (B) have 6db attenuation per octave; (C) and (D) have 12db attenuation per octave.

The advantage of the constant-resistance networks over the filter type networks is that they present a more nearly constant resistance to the amplifier feeding them. Networks (A) and (B) are seldom used because their attenuation is ordinarily too low.

POWER-HANDLING REQUIREMENTS.— One of the questions that arises in estimating the size of public address system required is the amount of audio power needed for adequate sound level in the room or auditorium. This of course varies not only with the size of the enclosure, but its function.

For example, a motion picture theatre requires a minimum intensity of 80db, where 0db = .0002dyne/sq. cm. This represents, in other words, a sound pressure of $.0002 \times 10^8 = 20{,}000$ dynes/sq. cm. in the theatre. But if the sound system is also to reinforce an orchestra in a large auditorium, then a level of 100db. will be required, or 10 times as great an intensity, and 100 times as much power.

On the other hand, a radio in a hotel need not be as loud as the sound system in a motion picture theatre, and an intensity of 70db is adequate. The value of 80 db is sufficient for the average home as well as being the minimum intensity for a sound motion picture theatre.

In the case of noisy places, like ball parks and railroad stations, an intensity of 20 to 40db above the general noise level is advisable, and the noise level should be preferably measured by a noise meter. However, a noise level of 60-70db can be taken for a noisy place, and the peak intensity (noise plus audio system's output) should not exceed 100db or so for comfort.

The *accustical* power in watts required to produce a sound *intensity* or pressure in dynes/sq. cm. corresponding to 80db depends upon the volume of the enclosure. Fig. 58, taken from Olson's book "Elements of Acoustical Engineering," shows the amount of acoustical power required to produce an intensity of 80db in auditoriums of various volumes. For example, if the enclosure has the dimensions of 12.5' × 20' × 40' or a volume of V = 12.5 × 20 × 40 = 10,000 cu. ft., then, from Fig. 58, acoustic power of about 0.013 watt is required for a sound intensity of 80db.

This is very small amount of acoustic power, and indicates how marvelously sensitive the ear is.

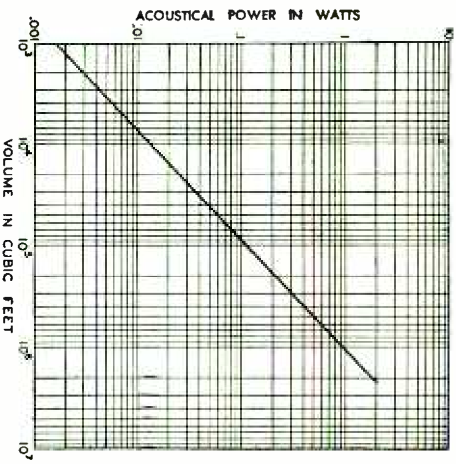


Fig. 58.—Acoustic power required to produce an intensity level of 80db as a function of the volume of the auditorium.

Suppose that a peak intensity of 90db instead of 80db is desired. Then the acoustic power will be 10db higher, or 10 times as great; that is, 0.13 watt.

How many watts ELECTRICAL INPUT to the loudspeaker unit (or units) is required to furnish 0.13 watt ACOUSTIC output? This, of course, depends upon the efficiency of the loudspeaker system. A commercial horn loudspeaker has an efficiency of from 25 to 50 per cent; a high-grade direct-radiator electrodynamic unit, such as a cone in a baffle, may have an efficiency of 5 per cent; and a small balanced-armature type of magnetic speaker may have an efficiency of but 1 per cent. Hence the electrical input power may vary over a wide range.

Suppose a horn type speaker of 25 per cent efficiency were used in the above auditorium. The electrical power input would be

$$W_e = 0.13/.25 = 0.52 \text{ watt}$$

which is very small. On the other hand, if a direct radiator electrodynamic speaker of 5% efficiency were used, five times as much electrical power would be required, or $5 \times 0.52 = 2.6$ watts.

