CHAPTER 20.3
MICROPHONES, LOUDSPEAKERS, AND EARPHONES

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MICROPHONES

Sound-Responsive Elements

The sound-responsive element in a microphone may have many forms (Fig. 20.3.1). It may be a stretched membrane (*a*), a clamped diaphragm (*b*), or a magnetic diaphragm held in place by magnetic attraction (*c*). In these the moving element is either an electric or magnetic conductor, and the motion of the element creates the electric or magnetic equivalent of the sound directly.

Other sound-responsive elements are straight (*d*) or curves (*e*) conical diaphragms with various shapes of annular compliance rings, as shown. The motion of these diaphragms is transmitted by a drive rod from the conical tip to a mechanical transducer below.

Other widely used elements are a circular piston (*f*) bearing a circular voice coil of smaller diameter and a corrugated-ribbon conductor (*g*) of extremely low mass and stiffness suspended in a magnetic field.

Transduction Methods

Microphones have a great variety of transduction methods shown in Fig. 20.3.2.

The loose-contact transducer (Fig. 20.3.2*a*) was the first achieved by Bell in magnetic form and later made practical by Edison's use of carbonized hard-coal particles. It is widely used in telephones. Its chief advantage is its self-amplifying function, in which diaphragm amplitude variations directly produce electric resistance and current variations. Disadvantages include noise, distortion, and instability.

Moving-iron transducers have great variety, ranging from the historic pivoted armature (Fig. 20.3.2b) to the modern ring armature driven by a nonmagnetic diaphragm (Fig. 20.3.2*h*). In all these types a coil surrounds some portion of the magnetic circuit. The reluctance of the magnetic circuit is varied by motion of the sound-responsive element, which is either moving iron itself (Fig. 20.3.2*c* and *d*) or is coupled mechanically to the moving iron (Fig. 20.3.2*e*–*h*). In some of the magnetic circuits that portion of the armature surrounded by the coil carries very little steady flux, operating on differential magnetic flux only. Output voltage is proportional to moving-iron velocity.

Electrostatic transducers (Fig. 20.3.2*i*) use a polarizing potential and depend on capacitance variation between the moving diaphragm and a fixed electrode for generation of a corresponding potential difference. The *electret microphone* is a special type of electrostatic microphone that holds polarization indefinitely without continued application of a polarizing potential, an important practical advantage for many applications.

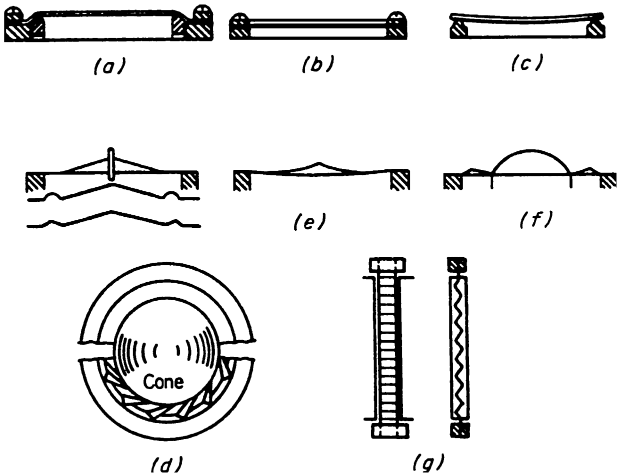


FIGURE 20.3.1 Sound-responsive elements in microphones.[1]

Piezoelectric transducers (Fig. 20.3.2*j*) create an alternating potential through the flexing of crystalline elements which, when deformed, generate a charge difference proportional to the deformation on opposite surfaces. Because of climatic effects and high electric impedance the rochelle salt commonly used for many years has been superseded by polycrystalline ceramic elements and by piezoelectric polymer.

Moving-coil transducers (Fig. 20.3.2*k*) generate potential by oscillation of the coil within a uniform magnetic field. The output potential is proportional to coil velocity.

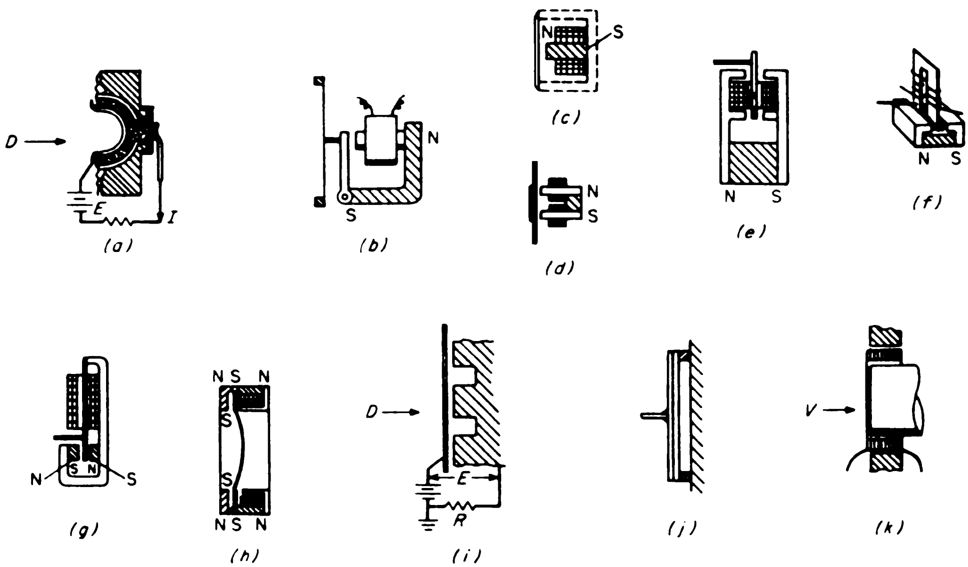


FIGURE 20.3.2 Microphone transduction methods.[1]

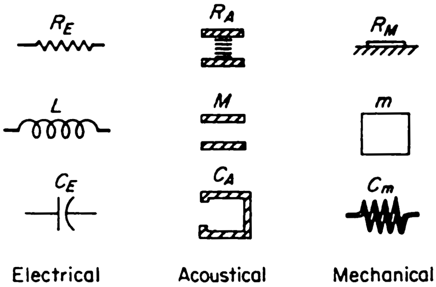


FIGURE 20.3.3 Equivalent basic elements in electrical, acoustical, and mechanical systems.[2]

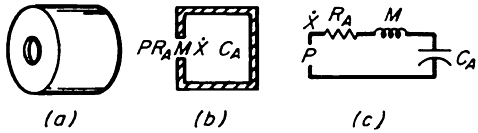


FIGURE 20.3.4 Helmholtz resonator in (*a*) perspective and (*b*) in section and (*c*) equivalent electric circuit.[2]

Equivalent Circuits

Electronics engineers understand electroacoustic and electromechanical design better with the help of equivalent or analogous electric circuits. Microphone design provides an ideal base for introduction of equivalent circuits because microphone dimensions are small compared with acoustical wavelengths over most of the audio-frequency range. This allows the assumption of lumped circuit constants.

Figure 20.3.3 shows equivalent symbols for the three basic elements of electrical, acoustical, and mechanical systems. In acoustical circuits the resistance is air friction or viscosity, which occurs in porous materials or narrow slots. Radiation resistance is another form of acoustical damping. Mechanical resistance is friction. Mass in the mechanical system is analogous to electric inductance. The acoustical equivalent is the mass of air in an opening or constriction divided by the square of its cross-sectional area. The acoustical analog of electric capacitance and mechanical-spring compliance is acoustical capacitance. It is the inverse of the stiffness of an enclosed volume of air under pistonlike action. Acoustical capacitance is proportional to the volume enclosed.

Figure 20.3.4 is an equivalent electric circuit for a Helmholtz resonator. Sound-pressure and air-volume current are analogous to electric potential and current, respectively. Other analog systems have been proposed. One frequently used has advantages for mechanical systems.

