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HARRY F. OLSON

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Preface to the Second Edition

Many and varied advances have been made in musical engineering in the 13 years since the appearance of the first edition. The most important developments have been made in the field of sound reproduction, subjective acoustics, and electronic music.

The most recent, significant, and important advance in the field of sound reproduction is the large-scale commercialization of stereophonic sound in the consumer or mass market complex. Stereophonic sound provides auditory perspective of the reproduced sound and thereby preserves a subjective illusion of the spatial distribution of the original sound sources. Three mediums for stereophonic sound reproduction have been commercialized on a large scale for the consumer's use in the home, namely, magnetic tape, disc and frequency-modulation radio.

The main purpose of sound reproduction is to provide the listener with the highest order of artistic and subjective resemblance to the condition of the live rendition. To achieve this objective requires the utmost degree of excellence of the physical performance of the equipment as directed by the psychological factors involved. The state of the art in sound reproduction has advanced to a stage where a high order of physical performance can be obtained. Up until very recent times, the application of the psychological characteristics as related to sound reproduction have lagged behind the physical considerations. The commercialization of stereophonic sound in the consumer complex has hastened the work in the subjective aspects of sound reproduction.

The properties of a musical tone described in the first edition of *Musical Engineering* remain the same. Once a tone has been described it is possible to produce this tone by purely electronic means. Thus, it will be seen that any tone produced by a voice or musical instrument can be produced by an electronic system. In addition, electronic synthesizers provide the capability of creating musical tones which cannot be produced by

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the melody. A chord consists of a series of three, four, five, six, or seven tones, arranged one above the other and bearing a harmonic relation to each other. The lowest note of a chord is numbered with what is termed a root as the reference point. The smallest chord is a triad consisting of three tones. Triads may be classified as follows: major, minor, diminished, and augmented.

A triad may be built on any scale tone as, for example, on C: the major triad on C is C-E-G, the minor triad on C is C-E $\flat$ -G, the diminished triad on C is C-E $\flat$ -G $\flat$, and the augmented triad on C is C-E-G $\sharp$; or on G: the major triad on G is G-B-D, the minor triad on G is G-B $\flat$ -D, the diminished triad on G is G-B $\flat$ -D $\flat$, and the augmented triad on G is G-B-D $\sharp$.

Harmony is a factor which must be considered in the process of composition. Harmony refers to tones sounding simultaneously, that is, chords, as contrasted to tones sounding consecutively as in the melody. Harmony is the process of fitting suitable chords to the melody employing chord constructions and combinations. A few of the chords have been given above. Harmony is a subject dealing with tones as related to consonance and dissonance (Sec. 7.4H). The laws of harmony have been derived by empirical and subjective means. Each era evolves new laws of harmony.

Counterpoint is the art of making two parts of a musical selection move together with such independence that each one appears to have a design of its own. It may be said to be the process of adding one or more parts or melodies to a given melody.

In the opinion of the author, it is beyond the scope of musical engineering at this time to outline the theory of music as related to the process of composition. It appears that the process of musical composition is art, rather than a science. It is an art in which the composer is a gifted person who does not obtain his creative ability to compose from a set of rules or studies. However, this does not mean that studies in the different aspects of composition are not an aid to those persons endowed with the creative ability to compose. The ultimate objective of a composer is the production of a composition which, when rendered, will be acceptable to a certain cross section of the population.

CHAPTER FOUR

Resonators and Radiators

4.1 INTRODUCTION

The musical tones used in Western music consist of certain definite and discrete fundamental frequencies. Therefore, musical instruments must be capable of producing these tones and discriminate against other tones. Highly resonant vibrating systems are excited by driving frequencies which correspond to the resonant frequencies of the system. There are a large number of vibrating systems which exhibit resonant characteristics that are suitable for use in musical instruments. These systems include strings, bars, membranes, plates, pipes, horns, and electronic generators. In some of these systems, the resonant frequencies are fixed, and in others the resonant frequencies are variable. The resonant systems are provided with radiating means for producing sound waves in air which correspond to the mechanical vibrations. It is the purpose of this chapter to describe resonant vibrating systems used in musical instruments to produce musical sound waves in the surrounding air.

As an aid in describing the action of vibrating systems in musical instruments, it is useful to employ analogies between electrical circuits and these vibrating systems. Therefore, in this book dynamical analogies will be employed to describe the action of the vibrating systems of musical instruments. As a consequence, one of the first considerations in developing the theory and action of vibrating systems will be a brief development of dynamical analogies between electrical, mechanical, and acoustical systems.

4.2 DYNAMICAL ANALOGIES^{1,2}

Analogies are useful when it is desired to compare an unfamiliar system with one that is better known. The relations and actions are more easily visualized, the mathematics more readily applied, and the analytical

¹ Olson, H. F., *Dynamical Analogies*, D. Van Nostrand Company, Inc., Princeton 1958.

² Olson, H. F., *Acoustical Engineering*, D. Van Nostrand Company, Inc., Princeton, 1957.

solutions more readily obtained in the familiar system. Analogies make it possible to extend the line of reasoning into unexplored fields.

All the analysis in musical engineering is concerned with vibrating systems. Although not generally so considered, the electrical circuit is the most common example and the most widely exploited vibrating system. The electrical circuit is a vibrating system in which the kinetic energy, potential energy, and dissipation may be expressed by general dynamic equations. This immediately suggests analogies between electrical circuits and other dynamical systems as, for example, mechanical and acoustical vibrating systems.

In view of the tremendous amount of study which has been directed toward the solution of circuits and the engineer's familiarity with electrical circuits, it seems logical to apply this knowledge to the solution of vibrating problems in the field of musical engineering by the same theory as that used in the solution of electrical circuits.

It is the purpose of this section to develop the analogies between elements in electrical, mechanical, and acoustical systems.

A. Definitions

A few of the terms used in dynamical analogies will be defined here.

Abvolt. An abvolt is the unit of electromotive force.

Instantaneous Electromotive Force. The instantaneous electromotive force between two points is the total instantaneous electromotive force. The unit is the abvolt.

Effective Electromotive Force. The effective electromotive force is the root mean square of the instantaneous electromotive force over a complete cycle between two points. The unit is the abvolt.

Maximum Electromotive Force. The maximum electromotive force for any given cycle is the maximum absolute value of the instantaneous electromotive force during that cycle. The unit is the abvolt.

Peak Electromotive Force. The peak electromotive force for any specified time interval is the maximum absolute value of the instantaneous electromotive force during that interval. The unit is the abvolt.

Dyne. A dyne is the unit of force or mechanomotive force.

Instantaneous Force (Instantaneous Mechanomotive Force). The instantaneous force at a point is the total instantaneous force. The unit is the dyne.

Effective Force (Effective Mechanomotive Force). The effective force is the root mean square of the instantaneous force over a complete cycle. The unit is the dyne.

Maximum Force (Maximum Mechanomotive Force). The maximum force for any given cycle is the maximum absolute value of the instantaneous force during that interval. The unit is the dyne.

Peak Force (Peak Mechanomotive Force). The peak force for any specified time interval is the maximum absolute value of the instantaneous force during that interval. The unit is the dyne.

Dyne per Square Centimeter. A dyne per square centimeter is the unit of sound pressure.

Static Pressure. The static pressure is the pressure that would exist in a medium with no sound waves present. The unit is the dyne per square centimeter.

Instantaneous Sound Pressure (Instantaneous Acoustomotive Force). The instantaneous sound pressure at a point is the total instantaneous pressure at the point minus the static pressure. The unit is the dyne per square centimeter.

Effective Sound Pressure (Effective Acoustomotive Force). The effective sound pressure at a point is the root-mean-square value of the instantaneous sound pressure over a complete cycle at the point. The unit is the dyne per square centimeter.

Maximum Sound Pressure (Maximum Acoustomotive Force). The maximum sound pressure for any given cycle is the maximum absolute value of the instantaneous sound pressure during that cycle. The unit is the dyne per square centimeter.

Peak Sound Pressure (Maximum Acoustomotive Force). The peak sound pressure for any specified time interval is the maximum absolute value of the instantaneous sound pressure in that interval. The unit is the dyne per square centimeter.

Abampere. An abampere is the unit of current.

Instantaneous Current. The instantaneous current at a point is the total instantaneous current at that point. The unit is the abampere.

Effective Current. The effective current at a point is the root-mean-square value of the instantaneous current over a complete cycle at that point. The unit is the abampere.

Maximum Current. The maximum current for any given cycle is the maximum absolute value of the instantaneous current during that cycle. The unit is the abampere.

Peak Current. The peak current for any specified time interval is the maximum absolute value of the instantaneous current in that interval. The unit is the abampere.

Centimeter per Second. A centimeter per second is the unit of velocity.

Instantaneous Velocity. The instantaneous velocity at a point is the total instantaneous velocity at that point. The unit is the centimeter per second.

Effective Velocity. The effective velocity at a point is the root-mean-square value of the instantaneous velocity over a complete cycle at that point. The unit is the centimeter per second.

Maximum Velocity. The maximum velocity for any given cycle is the maximum absolute value of the instantaneous velocity during that cycle. The unit is the centimeter per second.

Peak Velocity. The peak velocity for any specified time interval is the maximum absolute value of the instantaneous velocity in that interval. The unit is the centimeter per second.

Cubic Centimeter per Second. A cubic centimeter per second is the unit of volume current.

Instantaneous Volume Current. The instantaneous volume current at a point is the total instantaneous volume current at that point. The unit is the cubic centimeter per second.

Effective Volume Current. The effective volume current at a point is the root-mean-square value of the instantaneous volume current over a complete cycle at that point. The unit is the cubic centimeter per second.

Maximum Volume Current. The maximum volume current for any given cycle is the maximum absolute value of the instantaneous volume current during that cycle. The unit is the cubic centimeter per second.

Peak Volume Current. The peak volume current for any specified time interval is the maximum absolute value of the instantaneous volume current in that interval. The unit is the cubic centimeter per second.

Electrical Impedance. Electrical impedance is the complex quotient of the alternating electromotive force applied to the system by the resulting current. The unit is the abohm.

Electrical Resistance. Electrical resistance is the real part of the electrical impedance. This is the part responsible for the dissipation of energy. The unit is the abohm.

Electrical Reactance. Electrical reactance is the imaginary part of the electrical impedance. The unit is the abohm.

Inductance. Inductance in an electrical system is that coefficient which, when multiplied by 2π times the frequency, gives the positive imaginary part of the electrical impedance. The unit is the abhenry.

Electrical Capacitance. Electrical capacitance in an electrical system is that coefficient which, when multiplied by 2π times the frequency, is the reciprocal of the negative imaginary part of the electrical impedance. The unit is the abfarad.

Mechanical Impedance. Mechanical impedance is the complex quotient of the alternating force applied to the system by the resulting linear velocity in the direction of the force at its point of application. The unit is the mechanical ohm.

Mechanical Resistance. Mechanical resistance is the real part of the mechanical impedance. This is the part responsible for the dissipation of energy. The unit is the mechanical ohm.

Mechanical Reactance. Mechanical reactance is the imaginary part of the mechanical impedance. The unit is the mechanical ohm.

Mass. Mass in a mechanical system is that coefficient which, when multiplied by 2π times the frequency, gives the positive imaginary part of the mechanical impedance. The unit is the gram.

Compliance. Compliance in a mechanical system is that coefficient which, when multiplied by 2π times the frequency, is the reciprocal of the negative imaginary part of the mechanical impedance. The unit is the centimeter per dyne.

Acoustical Impedance. Acoustical impedance is the complex quotient of the alternating pressure applied to the system by the resulting volume current. The unit is the acoustical ohm.

Acoustical Resistance. Acoustical resistance is the real part of the acoustical impedance. This is the part responsible for the dissipation of energy. The unit is the acoustical ohm.

Acoustical Reactance. Acoustical reactance is the imaginary part of the acoustical impedance. The unit is the acoustical ohm.

Inertance. Inertance in an acoustical system is that coefficient which, when multiplied by 2π times the frequency, gives the positive imaginary part of the acoustical impedance. The unit is the gram per centimeter to the fourth power.

Acoustical Capacitance. Acoustical capacitance in an acoustical system is that coefficient which, when multiplied by 2π times the frequency is the reciprocal negative imaginary part of the acoustical impedance. The unit is the centimeter to the fifth power per dyne.

Element. An element or circuit parameter in an electrical system defines a distinct activity in its part of the circuit. In the same way, an element in a mechanical or acoustical system defines a distinct activity in its part of the system. The elements in an electrical circuit are electrical resistance, inductance, and electrical capacitance. The elements in a mechanical system are mechanical resistance, mass, and compliance. The elements in an acoustical system are acoustical resistance, inertance, and acoustical capacitance.

Electrical System. An electrical system is a system adapted for the transmission of electrical currents consisting of one or all of the following electrical elements: electrical resistance, inductance, and electrical capacitance.

Mechanical System. A mechanical system is a system adapted for the transmission of vibrations consisting of one or all of the following mechanical elements: mechanical resistance, mass, and compliance.

Acoustical System. An acoustical system is a system adapted for the transmission of sound consisting of one or all of the following acoustical

elements: acoustical resistance, inertance, and acoustical capacitance.

Electrical Abohm. An electrical resistance, reactance, or impedance is said to have a magnitude of one abohm when an electromotive force of one abvolt produces a current of one abampere.

Mechanical Ohm. A mechanical resistance, reactance, or impedance is said to have a magnitude of one mechanical ohm when a force of one dyne produces a velocity of one centimeter per second.

Acoustical Ohm. An acoustical resistance, reactance, or impedance is said to have a magnitude of one acoustical ohm when a pressure of one dyne per square centimeter produces a volume current of one cubic centimeter per second.

B. Elements

An element or circuit parameter in an electrical system defines a distinct activity in its part of the circuit. In an electrical system these elements are resistance, inductance, and capacitance. As indicated in the introduction to this chapter, the study of mechanical and acoustical systems is facilitated by the introduction of elements analogous to the elements in an electrical circuit. In this procedure, the first step is to develop the elements in these vibrating systems. It is the purpose of the section which follows to describe electrical, mechanical, and acoustical elements.

C. Resistance

1. Electrical Resistance. Electrical energy is changed into heat by the passage of an electrical current through a resistance. Energy is lost by the system when a charge q of electricity is driven through a resistance by a voltage e . Resistance is the circuit element which causes dissipation.

Electrical resistance r_E , in abohms, is defined as

$$r_E = \frac{e}{i} \quad (4.1)$$

where e = voltage across the electrical resistance, in abvolts

i = current through the electrical resistance, in abamperes

Equation (4.1) states that the electromotive force across an electrical resistance is proportional to the electrical resistance and the current.

2. Mechanical Resistance. Mechanical energy is changed into heat by a rectilinear motion which is opposed by linear resistance (friction). In a mechanical system, dissipation is due to friction. Energy is lost by the system when a mechanical resistance is displaced a distance x by a force f_M .

Mechanical resistance r_M , in mechanical ohms, is defined as

$$r_M = \frac{f_M}{u} \quad (4.2)$$

where f_M = applied force, in dynes

u = velocity at the point of application of the force, in centimeters per second

Equation (4.2) states that the driving force applied to a mechanical resistance is proportional to the mechanical resistance and the linear velocity.

3. Acoustical Resistance. In an acoustical system, dissipation may be due to the fluid resistance or radiation resistance. At this point the former type of acoustical resistance will be considered. Acoustical energy is changed into heat by the passage of a fluid through an acoustical resistance. The resistance is due to viscosity. Energy is lost by the system when a volume displacement X of a fluid or gas is driven through an acoustical resistance by a pressure p .

Acoustical resistance r_A , in acoustical ohms, is defined as

$$r_A = \frac{p}{U} \quad (4.3)$$

where p = pressure, in dynes per square centimeter

U = volume current, in cubic centimeters per second

Equation (4.3) states that the driving pressure applied to an acoustical resistance is proportional to the acoustical resistance and the volume current.

D. Inductance, Mass, Inertance

1. Inductance. Electromagnetic energy is associated with inductance. Electromagnetic energy increases as the current in the inductance increases. It decreases when the current decreases. It remains constant when the current in the inductance is a constant. Inductance is the electrical circuit element which opposes a change in current. Inductance L , in abhenrys, is defined as

$$e = L \frac{di}{dt} \quad (4.4)$$

where e = electromotive or driving force, in abvolts

di/dt = rate of change of current, in abamperes per second

Equation (4.4) states that the electromotive force across an inductance is proportional to the inductance and the rate of change of current.

2. Mass. Mechanical inertial energy is associated with mass in the mechanical system. Mechanical energy increases as the linear velocity

of a mass increases, that is, during linear acceleration. It decreases when the velocity decreases. It remains constant when the velocity is constant. Mass is the mechanical element which opposes a change of velocity. Mass m , in grams, is defined as

$$f_M = m \frac{du}{dt} \quad (4.5)$$

where du/dt = acceleration, in centimeters per second per second

f_M = driving force, in dynes

Equation (4.5) states that the driving force applied to the mass is proportional to the mass and the rate of change of velocity.

3. Inertance. Acoustical inertial energy is associated with inertance in the acoustical system. Acoustical energy increases as the volume current of an inertance increases. It decreases when the volume current decreases. It remains constant when the volume current of the inertance is a constant. Inertance is the acoustical element that opposes a change in volume current. Inertance M , in grams per (centimeter)⁴ is defined as

$$p = M \frac{dU}{dt} \quad (4.6)$$

where M = inertance, in grams per (centimeter)⁴

dU/dt = rate of change of volume current, in cubic centimeters per second per second

p = driving pressure, in dynes per square centimeter

Equation (4.6) states that the driving pressure applied to an inertance is proportional to the inertance and the rate of change of volume current.