Finally, if balanced-armature magnetic speakers were used, an input of $0.13/.01 = 13$ watts would be required, and several units in parallel would be required to handle this amount of power because of the limitations of this type of speaker.

Nevertheless, the power requirements are well within the rating of an ordinary audio output stage, such as one employing two 2A3 tubes in push-pull (which has a peak output of 15 watts). This means that of the three speakers, probably the electrodynamic unit would be preferred because of its better combination of efficiency, small space requirements, and first cost.

When we come to large installations however, the horn type speaker is preferred because of its superior efficiency. Consider for example an auditorium of 50' × 100' × 20' high, or $50 \times 100 \times 20 = 100{,}000$ cu. ft. volume. From Fig. 58, it is found to require an acoustic power (for 80db intensity) of 0.105 watt. Assume a peak intensity of 100 db instead of 80 db; the acoustic power will be 100 times as great, or $0.105 \times 100 = 10.5$ watts.

A bank of direct-radiator speakers of 5% efficiency would require an electrical input of $10.5/.05 = 210$ watts, which is a considerable amount of audio power. It would require special large tubes beyond those used in receivers and similar applications, or else it would require a number of ordinary audio power output stages in parallel.

On the other hand, if a horn type speaker were used, with an efficiency of 25 per cent (which is easily attained), the amplifier output would have to be but $10.5/.25 = 42$ watts. This is well within the rating of an amplifier using a pair of 6L6 tubes in push pull, as in a McIntosh amplifier. The difference in amplifier cost would be considerable, not only because only one-fifth as much power is required, but because the requirements have been scaled down to fit an amplifier using tubes and components readily available. Moreover, in such a large auditorium, the bulk of a horn type loudspeaker is ordinarily of no great consequence.

This all indicates the sphere of usefulness of the horn and direct radiator type of speaker. There are, of course, other advantages of a horn type speaker, such as the ability to direct or beam the sound in the direction desired, rather than to waste it in directions such as toward the ceiling, etc. This and other matters pertaining to speaker placement will be discussed in a subsequent section of this assignment.

OUTPUT TRANSFORMER TAPS.—In installations where different rooms are to be fed by individual banks of loudspeakers, it becomes necessary to divide the total audio power in accordance with the requirements of the various rooms, as determined in the preceding section. At the same time, the speakers must all reflect to the power output tubes the correct impedance so as to enable them to operate under optimum conditions.

in (B). This arrangement is perhaps more expensive. Finally, in (C) is shown a method whereby step-down to a moderately high impedance line (usually 500 ohms) occurs in the output transformer, and then at the end of the run further stepdown to the lower impedances of the various loudspeakers is obtained by means of a multi-tap auto transformer.

This method has been used by Western Electric, for example, in their sound motion picture systems in order to enable the sound intensity of each horn in the theatre to be adjusted so as to obtain equal loudness in all parts of the theatre. Also, in installations having long runs, as perhaps in an amusement park, the moderately high impedance line to the auto transformer prevents excessive voltage drop in the long conductor run. This is very much the same as the use of step-up and step-down transformers in the

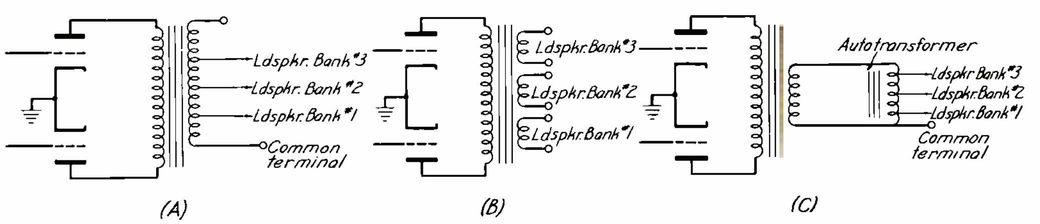


Fig. 59.—Method of feeding a number of loudspeaker banks from one amplifier.