Microphone Types and Equivalent Circuits

Different types of microphone respond to different properties of the acoustical *input* wave. Moreover, the electric *output* can be proportional to different internal mechanical variables.

Pressure Type, Displacement Response. Figure 20.3.5 shows a microphone responsive to the sound-pressure wave acting through a resonant acoustical circuit upon a resonant diaphragm coupled to a piezoelectric element responsive to displacement. (The absence of sound ports in the case or in the diaphragm keeps the microphone pressure responsive.) In the equivalent circuit the sound pressure is the generator. L_a and R_a represent

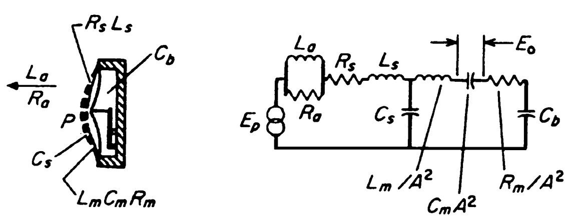
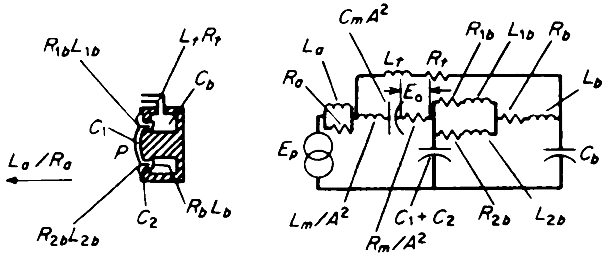


FIGURE 20.3.5 Pressure microphone, displacement response.[1]



$$\left(\text{If } \frac{A_1}{A_2} = \frac{C_1}{C_2} = \frac{R_{2b}}{R_{1b}} = \frac{L_{2b}}{L_{1b}}\right)$$

FIGURE 20.3.6 Pressure microphone, velocity response.[1]

the radiation impedance. L_s and R_s are the inertance and acoustical resistance of the holes; C_s is the capacitance of the volume in front of the diaphragm; L_m, C_m, and R_m are the mass, compliance, and resistance of the piezoelectric element and diaphragm lumped together; and C_b is the capacitance of the entrapped back volume of air. The electric output is the potential differential across the piezoelectric element. It is shown across the capacitance in the equivalent circuit because microphones of this type are designed to be stiffness-controlled throughout most of their operating range.

Pressure Type, Velocity Response. Figure 20.3.6 shows a moving-coil pressure microphone, which is a velocity-responsive transducer. In this microphone three acoustical circuits lie behind the diaphragm. One is behind the dome and another behind the annular rings. The third acoustical circuit lies beyond the acoustical resistance at the back of the voice-coil gap and includes a leak from the back chamber to the outside. This microphone is resistance-controlled throughout most of the range, but at low frequencies its response is extended by the resonance of the third acoustical circuit. Output potential is proportional to the velocity of voice-coil motion.

Pressure-Gradient Type, Velocity Response. When both sides of the sound-responsive element are open to the sound wave, the response is proportional to the *gradient* of the pressure wave. Figure 20.3.7 shows a ribbon conductor in a magnetic field with both sides of the ribbon open to the air. In the equivalent circuit there are two generators, one for sound pressure on each side. Radiation resistance and reactance are in series with each generator and the circuit constants of the ribbon. Usually the ribbon resonates at a very low frequency, making its mechanical response mass-controlled throughout the audio-frequency range. The electric output is proportional to the conductor velocity in the magnetic field. Gradient microphones respond differently to distant and close sound sources.

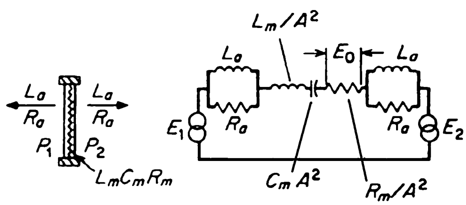


FIGURE 20.3.7 Gradient microphone, velocity response.[1]

Directional Patterns and Combination Microphones

Because of diffraction, a pressure microphone is equally responsive to sound from all directions as long as the wavelength is larger than microphone dimensions (see Fig. 20.3.8*a*). (At high frequencies it is somewhat directional along the forward axis of diaphragm or ribbon motion.)

By contrast a pressure-gradient microphone has a figure-eight directional pattern (Fig. 20.3.8*b*), which rotates about the axis of ribbon or diaphragm motion. A sound wave approaching a gradient microphone at 90° from the axis produces balanced pressure on the two sides of the ribbon and consequently no response.

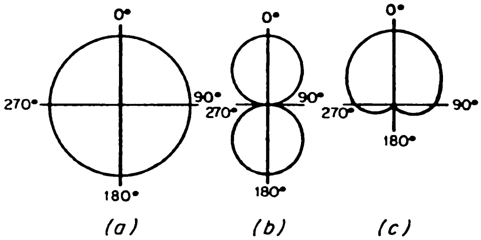


FIGURE 20.3.8 Directional patterns of microphones: (*a*) nondirectional; (*b*) bidirectional; (*c*) unidirectional.[2]

This defines the *null plane* of a gradient microphone. Outside this plane the microphone response follows a cosine law.

If the pressure and gradient microphones are combined in close proximity (see Fig. 20.3.9) and are connected electrically to add in equal (half-and-half) proportions, a heart-shaped cardioid pattern (Fig. 20.3.8*c*) is obtained. (The back of the ribbon in the pressure microphone is loaded by an acoustical resistance line.) By combining the two outputs in other proportions other limacon directional patterns can be obtained.

Phase-Shift Directional Microphones

Directional characteristics similar to those of the combination microphones can also be obtained with a single moving element by means of equivalent circuit analysis using acoustical phase-shift networks. Figure 20.3.10 shows a moving-coil, phase-shift microphone and its simplified equivalent circuit. The phase-shift network is composed of the rear-port resistance R_2 and inertance L_2, the capacitance of the volume under the diaphragm and within the magnet, and the impedance of the interconnecting screen. The microphone has a cardioid directional pattern.

Special-Purpose Microphones

Special-purpose microphones include two types that are superdirectional, two that overcome noise, and one without cables.

Line microphones use an approximate line of equally spaced pickup points connected through acoustically damped tubes to a common microphone diaphragm. The phase relationships at these points for an incident plane wave combine to give a sharply directional pattern along the axis if the line segment is at least one wavelength.

Parabolic microphones face a pressure microphone unit toward a parabolic reflector at its focal point, where sounds from distant sources along the axis of the parabola converge. They are effective for all wavelengths smaller than the diameter of the reflector.

Noise-canceling microphones are gradient microphones in which the mechanical system is designed to be stiffness-controlled rather than mass-controlled. For distant sound sources the resulting response is greatly attenuated at low frequencies. However, for a very close sound source, the response-frequency characteristic is uniform because the *gradient* of the pressure wave near a point source decreases with increasing frequency. Such a microphone provides considerable advantage for nearby speech over distant noise on the axis of the microphone.

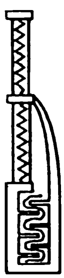


FIGURE 20.3.9 Combination unidirectional microphone.[1]

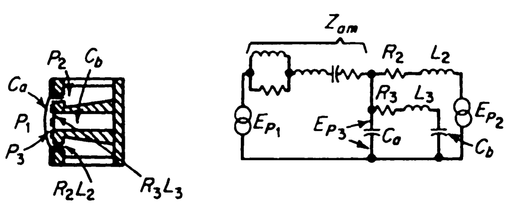


FIGURE 20.3.10 Phase-shift unidirectional microphone.[1]

Contact microphones are used on string and percussion musical instruments, on seismic-vibration detectors, and for pickup of body vibrations including speech. The throat microphone was noted for its convenience and its rejection of airborne noise. Most types of throat microphone are inertia-operated, the case receiving vibration from the throat walls actuated by speech sound pressure in the throat. The disadvantage is a deficiency of speech sibilant sounds received back in the throat from the mouth.