Inertance may be expressed as

$$M = \frac{m}{S^2} \quad (4.7)$$

where m = mass, in grams

S = cross-sectional area, in square centimeters, over which the driving pressure acts to drive the mass

The inertance of the circular tube is

$$M = \frac{\rho l}{\pi R^2} \quad (4.8)$$

where R = radius of the tube, in centimeters

l = effective length of the tube, that is, length plus end correction, in centimeters

ρ = density of the medium in the tube, in grams per cubic centimeter

E. Electrical Capacitance, Compliance, Acoustical Capacitance

1. Electrical Capacitance. Electrostatic energy is associated with the separation of positive and negative charges, as in the case of the charges on the two plates of an electrical capacitance. Electrostatic energy increases as the charge of opposite polarity are separated. It is constant and stored when the charges remain unchanged. It decreases as the charges are brought together and the electrostatic energy is released. Electrical capacitance is the electrical-circuit element which opposes a change in voltage. Electrical capacitance C_E , in abfarads, is defined as

$$i = C_E \frac{de}{dt} \quad (4.9)$$

Equation (4.9) may be written

$$e = \frac{1}{C_E} \int i dt = \frac{q}{C_E} \quad (4.10)$$

where q = charge on electrical capacitance, in abcoulombs

e = electromotive force, in abvolts

Equation (4.10) states that the charge on an electrical capacitance is proportional to the electrical capacitance and the applied electromotive force.

2. Compliance. Mechanical potential energy is associated with the compression of a spring or compliant element. Mechanical energy increases as the spring is compressed. It decreases as the spring is allowed to expand. It is constant and is stored when the spring remains immovably compressed. Compliance is the mechanical element which opposes a change in the applied force. Compliance C_M , in centimeters per dyne, is defined as

$$f_M = \frac{x}{C_M} \quad (4.11)$$

where x = displacement, in centimeters

f_M = applied force, in dynes

Equation (4.11) states that the displacement of a compliance is proportional to the compliance and the applied force.

Stiffness is the reciprocal of compliance.

3. Acoustical Capacitance. Acoustical potential energy is associated with the compression of a fluid or gas. Acoustical energy increases as the gas is compressed. It decreases as the gas is allowed to expand. It is constant and is stored when the gas remains immovably compressed. Acoustical capacitance is the acoustic element which opposes a change

in the applied pressure. The pressure,³ in dynes per square centimeter, in terms of the condensation, is

$$p = c^2 \rho s \quad (4.12)$$

where c = velocity, in centimeters per second

ρ = density, in grams per cubic centimeter

s = condensation, defined in Eq. (4.13)

The condensation in a volume V due to a change in volume from V to V' is

$$s = \frac{V - V'}{V} \quad (4.13)$$

The change in volume $V - V'$, in cubic centimeters, is equal to the volume displacement, in cubic centimeters.

$$V - V' = X \quad (4.14)$$

where X = volume displacement, in cubic centimeters

From Eqs. (4.12), (4.13), (4.14), the pressure is

$$p = \frac{\rho c^2}{V} X \quad (4.15)$$

Acoustical capacitance C_A , in (centimeters)⁵ per dyne, is defined as

$$p = \frac{X}{C_A} \quad (4.16)$$

where p = sound pressure, in dynes per square centimeter

X = volume displacement, in cubic centimeters

Equation (4.16) states the volume displacement in an acoustical capacitance is proportional to the pressure and the acoustical capacitance.

From Eqs. (4.15) and (4.16) the acoustical capacitance of a volume is

$$C_A = \frac{V}{\rho c^2} \quad (4.17)$$

where V = volume, in cubic centimeters

F. Representation of Electrical, Mechanical, and Acoustical Elements

Electrical, mechanical, and acoustical elements have been defined in the preceding sections. Figure 4.1 illustrates schematically the four elements in each of the four systems.

The electrical elements, electrical resistance, inductance, and electrical capacitance, are represented by the conventional symbols.

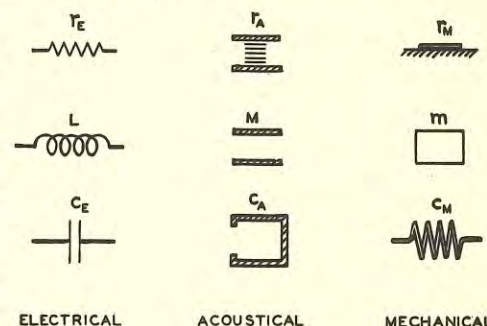
³ Olson, H. F., *Acoustical Engineering*, D. Van Nostrand Company, Inc., Princeton, 1957.

Mechanical resistance is represented by sliding friction which causes dissipation. Acoustical resistance is represented by narrow slits which cause dissipation due to viscosity when fluid is forced through the slits. These elements are analogous to electrical resistance in the electrical system.

Inertia in the mechanical system is represented by a mass. Inertance in the acoustical system is represented as the fluid contained in a tube in which all the particles move with the same phase when actuated by a force due to pressure. These elements are analogous to inductance in the electrical system.

Compliance in the mechanical system is represented as a spring. Acoustical capacitance in the acoustical system is represented as a volume which acts as a stiffness or spring element. These elements are analogous to electrical capacitance in the electrical system.

FIG. 4.1. Graphical representation of the three basic elements in electrical, mechanical, and acoustical systems. In the electrical system, r_E = electrical resistance, L = inductance, C_E = electrical capacitance. In the mechanical system, r_M = mechanical resistance, m = mass, C_M = compliance. In the acoustical system, r_A = acoustical resistance, M = inertance, C_A = acoustical capacitance.



4.3 ELECTRICAL, MECHANICAL, AND ACOUSTICAL SYSTEM OF ONE DEGREE OF FREEDOM^{4,5}

A resonant system must contain at least one element in which kinetic energy may be stored and another element in which potential energy may be stored. At resonance, energy flows from one type of element to the other, and vice versa. The simplest resonators in electrical, mechanical, and acoustical systems are shown in Fig. 4.2. The element in which kinetic energy may be stored in each of these systems is inductance, mass, and inertance. The element in which potential energy may be stored in each of these systems is electrical capacitance, compliance, and acoustical capacitance. A practical vibrating system without dissipation is impossible of attainment. The element in which dissipation occurs in each of

⁴ Olson, H. F., *Dynamical Analogies*, D. Van Nostrand Company, Inc., Princeton, 1958.

⁵ Olson, H. F., *Acoustical Engineering*, D. Van Nostrand Company, Inc., Princeton, 1957.

the systems is termed electrical resistance, mechanical resistance, and acoustical resistance.

The current, velocity, and volume current in each of the systems is Electrical:

$$\dot{q} = i = \frac{E e^{j\omega t}}{r_E + j\omega L - \frac{j}{\omega C_E}} = \frac{e}{z_E} \quad (4.18)$$

Mechanical:

$$\dot{x} = \frac{F e^{j\omega t}}{r_M + j\omega m - \frac{j}{\omega C_M}} = \frac{f_M}{z_M} \quad (4.19)$$

Acoustical:

$$\dot{X} = \frac{P e^{j\omega t}}{r_A + j\omega M - \frac{j}{\omega C_A}} = \frac{p}{z_A} \quad (4.20)$$

The current, velocity, or volume-current response to an electromotive force, force, or pressure as a function of the frequency is shown in Fig. 4.2.

The vector electrical impedance is

$$z_E = r_E + j\omega L - \frac{j}{\omega C_E} \quad (4.21)$$

The vector mechanical impedance is

$$z_M = r_M + j\omega m - \frac{j}{\omega C_M} \quad (4.22)$$

The vector acoustical impedance is

$$z_A = r_A + j\omega M - \frac{j}{\omega C_A} \quad (4.23)$$

For a certain value of L and C_E , m and C_M , and M and C_A there will be a

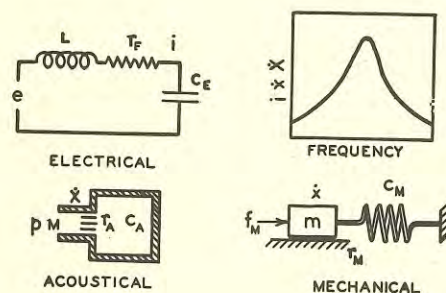


FIG. 4.2. Electrical, mechanical, and acoustical systems of 1 degree of freedom and the current, velocity, and volume-current response characteristics. In the electrical system, e = electromotive force, L = inductance, C_E = electrical capacitance, r_E = electrical resistance, i = current. In the mechanical system, f = force, m = mass, C_M = compliance, r_M = mechanical resistance, $\dot{x}$ = velocity. In the acoustical system, p = pressure, M = inertance, C_A = acoustical capacitance, r_A = acoustical resistance, $\dot{X}$ = volume current.

certain frequency at which the imaginary component of the impedance is zero. This frequency is called the resonant frequency. At this frequency, the ratio of the current to the applied voltage, or the ratio of the velocity to the applied force, or the ratio of the volume current to the applied pressure is a maximum. At the resonant frequency the current and voltage, the velocity and force, and the volume current and pressure are in phase.

The resonant frequency f_r , in the three systems, is Electrical:

$$f_r = \frac{1}{2\pi \sqrt{LC_E}} \quad (4.24)$$

Mechanical:

$$f_r = \frac{1}{2\pi \sqrt{mC_M}} \quad (4.25)$$

Acoustical:

$$f_r = \frac{1}{2\pi \sqrt{MC_A}} \quad (4.26)$$

From the equations in this section it will be seen that the current, velocity, and volume current increase as the magnitude of the dissipative or resistive element decreases. The sharpness of the resonance also increases as the resistance is decreased. The response characteristics of the resonant systems for different values of the ratio

$$Q_r = \frac{\omega_r L}{r_E} = \frac{\omega_r m}{r_M} = \frac{\omega_r M}{r_A} \quad (4.27)$$

where $\omega_r = 2\pi f_r$

f_r = resonant frequency

and as a function of $f \div f_r$ are shown in Fig. 4.3.

4.4 HELMHOLTZ RESONATOR

The Helmholtz resonator consists of an enclosed volume of air coupled to the outside free air by means of an aperture (Fig. 4.4). The Helmholtz resonator, being a system of one degree of freedom, is the simplest type of resonant system used in musical instruments. The resonant system of the whistle, ocarina, or celesta or the body of the guitar, violin, etc., constitutes a Helmholtz resonator.

Perspective and sectional views and the acoustical circuit of a Helmholtz resonator are shown in Fig. 4.4. Since the dimensions of the resonator and the hole for the fundamental resonant frequency are small compared to the wavelength in the systems in which this type of resonator is effectively employed, the lumped-constant representation may be used

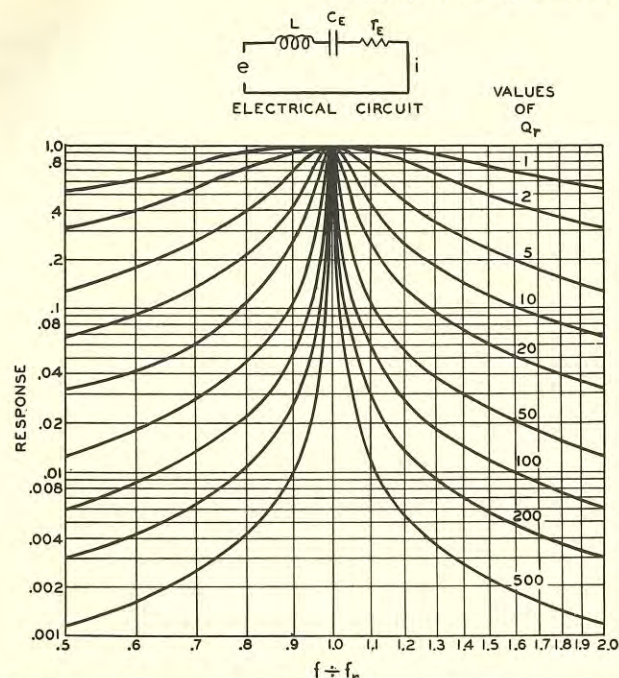


FIG. 4.3. The current-response characteristic of a simple circuit as a function of the ratio f/f_r , where f_r = the resonant frequency and f = the frequency under consideration. The numbers refer to the value Q_r , $Q_r = 2\pi f_r L/r_E$. The above characteristics are applicable to acoustical and mechanical systems by the substitution of the elements and quantities which are analogous to the electrical system. (After Olson, *Acoustical Engineering*, D. Van Nostrand Company, Inc., Princeton, 1957.)

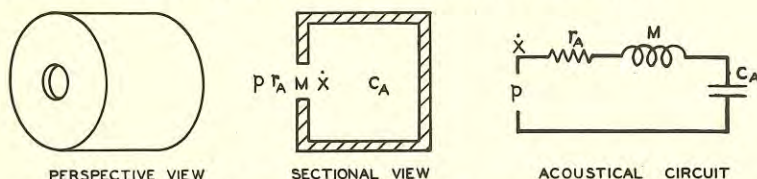


FIG. 4.4. Perspective and sectional views and the acoustical circuit of a Helmholtz resonator. In the acoustical circuit, p = actuating pressure, r_A = acoustical radiation resistance, M = inertance of the aperture, C_A = acoustical capacitance of the volume, $\dot{X}$ = volume current.

to describe the performance of the system. The expressions for the constants of the resonator in terms of the dimensions are as follows:

The acoustical radiation resistance r_A , in acoustical ohms, is given by

$$r_A = \frac{\rho \omega^2}{4\pi c} \quad (4.28)$$

where ρ = density of air, in grams per cubic centimeter

c = velocity of sound, in centimeters per second

$\omega = 2\pi f$

f = frequency, in cycles per second

The inertance M , in grams per (centimeter)⁴, for a circular orifice is given by

$$M = \frac{\rho(l + 1.7R)}{\pi R^2} \quad (4.29)$$

where ρ = density of air, in grams per cubic centimeter

l = thickness of the boundary wall of the orifice, in centimeters

R = radius of the orifice, in centimeters

The acoustical capacitance C_A , in (centimeters)⁵ per dyne, for the enclosed volume is given by

$$C_A = \frac{V}{\rho c^2} \quad (4.30)$$

where ρ = density of air, in grams per cubic centimeter

c = velocity of sound, in centimeters per second

V = volume of the enclosure, in cubic centimeters

The volume current $\dot{X}$, in cubic centimeters per second, through the aperture is given by

$$\dot{X} = \frac{p}{r_A + j\left(\omega M - \frac{1}{\omega C_A}\right)} \quad (4.31)$$

where p = actuating pressure, in dynes per square centimeter

It will be seen that the volume current is a maximum when

$$\omega = \sqrt{\frac{1}{MC_A}} \quad (4.32)$$

This is the resonant frequency of the resonator.

In some of the acoustical systems the orifice is not circular. Under these conditions, Eq. (4.29) does not apply. An approximate expression for the inertance of an orifice of any shape is given by

$$M = \frac{\rho(l + \sqrt{A})}{A} \quad (4.33)$$

where ρ = density of air, in grams per cubic centimeter

l = thickness of the boundary wall of the orifice, in centimeters

A = area of the orifice, in square centimeters

The shapes of the response characteristics of the Helmholtz resonator are the same as those of the system for a single degree of freedom shown in Fig. 4.3.

4.5 STRINGS⁶⁻¹⁰

Strings are used as the resonant system in many musical instruments. In string musical instruments the transverse and not the longitudinal vibrations are used. In the transverse vibrations, each particle of the string vibrates in a plane perpendicular to the line of the string. It is assumed that the mass per unit length is a constant, that it is perfectly flexible (the stiffness being negligible), and that it is connected to massive

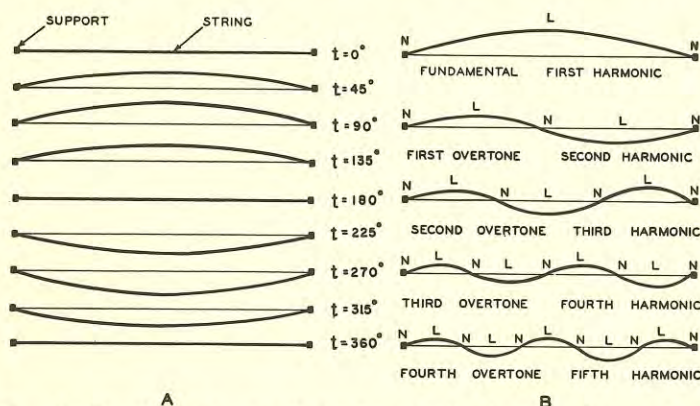


FIG. 4.5. A. The shape of a stretched string vibrating in the fundamental mode for eight equal intervals of a complete cycle. B. The first five modes of vibration of a stretched string. The nodes and loops are indicated by N and L.

nonyielding supports (Fig. 4.5). Since the string is fixed at the supports, nodes will occur at these points. The fundamental resonant frequency of the string is given by

$$f_r = \frac{1}{2l} \sqrt{\frac{T}{m}} \quad (4.34)$$

where T = tension, in dynes

m = mass per unit length, in grams

l = length of the string, in centimeters

The relative positions of a string vibrating in the fundamental mode for a complete cycle are shown in Fig. 4.5A. The shape of the string for all positions during a cycle is sinusoidal.

⁶ Rayleigh, *Theory of Sound*, Macmillan & Co., Ltd., London, 1926.

⁷ Crandall, *Theory of Vibrating Systems and Sound*, D. Van Nostrand Company, Inc., Princeton, 1926.

⁸ Wood, *A Text Book of Sound*, George Bell & Sons, Ltd., London, 1930.

⁹ Morse, *Vibration and Sound*, McGraw-Hill Book Company, Inc., New York, 1948.

¹⁰ Lamb, *Dynamical Theory of Sound*, Edward Arnold & Co., London, 1931.

In addition to the fundamental, other modes of vibration may occur, the frequencies being 2, 3, 4, 5, etc., times the fundamental. In other words, the resonant vibration frequencies of a string correspond to the terms in the harmonic series. The first few modes of vibration of a string are shown in Fig. 4.5B. The points which are at rest are termed nodes and are marked N. The points between the nodes where the amplitude is a maximum are termed antinodes or loops and are marked L.