This is accomplished by connecting each speaker bank to the proper secondary tap on the output transformer (assuming the taps are available). This is indicated in Fig. 59(A). Alternatively, individual secondaries can be used, as shown

transmission of 60-cycle power by feeders, and then further step-down by line transformers to the 110-220-volt service to each customer.

The secondary windings or taps on the output transformer are usually designated by so many ohms impedance.

For example, one tap or secondary winding may be marked 8 ohms impedance; another, 15 ohms impedance; and so on. THIS MEANS THAT AN 8-OHM VOICE COIL OR 15-OHM VOICE COIL IS TO BE CONNECTED TO THE TAP, AS THE CASE MAY BE. IT DOES NOT MEAN THAT THE APPARENT SOURCE IMPEDANCE, LOOKING INTO THAT TAP, IS 8 OHMS OR 15 OHMS. The apparent source impedance should be much LOWER, in order that the loudspeaker, if of the direct-radiator type, be adequately damped at its low-frequency resonance. This can be accomplished either by using low impedance tubes, such as triodes, or by employing inverse voltage feedback, as in the case of pentodes.

To determine the proper taps to which the various loudspeaker bands should be connected, the following procedure should be employed. This procedure is best illustrated by an example. Suppose that a hotel requires a loudspeaker bank in a ball room of dimensions 20' × 50' × 100'; another loudspeaker bank in the bar, of dimensions 12.5' × 20' × 50'; a third in the cocktail lounge, of dimensions 10' × 20' × 50'; and finally loudspeakers in 20 rooms, each of dimensions 8' × 10' × 15'.

What power is required in each room, what is the total power, and how should the loudspeakers be connected to the output transformer? First as to the power. From Fig. 58 the acoustic power for each room can be found:

	Dimensions in ft.	Volume cu. ft.	Acoustic Power watts	Desired Intensity db.	Peak Acoustic Power watts	Type Loudspeaker	Eff. %	Electrical Power watts	Speaker Impedance ohms
Ball Room	20×50××100	100,000	0.12	90	1.2	Horn	30	4.0	10
Bar	12.5×20×50	12,500	0.015	90	0.15	Direct Radiator	5	3.0	15
Cocktail Lounge	10×20×50	10,000	0.013	80	0.013	Direct Radiator	5	0.26	15
Hotel Rooms	8×10×15	1,200	0.002	70	0.0002	Direct Radiator	2	0.01	7.5

Since there are 20 rooms, the total power for the speakers will be 20 × .01 = 0.2 watt. Assume for the moment that these are connected in series, so that the total impedance will be 20 × 7.5 = 150 ohms.

The total power for the entire installation will be 4 + 3 + .26 + 0.2 = 7.46 watts. A 10-watt amplifier would appear to be adequate; its volume control could be turned down to furnish the desired output of 7.46 watts.

The secondary tap impedances required can be found from the following formula:

Let P_t = total power
P_1 = power required for a given loudspeaker bank
R_1 = impedance of this bank, and
R_t = impedance designation of tap to which the bank is connected.

Then

$$R_t = \frac{P_1}{P_t} R_1 \qquad (11)$$

Thus, the tap for the ball room speaker is

$$R_t = \frac{4}{7.46} \times 10 = 5.36 \text{ ohms}$$

A 5-ohm tap would probably be close enough.

In the case of the bar loudspeaker,

$$R_t = \frac{3}{7.46} \times 15 \quad 6.04 \text{ ohms}$$

The cocktail lounge loudspeaker requires a tap

$$R_t = \frac{.26}{7.46} \times 15 = 0.523 \text{ ohm}$$

This is a rather low tap, and may require a special matching transformer. Finally, the loudspeakers for the hotel rooms, if connected all in series, would require a tap

$$R_t = \frac{.2}{7.46} \times 150 = 4.02 \text{ ohms}$$

It will be noted that had the speakers been connected in parallel instead of series, the total impedance would have been 7.5 ÷ 20 = 0.375 ohms, and the required tap would have been

$$R_t = \frac{.2}{7.46} \times 0.375 = 0.001005 \text{ ohm}$$

which would be far too low for any practical output transformer.