Wireless microphones have obvious operational advantages over those with microphone cords. A wireless microphone contains a small, low-power radio transmitter with a nearby receiver connected to an audio communication system. Any of the microphone types can be so equipped. The potential disadvantage is in rf interference and field effects.

Microphone Use in Recordings

The choice of microphone type and placement greatly affects the sound of a recording. For speech and dialogue recordings pressure microphones are usually placed near the speakers in order to minimize ambient-noise pickup and room reverberation. Remote pressure microphones are also used when a maximum room effect is desired.

In the playback of monophonic recordings room effects are more noticeable than they would have been to a listener standing at the recording microphone position because single-microphone pickup is similar to single-ear (monaural) listening, in which the directional clues of localization are lost. Therefore microphones generally need to be closer in a monophonic recording than in a stereophonic recording.

In television pickup of speech, where a boom microphone should be outside the camera angle, unidirectional microphones are often used because of their greater ratio of direct to generally reflected sound response.

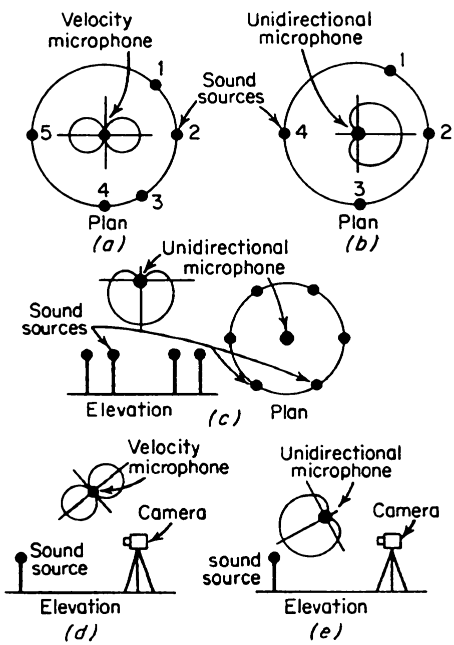


FIGURE 20.3.11 Use of directional microphones.[2]

Both velocity (gradient) microphones and unidirectional microphones can be used to advantage in broadcasting and recording. Figure 20.3.11*a* shows how instruments may be placed around a figure-eight directivity pattern to balance weaker instruments 2 and 5 against stronger instruments 1 and 3 with a potential noise source at point 4. In Fig. 20.3.11*b* source 2 is favored, with sources 1 and 3 somewhat reduced and source 4 highly discriminated against by the cardioid directional pattern. In Fig. 20.3.11*c* an elevated unidirectional microphone aimed downward responds uniformly to sources on a circle around the axis while discriminating against mechanical noises at ceiling level. Figure 20.3.11*d* places the camera noise in the null plane of a figure-eight pattern, and Fig. 20.3.11*e* shows a similar use for the unidirectional microphone. Camera position is less critical for the cardioid microphone than for the gradient microphone.

Early classical stereo recordings used variations of two basic microphone arrangements. In one scheme two unidirectional microphones were mounted close together with their axes angled toward opposite ends of the sound field to be recorded. This retained approximately the same arrival time and phase at both microphones, depending chiefly on the directivity patterns to create the sound difference in the two channels.

In the second scheme the two microphones (not necessarily directional) were separated by distances of 5 to 25 ft, depending on the size of the sound field to be recorded. Microphone axes (if directional) were again directed toward the ends of the sound field or group of sound sources. In this arrangement the time of arrival and phase differences were more important, and the effect of directivity was lessened. Each approach had its advantages and disadvantages.

With the arrival of tape recorders having many channels a trend has developed toward the use of more microphones and closer microphone placement. This offers much greater flexibility in mixing and rerecording, and it largely removes the effect of room reverberation from the recording. This may be either an advantage or a disadvantage depending on the viewpoint. Reverberation can be added later.

In sound-reinforcement systems for dramatic productions and orchestras the use of many microphones again offers operating flexibility. However, it also increases the probability of operating error, increased system noise, and acoustical feedback, making expert monitoring and mixing of the microphone outputs necessary.

An attractive alternative for multimicrophone audio systems is the use of independent voice-operated electronic control switches in each microphone channel amplifier, in combination with an automatic temporary reduction of overall system gain as more channels switch on, in order to prevent acoustical feedback. Automatic mixers have been devised to minimize speech signal dropouts, and to prevent the inadvertent operation of channel control switches by background noises.

Microphone Mounting

On podiums and lecterns microphones are typically mounted on fixed stands with adjustable arms. On stages they are mounted on adjustable floor stands. In mobile communication and in other situations where microphone use is occasional, handheld microphones are used during communication and are stowed on hangers at other times. For television and film recording, where the microphone must be out of camera sight, the microphones are usually mounted on booms overhead and are moved about during the action to obtain the best speech-to-noise ratio possible at the time. In two-way communication situations which require the talker to move about or to turn his head frequently, the microphone can be mounted on a boom fastened to his headset. This provides a fixed close-talking microphone position relative to the mouth, a considerable advantage in high-ambient-noise levels.

Microphone Accessories

Noise shields are needed for microphones in ambient noise levels exceeding 110 dB. Noise shields are quite effective at high frequencies, where the random-noise discrimination of noise-canceling microphones diminishes. Noise shields and noise-canceling microphones complement each other.

Windscreens are available for microphone use in airstreams or turbulence. Without them aerodynamically induced noise is produced by turbulence at the microphone grille or openings. Large windscreens are more effective than small ones because they move the turbulence region farther from the microphone.

Special sponge-rubber mountings for the microphone and cable to reduce extraneous vibration of the microphone are often used. Many microphone stands and booms have optional suspension mounting accessories to reduce shock and vibration transmitted through the stand or boom to the microphone.

Special Properties of Microphones

The source impedance of a microphone is important not only to the associated preamplifier but also to the allowable length of microphone cable and the type and amount of noise picked up by the cable. High-impedance microphones (10 kΩ or more) cannot be used more than a few feet from the preamplifier without pickup from stray fields. Microphones having an impedance of a few ohms or less are usually equipped with stepup transformers to provide a line impedance in the range of 30 to 600 Ω, which extensive investigation has established as the most noise-free line-impedance range.

The microphone unit itself can be responsive to hum fields at power-line frequencies unless special design precautions are taken. Most microphones have a hum-level rating based on measurement in a standard alternating magnetic field.

For minimum electrical noise balanced and shielded microphone lines are used, with the shield grounded only at the amplifier end of the line.

Microphone linearity should be considered when the sound level exceeds 100 dB, a frequent occurrence for loud musical instruments and even for close speech. Close-talking microphones, especially of the gradient type, are particularly susceptible to noise from breath and plosive consonants.

Specifications

Microphone specifications typically include many of the following items: type or mode of operation, directivity pattern, frequency range, uniformity of response within the range, output level at one or more impedances for a standard sound-pressure input (for example, 1 Pa or 10 dyn/cm^2), recommended load impedance, hum output level for a standard magnetic field (for example, 10^{-3} G), dimensions, weight, finish, mounting, power supply (if necessary), and accessories.

LOUDSPEAKERS

Introduction

A loudspeaker is an electroacoustic transducer intended to radiate acoustic power into the air, with the acoustic waveform equivalent to the electrical input waveform. An earphone is an electroacoustic transducer intended to be closely coupled acoustically to the ear. Both the loudspeaker and earphone are receivers of audio-electronic signals. The principal distinction between them is the acoustical loading. An earphone delivers sound to air in the ear. A loudspeaker delivers sound indirectly to the ear through the air.

The transduction methods of loudspeakers and earphones are historically similar and are treated together. An overview of loudspeaker developments of the closing 50 years of the last millennium is given by Gander.[3] However, since loudspeakers operate primarily into radiation resistance and earphones into acoustical capacitance, the design, measurement, and use of the two types of electroacoustic transducers will be discussed separately.