The string can be represented by a lumped-constant analogy shown in Fig. 4.6. The analogy is a multiresonant system. The resonant frequencies are determined by the constants of the system.

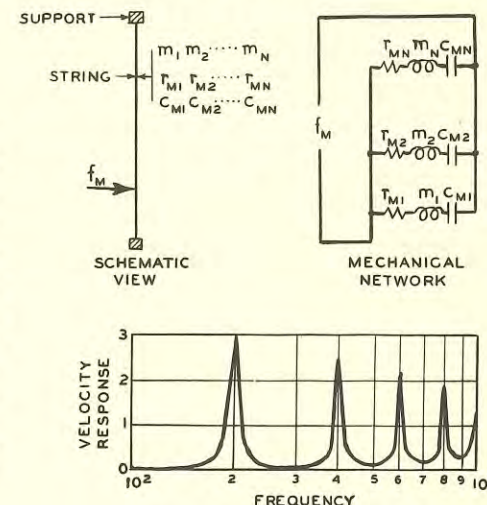


FIG. 4.6. The vibrating system, the mechanical network, and a velocity-response frequency characteristic of a string. In the mechanical system, $m_1, m_2, m_3, \dots, m_N$; $C_{M1}, C_{M2}, C_{M3}, \dots, C_{MN}$; and $r_{M1}, r_{M2}, r_{M3}, \dots, r_{MN}$ represent the masses, the compliances, and the mechanical resistances of the different modes of vibration; f_M = driving force. The velocity-response frequency characteristic shown depicts the response as a function of frequency of a typical string.

The response-frequency characteristic of a string is shown in Fig. 4.6. It will be seen that the harmonics are integral multiples of the fundamental.

Strings are used as the resonant system in the violin, viola, violon cello, contrabass, banjo, guitar, ukulele, mandolin, zither, piano, and many other string musical instruments.

4.6 TRANSVERSE VIBRATION OF BARS¹¹⁻¹⁴

In the preceding section the perfectly flexible string was considered where the restoring force due to stiffness is negligible compared with that due to

¹¹ Rayleigh, *Theory of Sound*, Macmillan & Co., Ltd., London, 1926.

¹² Wood, *A Text Book of Sound*, George Bell & Sons, Ltd., London, 1930.

¹³ Morse, *Vibration and Sound*, McGraw-Hill Book Company, Inc., New York, 1948.

¹⁴ Lamb, *Dynamical Theory of Sound*, Edward Arnold & Co., London, 1931.

tension. The bar under no tension is the other limiting case, the restoring force being entirely due to stiffness. For the cases to be considered it is assumed that the bars are straight, the cross section is uniform and symmetrical about a central plane, and, as in the case of the string, only the transverse vibrations will be considered.

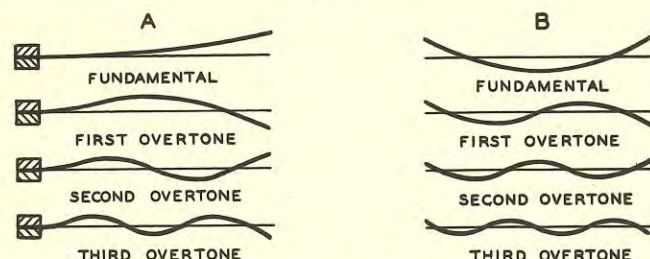


FIG. 4.7. Modes of transverse vibrations of bars. A. A bar clamped at one end and free at the other. B. A bar free at both ends.

A. Bar Clamped at One End

Consider a bar clamped in a rigid support at one end with the other end free (Fig. 4.7A). The fundamental frequency is given by

$$f_1 = \frac{0.5596}{l^2} \sqrt{\frac{QK^2}{\rho}} \quad (4.35)$$

where l = length of the bar, in centimeters

ρ = density, in grams per cubic centimeter

Q = Young's modulus, in dynes per square centimeter

K = radius of gyration

For a rectangular cross section, the radius of gyration is

$$K = \frac{a}{\sqrt{12}} \quad (4.36)$$

where a = thickness of the bar, in centimeters, in the direction of vibration

For a circular cross section,

$$K = \frac{a}{2} \quad (4.37)$$

where a = radius of the bar, in centimeters

For a hollow circular cross section,

$$K = \frac{\sqrt{a^2 + a_1^2}}{2} \quad (4.38)$$

where a = outside radius of the pipe, in centimeters

a_1 = inside radius of the pipe, in centimeters

The modes of vibration of a bar clamped at one end are shown in Fig. 4.7A. Table 4.1 gives the position of the nodes and the frequencies of the overtones. It will be seen that the overtones are not harmonics.

TABLE 4.1

No. of tone	No. of nodes	Distances of nodes from free end in terms of the length of the bar	Frequencies as a ratio of the fundamental
1	0		f_1
2	1	0.2261	$6.267f_1$
3	2	0.1321, 0.4999	$17.55f_1$
4	3	0.0944, 0.3558, 0.6439	$34.39f_1$

The first overtone of a bar or reed has a higher frequency than the sixth harmonic of a string. The reed organ, harmonica, clarinet, oboe, bassoon, saxophone, and tuning fork are examples of instruments employing a bar or reed clamped at one end.

B. Bar Free at Both Ends

Consider a perfectly free bar (Fig. 4.7B). The fundamental frequency is given by

$$f_1 = \frac{1.133\pi}{l^2} \sqrt{\frac{QK^2}{\rho}} \quad (4.39)$$

where l = length of the bar, in centimeters. All the other quantities are the same as in Eq. (4.38).

The modes of vibration of a perfectly free bar are shown in Fig. 4.7B. Table 4.2 gives the position of the nodes and the frequencies of the overtones.

TABLE 4.2

No. of tone	No. of nodes	Distances of nodes from one end in terms of the length of the bar	Frequencies as a ratio of the fundamental
1	2	0.2242, 0.7758	f_1
2	3	0.1321, 0.50, 0.8679	$2.756f_1$
3	4	0.0944, 0.3558, 0.6442, 0.9056	$5.404f_1$
4	5	0.0734, 0.277, 0.5, 0.723, 0.9266	$8.933f_1$

The most common examples of musical instruments employing a free bar are the xylophone, marimba, glockenspiel, and chimes.

4.7 STRETCHED MEMBRANES¹⁵⁻¹⁹

The ideal membrane is assumed to be flexible and very thin in cross section and stretched in all directions by a force which is not affected by the motion of the membrane. For cases of practical interest the membrane in musical instruments is assumed to be rigidly clamped and stretched by a massive surround. The considerations will be confined to circular membranes, because no other shapes are used in musical instruments.

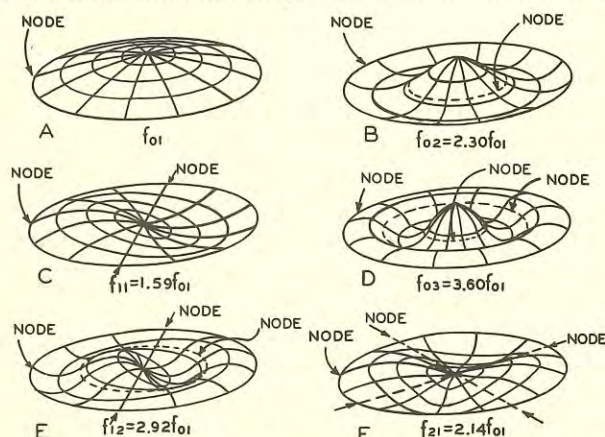


FIG. 4.8. Modes of vibration of a stretched circular membrane. (After Morse *Vibration and Sound*, McGraw-Hill Book Company, Inc., New York, 1948.)

The fundamental frequency of a circular stretched membrane is given by

$$f_{01} = \frac{0.382}{R} \sqrt{\frac{T}{m}} \quad (4.40)$$

where m = mass, in grams per square centimeter of area

R = radius of the membrane, in centimeters

T = tension, in dynes per centimeter

The fundamental vibration is with the circumference as a node and a maximum displacement at the center of the circle (Fig. 4.8A). The frequencies of the next two overtones with nodal circles are

$$f_{02} = 2.30f_{01}$$

$$f_{03} = 3.60f_{01}$$

¹⁵ Lamb, *Dynamical Theory of Sound*, Edward Arnold & Co., London, 1931.

¹⁶ Rayleigh, *Theory of Sound*, Macmillan & Co., Ltd., London, 1926.

¹⁷ Morse, *Vibration and Sound*, McGraw-Hill Book Company, Inc., New York, 1948.

¹⁸ Wood, *A Text Book of Sound*, George Bell & Sons, Ltd., London, 1930.

¹⁹ Crandall, *Theory of Vibrating Systems and Sound*, D. Van Nostrand Company, Inc., Princeton, 1926.

and are shown in Figs. 4.8B and 4.8D. The frequencies of the first and second overtones with nodal diameters are

$$f_{11} = 1.59f_{01}$$

$$f_{21} = 2.14f_{01}$$

These modes are shown in Figs. 4.8C and 4.8F. Following these simpler forms of vibration are combinations of nodal circles and nodal diameters. The frequency of one nodal circle and one nodal diameter, Fig. 4.8E, is

$$f_{12} = 2.92f_{01}$$

A stretched circular membrane is used in the banjo, tambourine, and all types of drums. In this case, the air enclosure as well as the characteristics of the membrane controls the modes of vibration.

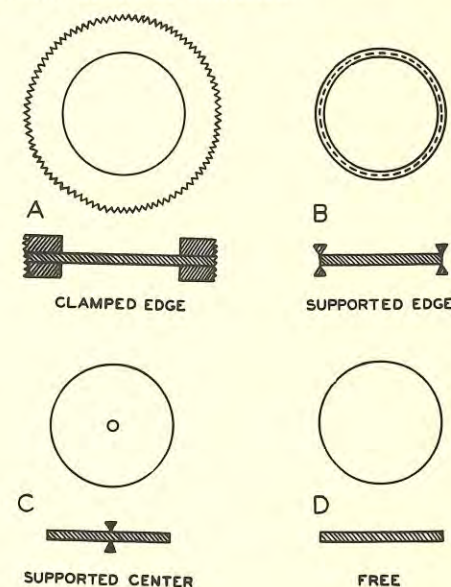


FIG. 4.9. Circular plates. A. A circular plate clamped at the edge. B. A circular plate supported at the edge. C. A circular plate supported at the center. D. A free circular plate. (After Olson, *Acoustical Engineering*, D. Van Nostrand Company, Inc., Princeton, 1957.)

4.8 CIRCULAR PLATES²⁰⁻²⁴

The circular plates shown in Fig. 4.9 are assumed to be of uniform cross section and under no tension. It is the purpose of this section to consider the vibration of circular plates for the various support means of Fig. 4.9.

²⁰ Rayleigh, *Theory of Sound*, Macmillan & Co., Ltd., London, 1926.

²¹ Morse, *Vibration of Sound*, McGraw-Hill Book Company, Inc., New York, 1948.

²² Wood, *A Text Book of Sound*, George Bell & Sons, Ltd., London, 1930.

²³ Crandall, *Theory of Vibrating Systems and Sound*, D. Van Nostrand Company, Inc., Princeton, 1926.

²⁴ Lamb, *Dynamical Theory of Sound*, Edward Arnold & Co., London, 1931.

A. Circular Clamped Plate

Consider a circular clamped plate, as shown in Fig. 4.9A. The fundamental frequency is given by

$$f_{01} = \frac{0.467t}{R^2} \sqrt{\frac{Q}{\rho(1-\sigma^2)}} \quad (4.41)$$

where t = thickness of the plate, in centimeters

R = radius of the plate up to the clamping boundary, in centimeters

ρ = density, in grams per cubic centimeters

σ = Poisson's ratio

Q = Young's modulus, in dynes per square centimeter

The fundamental frequency is with the circumference as a node and with maximum displacement at the center (Fig. 4.10A). The frequencies of the next two overtones with nodal circles, Figs. 4.10B and 4.10D, are

$$f_{02} = 3.91f_{01}$$

$$f_{03} = 8.75f_{01}$$

The frequencies of the first and second overtones with nodal diameters, Figs. 4.10C and 4.10F, are

$$f_{11} = 2.09f_{01}$$

$$f_{21} = 3.43f_{01}$$

Following these simpler forms of vibration are the combination of nodal circles and nodal diameters. The frequency of the first overtone with one nodal diameter and one nodal circle, Fig. 4.10E, is

$$f_{12} = 5.98f_{01}$$

The clamped circular plate is used in telephone receivers and small condenser microphones.

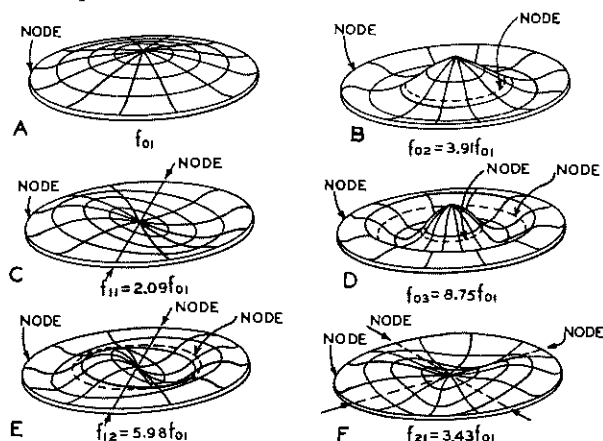


FIG. 4.10. Modes of vibration of a clamped circular plate.

B. Circular Free Plate

Consider a circular plate under no tension, uniform in cross section, and perfectly free (Fig. 4.9D). For a vibration with nodal circle, Fig. 4.11A, the frequency is

$$f = \frac{0.412t}{R^2} \sqrt{\frac{Q}{\rho(1-\sigma^2)}} \quad (4.42)$$

where t = thickness of the plate, in centimeters

R = radius of the plate, in centimeters

ρ = density, in grams per cubic centimeter

σ = Poisson's ratio

Q = Young's modulus, in dynes per square centimeter

For a vibration with two nodal diameters, Fig. 4.11B, the frequency is

$$f = \frac{0.193t}{R^2} \sqrt{\frac{Q}{\rho(1-\sigma^2)}} \quad (4.43)$$

The gong is an example of the circular free plate.

C. Circular Plate Supported at the Center

Consider a circular plate under no tension, uniform in cross section, edges perfectly free, and supported at the center (Fig. 4.9C). The frequency, for the umbrella mode, shown in Fig. 4.11C, is

$$f = \frac{0.172t}{R^2} \sqrt{\frac{Q}{\rho(1-\sigma^2)}} \quad (4.44)$$

The cymbal is an example of a circular plate supported in the center.

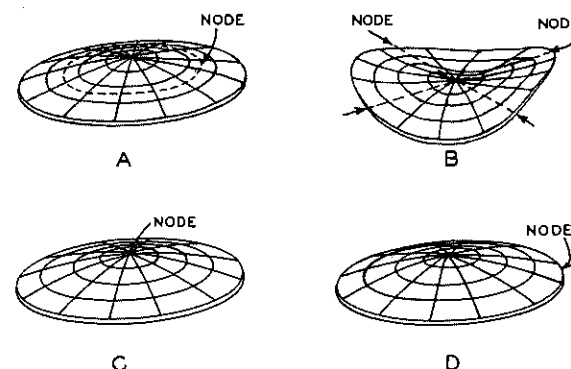


FIG. 4.11. Modes of vibrations of plates. A. Circular free plate with a circular node. B. Circular free plate with two nodal diameters. C. Circular plate supported at the center. D. Circular plate supported at the outside.

D. Circular Plate Supported at the Outside

Consider a plate under no tension, uniform in cross section, edges simply supported at the periphery (Fig. 4.9B). The fundamental frequency for a vibration with an outside edge node, shown in Fig. 4.11D, is

$$f = \frac{0.233t}{R^2} \sqrt{\frac{Q}{\rho(1-\sigma^2)}} \quad (4.45)$$

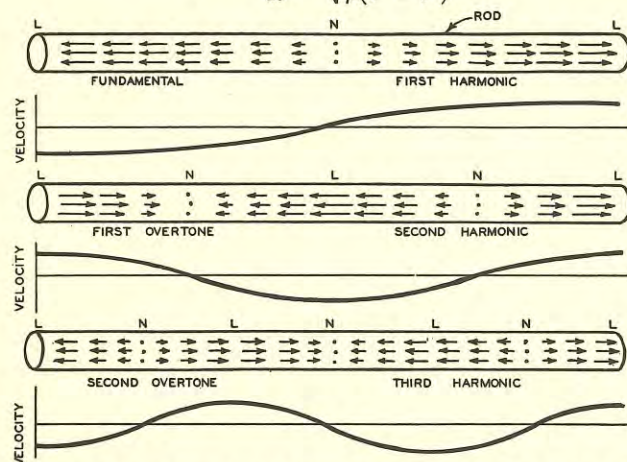


FIG. 4.12. Modes of longitudinal vibrations of a free rod. The nodes and loops are indicated by *N* and *L*. The arrows indicate the directions and relative magnitudes of the motion. The graphs depict the relative amplitude of the motion along the rod.

4.9 LONGITUDINAL VIBRATION OF BARS²⁵⁻²⁸

Consider an entirely free rod of homogeneous material and constant cross section. The simplest mode of longitudinal vibration of a free rod is one in which a loop occurs at each end and a node in the middle, that is, when the length of the rod is one-half wavelength. The fundamental frequency of longitudinal vibration of a free rod, Fig. 4.12, may be obtained from Eq. (1.3) as follows:

$$f_1 = \frac{c}{\lambda} = \frac{c}{2l} = \frac{1}{2l} \sqrt{\frac{Q}{\rho}} \quad (4.46)$$

where *l* = length of the rod, in centimeters

ρ = density of the material, in grams per cubic centimeter (see Table 1.1)

Q = Young's modulus, in dynes per square centimeter

c = velocity of sound, in centimeters per second

λ = wavelength of the sound wave, in centimeters

²⁵ Rayleigh, *Theory of Sound*, Macmillan & Co., Ltd., London, 1926.