On the other hand, it is somewhat awkward, circuit-wise, to have all the speakers in series. In particular, if one speaker becomes open-circuited, all the other

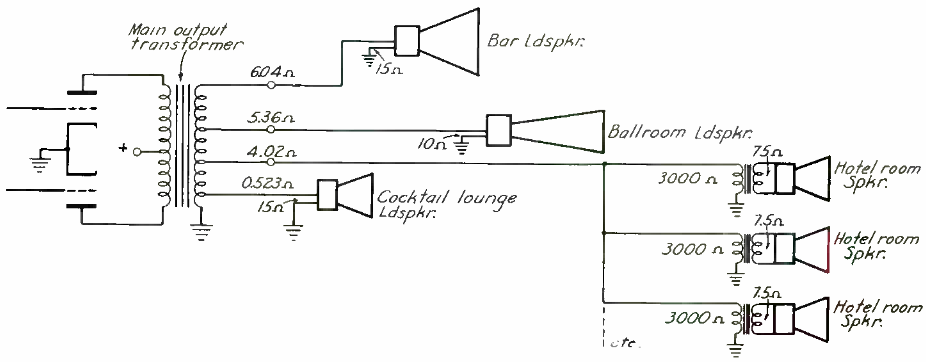


Fig. 60.—Multiple speaker installation showing taps to be used and speaker connections.

speakers will be cut off, too. Hence it might be preferable to employ a separate individual output transformer for each loudspeaker. These might transform the 7.5-ohm impedance of the voice coil to say 3000 ohms. Then the primaries could be all connected in parallel to give a joint impedance of 3000/20 = 150 ohms, and the group then connected to a 4-ohm tap on the main output transformer.

This is illustrated in Fig. 60. The twenty hotel room speakers are shown connected to the 4.02-ohm tap through individual 3000 to 7.5 ohm step-down transformers. If a 0.523-ohm tap is not available for the Cocktail Lounge Speaker, a step-down transformer of say 125 to 15 ohms could be used at the speaker terminals, similar to those employed for the hotel room speakers.

In this case the Cocktail Lounge speaker would present an impedance 125/15 = 8 1/3 times as great, and could therefore be connected to a tap 0.523 × 8 1/3 = 4.36 ohms. Presumably the 4.02-ohm tap could then also supply this speaker, with but little reduction in power and small mismatch in the impedance presented to the output tubes. However, each transformer produces some losses; this may vary from about 20% or 1 db for a small transformer to only 5 or 10% for a larger transformer.

It is to be stressed that the above design formula Eq. (11) divides the power in the proper proportions to the various speakers, and at the same time permits the totality of speakers to present the desired impedance to the output tubes. Thus, each speaker in Fig. 60 is connected to a tap of lower impedance designation than the impedance of the speaker. For example, the Bar loudspeaker has an impedance of 15 ohms, yet it is connected to a 6.04-ohm tap.

If 6.04 ohms were connected to this tap, the proper impedance would be presented to the output tubes; for a pair of 2A3 tubes in push pull the step-up ratio to the primary would be 3000 ohms. If, instead, 15 ohms is connected to this tap then 15/6.04 × 3000 = 7,450 ohms is presented to the tubes. However, the 10-ohm Ball Room speaker, when connected to the 5.36-ohm tap, presents to the tubes an impedance of 10/5.36 × 3000 = 5,600 ohms, and this is in PARALLEL with the 7450-ohm impedance.

The Cocktail Lounge speaker and the hotel room bank also present individually to the tubes impedances in excess of 3000 ohms, and in parallel with the other two impedances. If now all these reflected impedances are combined in parallel, it will be found that the joint impedance reduces to 3000 ohms once again, and this is exactly what the tubes should face for maximum so-called undistorted power output. At the same time, each loudspeaker receives its proper share of the 7.46 total power output.