Transduction Methods

Early transducers for sound reproduction were of the mechanoacoustic type. Vibrations received by a stylus in the undulating groove of a record were transmitted to a diaphragm, placed at the throat of a horn for better acoustical impedance matching to the air, all without the aid of electronics. Electro-acoustics and electronics introduced many advantages and a variety of transduction methods including moving-coil, moving-iron, electrostatic, magnetostrictive, and piezoelectric (Fig. 20.3.12).

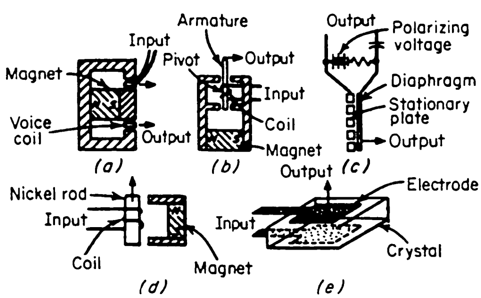


FIGURE 20.3.12 Loudspeaker (and earphone) transduction methods: (*a*) moving-coil; (*b*) moving-iron; (*c*) electrostatic; (*d*) magnetostrictive; (*e*) piezoelectric.[4]

Most loudspeakers are moving-coil type today, although moving-iron transducers were once widely used. Electrostatic loudspeakers are used chiefly in the upper range of audio frequencies, where amplitudes are small. Magnetostrictive and piezoelectric loudspeakers are used for underwater sound. All the transducer types are used in earphones except magnetostrictive.

Moving-Coil. The mechanical force on the moving coil of Fig. 20.3.12*a* is developed by the interaction of the current in the coil and the transverse magnetic field disposed radially across the gap between the magnet cap and the iron housing, which completes the magnetic circuit. The output force along the axis of the circular coil is applied to a sound radiator.

Moving-iron transducers reverse the mechanical roles of the coil and the iron. The iron armature surrounded by the stationary coil is moved by mechanical forces developed within the magnetic circuit. Moving-iron magnetic circuits have many forms. As an example in the balanced armature system (Fig. 20.3.12*b*) the direct magnetic flux passes only transversely through the ends of the armature centered within the two magnetic gaps. Coil current polarizes the armature ends oppositely, creating a force moment about the pivot point. The output force is applied from the tip of the armature to an attached sound radiator. In a balanced-diaphragm loudspeaker the armature is the radiator.

Electrostatic. In the electrostatic transducer (Fig. 20.3.12*e*) there is a dc potential difference between the conductive diaphragm and the stationary perforated plate nearby. Audio signals applied through a blocking capacitor superimpose an alternating potential, resulting in a force upon the diaphragm, which radiates sound directly.

Magnetostrictive transducers (Fig. 20.3.12*d*) depend on length fluctuations of a nickel rod caused by variations in the magnetic field. The output motion may be radiated directly from the end of the rod or transmitted into the attached mechanical structure.

Piezoelectric transducers are of many forms using crystals or polycrystalline ceramic materials. In simple form (Fig. 20.3.12*e*) an expansion-contraction force develops along the axis joining the electrodes through alternation of the potential difference between them.

Sound Radiators

The purpose of a sound radiator is to create small, audible air-pressure variations. Whether they are produced within a closed space by an earphone or in open air by a loudspeaker, the pressure variations require air motion or current.

Pistons, Cones, Ports. Expansion and contraction of a sphere is the classical configuration but most practical examples involve rectilinear motion of a piston, cone, or diaphragm. In addition to the primary direct radiation from moving surfaces, there is also indirect or secondary radiation from enclosure ports or horns to which the direct radiators are acoustically coupled.

Attempts have been made to develop other forms of sound radiation such as oscillating airstreams and other aerodynamic configurations with incidental use, if any, of moving mechanical members.

Directivity. Figure 20.3.13 shows the directional characteristics of a rigid circular piston for different ratios of piston diameter and wavelength of sound. (In three dimensions these curves are symmetrical around the axis of piston motion.) For a diameter of one-quarter wavelength the amplitude decreases 10 percent (approximately

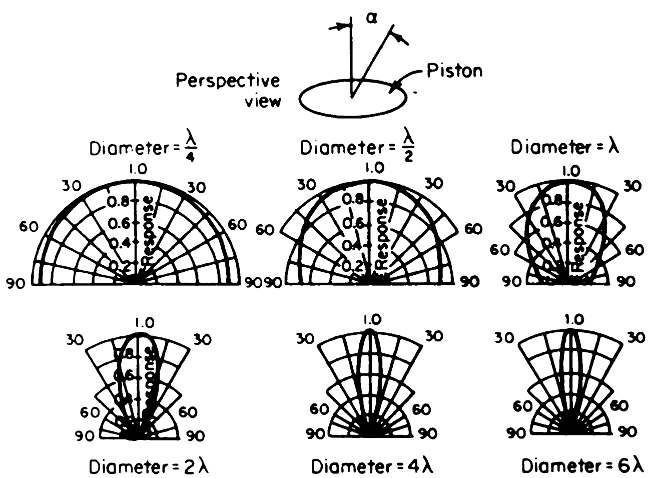


FIGURE 20.3.13 Directional characteristics of rigid circular pistons of different diameters or at different sound wavelengths.[2]

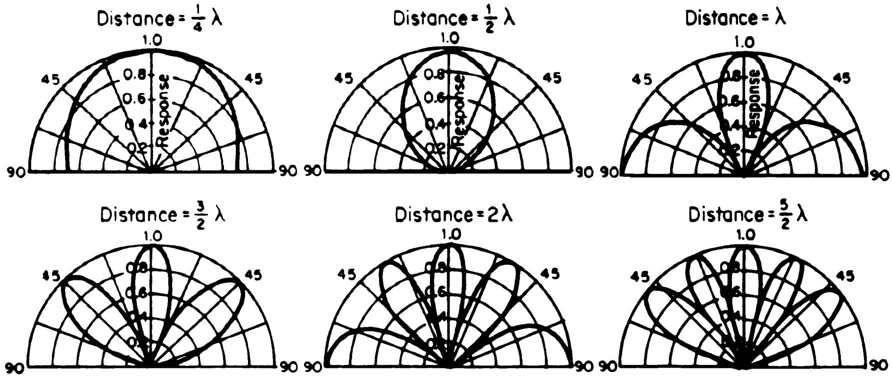


FIGURE 20.3.14 Directional characteristics of two equal small in-phase sound sources separated by different distances or different sound wavelengths.[2]

1 dB sound level) at 90° off axis. For a four-wavelength diameter the same drop occurs in only 5°. (The beam of an actual loudspeaker cone is less sharp than this at high frequencies, where the cone is not rigid.) Note that all the polar curves are smooth when the single-source piston vibrates as a whole.

Radiator Arrays. When two separate, identical small-sound sources vibrate in phase, the directional pattern becomes narrower than for one source. Figure 20.3.14 shows that for a separation of one-quarter wavelength the two-source beam is only one-half as wide as for a single piston. At high frequencies the directional pattern becomes very complex. (In three dimensions these curves become surfaces of revolution about the axis joining the two sources.)

Arrays of larger numbers of sound radiators in close proximity are increasingly directional. Circular-area arrays have narrow beams which are symmetrical about an axis through the center of the circle. Line arrays, e.g., column loudspeakers, are narrowly directional in planes containing the line and broadly directional in planes perpendicular to the line.

Direct-Radiator Loudspeakers

Most direct-radiator loudspeakers are of the moving-coil type because of simplicity, compactness, and inherently uniform response-frequency trend. The uniformity results from the combination of two simple physical principles: (1) the radiation resistance increases with the square of the frequency, and hence the radiated sound power increases similarly for constant velocity amplitude of the piston or cone; (2) for a constant applied force (voice-coil current) the mass-controlled (above resonance) piston has a velocity amplitude which decreases with the square of the frequency. Consequently a loudspeaker designed to resonate at a low frequency combines decreasing velocity with increasing radiation resistance to yield a uniform response within the frequency range where the assumptions hold.