²⁶ Morse, *Vibration of Sound*, McGraw-Hill Book Company, Inc., New York, 1948.

²⁷ Wood, *A Text Book of Sound*, George Bell & Sons, Ltd., London, 1930.

²⁸ Lamb, *Dynamical Theory of Sound*, Edward Arnold & Co., London, 1931.

The overtones of the free rod are harmonics of the fundamental, that is, $f_2 = 2f_1$, $f_3 = 3f_1$, $f_4 = 4f_1$, etc. (Fig. 4.12).

The fundamental resonance frequency occurs when the length of the rod is one-half wavelength. This fact provides a means of computing the velocity of sound when the density, Young's modulus, and the frequency are known, or the frequency of sound when the velocity, density, and Young's modulus are known.

Rods in which the longitudinal waves are excited by striking the ends are used as standards of high-frequency sounds, 5,000 cycles and above, where a tuning fork is not very satisfactory.

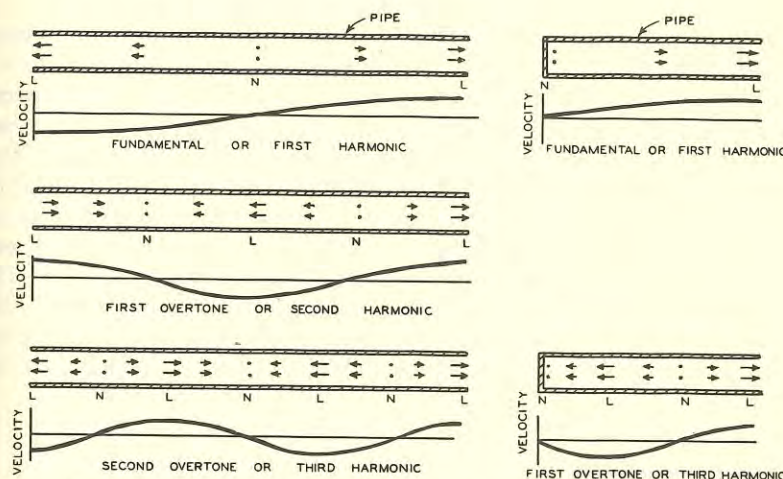


FIG. 4.13. Modes of vibration of the air column in a pipe open at both ends and in a pipe closed at one end and open at the other end. The velocity nodes and loops are indicated by *N* and *L*. The arrows indicate the directions and the relative magnitudes of the motion of the air particles in the pipe. The graphs depict the relative amplitudes of the air particles in the pipe.

4.10 OPEN AND CLOSED PIPES²⁹

The vibrations of a column of gas or fluid in a cylindrical tube are analogous to the longitudinal vibrations in a solid bar. For the open pipe there must be a loop of displacement at the open ends.

The fundamental resonant frequency of a pipe open at both ends (Fig. 4.13) is

$$f_1 = \frac{c}{\lambda} = \frac{c}{2l} \quad (4.47)$$

²⁹ Olson, H. F., *Acoustical Engineering*, D. Van Nostrand Company, Inc., Princeton, 1957.

where l = length of the pipe, in centimeters

c = velocity of sound, in centimeters per second

λ = wavelength, in centimeters

The resonant overtones of an open pipe are harmonics of the fundamental, that is, $f_2 = 2f_1$, $f_3 = 3f_1$, $f_4 = 4f_1$, etc. (Fig. 4.13).

The fundamental resonant frequency of a pipe closed at one end and open at the other end (Fig. 4.13) is

$$f_1 = \frac{c}{\lambda} = \frac{c}{4l} \quad (4.48)$$

The resonant overtones of the pipe closed at one end are the odd harmonics, that is, $f_2 = 3f_1$, $f_3 = 5f_1$, etc. (Fig. 4.13).

In the above considerations there is no end correction applied. There are two components of acoustical impedance at the end of a pipe, namely, the acoustical resistance due to radiation and the positive acoustical reactance due to the quadrature motion of the air beyond the open end of the pipe. The end correction depends upon whether the end of the pipe is flanged or unflanged. The unflanged pipe radiates into a solid angle of 4π steradians, while the flanged pipe radiates into a solid angle of 2π steradians. The end correction of an unflanged pipe is $0.62R$, where R is the radius of the pipe. The end correction of a flanged pipe is $0.82R$. These lengths must be added to the actual length of the pipe to obtain the effective length. The effective length of the pipe is used in accurate determinations of the resonant frequencies of pipes from the physical dimensions.

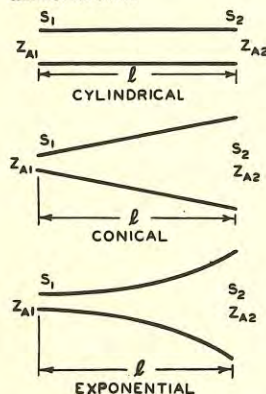


FIG. 4.14. Finite cylindrical, conical, and exponential horns. z_{A1} = input acoustical impedance at the throat; S_1 = cross-sectional area at the throat, in square centimeters; z_{A2} = terminating acoustical impedance at the mouth; S_2 = cross-sectional area at the mouth, in square centimeters; l = length, in centimeters.

The resonant frequencies and the response of a pipe may be determined theoretically by employing the expressions for the acoustical impedance of a pipe. The elements of a pipe are shown in Fig. 4.14. The expression³⁰ for the acoustical impedance z_{A1} , at the throat, in terms of the length and

³⁰ Olson, H. F., *Acoustical Engineering*, D. Van Nostrand Company, Inc., Princeton, 1957.

cross-sectional area of the pipe and the acoustical impedance z_{A2} , at the mouth, is

$$z_{A1} = \frac{\rho c}{S_1} \left[\frac{S_1 z_{A2} \cos(kl) + j\rho c \sin(kl)}{jS_1 z_{A2} \sin(kl) + \rho c \cos(kl)} \right] \quad (4.49)$$

where ρ = density of the medium, in grams per cubic centimeter

$$k = \frac{2\pi}{\lambda}$$

λ = wavelength, in centimeters

c = velocity of sound, in centimeters per second

S_1 = cross-sectional area of the pipe, in square centimeters

l = length of the pipe, in centimeters

z_{A2} = acoustical impedance at the mouth, in acoustical ohms

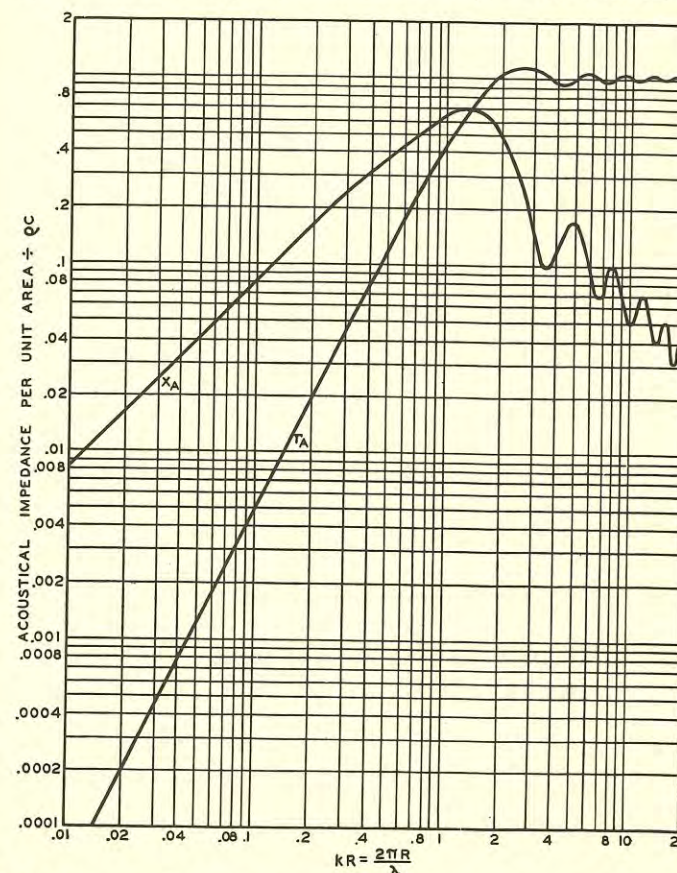


FIG. 4.15. The acoustical-resistance r_A and the acoustical-reactance x_A load per unit area divided by ρc as a function of kR for a piston of radius R set in an infinite baffle.

The acoustical impedance z_{A2} , at the mouth or open end of the flanged pipe, is assumed to be the same as the load upon a piston in an infinite baffle. The acoustical-resistance and acoustical-reactance components of the air load upon a piston are shown in Fig. 4.15. Note that the characteristics depicted in Fig. 4.15 are the acoustical resistance and acoustical reactance per unit area divided by ρc . The acoustical impedance z_{A2} , in acoustical ohms, at the mouth of the horn is given by

$$z_{A2} = r_{A2} + x_{A2} \quad (4.50)$$

where r_{A2} = acoustical resistance at mouth, in acoustical ohms

x_{A2} = acoustical reactance at the mouth, in acoustical ohms

The acoustical resistance r_{A2} , in acoustical ohms, at the mouth of the horn is given by

$$r_{A2} = r_A \frac{\rho c}{S_2} \quad (4.51)$$

where r_A = unit acoustical-resistance component depicted in Fig. 4.15

ρ = density of air, in grams per cubic centimeter

c = velocity of sound, in centimeters per second

S_2 = area of the mouth, in square centimeters

The acoustical reactance x_{A2} , in acoustical ohms, at the mouth of the horn is given by

$$x_{A2} = x_A \frac{\rho c}{S_2} \quad (4.52)$$

where x_A = unit acoustical-reactance component depicted in Fig. 4.15.

The acoustical-impedance characteristics of a pipe 3 inches in diameter and 12 inches in length are shown in Fig. 4.16. It will be seen that the maxima of acoustical resistance and the zero values of acoustical reactance occur at practically the same frequencies. Furthermore, these frequencies are in the ratio of 1:3:5, etc.

The theoretical acoustical-impedance characteristics of pipes and horns are considered in this chapter. Experimental output response-frequency characteristics of these systems are also given. The response can be predicted from the acoustical-impedance characteristics, provided that the generator acoustical impedance is known.

The acoustical circuit consisting of a pipe or horn and the generator is shown in Fig. 4.17. The volume current X_1 , in cubic centimeters per second, is given by

$$\dot{X}_1 = \frac{p}{z_{AG} + r_{A1} + x_{A1}} \quad (4.53)$$

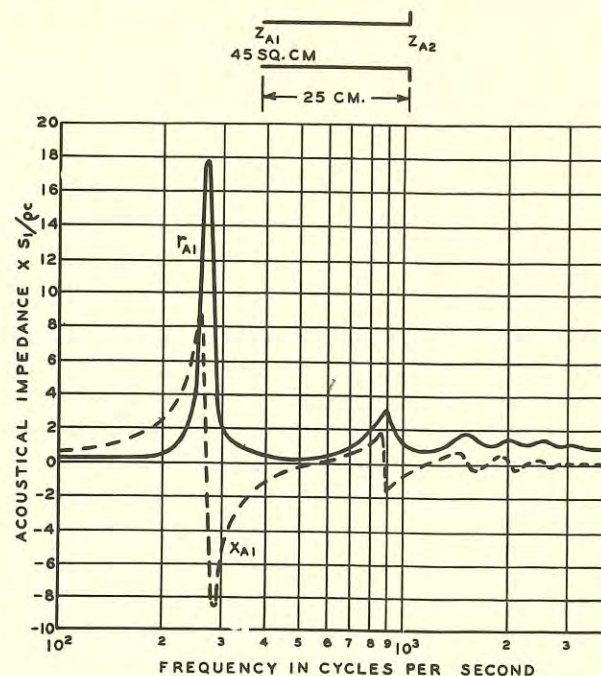


FIG. 4.16. The throat acoustical-resistance and acoustical-reactance frequency characteristics of the finite pipe shown above. r_{A1} = acoustical resistance, x_{A1} = acoustical reactance. Note: The characteristics shown are the throat acoustical resistance and acoustical reactance multiplied by S_1 and divided by ρc . (After Olson, *Acoustical Engineering*, D. Van Nostrand Company, Inc., Princeton, 1957.)

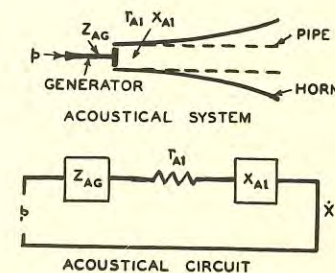


FIG. 4.17. An acoustical generator coupled to a horn or pipe and the acoustical circuit of the system. In the acoustical circuit, p = driving sound pressure, z_{AG} = acoustical impedance of the generator, r_{A1} = acoustical resistance at the throat of the horn or pipe, x_{A1} = acoustical reactance at the throat of the horn or pipe.

where p = sound pressure of the generator, in dynes per square centimeter

z_{AG} = acoustical impedance of the generator, in acoustical ohms

r_{A1} = acoustical impedance at the throat, in acoustical ohms

x_{A1} = acoustical reactance at the throat, in acoustical ohms

The sound output P is given by

$$P = r_{A1} |\dot{X}_1|^2 \quad (4.54)$$

The sound output can be obtained from Eqs. (4.53) and (4.54). The acoustical-impedance characteristics for different pipes and horns are given in this chapter. Examining these characteristics, it will be seen that the maximum of r_{A1} corresponds very closely to the frequency at which the acoustical reactance x_{A1} is zero. Under these conditions, it will be seen from Eqs. (4.53) and (4.54) that the maximum sound output will occur at or very near these frequencies provided that the acoustical impedance of the generator is relatively large or is an acoustical resistance. The experimental response-frequency characteristics of pipes and horns, depicted in this chapter, were obtained by coupling the throat to a piston vibrating with a constant velocity. Under these conditions, the acousti-

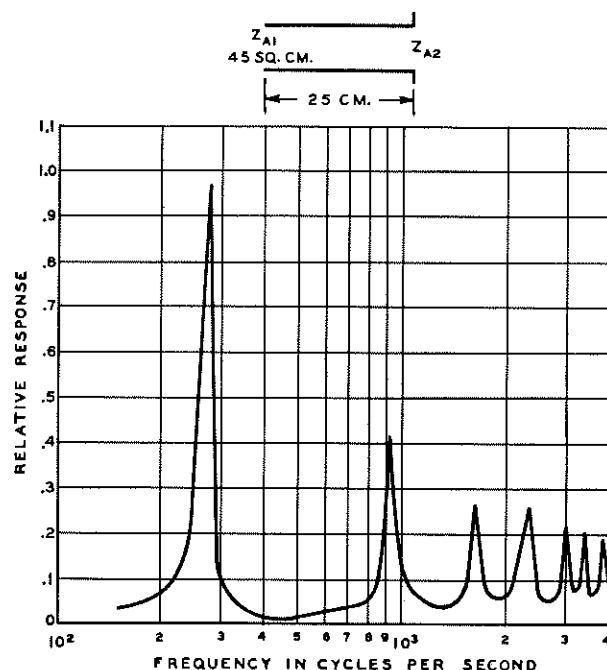


FIG. 4.18. The measured output response-frequency characteristic of the pipe shown above when driven by a piston moving with constant velocity.

cal impedance of the generator is for all practical purposes infinite. The experimental response-frequency characteristic of the pipe of Fig. 4.16 is shown in Fig. 4.18. Comparing Figs. 4.16 and 4.18, it will be seen that resonance occurs when the reactance is zero. It will also be seen that the resonant frequencies are in the ratio 1, 3, 5, etc.

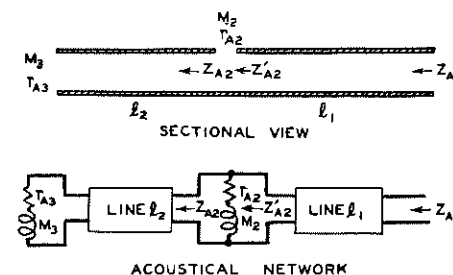
Pipes are used for the resonant system in many musical instruments. In the pipe organ the pipes are fixed in length. Therefore, a pipe is required for each frequency. However, in other instruments, where a single tube is used to cover a range of frequencies, some means must be provided to vary the effective length of the pipe and thereby effect a corresponding change in the resonant frequency. A change in the effective length of the pipe may be accomplished by means of holes in the side of the pipe, which may be open or closed. Examples of musical instruments employing this system are the piccolo, flute, clarinet, saxophone, oboe, and bassoon.

A pipe with a hole in the side and the acoustical network for this system are shown in Fig. 4.19. The acoustical impedance z_{A2} of the pipe length l_2 at the mid-point of the pipe may be obtained in terms of the length l_2 and the acoustical impedance z_{A3} from Eq. (4.49). The acoustical impedance z_{A1} may also be obtained from Eq. (4.49). However, the mouth acoustical impedance z'_{A2} for the length l_1 is now the combination of the hole inductance M_2 in series with the acoustical resistance r_{A2} , in parallel with z_{A2} . The acoustical impedance z'_{A2} is

$$z'_{A2} = \frac{z_{A2}(r_{A2} + j\omega M_2)}{z_{A2} + r_{A2} + j\omega M_2} \quad (4.55)$$

To obtain the acoustical impedance z_{A1} , from Eq. (4.49), the mouth impedance z'_{A2} is used with the length l_1 .