PRACTICAL INSTALLATIONS

LARGE AUDITORIUMS.—Large auditoriums, such as sound motion picture theatres, may require a multiplicity of horn type loudspeakers to cover adequately all the seats in the house. To determine the number and placement, it is necessary also to know the directional patterns of the speakers; i.e.—the angular flare of the beam in both dimensions.

In Fig. 61 are shown two speakers, and the beams of sound that they radiate. These beams, it will be observed, are arranged to overlap so as to contribute to the sound intensity of seats along the center line of the theatre. For example, in the plan view seats AA, at the front of the theatre and on the axis of either horn, will apparently receive more sound than rear seats EE, simply because the latter are relatively much farther away from the speakers.

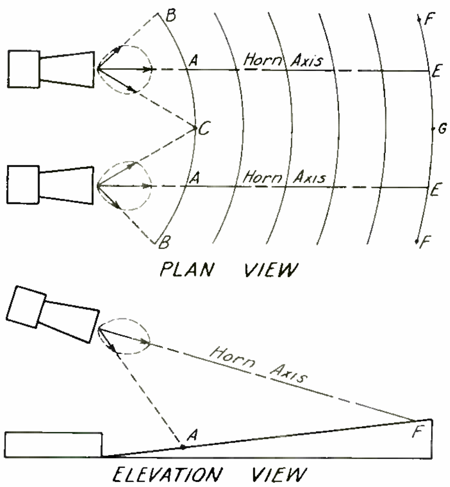


Fig. 61.—Two-speaker layout, showing how the directional patterns can produce a fairly uniform sound intensity throughout the theatre.

But a glance at the elevation view will show that the polar diagrams of sound intensity (drawn in dotted lines) are such that the vector intensity toward A is appreciably less than that toward E, simply because the horn axis in this view is shown directed toward seat E. This compensates for the difference in distance, and can make the sound intensity at the front and rear more nearly equal.

Refer once more to the plan view. Seats BB receive very nearly as much sound energy as seats AA, simply because of the flattened shape of the polar diagram in this plane. If the intensity, for example, is but 71% of that at A, the power is $(.71)^2 = 1/2$ or 3db less than that at A, and this is a small variation. Furthermore, as regards end seats BB and FF, the direction of the horn axes, as indicated in the elevation view, tends to compensate for the difference in distances, just as in the case of seats AA and EE.

Next note a front center seat such as C. It receives contributions from both horns. Assuming that the two voice coils receive their electrical energy in phase, the acoustic intensity at C is the arithmetic sum of the contributions from the two horns. If the intensity is less in this direction, say 71%, then the total intensity is $2 \times .71 = 1.41$, and the power is $(1.41)^2 = 2$ times that at A, or 3db greater. This is not of any great consequence in view of the equalizing effect of the reverberation inherent in the theatre. Finally, owing to the orientation of the horn axes, center rear seat G can receive as much sound as C, just as in the case of the other front and rear seats.

Another arrangement is shown in Fig. 62. Here an upper and lower loudspeaker are employed; two or more of course can be employed side by side in either position. The arrangement shown can give perhaps more even distribution than that shown in

Fig. 61 because the electrical inputs to the two speakers can be adjusted so as to equalize the sound outputs, by the use of different taps on the output transformer. It should be evident that a variety of adjustments and compensations are thus possible.

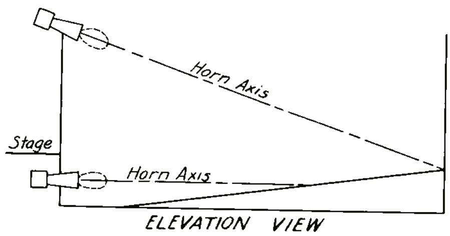


Fig. 62.—By using an upper and lower speaker, and proportioning their electrical inputs, very even distribution can be obtained.