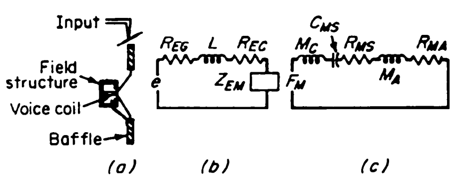


FIGURE 20.3.15 (*a*) Structure, (*b*) electric circuit, and (*c*) equivalent mechanical circuit for a direct-radiator moving-coil loudspeaker in a baffle.[4]

Equivalent Electric Circuits. Figure 20.3.15 shows a cross-sectional view of a direct-radiator loudspeaker mounted in a baffle, the electric voice-coil circuit, and the equivalent electric circuit of the mechanoacoustic system. In the voice-coil circuit e is the emf and R_{EG} the resistance of the generator, e.g., power-amplifier output, L and R_{EC} are the inductance and resistance of the voice coil. Z_{EM} is the motional electric impedance from the mechanoacoustic system.

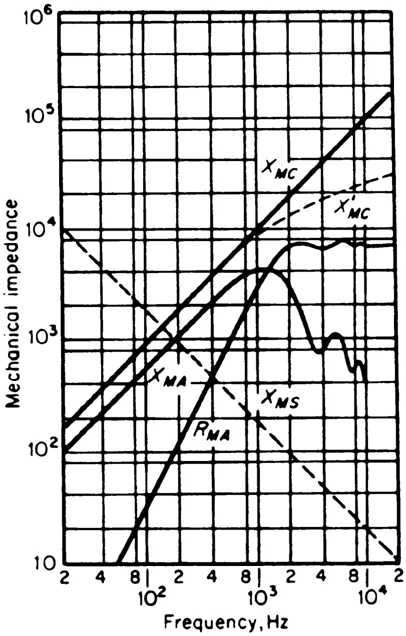


FIGURE 20.3.16 Components of a mechanical impedance of a typical 4-in loudspeaker.[4]

F_M is the driving force resulting from interaction of the voice-coil current field with the gap magnetic field. M_C is the combined mass of the cone and voice coil. C_{MS} is the compliance of the cone-suspension system. R_{MS} is the mechanical resistance. The mass M_A and radiation resistance R_{MA} of the air load complete the circuit.

Figure 20.3.16 summarizes these mechanical impedance factors for a 4-in direct-radiator loudspeaker of conventional design. Above resonance (where the reactance of the suspension system equals the reactance of the cone-coil combination) the impedance-frequency characteristic is dominated by M_C. From the resonance frequency of about 150 Hz to about 1500 Hz the conditions for uniform response hold.

Efficiency. Since R_{MA} is small compared to the magnitudes of the reactive components, the efficiency of the loudspeaker in this frequency range can be expressed as

$$\text{Efficiency} = \frac{100(Bl)^2 R_{MA}}{R_{EC}(X_{MA} + X_{MC})^2} \text{ percent} \tag{1}$$

where B = gap flux density (G)
l = voice-coil conductor length (cm)
R_{EC} = voice-coil electric resistance (abohms)

Since R_{MA} is proportional to the square of the frequency and both X_{MA} and X_{MC} increase with frequency, the efficiency is theoretically uniform.

All this has assumed that the cone moves as a whole. Actually wave motion occurs in the cone. Consequently at high frequencies the mass reactance is somewhat reduced (as shown in the dashed curve of Fig. 20.3.16), tending to improve efficiency beyond the frequency where radiation resistance becomes uniform.

Magnetic Circuit. Most magnets now are a high-flux, high-coercive permanent type, either an alloy of aluminum, cobalt, nickel, and iron, or a ferrite of iron, cobalt, barium, and nickel. The magnet may be located in the core of the structure or in the ring, or both. However, magnetization is difficult when magnets are oppositely polarized in the core and ring.

Air-gap flux density varies widely in commercial designs from approximately 3000 to 20,000 G. Since most of the reluctance in the magnetic circuit resides in the air gap, the minimum practical voice-coil clearance in the gap compromises the maximum flux density. Pole pieces of heat-treated soft nickel-iron alloys, dimensionally tapered near the gap, are used for maximum flux density.

Voice Coils. The voice coil is a cylindrical multilayer coil of aluminum or copper wire or ribbon. Aluminum is used in high-frequency loudspeakers for minimum mass and maximum efficiency. Voice-coil impedance varies from 1 to 100 Ω with 4, 8, and 16 Ω standard. For maximum efficiency the voice-coil and cone masses are equal. However, in large loudspeakers the cone mass usually exceeds the voice-coil mass. Typically the voice-coil mass ranges from tenths of a gram to 5 g or more.

Cones. Cone diameters range from 1 to 18 in. Cone mass varies from tenths of a gram to 100 g or more. Cones are made of a variety of materials. The most common is paper deposited from pulp on a wire-screen form in a felting process. For high-humidity environment cones are molded from plastic materials, sometimes with a cloth or fiber-glass base. Some low-frequency loudspeaker cones are molded from low-density plastic foam to achieve greater rigidity with low density.

So far piston action has been assumed in which the cone moves as a whole. Actually at high frequencies the cone no longer vibrates as a single unit. Typically there is a major dip in response resulting from quarter-wave

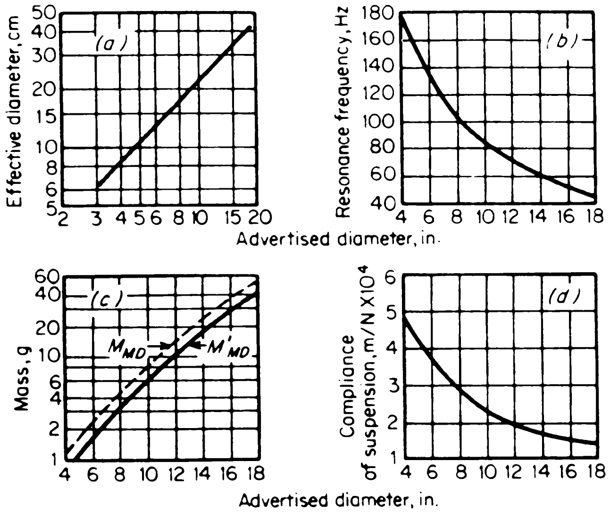


FIGURE 20.3.17 Typical cone and coil design values.[5]

reflection from the circular rim of the cone back to the voice coil. For loudspeaker cones in the range of 8 to 15 in. diameter this dip usually occurs in the range of 1 to 2 kHz.

Typical Commercial Design Values. Figure 20.3.17 shows typical values for several cone and voice-coil design parameters for a range of loudspeaker diameters. These do not apply to extreme cases, such as high-compliance loudspeakers or high-efficiency horn drivers. The effective piston diameter (Fig. 20.3.17*a*) is less than the loudspeaker cone diameter because the amplitude falls off toward the edges. A range of resonance frequencies is available for any cone diameter, but Fig. 20.3.17*b* shows typical values. In Fig. 20.3.17*c* typical cone mass is M including the voice coil and M' excluding the voice coil. Figure 20.3.17*d* shows typical cone-suspension compliance.

Impedance. A major peak results from motional impedance at primary mechanical resonance. Impedance is usually uniform above this peak until voice-coil inductance becomes dominant over resistance.

Power Ratings. Different types of power rating are needed to express the performance capabilities of loudspeakers. The large range of typical loudspeaker efficiency makes the acoustical power-delivering capacity quite important. The electrical power-receiving capacity (without overload or damage) determines the choice of power amplifier.

Loudspeaker efficiencies are seldom measured but are often compared by measuring the sound-pressure level at 4 ft on the loudspeaker axis for 1-W audio input. High-efficiency direct radiators provide 95 to 100 dB. Horn loudspeakers are typically higher by 10 dB or more, being both more efficient and more directional.