FIG. 4.19. A pipe with a hole in the side and the acoustical network of the vibrating system. z_{A1} = acoustical impedance at the throat, r_{A2} and M_2 = acoustical resistance and inductance of the hole, r_{A3} and M_3 = acoustical resistance and inductance of the mouth of the pipe, l_1 and l_2 = lengths of the two sections of pipes, z_{A2} = acoustical impedance of the section l_2 considered as a simple pipe.



To illustrate the effect of a hole in the side of a pipe, the acoustical impedance and response-frequency characteristics of a pipe with and

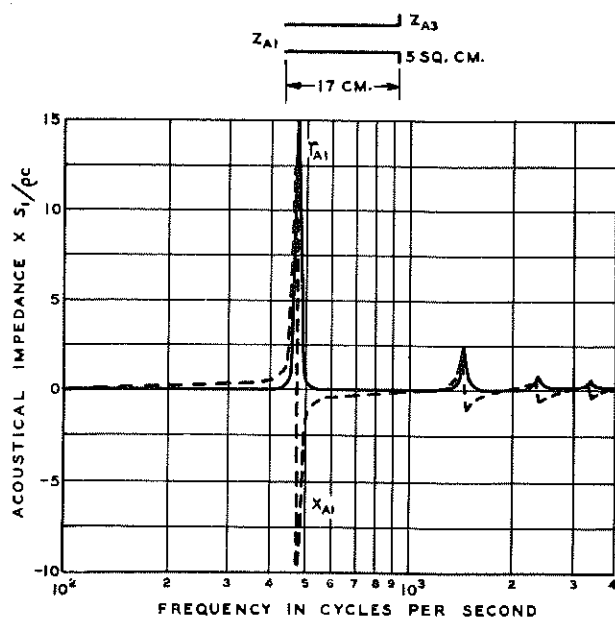


FIG. 4.20. The throat acoustical-resistance and acoustical-reactance frequency characteristic of the finite pipe shown above. r_{A1} = acoustical resistance, x_{A1} = acoustical reactance. Note: The characteristics shown are the throat acoustical resistance and acoustical reactance multiplied by S_1 and divided by ρc .

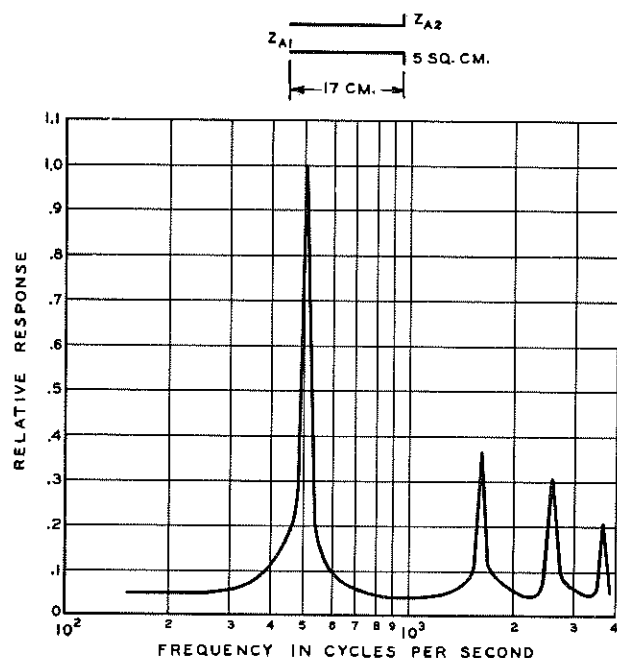


FIG. 4.21. The measured output response-frequency characteristic of the pipe shown above when driven by a piston moving with constant velocity.

without a hole will now be considered. The acoustical-impedance characteristic and the measured response-frequency characteristic of a pipe 1 inch in diameter and 6 inches in length are shown in Figs. 4.20 and 4.21. The fundamental resonant frequency is 500 cycles, both from the theoretical and experimental characteristics.

The acoustical-impedance characteristic of a pipe with a hole in the side is shown in Fig. 4.22. The measured response-frequency characteristic is shown in Fig. 4.23. The fundamental resonant frequency is 800 cycles. Comparing Figs. 4.20 and 4.21 with Figs. 4.22 and 4.23, it will be seen that the fundamental resonant frequency has been shifted from 500 cycles for a pipe without a hole to 800 cycles for a pipe with a hole. This example illustrates how the resonant frequency of a pipe can be varied by means of a hole in the side of the pipe with provisions for opening or closing the hole.

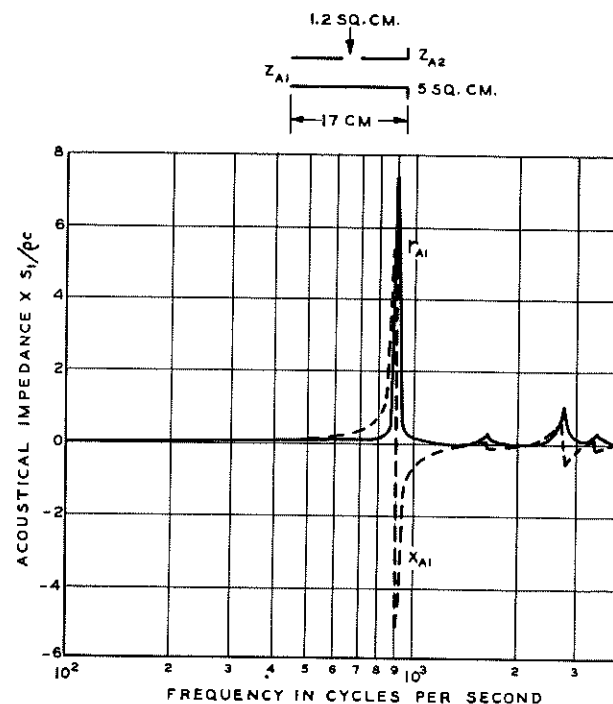


FIG. 4.22. The throat acoustical-resistance and acoustical-reactance frequency characteristics of the finite pipe with a hole as shown above. r_{A1} = acoustical resistance, x_{A1} = acoustical reactance. Note: The characteristics shown are the throat acoustical resistance and acoustical reactance multiplied by S_1 and divided by ρc .

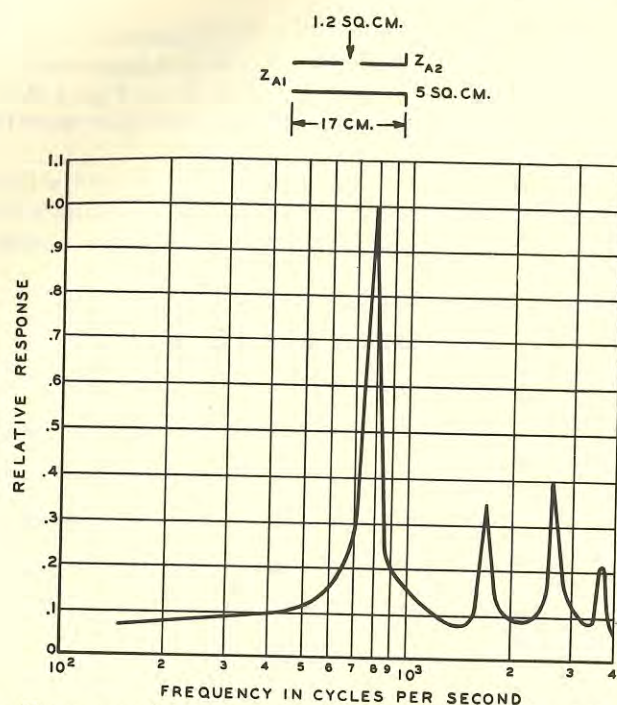


Fig. 4.23. The measured output response-frequency characteristic of the finite pipe with a hole as shown above when driven by a piston moving with a constant velocity.

4.11 HORNS³¹⁻³⁸

A horn is an acoustical transducer consisting of a tube of varying sectional area. Horns have been used for centuries for increasing the radiation from a sound source. The shape or the rate of flare is the distinguishing characteristic between various horns. The most common rates of flare are parabolic, conical, exponential, and hyperbolic. In horn-type musical instruments, the horn may be any one of these or some combinations of

³¹ Webster, A. G., *Jour. Nat. Acad. Sci.*, Vol. 5, p. 275, 1919.

³² Stewart, G. W., *Phys. Rev.*, Vol. 16, No. 4, p. 313, 1920.

³³ Goldsmith and Minton, *Proc. Inst. Radio Engrs.*, Vol. 12, No. 4, p. 423, 1924.

³⁴ Slepian and Hanna, *Jour. Amer. Inst. Elec. Engrs.*, Vol. 43, p. 250, 1924.

³⁵ Ballantine, G., *Jour. Franklin Inst.*, Vol. 203, No. 1, p. 85, 1927.

³⁶ Crandall, *Theory of Vibrating Systems and Sound*, D. Van Nostrand Company, Inc., Princeton, 1926.

³⁷ Stewart and Lindsay, *Acoustics*, D. Van Nostrand Company, Inc., Princeton, 1930.

³⁸ Olson, H. F., *Acoustical Engineering*, D. Van Nostrand Company, Inc., Princeton, 1957.

these. It is beyond the scope of these considerations to consider all types of flare. A consideration of the acoustical-impedance and response-frequency characteristics of the conical and exponential horns, shown in Fig. 4.14, will be carried out and will illustrate the action of horns.

The equation expressing the cross-sectional area as a function of the distance along the axis in the conical horn is given by

$$S = S_1 x^2 \quad (4.56)$$

where S = cross-sectional area at any point along the axis

x = distance along the axis

S_1 = area at the throat of the horn or at the point $x = x_1$

The expression³⁹ for the acoustical impedance z_{A1} , at the throat S_1 , of a conical horn in terms of the dimensions and the acoustical impedance z_{A2} , at the mouth, is given by

$$z_{A1} = \frac{\rho c}{S_1} \left[\frac{jz_{A2} \frac{\sin k(l - \theta_2)}{\sin k\theta_2} + \frac{\rho c}{S_2} \sin kl}{z_{A2} \frac{\sin k(l + \theta_1 - \theta_2)}{\sin k\theta_1 \sin k\theta_2} - \frac{j\rho c \sin k(l + \theta_1)}{S_2 \sin k\theta_1}} \right] \quad (4.57)$$

where S_1 = area of the throat, in square centimeters

S_2 = area of the mouth, in square centimeters

l = length of the horn, in centimeters

$k\theta_1 = \tan^{-1} kx_1$

$k\theta_2 = \tan^{-1} kx_2$

x_1 = distance from the apex to the throat, in centimeters

x_2 = distance from the apex to the mouth, in centimeters

$$k = \frac{2\pi}{\lambda}$$

The acoustical-impedance characteristics at the throat of a conical horn are shown in Fig. 4.24. The acoustical impedance z_{A2} at the mouth of the horn is assumed to be the same as the load upon a piston in an infinite baffle [see Fig. 4.15 and Eqs. (4.50), (4.51), and (4.52)]. The experimental response-frequency characteristic of the same horn is shown in Fig. 4.25. Comparing Figs. 4.24 and 4.25, it will be seen that the resonant frequencies occur when the acoustical reactance is zero.

The equation expressing the cross-sectional area of an exponential horn as a function of the distance along the axis is given by

$$S = S_1 e^{mx} \quad (4.58)$$

where S_1 = cross-sectional area at the throat of the horn

x = distance along the axis

m = flare constant of the horn

S = cross-sectional area at a point x along the axis of the horn

³⁹ Olson, H. F., *Acoustical Engineering*, D. Van Nostrand Company, Inc., Princeton, 1957.

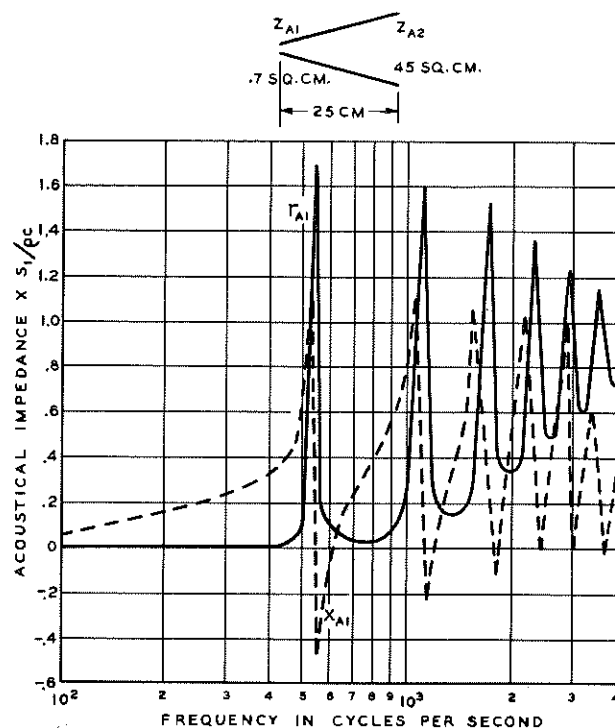


FIG. 4.24. The throat acoustical-resistance and acoustical-reactance frequency characteristics of the finite conical horn shown above. r_{A1} = acoustical resistance, z_{A1} = acoustical reactance. Note: The characteristics shown are the throat acoustical resistance and acoustical reactance multiplied by S_1 and divided by ρc .

The expression⁴⁰ for the acoustical impedance z_{A1} , at the throat S_1 , of an exponential horn in terms of the dimensions, the flare constant, and the acoustical impedance z_{A2} , at the mouth, is given by

$$z_{A1} = \frac{\rho c}{S_1} \left\{ \frac{S_2 z_{A2} [\cos(bl + \theta)] + j\rho c [\sin(bl)]}{jS_2 z_{A2} [\sin(bl)] + \rho c [\cos(bl - \theta)]} \right\} \quad (4.59)$$

where S_1 = area of the throat, in square centimeters

S_2 = area of the mouth, in square centimeters

l = length of the horn, in centimeters

z_{A2} = acoustical impedance of the mouth, in acoustical ohms

$\theta = \tan^{-1} a/b$

$a = m/2$

$b = \frac{1}{2} \sqrt{4k^2 - m^2}$

⁴⁰ Olson, H. F., *Acoustical Engineering*, D. Van Nostrand Company, Inc., Princeton, 1957.

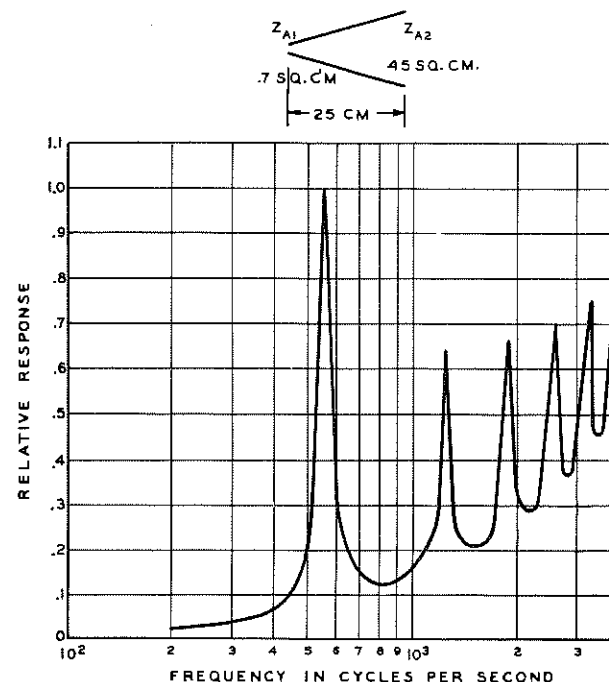


FIG. 4.25. The measured output response-frequency characteristic of the finite conical horn shown above driven by a piston moving with constant velocity.

The acoustical-impedance characteristics at the throat of an exponential horn are shown in Fig. 4.26. The acoustical impedance z_{A2} at the mouth of the horn is assumed to be the same as the load upon a piston in an infinite baffle [see Fig. 4.15 and Eqs. (4.50), (4.51), and (4.52)]. The measured response-frequency characteristic of the same horn is shown in Fig. 4.27. Comparing Figs. 4.26 and 4.27, it will be seen that the resonant frequencies occur when the acoustical reactance is zero.

In musical instruments employing horns, the resonant frequency is varied by means of holes in the side or by varying the length of the horn. The length of the horn is varied in the trumpet, French horn, tuba, and trombone to effect a change in the resonant frequency. In the first three instruments, the length is varied in discrete steps by introducing additional lengths by means of valves, as shown in Fig. 4.28A. The use of the three additional side tubes shown in Fig. 4.28A together with valves for introducing or eliminating any of the additional side tubes makes it possible to obtain eight resonant frequencies. The resonant frequencies correspond to the following lengths, $L, L + A, L + B, L + C, L +$

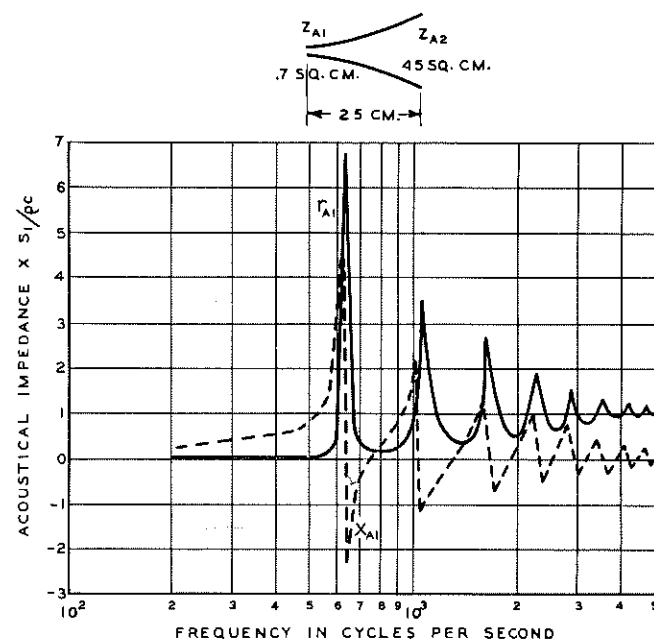


FIG. 4.26. The throat acoustical-resistance and acoustical-reactance frequency characteristics of the finite exponential horn shown above. r_{A1} = acoustical resistance, x_{A1} = acoustical reactance. Note: The characteristics shown are the throat acoustical resistance and acoustical reactance multiplied by S_1 and divided by ρc .