MULTIPLE SPEAKER ARRANGEMENT.—It is not always important that the source of sound be from a single area such as the screen of a motion picture theatre in order to obtain a maximum of realism. For example, in a stadium of horseshoe shape, see Fig. 63, it is rather difficult to cover uniformly the wide angle of seats involved with a single speaker or even speaker bank.

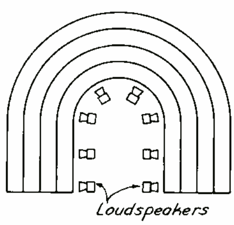


Fig. 63.—Use of multiple speakers in a stadium.

Instead, a series of smaller speakers, each covering a limited angle and group of seats, is preferable. There is no need to create the illusion of the sound coming from a stage or single source, and coverage is more complete and uniform by the use of a number of speakers.

Of course, if there is also a loud single source, some illusion of the sound originating at that point is obtained even though the additional speakers are employed. In any event, the use of multiple speakers is justified in the case of outdoor motion picture theatres where a speaker is placed beneath a grill over which each automobile is parked, since the car windows can then be kept closed in the winter time.

Any attempt to use a single loudspeaker bank may require not only considerable power handling capacity on its part, but also produce loud and disturbing sounds in the neighborhood. It must not be overlooked that at double the distance to the last seats of the theatre, the sound level from a stage speaker decreases only 6db (inverse square law). Similar considerations hold in the case of outdoor theatres having regular stage presentations; multiple speakers (see Fig. 64) provide good coverage of the audience without excessive sound beyond the confines of the theatre.

LOUDSPEAKERS FOR USE IN NOISY SURROUNDINGS.—When it is desired to use a public address or paging system in very noisy surroundings, where the noise even begins to approach the pain level, then special means must be taken. In the first place, it has been found that although most of the energy of speech, and even music, is in the region from 250 to perhaps 500 c.p.s., nevertheless the intelligibility of speech resides mainly in the higher frequencies that constitute the consonant rather than the vowel sounds.

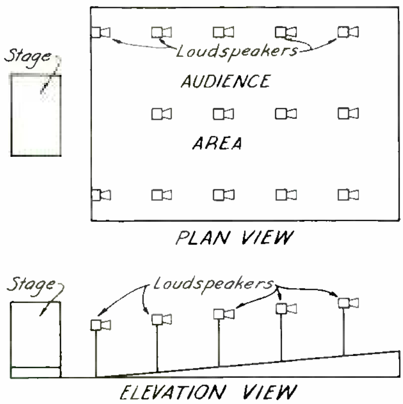


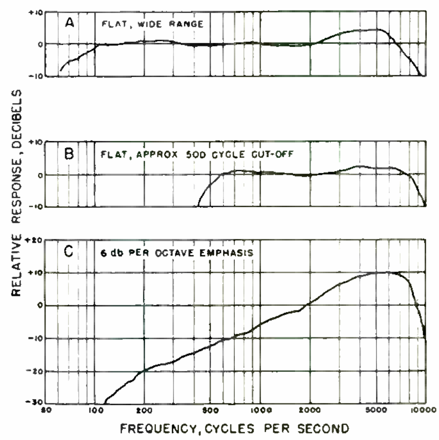
Fig. 64.—Use of multiple speakers to cover an outdoor theatre. Each covers a small portion of the audience.

At the same time it is found that the energy in the higher frequencies falls off at the rate of about 6db per octave. Suppose now that the noise has a fairly uniform energy distribution in the spectrum. Then its low-frequency components are found to be particularly effective in masking the high frequency components of speech; that is, the very components that are most important for intelligibility.

Indeed, it is found that the low-frequency components of speech can be deleted, even as far up as 400-500 c.p.s. without affecting the articulation or ability to understand speech. Hence, in high noise levels, three possibilities occur:

1. Transmit all components (flat response) at a much higher level.
2. Transmit flat all components above about 500 c.p.s.
3. Transmit the speech components with a response that rises 6db per octave.