Loudspeakers are also rated by the maximum rms power output of amplifiers which will not damage the loudspeaker or drive it into serious distortion on peaks. Such ratings usually assume that the amplifier will seldom be driven to full power. For example, a 30-W amplifier will seldom be required to deliver more than 10 W rms of music program material. Otherwise music peaks would be clipped and sound distorted.

However, in speech systems for high-ambient-noise levels the speech peaks may be clipped intentionally, causing the loudspeaker to receive the full 30 W much of the transmission time. Then the loudspeaker must handle large excursions without mechanical damage to the cone suspension and without destroying the cemented coil or charring the form.

Distortion. Nonlinear distortion in a loudspeaker is inherently low in the mass-controlled range of frequencies. However, distortion is produced by nonlinear cone suspension at low frequencies, voice-coil motion beyond the limits of uniform air-gap flux, Doppler shift modulation of high-frequency sound by large cone velocity at low frequencies, and nonlinear distortion of the air near the cone at high powers (particularly in horn drivers). Methods for controlling these distortions follow.

1. When a back enclosure is added to a loudspeaker, the acoustical capacitance of the enclosed volume is represented by an additional series capacitor in the mechanical circuit of Fig. 20.3.15. Insufficient volume stiffens the cone acoustically, raising the resonance frequency and limiting the low-frequency range of the loudspeaker. It is convenient to reduce nonlinear distortion at low frequencies by increasing the cone-suspension compliance and depending on the back enclosure to provide the system stiffness. Since an enclosed volume is more linear than most mechanical springs, this lowers low-frequency distortion.
2. Distortion from inhomogeneity of the air-gap flux can be reduced by making the voice-coil length either considerably smaller or larger than the gap width. This stabilizes the total number of lines passing through the coil, but it also reduces loudspeaker efficiency.
3. Doppler distortion can be eliminated only by separating the high and low frequencies in a multiple loudspeaker system.
4. Air-overload distortion can be avoided by increasing the radiating area.

Loudspeaker Mountings and Enclosures

Figure 20.3.18 shows a variety of mountings and enclosures. An unbaffled loudspeaker is an acoustic doublet for wavelengths greater than the rim diameter. In this frequency range the acoustical power output for constant cone velocity is proportional to the fourth power of the frequency.

Baffles. In order to improve efficiency at low frequencies it is necessary to separate the front and back waves. Figure 20.3.18*a* is the simplest form of baffle. The effect of different baffle sizes is given in Fig. 20.3.19. Response dips occurring when the acoustic path from front to back is a wavelength are eliminated by irregular baffle shape or off-center mounting.

Enclosures. The widely used open-back cabinet (Fig. 20.3.18*b*) is noted for a large response peak produced by open-pipe acoustical resonance. A closed cabinet (Fig. 20.3.18*c*) adds acoustical stiffness at low frequencies where the wavelength is larger than the enclosure. At higher frequencies the internal acoustical resonances create response irregularities requiring internal acoustical absorption.

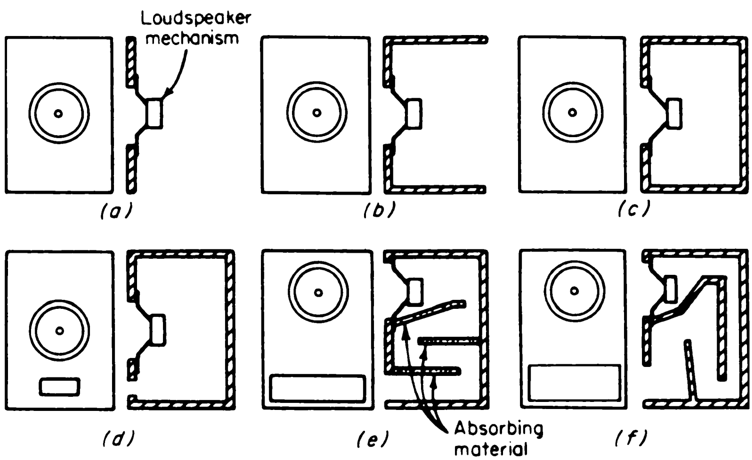


FIGURE 20.3.18 Mountings and enclosures for direct-radiator loudspeaker: (*a*) flat baffle; (*b*) open-back cabinet; (*c*) closed cabinet; (*d*) ported closed cabinet; (*e*) labyrinth; (*f*) folded horn.[4]

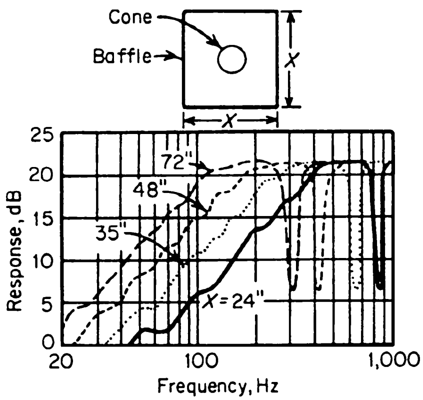


FIGURE 20.3.19 Response frequency for loudspeaker in 2-, 3-, 4-, and 6-ft square baffles.[6]

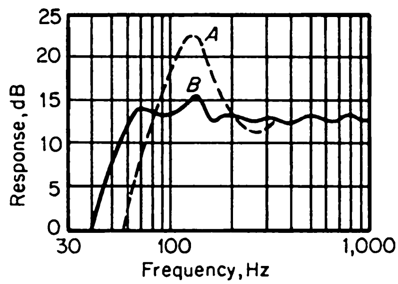


FIGURE 20.3.20 Response frequency for loudspeaker in closed (*A*) and ported (*B*) cabinets.[6]

Ported Enclosures (Fig. 20.3.18d). Enclosure volume can be minimized without sacrificing low-frequency range by providing an appropriate port in the enclosure wall. Acoustical inertance of the port should resonate with the enclosure capacitance at a frequency about an octave below cone-resonance frequency. *B*, Fig. 20.3.20, shows that this extends the low-frequency range. This is most effective when the port area equals the cone-piston area. Port inertance can be increased by using a duct. An extreme example of ducting is the acoustical labyrinth (Fig. 20.3.18*e*). When duct work is shaped to increase cross section gradually, the labyrinth becomes a low-frequency horn (Fig. 20.3.18*f*).

Direct-radiator loudspeaker efficiency is typically 1 to 5 percent. Small, highly damped types with miniature enclosures may be only 0.1 percent. Transistor amplifiers easily provide the audio power for domestic loudspeakers. However, in auditorium, outdoor, industrial, and military applications much higher efficiency is required.

Horn Loudspeakers

Higher efficiency is obtained with an acoustic horn, which is a tube of varying cross section having different terminal areas to provide a change of acoustic impedance. Horns match the high impedance of dense diaphragm material to the low air impedance. Horn shape or taper affects the acoustical transformer response. Conical, exponential, and hyperbolic tapers have been widely used. The potential low-frequency cutoff of a horn depends on its taper rate. Impedance transforming action is controlled by the ratio of mouth to throat diameter.

Horn Drivers. Figure 20.3.21 shows horn-driving mechanisms and straight and folded horns of large- and small-throat types. A large-throat driver (Fig. 20.3.21*a*) resembles a direct-radiator loudspeaker with a voice-coil diameter of 2 to 3 in. and a flux density around 15,000 G. A small-throat driver (Fig. 20.3.21*b*) resembles a moving-coil microphone structure. Radiation is taken from the back of the diaphragm into the horn throat through passages which deliver in-phase sound from all diaphragm areas. Diaphragm diameters are 1 to 4 in. with throat diameters of $^1/_4$ to 1 in. Flux density is approximately 20,000 G.

Large-Throat Horns. These are used for low-frequency loudspeaker systems. A folded horn (Fig. 20.3.21*c*) is preferred over a straight horn (Fig. 20.3.21*d*) for compactness.