$A + B$, $L + A + C$, $L + B + C$, and $L + A + B + C$, where L is the length of the horn alone and A , B , and C are the lengths of the side tubes. The trombone uses telescopic tubing to vary the length, as shown in Fig. 4.28B. The length can be varied in a uniform and continuous manner. As a consequence, an infinite number of resonant frequencies may be obtained within the range of the instrument.

The acoustical impedance of a horn with an additional length of cylindrical pipe can be determined from Eqs. (4.49) and (4.57) or (4.59). For example, if a pipe is added to the throat of an exponential horn, the acoustical impedance z_{A1} at the opening of the pipe is determined from Eq. (4.49). The acoustical impedance z_{A2} of the pipe is the acoustical impedance at the throat of the exponential horn given by Eq. (4.59).

The acoustical-impedance characteristic of the exponential horn of Fig. 4.26 with an additional cylindrical length of 6 inches attached to the throat is shown in Fig. 4.29. Comparing Figs. 4.26 and 4.29, it will be seen that the impedance characteristic has been shifted downward in

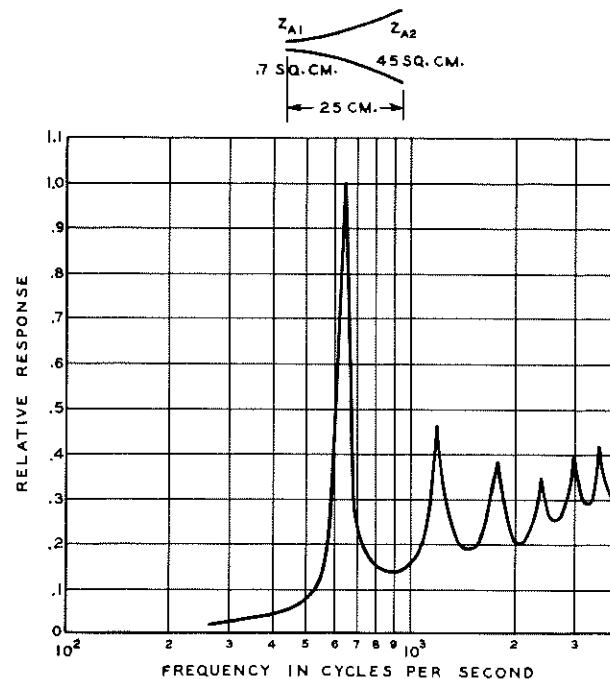


FIG. 4.27. The measured output response-frequency characteristic of the finite exponential horn shown above driven by a piston moving with constant velocity.

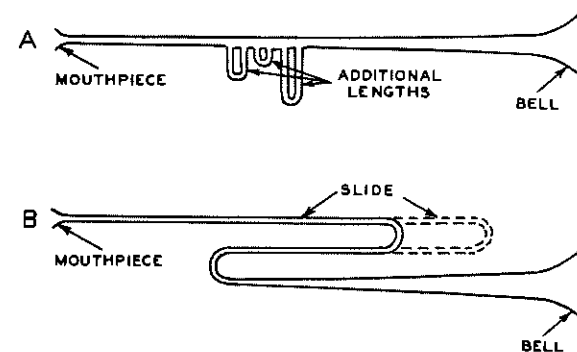


FIG. 4.28. Methods for varying the length of a horn, thereby changing the resonant frequency. In the system of *A* above, the length of the air column may be varied in discrete steps by the addition of the three tube lengths. In the system of *B* above, the length of the air column may be continuously varied by means of the telescopic slide.

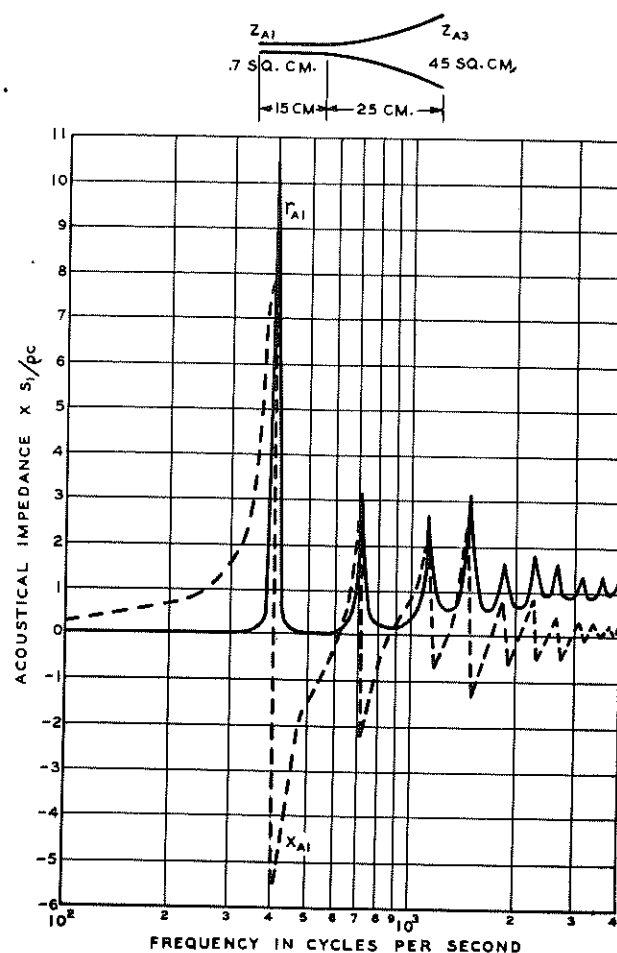


FIG. 4.29. The throat acoustical-resistance and acoustical-reactance frequency characteristics of the finite exponential horn with a 6-inch length of tubing added at the throat as shown above. r_{A1} = acoustical resistance, x_{A1} = acoustical reactance. Note: The characteristics shown are the throat acoustical resistance and acoustical reactance multiplied by S_1 and divided by ρc .

frequency by the additional length. The measured response-frequency characteristic of the exponential horn with the additional cylindrical length is shown in Fig. 4.30. Comparing Figs. 4.27 and 4.30, it will be seen that the fundamental resonant frequency has been lowered from 650

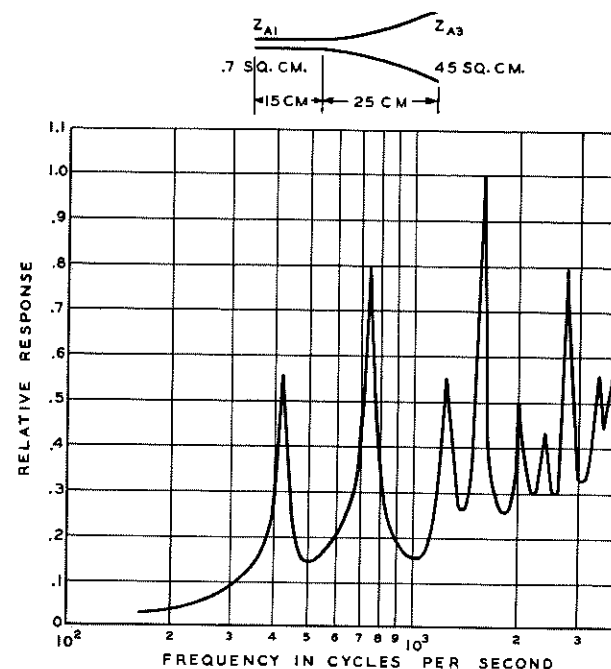


FIG. 4.30. The measured output response-frequency characteristic of the finite exponential horn, with a 6-inch length of tubing added at the throat as shown above, driven by a piston moving with constant velocity.

cycles for the exponential horn alone to 420 cycles for the exponential horn with the additional cylindrical length of pipe.

The resonant frequency of a horn may also be varied by means of holes in the side of the horn. The theory and the action is similar to that of a hole in the side of a cylindrical pipe considered in Sec. 4.10. The acoustical-impedance frequency characteristics of the exponential horn of Fig. 4.26 with a hole in the side at the mid-point along the axis is shown in Fig. 4.31. Comparing the impedance frequency characteristics of Figs. 4.26 and 4.31, it will be seen that the acoustical-impedance frequency characteristic has been shifted upward in frequency. The measured response-frequency characteristic of a horn with a hole in the side is shown in Fig. 4.32. Comparing Figs. 4.27 and 4.32, it will be seen that the fundamental resonant frequency has been raised from 650 cycles for the exponential horn without a hole to 900 cycles for the exponential horn with a hole in the side.

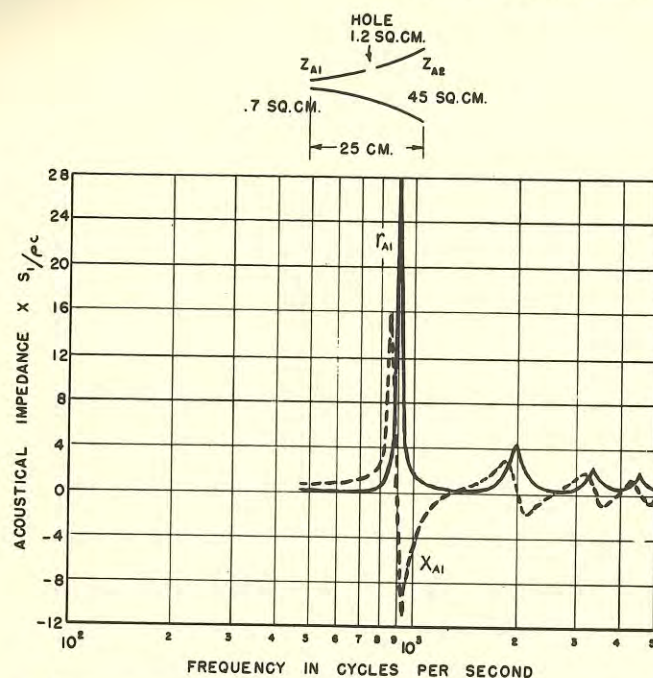


FIG. 4.31. The throat acoustical-resistance and acoustical-reactance frequency characteristics of the finite exponential horn with a hole in the side, as shown above. r_{A1} = acoustical resistance, x_{A1} = acoustical reactance. Note: The characteristics shown are the throat acoustical resistance and acoustical reactance multiplied by S_1 and divided by ρc .

4.12 DIRECTIONAL CHARACTERISTICS OF SOUND SOURCES

The directional characteristic of a sound source is the output as a function of the angle with respect to some axis of the system. The characteristics are usually plotted as a system of polar characteristics for various frequencies. The radiating characteristics of sound sources of musical instruments are extremely complex, because, in many instances, each instrument contains many different sound sources. However, a consideration of some of the simplest sound-radiating systems will give some insight into the directional performance of sound sources. It is the purpose of this section to consider the directional characteristics of a few of the simplest sound-radiating systems employed in musical instruments.

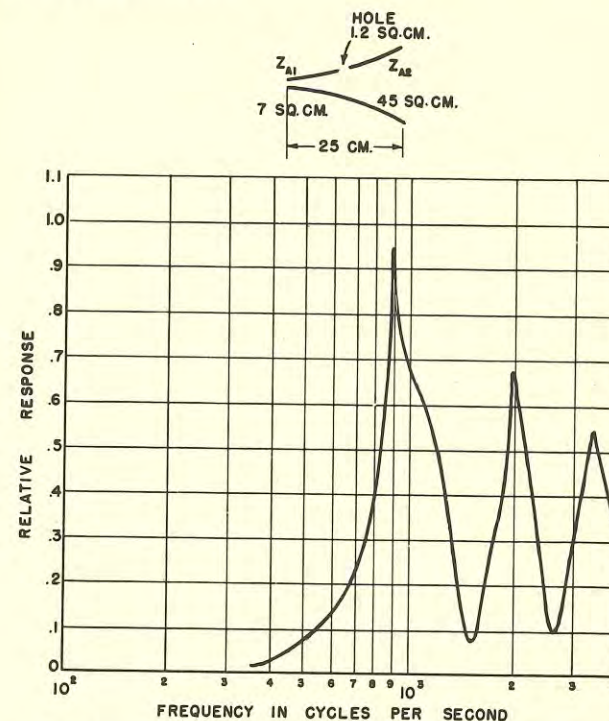


FIG. 4.32. The measured output response-frequency characteristic of the finite exponential horn with a hole in the side as shown above, driven by a piston moving with constant velocity.

A. Series of Point Sources⁴¹⁻⁴⁴

In some musical instruments there are several small sound sources as, for example, in the case of the wind instruments with side holes or a bank of organ pipes. The directional characteristics exhibited by these systems are very complicated because of the unsymmetrical distribution of the sound sources in space and the different amplitude and phase relations between the sound sources. However, a consideration of the directional characteristics of a series of equal point sources located on a line will give

⁴¹ Wolff, I., and Malter, L., *Jour. Acoust. Soc. Amer.*, Vol. 2, No. 2, p. 201, 1930.

⁴² Stenzel, H., *Elek. Nachr.-Tech.* Vol. 4, No. 6, p. 239, 1927.

⁴³ Stenzel, H., *Elek. Nachr.-Tech.* Vol. 6, No. 5, p. 165, 1929.

⁴⁴ Olson, H. F., *Acoustical Engineering*, D. Van Nostrand Company, Inc., Princeton, 1957.

some idea of the directional performance of more complicated systems.

The directional characteristic of a source made up of any number of equal point sources, vibrating in phase, located on a straight line, and separated by equal distances is given by

$$R_a = \frac{\sin \left(\frac{n\pi d}{\lambda} \sin \alpha \right)}{n \sin \left(\frac{\pi d}{\lambda} \sin \alpha \right)} \quad (4.60)$$

where R_a = ratio of the pressure for an angle α to the pressure for an angle $\alpha = 0$. The direction $\alpha = 0$ is normal to the line

n = number of sources

d = distances between the sources, in centimeters

λ = wavelength, in centimeters

The directional characteristics of a two point sound source are shown in Fig. 4.33. It will be noted that the directional pattern is quite complex even though there are only two sources.

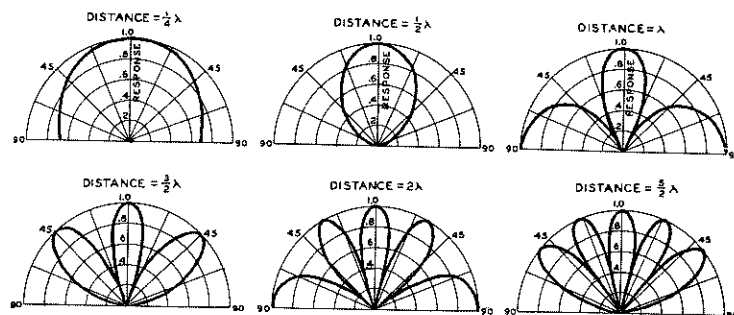


FIG. 4.33. Directional characteristics of two separated equal small sources vibrating in phase as a function of the distance between the sources and the wavelength. The polar graph depicts the pressure, at a fixed distance, as a function of the angle. The pressure for the angle 0 degrees is arbitrarily chosen as unity. The direction corresponding to the angle 0 degrees is perpendicular to the line joining the two sources. The directional characteristics in three dimensions are surfaces of revolution about the line joining the two sources as an axis. (After Olson, *Acoustical Engineering*, D. Van Nostrand Company, Inc., Princeton, 1957.)

B. Plane Circular Surface Source⁴⁵⁻⁴⁷

In some musical instruments the radiating surface consists of a circular flat vibrating surface as, for example, drum, banjo, and tambourine.

⁴⁵ Stenzel, H., *Elek. Nachr.-Tech.* Vol. 4, No. 6, p. 1, 1927.

⁴⁶ Wolff, I., and Malter, L., *Jour. Acoust. Soc. Amer.*, Vol. 2, No. 2, p. 201, 1930.

⁴⁷ Olson, H. F., *Acoustical Engineering*, D. Van Nostrand Company, Inc., Princeton, 1957.

The diaphragm of a loudspeaker is also an example of a circular surface source.

The directional characteristics of a circular surface source with all parts of the surface vibrating with the same strength and phase are

$$R_a = \frac{2J_1 \left(\frac{2\pi R}{\lambda} \sin \alpha \right)}{\frac{2\pi R}{\lambda} \sin \alpha} \quad (4.61)$$

where R_a = ratio of the pressure for an angle α to the pressure for an angle $\alpha = 0$

J_1 = Bessel function of the first order

R = radius of the circle, in centimeters

α = angle between the axis of the circle and the line joining the point of observation and the center of the circle

λ = wavelength, in centimeters

The directional characteristics of a plane circular surface source as a function of the diameter and wavelength are shown in Fig. 4.34. The directional pattern becomes sharper as the ratio of the diameter to the wavelength increases. This is typical of most acoustical radiating systems.

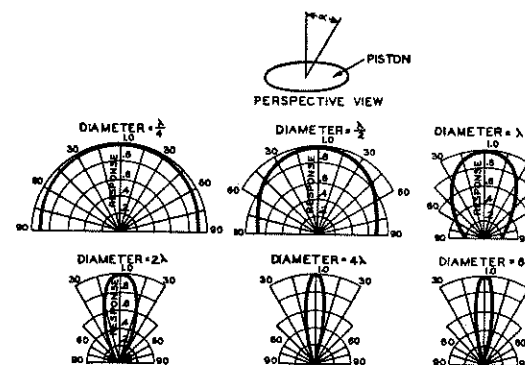


FIG. 4.34. Directional characteristics of a circular piston source as a function of the diameter and wavelength. The polar graph depicts the pressure, at a large fixed distance, as a function of the angle. The pressure for the angle 0 degrees is arbitrarily chosen as unity. The direction corresponding to the angle 0 degrees is the axis. The axis is the center line perpendicular to the plane of the piston. The directional characteristics in three dimensions are surfaces of revolution about the axis.