These three possible characteristics are illustrated in Fig. 65.



(Courtesy Jensen Radio Mfg. Co.)
Fig. 65.—The three different response characteristics that can be used to override noise.

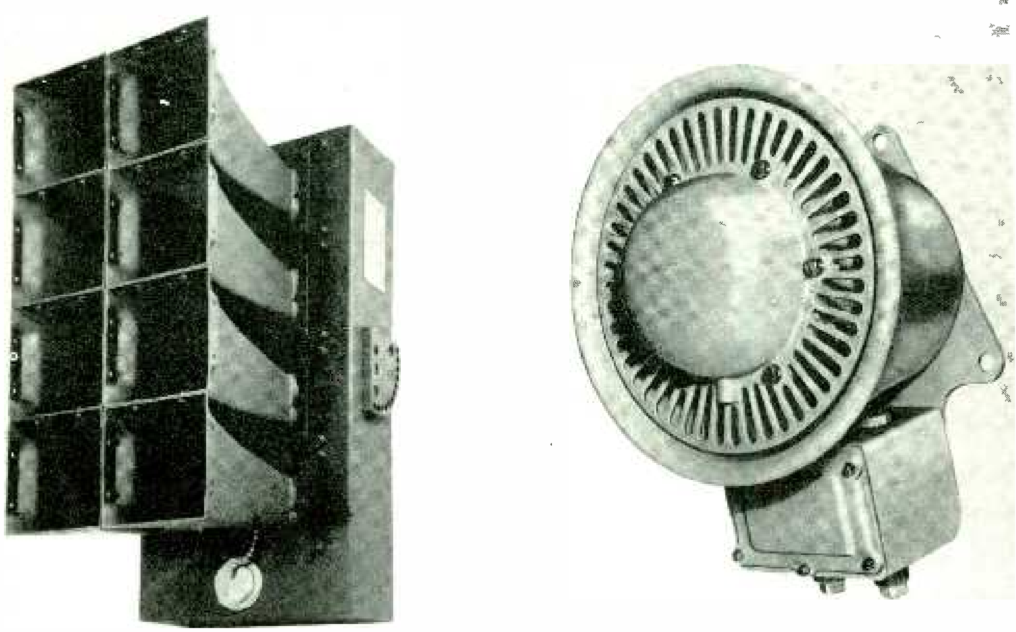
The flat characteristic (A) of increased level is an obvious solution, but is objectionable in that the pain level may be reached before intelligibility is attained, and the power requirements may be excessive. The effect is to "drown out" the noise.

The flat system cut off at 500 c.p.s. (B) require less power than that of (A), but is perhaps not quite as economical as the one shown in (C). The latter appears to "cut through" the noise and provides for very intelligible speech, although in the absence of noise it sounds very "thin" and sometimes even "harsh".

So far as audio power is concerned, the emphasized characteristic of (C) requires an electrical level 4.5db less than the flat system for equal loudness, or the latter requires 2.8 times as much power as the emphasized system. The flat system cut off as in (B) is comparable to the emphasized system providing cut off is at 500 c.p.s. or perhaps even higher.

The actual level of the system should be from 20 to 40db above the general noise level, and in any case should be 80db or higher. This is in contrast to speakers used in hotel rooms and for paging purposes in quiet localities, where a 70db intensity level is satisfactory.

As an example, Fig. 66 shows a loudspeaker designed for Navy shipboard service during battle conditions.



(Courtesy Jensen Radio Mfg. Co.)

Fig. 66.—Multiple straight horns (A) and re-entrant horn (B) loudspeakers used for speech reproduction in high noise level surroundings.

This particular speaker operates on a type (B) characteristic, but its low-frequency cutoff is considerably above 500 c.p.s.

RESUME'

In this assignment, there was discussed first the nature of sound, the mechanism of the ear and how it hears, and therefore the requirements for a loudspeaker system. Next the iron-diaphragm and balanced-armature units were described; these are used only in special applications, notably in telephone receivers.