Small-Throat Horns. A folded horn (Fig. 20.3.21*e*) with sufficient length and gradual taper can operate efficiently over a wide frequency range. This horn is useful for outdoor music reproduction in a range of 100 to 5000 Hz. Response smoothness is often compromised by segment resonances. Extended high-frequency range requires a straight-axis horn (Fig. 20.3.21*f*).

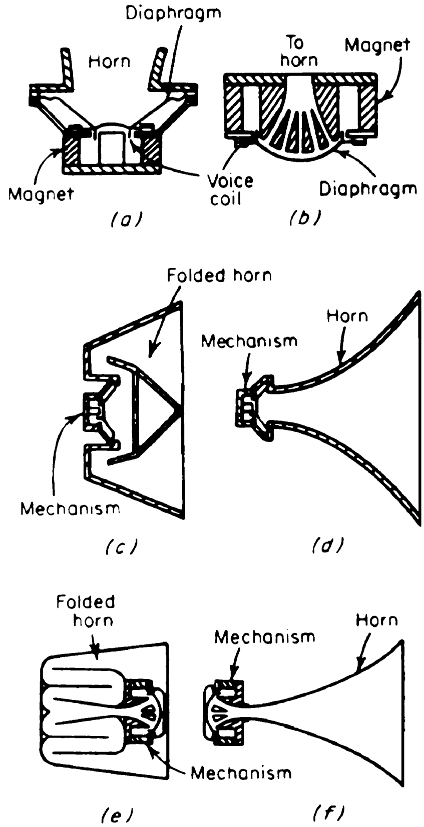


FIGURE 20.3.21 Horns and horn drivers: (*a*) large-throat driver; (*b*) small-throat driver; (*c*) folded large-throat horn; (*d*) straight large-throat horn; (*e*) folded small-throat horn; (*f*) straight small-throat horn.[4]

Horn Directivity. Large-mouth horns of simple exponential design produce high-directivity radiation that tends to narrow with increasing frequency (as in Fig. 20.3.13). In applications requiring controlled directivity over a broad angle and a wide frequency range, a horn array (shown in Fig. 20.3.22*a*) can be used, with numerous small horn mouths spread over a spherical surface and throats converging together. Figure 20.3.22*b* shows the directional characteristics. Single sectoral horns with radial symmetry can provide cylindrical wavefronts with smoother directional characteristics which are controlled in one plane. Recent rectangular or square-mouth "quadric" horns, designed by computer to have different conical expansion rates in horizontal and vertical planes, provide controlled directivity in both planes over a wide frequency range.

Special Loudspeakers

Special types of loudspeakers for limited applications include the following.

Electrostatic high-frequency units have an effective spacing of about 0.001 in. between a thin metalized coating on plastic and a perforated metal backplate. This spacing is necessary for sensitivity comparable to moving-coil loudspeakers, but it limits the amplitude and the frequency range. Extension of useful response to the lower frequencies can be obtained with larger spacing, for example, 1/16 in., with a polarizing potential of several thousand volts. This type of unit employs push-pull operation.

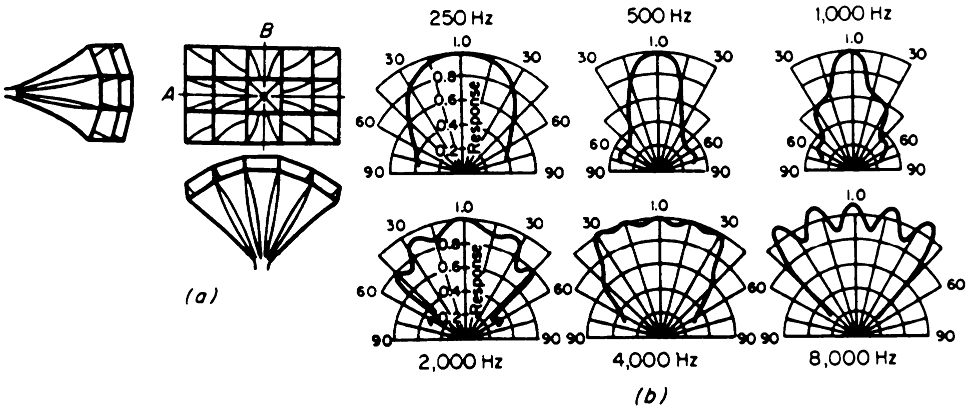


FIGURE 20.3.22 Horn array (cellular) and directional characteristics: (*a*) array; (*b*) horizontal directional curves.[6]

Modulated-airflow loudspeakers have an electromechanical mechanism for modulating the airstream from a high-pressure pneumatic source into a horn. Low audio power controls large acoustical power in this system. A compressor is also needed. Nonlinear distortion in the air and reduced speech intelligibility have been limitations of this high-power system.

Loudspeaker Specifications and Measurements

Typical loudspeaker specifications are shown in Table 20.3.1 for a variety of loudspeaker types.

Loudspeaker impedance is proportional to the voltage across the voice coil when driven by a high-impedance constant-current source. Continuous power ratings are obtained from sustained life tests with

TABLE 20.3.1 Characteristics of a Variety of Loudspeaker Types

Company	Altec	Altec	Bozak	RCA
Model no.	775C	1505B horn 290D driver	CM-109–23	LC1B
Type	Direct radiator	Cellular horn (3 × 5)	Three-way column	Duo-cone
Sensitivity (at 4 ft for 1 W), dB	95	110	106	95
Frequency range, Hz	40–15,000	300–8,000	65–13,000	25–16,000 (± 4 dB)
Impedance, Ω	8	4	8	15
Power rating, W	15	100	200	20
Distribution angle, deg	90	105 horizontal 60 vertical	90 horizontal 30 vertical	120
Voice-coil diameter, in.	2	2.8	(3 sizes)	(2 cones)
Cone resonance, Hz	52	…	(3 sizes)	22
Crossover frequency, Hz	…	500	800, 2,500	1,600
Diameter, in.	$8^3/_8$	$18^1/_2$ high $30^1/_2$ wide	57 in. high $22^3/_4$ wide	17
Depth, in.	$2^1/_4$	30	$15^3/_4$	$7^1/_2$
Weight, lb	$3^3/_4$	43	250	21

typical audio-program material restricted to the frequency range appropriate for the loudspeaker type. Sensitivity, response-frequency characteristics, frequency range, and directivity are most effectively measured under anechoic conditions using calibrated laboratory microphones and high-speed level recorders. However, data so measured should not be expected to be exactly reproducible under room-listening conditions.

Distortion measurements in audio-electronic systems are generally of three types shown in Fig. 20.3.23. For harmonic distortion a single sinusoidal signal A is supplied to the loudspeaker and wave analysis at the harmonic frequencies determines the percent distortion.

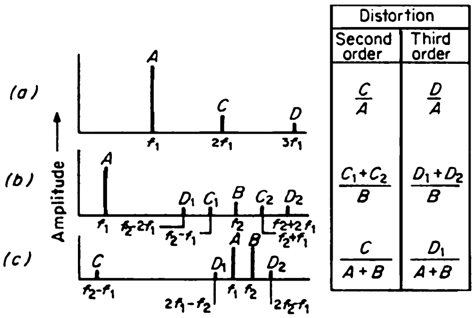


FIGURE 20.3.23 Methods of measuring nonlinear distortion: (*a*) harmonic; (*b*) intermodulation method of SMPTE; (*c*) intermodulation method of CCIF.[7]

Both intermodulation methods supply two sinusoidal signals of different frequency to the loudspeaker. In the older Society of Motion Picture and Television Engineers (SMPTE) method the frequencies are widely separated, and the distortion is expressed in terms of sum and difference frequencies around the higher test frequency. This method is meaningful for wide-range loudspeaker systems.

The CCIF (International Telephone Consultative Committee) method is more applicable to narrow-range systems and loudspeakers receiving input at high frequencies. It supplies two high frequencies to the loudspeaker and checks the low difference frequency.