C. Plane Rectangular Surface Source⁴⁸

In some musical instruments the radiating surface consists of a flat radiating surface, as, for example, the soundboard in the piano and the harp, the flat surfaces of the body of the guitar and ukulele. The directional characteristics of a plane rectangular surface source approximates the performance of these systems.

The directional characteristics of a rectangular surface source with all parts of the surface vibrating with the same strength and phase are

$$R_a = \frac{\sin\left(\frac{\pi l_a}{\lambda} \sin \alpha\right) \sin\left(\frac{\pi l_\beta}{\lambda} \sin \beta\right)}{\frac{\pi l_a}{\lambda} \sin \alpha \frac{\pi l_\beta}{\lambda} \sin \beta} \quad (4.62)$$

where l_a = length of the rectangle

l_β = width of the rectangle

α = angle between the normal to the surface source and the projection of the line joining the middle of the surface and the observation point on the plane normal to the surface and parallel to l_a

β = angle between the normal to the surface source and the projection of the line joining the middle of the surface and the observation point on the plane normal to the surface and parallel to l_β

The directional characteristics of a plane surface source are shown in Fig. 4.35. The characteristics are for either $\alpha = 0$ or $\beta = 0$. If $\beta = 0$, the directional pattern is taken in the plane normal to the vibrating surface and parallel to l_a . The length given above each characteristic is then l_a . If $\alpha = 0$, the directional pattern is taken in the plane normal to the vibrating surface and parallel to l_β . The length given above each characteristic is then l_β .

D. Cylindrical Surface Source⁴⁹

In some of the musical instruments the soundboard, or body of the instrument, is a curved surface. The directional characteristics of this type of surface approximates that of a cylindrical surface. A cylindrical surface radiator is shown in Fig. 4.36. In this radiator it will be assumed that all parts of the surface vibrate, with equal amplitudes and in the same phase, in directions along the radii of the cylinder. The directional characteristics of interest lie in a plane normal to the axis of the cylinder.

^{48, 49} Olson, H. F., *Acoustical Engineering*, D. Van Nostrand Company, Inc., Princeton, 1957.

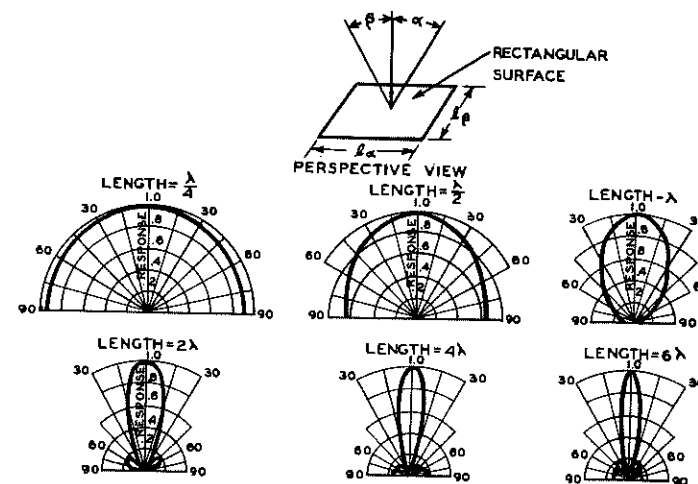


FIG. 4.35. Directional characteristics of a rectangular surface source as a function of the length and the wavelength. The polar graph depicts the pressure, at a large fixed distance, as a function of the angle. The pressure for the angle 0 degrees is arbitrarily chosen as unity. The direction corresponding to the angle 0 degrees is perpendicular to the surface.

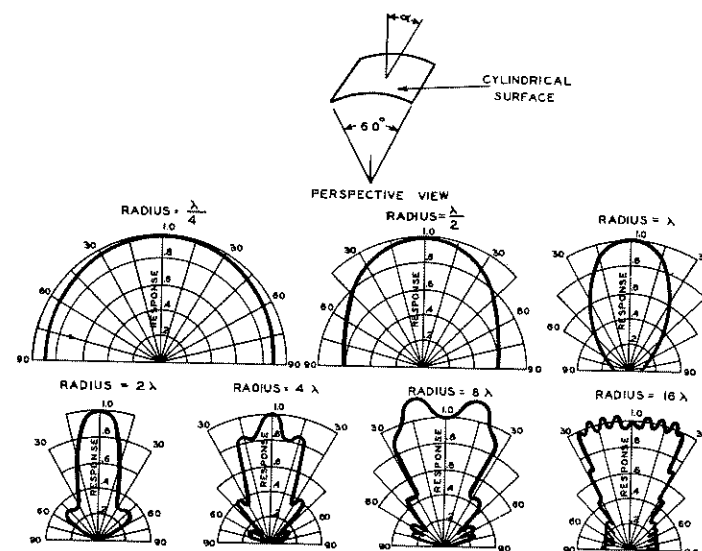


FIG. 4.36. The directional characteristics of a cylindrical surface source which subtends an angle of 60 degrees. The polar diagram depicts the pressure, at a large fixed distance, as a function of the angle. The direction corresponding to the angle 0 degrees is perpendicular to the surface at the mid-point. The observation points, for the angle α , are located in the plane normal to the axis of the cylindrical surface. The pressure for the angle 0 degrees is arbitrarily chosen as unity.

For an observation point located at a large distance from the surface, this directional characteristic is the same as that of one of the circular arcs of the cylindrical surface. A circular arc source may be assumed to be made up of a large number of point sources of equal strength separated by very small and equal distances on the arc of a circle, with all the points vibrating with the same phase. The directional characteristic of this type of vibrator may be expressed as follows:

$$R_a = \frac{1}{2m+1} \left| \sum_{k=-m}^{k=m} \cos \left[\frac{2\pi R}{\lambda} \cos (\alpha + k\theta) \right] + j \sum_{k=-m}^{k=m} \sin \left[\frac{2\pi R}{\lambda} \cos (\alpha + k\theta) \right] \right| \quad (4.63)$$

where R_a = ratio of the pressure for an angle α to the pressure for an angle $\alpha = 0$

α = angle between the radius drawn through the central point and the line joining the source and the distant observation point

λ = wavelength, in centimeters

R = radius of the arc, in centimeters

$2m+1$ = number of points

θ = angle subtended by any two points at the center of the arc

k = variable

The directional characteristics of cylindrical surface in a plane normal to the axis of the cylinder for an arc of 60 degrees are shown in Fig. 4.36. The interesting feature of the directional characteristics of a cylindrical surface is that the directional characteristics are very broad for wavelengths large compared to the dimensions, are narrow for wavelengths comparable to the dimensions, and are broad again for wavelengths small compared to the dimensions. The dimensions of the surface must be several wavelengths in order to yield the wedge-shaped directional characteristic.

E. Horn Source^{50,51}

In the lip-reed or brass instruments and some of the other wind instruments the radiating system is a horn. The directional characteristics of a horn depend upon the shape, mouth opening, and frequency. It is

⁵⁰ Olson, H. F., *RCA Rev.*, Vol. 1, No. 4, p. 68, 1937.

⁵¹ Olson, H. F., *Acoustical Engineering*, D. Van Nostrand Company, Inc., Princeton, 1957.

the purpose of this section to examine and consider some of the factors which influence the directional characteristics of a horn.

The phase and particle velocity of the various incremental areas which may be considered to constitute the mouth determine the directional characteristics of the horn. The particular complexion of the velocities and phase of these areas is governed by the flare and dimensions and shape of the mouth. The mouth of the horn plays a major role in determining the directional characteristics in the range where the wavelength is greater than the mouth diameter. The flare is the major factor in determining the directional characteristics in the range where the wavelength is less than the mouth diameter.

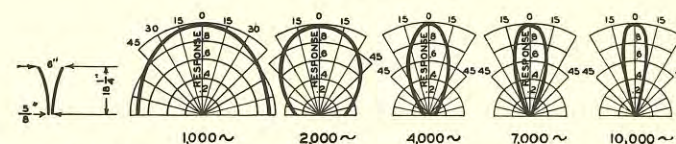


FIG. 4.37. Directional characteristics of a small exponential horn of circular cross section of the dimensions given above. The polar graph depicts the pressure, at a fixed distance, as a function of the angle. The pressure for the angle 0 degrees is arbitrarily chosen as unity. The direction corresponding to 0 degrees is the axis of the horn. The directional characteristics in three dimensions are surfaces of revolution about the horn axis.

The directional characteristics of a small exponential horn are shown in Fig. 4.37. As in the case of other radiating systems the directional characteristic becomes sharper with an increase of the frequency. The directional characteristics shown in Fig. 4.37 are typical of the type of characteristic obtained upon horn musical instruments. For horns of smaller or larger dimensions, the frequency given in Fig. 4.37 should be multiplied by the ratio obtained by dividing the linear dimensions of the horn in Fig. 4.37 by the linear dimensions of the horn in question.

recommended for measurements at the lower levels, and curve *B* for measurements around 70 decibels above the threshold. For very loud sounds (80 to 100 decibels), the flat characteristic of curve *C* should be used.

The noise meter may be used for noise analysis in offices, factories, restaurants, homes, etc. In these measurements, a large number of observations should be made in various positions. The noise meter may also be used to measure the transmission or attenuation by walls, floors, ceilings, and doors.

The highest noise level which can be tolerated within an enclosure over long periods of time without annoyance, undue fatigue, or discomfort due to the noise is termed the tolerable noise level. A list of tolerable noise levels for different enclosures are shown in Table 8.5. This table gives

TABLE 8.5. TOLERABLE AVERAGE NOISE LEVELS
UNOCCUPIED STUDIOS, THEATERS, AND ROOMS

Type of Enclosure	Noise Level, Decibels
Phonograph recording studio	20 to 25
Radio studio	20 to 30
Television studio	25 to 35
Legitimate theater	25 to 35
Auditoriums	25 to 40
Motion picture theaters	25 to 35
Homes	35 to 45
Hotels	40 to 50
Private and small offices	40 to 45
Large offices	45 to 50
Restaurants	45 to 55

the average noise level as measured on the standard noise meter described above. The values in this table assume that there are no periodic or discrete sounds of a certain frequency, which would of course stand out in the operation of recording, radio, and television studios, and still be low enough so that the average limits of Table 8.5 would be satisfied. It should be mentioned in passing that the lower limit in this table is the most desirable, and attempts should be made to achieve this noise level.

CHAPTER NINE

Sound-reproducing Systems

9.1 INTRODUCTION

Reproduction of sound¹ involves the picking up of sound at one location and reproducing it at the same location or at another location either at the same time or at some subsequent time. In the reproduction of sound, sound waves are converted into the corresponding waves in some other medium. This medium may be the undulations in a phonograph record, the magnetic variations in a magnetic tape, the density or area variations in a photographic film, the electrical waves in a wire, or the electromagnetic waves in the ether. The variations in these different media correspond to the variations in the sound pressure in the original sound wave. To complete the process of the reproduction of sound, the variations in these various media are transformed back into sound waves. These sound waves correspond to the variations in the different media and hence correspond to the original sound waves. This process is termed the reproduction of sound. The reproduction of sound requires a number of different transducers for accomplishing the various energy conversions required in these systems. There are two electroacoustic transducers which are almost universally used in sound reproduction today, namely, the microphone for converting sound waves into electrical variations and the loudspeaker for converting electrical variations into sound waves. In addition to these two there are the various transducers for further conversion into the various media. It is the purpose of this chapter to consider microphones, loudspeakers, and complete sound-reproducing systems.

9.2 MICROPHONES

A microphone is an electroacoustic transducer actuated by energy in an acoustical system and delivering energy to an electrical system, the waveform in the electrical system being substantially equivalent to that in the

¹ For a more complete discussion on microphones, loudspeakers, and sound-reproducing systems and references, see Olson, H. F., *Acoustical Engineering*, D. Van Nostrand Company, Inc., Princeton, 1957. Also Olson, H. F., *Advances in Sound Reproduction*, Proceedings 5th Congress International D'Acoustique, Liege, 1965.

acoustical system. A pressure microphone is a microphone in which the electrical response corresponds to the variations in pressure in the actuating sound wave. A velocity microphone is a microphone in which the electrical response corresponds to the particle velocity resulting from the propagation of a sound wave through an acoustical medium. All microphones in use today may be classified as follows: pressure, velocity, or combination pressure and velocity. For the conversion of the acoustical variations into the corresponding electrical variations the following transducers may be used: carbon, magnetic, dynamic, condenser, crystal, magnetostrictive, electronic, and hot-wire.

It is beyond the scope of this book to consider in detail all types of microphones. Therefore, the considerations of microphones will be confined to the following, namely, the pressure-responsive carbon, crystal, dynamic, magnetic, and condenser microphones; the velocity-responsive dynamic ribbon microphone; and various unidirectional microphones.

A. Pressure Microphones

1. Carbon Microphone. A carbon microphone is a microphone which depends for its operation on the variation in resistance of carbon contacts. The high sensitivity of this microphone is due to the relay action of the carbon contacts. The carbon microphone is almost universally employed in telephonic communications, where the prime requisite is sensitivity rather than low distortion and uniform response over a wide frequency range.

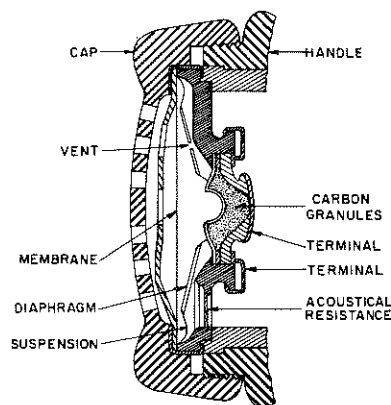


FIG. 9.1. Sectional view of a carbon-pressure microphone.

A sectional view of a carbon microphone is shown in Fig. 9.1. The carbon button consists of a cylindrical cavity filled with carbon granules. The carbon granules make contact with the central portion of the dia-

phragm and base of the carbon granule cavity. A displacement of the diaphragm produces a change in the electrical resistance between the two terminals of the carbon chamber. The resistance decreases as the motion of the diaphragm compresses the carbon granules and increases as the motion of the diaphragm expands the volume occupied by the carbon granules. If the carbon microphone is connected in a series electrical circuit consisting of a fixed electrical resistance and battery, the current will be an inverse function of the resistance of the carbon microphone. The voltage developed across the fixed electrical resistor is proportional to the current. Under these conditions, since the motion of the diaphragm is proportional to the impinging sound pressure, the alternating voltage developed across the fixed electrical resistor will correspond to the pressure undulations in the sound wave.

2. Crystal Microphone. A crystal or piezoelectric microphone is a microphone which depends upon the generation of a voltage by the deformation of a crystal having piezoelectric properties. A piezoelectric crystal generates a voltage when it is deformed. Rochelle salt exhibits the greatest piezoelectric activity of all the known crystals. A barium titanate ceramic element may be used in place of the Rochelle salt crystal element. The action and the general performance is the same except that a microphone with a ceramic element exhibits slightly lower sensitivity. The barium titanate element will operate at higher temperatures than the Rochelle salt element without suffering permanent damage.

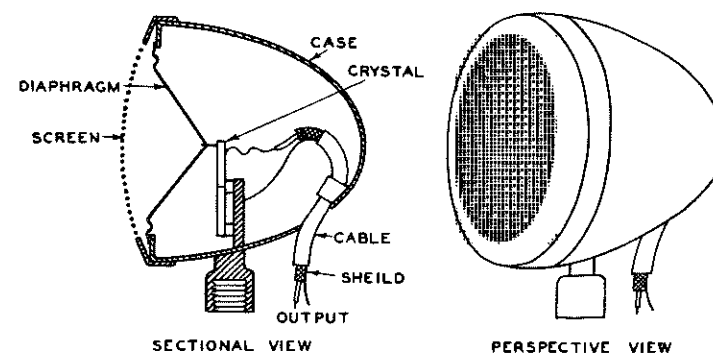


Fig. 9.2. Sectional and perspective views of a crystal-pressure microphone.

Perspective and cross-sectional views of a crystal microphone are shown in Fig. 9.2. The apex of the diaphragm is coupled to one corner of the crystal. The remaining three corners are attached to the case of the microphone. The motion of the diaphragm due to an impinging sound

wave deforms the crystal which produces an alternating voltage which corresponds to the pressure undulations in the sound wave. The crystal microphone has a high electrical impedance and is therefore coupled directly to the grid of the first vacuum tube in the amplifier or a high-impedance field-effect transistor.

3. Dynamic Microphone. A dynamic microphone is a microphone in which the output results from the motion of a conductor in a magnetic field. When a conductor moves in a magnetic field, a voltage is generated which is proportional to the number of magnetic flux lines which are cut per second.

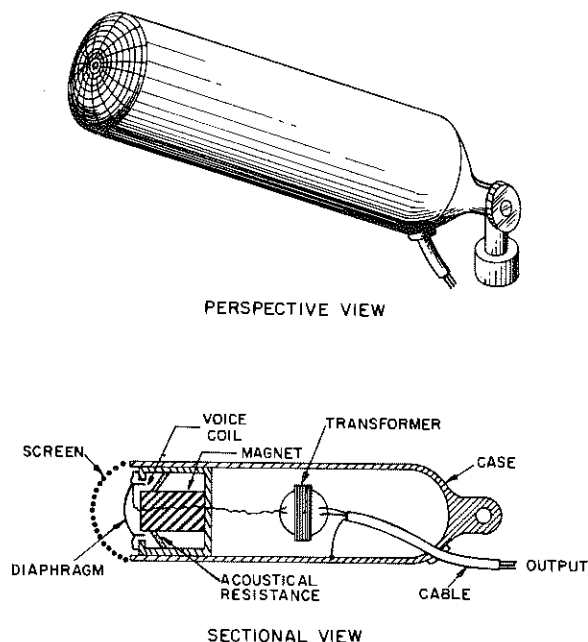


FIG. 9.3. Perspective and sectional views of a dynamic-pressure microphone.