The more universally used moving coil or dynamic loudspeaker was then taken up, and its advantages both as to freedom from distortion and its inherently high efficiency pointed out. Its use as a direct-radiator type of speaker was next studied, and the many forms employing multiple cones and voice coils analyzed for high-fidelity use.

The next topic taken up was that of loudspeaker baffles, and the behavior of the flat, box, and reflex types were discussed. Following this, horn-type speakers were studied; the various horn formulas presented, and the effect of folding of the horn on the high-frequency response indicated. Coordinated with this was the analysis of horn-type units both for woofer and tweeter applications.

Following this, the various cross-over networks were discussed, as well as considerations concerning the amount of acoustical and electrical power required for a given size auditorium. This section concluded with an analysis of the method of connecting loudspeaker loads to the output transformer taps so as to divide the power as desired between the various loudspeaker banks and yet match the output tubes' impedance.

Finally, practical installations were studied, with special reference to large auditoriums, multiple speaker layouts, and loudspeaker installations for noisy localities.

LOUDSPEAKERS

EXAMINATION

1. (A) What wavelength does a 50-cycle sound wave have in air at 20°C?

   (B) How does this wavelength compare with that in water?

   (C) Two observers are located at distances of 150 and 650 feet from a source of sound waves. Compare the sound intensity at their respective receiving positions.

2. (A) What is the basic objection to the use of a small cone, such as one 2 inches in diameter, in a low-frequency direct-radiator unit?

   (B) What is the basic objection to the use of a large 16-inch cone as a high-frequency unit, assuming that it can still move as a piston at such high frequencies?

3. (A) What is the advantage of a coaxial-type loudspeaker system, such as the duo-cone type, over the system employing for example a tweeter mounted directly above the woofer system?

   (B) How can the cross-over network for the latter type system minimize this objection?

4. (A) Determine the dimensions of a symmetrical square baffle to be used with a ten-inch cone speaker for effective reproduction of frequencies down to 80 c.p.s.

(B) What is the objection to the use of a small box that totally encloses the rear of a loudspeaker unit with little volume to spare?

5. (A) What is the basic action of a reflex port in an enclosed baffle, and what improvement does it produce in the low-frequency response?

(B) What effect does increased acoustic damping of the cone at the low-frequency end of the spectrum have on the distortion generated at high signal levels? Why?

(C) What is the purpose of the acoustic damping material employed in reflex and closed cabinet baffles?

6. (A) What is the fundamental action of a horn?

(B) In the process of obtaining a transformer action between the diaphragm of the horn unit and the throat of the horn, what other acoustic reactance is inherently obtained? How is this utilized in conjunction with the mass of the diaphragm and voice coil?

7. An exponential horn is to operate down to 80 c.p.s. The mouth or bell is to be a rectangle whose width is 2.5 times its height. The throat is circular and has a diameter of 0.55 inch. Calculate the length of the horn and the mouth dimensions.

8. (A) In a tweeter unit, what do the high-frequency channels do?

8. (B) What advantage does the cellular horn have at high frequencies over an ordinary horn?

9. A woofer and tweeter combination are to have a cross-over frequency at 400 c.p.s. The voice-coil impedance is 8 ohms in either case. A series-type network is desired, and an attenuation of 18db is required. Choose the appropriate network to accomplish this result, and calculate its constants.

10. (A) An auditorium has the following dimensions: 50×100×200 feet. An acoustic level of 86db* is required, and the efficiency of the loud-speaker system is 30 per cent. Calculate the required electrical power input in watts.

*Note that the power doubles for every 3db increase in level.

10. (B) Two loudspeakers are to be fed from an amplifier. One has a voice-coil impedance of 15 ohms, and is to receive 6 watts; the other has a voice-coil impedance of 8 ohms, and is to receive 4 watts. What is the impedance of each tap to which the units are to be connected?