Transient intermodulation distortion, resulting from nonlinear response to steep wavefronts, is measured by adding square-wave (3.18-kHz) and sine-wave (15-kHz) inputs, with a 4:1 amplitude ratio, and observing the multiple sum- and difference-frequency components added to the output spectrum.

EARPHONES

The transduction methods are the same as for loudspeakers. Telephone and hearing aid receivers are usually moving-iron. Most military headsets are now moving-coil. Piezoelectric, moving-coil, and electrostatic types are used for listening to recorded music.

Equivalent Electric Circuits

Figure 20.3.24 shows a cross section of a moving-coil earphone and the equivalent electric circuit. The voice-coil force drives the voice coil and diaphragm. (Mechanical resonance of earphone diaphragms occurs at a high audio frequency in contrast to loudspeakers.) Diaphragm motion creates sound pressure in several spaces behind the diaphragm and the voice coil and between the diaphragm and the earcap. Inertance and resistance of the connecting holes and clearances combine with the capacitance of the spaces to add acoustical resonances. Z is the acoustical impedance of the ear.

Idealized Ear Loading

The ear is approximately an acoustical capacitance. However, acoustical leakage adds a parallel resistance-inertance path affecting low-frequency response. At high frequencies the ear canal-length resonance is a factor.

Since the ear is a capacitance, the goal of earphone design is a constant diaphragm amplitude throughout the frequency range. This requires a stiffness-controlled system or a high-resonance frequency. The potential across the ear is analogous to sound pressure within the ear cavity. This sound pressure is proportional to

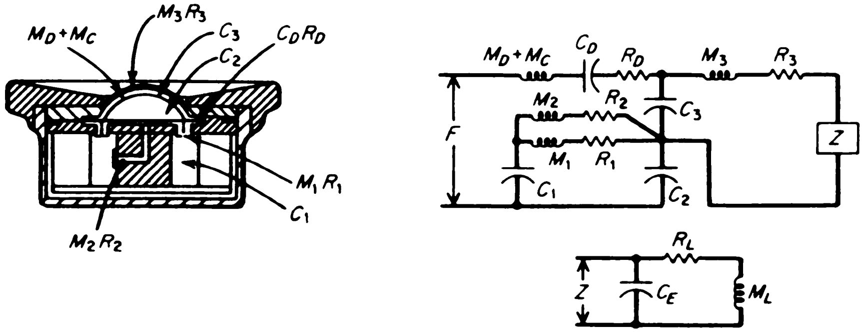


FIGURE 20.3.24 Moving-coil earphone cross section and equivalent electric circuit.[8]

diaphragm area and inversely proportional to enclosed volume. Earphone loading conditions are extremely varied for different types of earphone mountings.

Earphone Mountings

The most widely used earphone is the single receiver unit on a telephone handset. It is intended to be held against the ear but is often tilted away, leaving considerable leakage.

Headsets provide better communication than handsets because they supply sound to both ears and shield them.

A remote earphone can drive the ear canal through a small acoustic tube. The length may be an inch or two for hearing aids and several feet for music listening on aircraft.

Efficiency, Impedance, and Driving Circuits

Moving-iron earphones and microphones can be made efficient enough to operate as sound-powered (batteryless) telephones. Efficient magnet structures, minimum mechanical and acoustical damping, and minimum volume of acoustical coupling are required for this purpose. In some earphone applications overall efficiency is less critical, and wearer comfort is important.

Insert earphones need less efficiency than external earphones because the enclosed volume is much smaller; however, they require moderate efficiency to save the amplifier batteries.

Circumaural earphones are frequently driven by amplifiers otherwise used for loudspeakers. Here efficiency is less important than power-delivering capacity.

Typically 1 mW of audio power to an earphone will produce 100 to 110 dB in a standard 6-cm^3 coupler. The same earphone will produce less sound level in an earmuff than in an ear cushion and more when coupled to an ear insert.

The shape of the enclosed volume also affects response. The farther the driver is from the eardrum the lower the frequency of standing-wave resonance. Small tube diameters produce high-frequency attenuation.

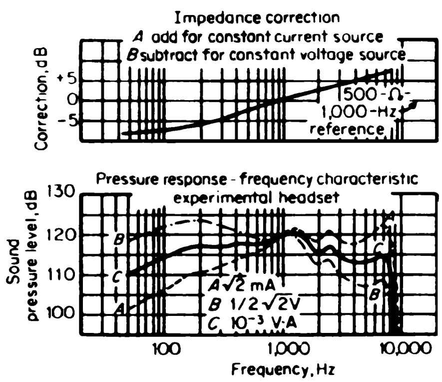


FIGURE 20.3.25 Effect of source impedance upon earphone response curve: (*a*) constant current; (*b*) constant voltage; (*c*) constant power.[9]

The response-frequency characteristic of moving-iron or piezoelectric earphones is quite dependent on source impedance. A moving-iron earphone having uniform response when driven at constant power will have a rising response (with increasing frequency) at constant current and a falling response at constant voltage (Fig. 20.3.25).

Real-Ear Response

The variety of earphone-coupling methods and the variability of outer-ear geometry (among different listeners) make response data from artificial ears only indicative, not definitive. Out of necessity a real-ear response-measuring technique was developed. A listener adjusts headset input to match headset loudness to an external calibrated sound wave in an anechoic chamber. From matching data at numerous frequencies an equivalent free-field sound-pressure level can be plotted for constant input to the earphone. This curve usually differs from a sound-level curve on a simple earphone coupler. The reason is that probe measurements of sound at the eardrum and outside the ear in a free field differ because of ear amplification and diffraction about the head (Fig. 20.3.26).

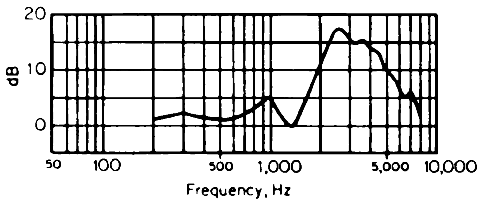


FIGURE 20.3.26 Relative level of sound pressures at the listener's eardrum and in the free sound field.[10]

Acoustic attenuation by earphones can be measured either by threshold shift or by matching the loudness of tones heard from an external loudspeaker, with and without the headset on. The sound-level difference is plotted as attenuation in decibels.

Monaural, Diotic, and Binaural Listening

A handset earphone provides monaural listening. Diotic listening with the same audio signal in both earphones localizes sound within the head. This is not unpleasant and may actually be an aid to concentration. In natural-binaural listening the ears receive sound differently from the same source unless it is directly on the listening axis. Usually there are differences in phase, arrival time, and spectrum (because of diffraction about the head).

Recordings provide true binaural effects only if the two recording microphones are on an artificial head. Stereophonic microphones are usually separated much farther, so that headset listening gives an exaggerated effect. For some listeners this is an enhancement.

REFERENCES

1. Bauer, B. B. *Proc. IRE*, 1962, Vol. 50, 50th Anniversary Issue, p. 719.
2. Olson, H. F. "Musical Engineering," McGraw-Hill, 1952.
3. Gander, M. R. Fifty Years of Loudspeaker Developments as Viewed Through the Perspective of the Audio Engineering Society, *J. Audio Eng. Soc.*, 1998, Vol. 46, No. 1/2, pp. 43–58.
4. Olson, H. F. *Proc. IRE*, 1962, Vol. 50, 50th Anniversary Issue, p. 730.
5. Beranek, L. L. "Acoustics," Acoustical Society of America, 1986.
6. Olson, H. F. "Elements of Acoustical Engineering," Van Nostrand, 1947.
7. Beranek, L. L. *Proc. IRE*, 1962, Vol. 50, p. 767.
8. Anderson, L. J. *J. Soc. Motion Pict. Eng.*, 1941, Vol. 37, p. 319.
9. Martin, D. W., and L. J. Anderson *J. Acoust. Soc. Am.*, 1947, Vol. 19, p. 63.
10. Wiener, F. M., and D. A. Ross *J. Acoust. Soc. Am.*, 1946, Vol. 18, p. 401.