Perspective and cross-sectional views of a dynamic microphone are shown in Fig. 9.3. The diaphragm is coupled to a coil located in a magnetic field. The motion of the diaphragm due to an impinging sound wave is transmitted to the coil. The motion of the coil generates an alternating voltage which corresponds to the pressure variations in the sound wave. The electrical impedance of the dynamic microphone is relatively low, being of the order of 10 to 50 ohms. Therefore, a trans-

former is used to couple the output of the microphone to the grid of a vacuum tube. The low electrical impedance of the dynamic microphone makes it suitable for direct coupling to a transistor amplifier. Very small dynamic microphones of the type shown in Fig. 9.3 have been built for personal use and worn in the manner of a lavalier.

4. Magnetic Microphone. A magnetic microphone is a microphone which depends for its operation on variations in the reluctance of a magnetic circuit. Perspective and sectional views of a magnetic microphone are shown in Fig. 9.4. The variation in reluctance in the magnetic circuit

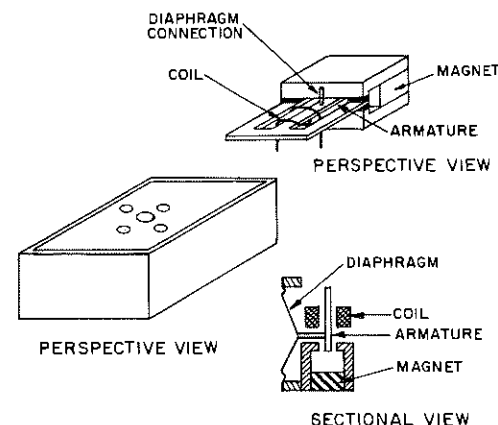


FIG. 9.4. Perspective and sectional views of a magnetic-pressure microphone.

produces a variation in magnetic flux and leads to the induction of a voltage in the coil surrounding the magnetic circuit. In the magnetic microphone shown in Fig. 9.4 the diaphragm is coupled to the armature. Motion of the armature produces variation in the magnetic flux in the armature and leads to the induction of a voltage which corresponds to the motion of the diaphragm. The electrical impedance can be of almost any value up to 5,000 ohms. The electrical impedance can be selected for coupling directly to a transistor. One of the main uses for the magnetic microphone is in hearing aids.

5. Condenser Microphone (Electrostatic Microphone). A condenser or electrostatic microphone is a microphone which depends for its operation on the variations in electrical capacitance. Perspective and sectional views of a condenser microphone are shown in Fig. 9.5. The elements of the condenser microphone consist of the diaphragm and fixed back plate. The impinging sound wave produces motion of the diaphragm and a corre-

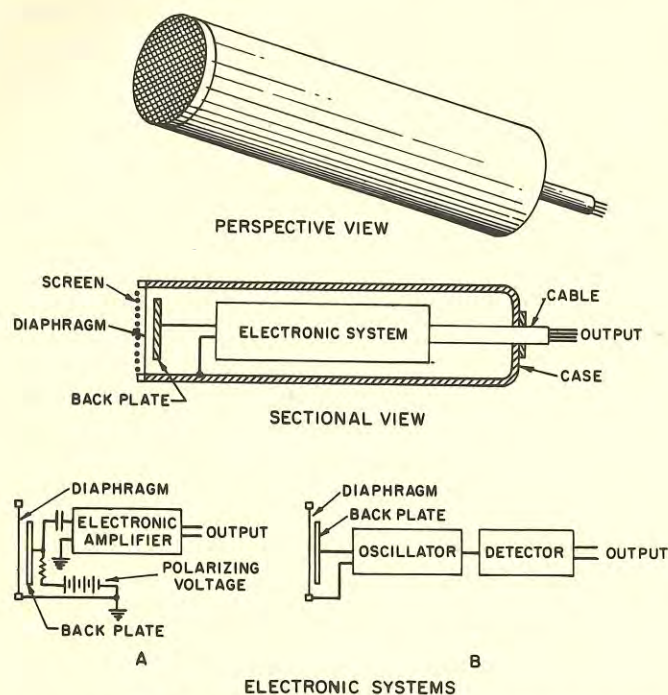


FIG. 9.5. Perspective and sectional views of a condenser or electrostatic-pressure microphone. A. Polarized condenser and electronic amplifier. B. Condenser as an element of an oscillator with a detector.

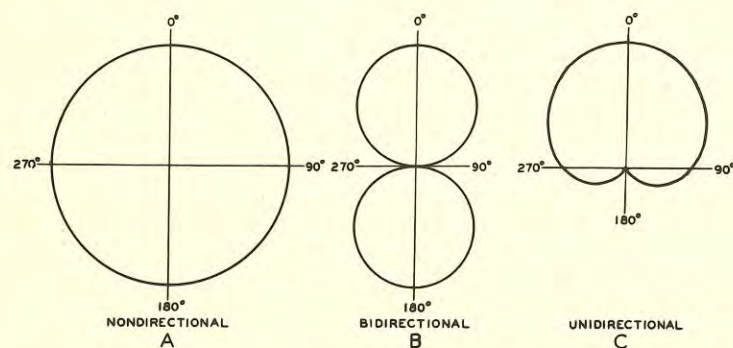


FIG. 9.6. Directional patterns of microphones.

sponding variation in capacitance. In Fig. 9.5A, a polarizing voltage is applied between the diaphragm and the back plate. Under these conditions the electrical output of the diaphragm corresponds to the motion of the diaphragm. The output is amplified by a vacuum tube or transistor amplifier. In Fig. 9.5B, the capacitance of the diaphragm and back plate is a part of the resonant circuit of vacuum tube or transistor oscillator. Motion of the diaphragm produces a variation in the frequency of the oscillator. The output of the oscillator is detected by means of a frequency-modulation detector. The resultant electrical output of the detector corresponds to the motion of the diaphragm.

The response of the carbon, crystal, magnetic, condenser and dynamic microphones described in the preceding sections corresponds to the pressure component in a sound wave. The directivity pattern of these microphones is a nondirectional characteristic, as shown in Fig. 9.6A. This means that the efficiency of pickup is the same in all directions. There is some deviation from this at the high frequencies, where the dimensions of the microphone become comparable to the wavelength of the impinging sound wave. That is to say, the diffraction of the sound wave by the case of the microphone makes the unit somewhat directional in the high-frequency portion of the audio-frequency range.

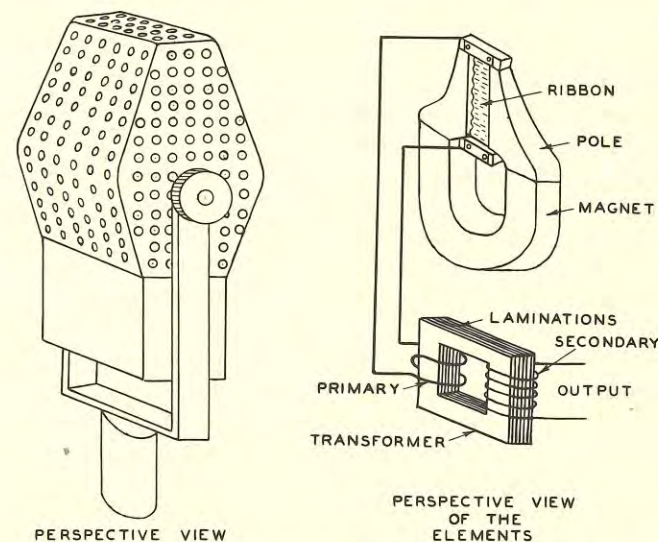


FIG. 9.7. Perspective views of the exterior and the elements of a ribbon-velocity microphone.

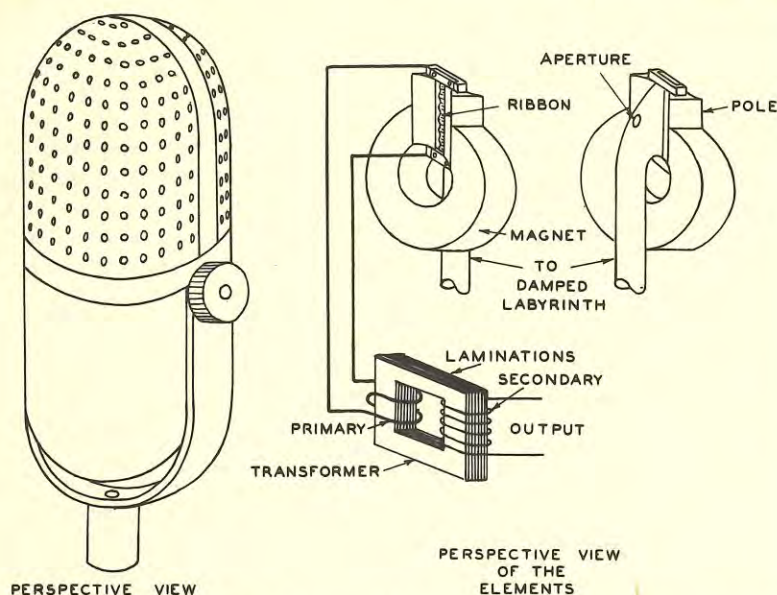


FIG. 9.8. Perspective views of the exterior and the elements of a ribbon-unidirectional microphone.

B. Velocity Microphone

The response of the carbon, crystal, magnetic, condenser and dynamic microphones corresponds to the pressure variations in a sound wave. The response of a velocity microphone corresponds to the particle velocity in a sound wave. The most common type of velocity microphone consists of a ribbon suspended in a magnetic field with both sides of the ribbon freely accessible to the surrounding air (Fig. 9.7). The ribbon is driven by the difference in sound pressure on the two sides. This difference in pressure corresponds to the particle velocity in the sound wave. Therefore, the motion of the ribbon is proportional to the particle velocity in the sound wave. The resultant motion of the ribbon in a magnetic field produces a voltage which corresponds to the particle velocity in the sound wave. The electrical impedance of the ribbon is very low, being about one-fourth ohm. Therefore, a transformer is located in the microphone case to step up the electrical impedance to a value suitable for transmission over a line to the vacuum-tube or transistor amplifier.

The directivity pattern of the velocity microphone is a bidirectional characteristic, as shown in Fig. 9.6B. The directional pattern of the velocity microphone is useful in discriminating against undesirable sounds,

in balancing the instruments of an orchestra, and in monologue and dialogue pickup.

C. Unidirectional Microphones

A unidirectional microphone is a microphone with a substantially unidirectional pattern over the response-frequency range. A unidirectional microphone may be constructed by combining a bidirectional microphone and a nondirectional microphone or by combining a single-element microphone with an appropriate acoustical-delay system.

The elements of a single-unit unidirectional microphone are shown in Fig. 9.8. The back of the ribbon is coupled to an acoustical resistance in the form of a damped pipe which is folded to fit in the lower part of the case. The addition of the aperture in the portion of the pipe covering the back of the ribbon introduces an acoustical element in the form of an inductance which shifts the phase so that the response from the back is very low. The directional pattern under these conditions is a cardioid, as shown in Fig. 9.6C. This characteristic is useful in picking up sounds from the front and discriminating against sounds originating from the rear. The unidirectional microphone is almost universally used in sound-motion-

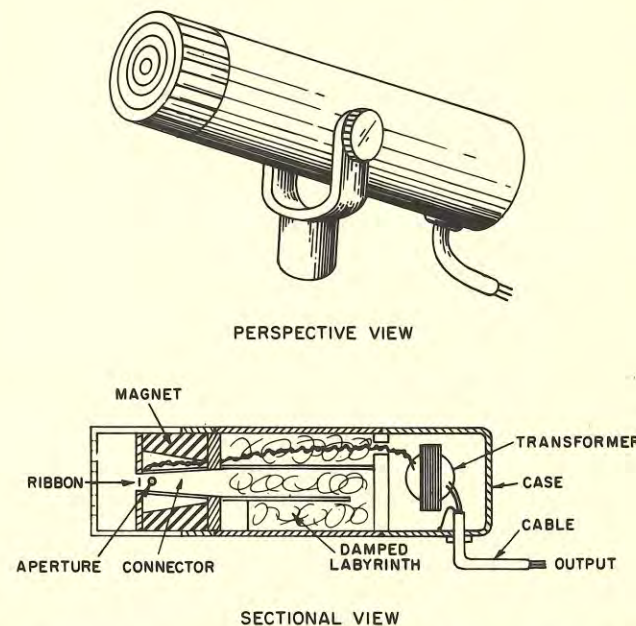


FIG. 9.9. Perspective and sectional views of an axial-ribbon unidirectional microphone.

picture recording, television, stage-sound pickup, and sound-reinforcing systems.

Other directional patterns may be obtained from the microphone shown in Fig. 9.8. For example, if the aperture is closed, a pressure-responsive microphone is obtained with a nondirectional characteristic, as shown in Fig. 9.6A. If the aperture is made very large, a velocity microphone is obtained with a bidirectional characteristic, as shown in Fig. 9.6B.

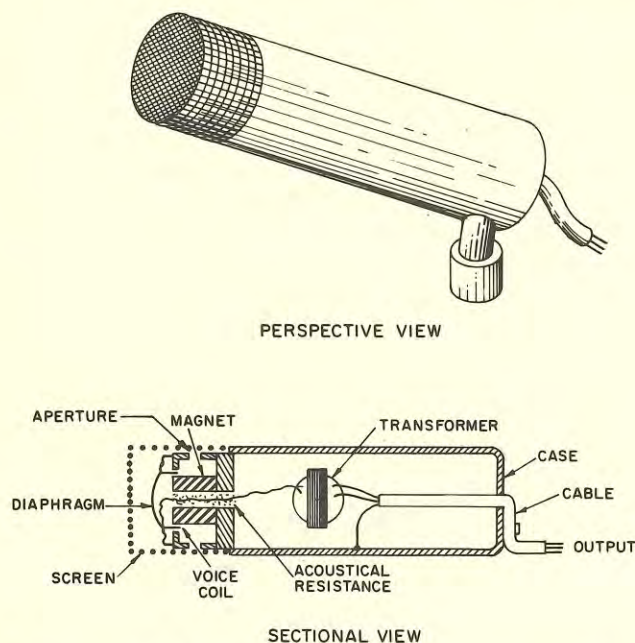


FIG. 9.10. Perspective and sectional views of a dynamic-unidirectional microphone.

The perspective and sectional views of Fig. 9.9 depicts another version of the single-unit unidirectional microphone employing a ribbon coupled to an acoustical resistance in the form of a damped folded pipe. The directivity pattern is the cardioid of Fig. 9.6C. The maximum sensitivity of the unidirectional microphone of Fig. 9.9 is along the axis of the cylinder.

A unidirectional microphone employing a dynamic transducer is illustrated by the perspective and sectional views shown in Fig. 9.10. The back of the diaphragm is coupled to the inertance of the ports in the side of the case and to an acoustical resistance and acoustical capacitance of the air volume of the case. The phase shift introduced by the acoustical

network consisting of the inertance, acoustical resistance and acoustical capacitance is the same as the distance from the ports to the front of the diaphragm. As a result, for sound arriving from the back, the forces on the front and back of the diaphragm are practically equal in phase and amplitude and the response is low. The phase shift is at a maximum for sound arriving in a forward direction with a resultant maximum sensitivity. The directivity pattern of the dynamic-unidirectional microphone of Fig. 9.10 is the cardioid of Fig. 9.6C.

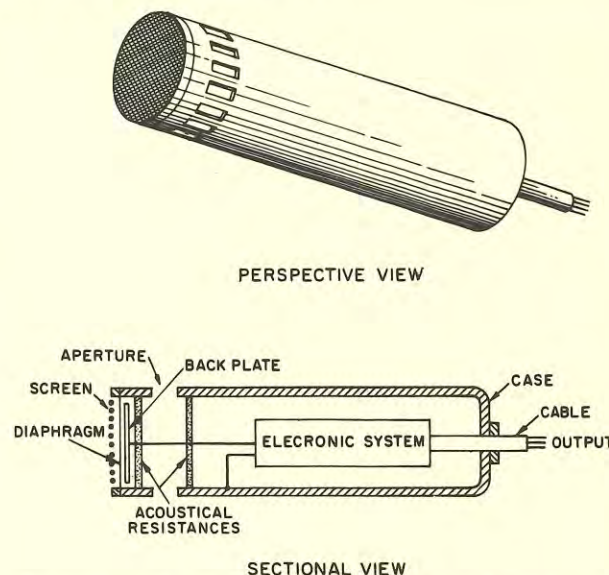


FIG. 9.11. Perspective and sectional views of a condenser or electrostatic-unidirectional microphone.

A unidirectional microphone employing a condenser or electrostatic transducer is depicted by the perspective and sectional views of Fig. 9.11. An acoustical resistance is located behind the diaphragm to provide resistance control of the vibrating system. The back of the diaphragm is coupled to the inertance of the ports in the side of the case and an acoustical resistance and acoustical capacitance of the air volume in the case. The acoustical network is similar to that of the dynamic-unidirectional microphone. The directional pattern is the cardioid of Fig. 9.6C. The maximum sensitivity is along the cylindrical axis of the microphone. The electronic systems which may be used with this microphone are the same as those for the condenser microphone of Fig. 9.5.

A line microphone is a microphone consisting of a large number of equally spaced pickup points arranged on a line. The arrangement may consist of a series of tubes graduated in length and connected to a diaphragm of a condenser microphone as shown in Fig. 9.12A or a pipe with openings along the length with one end closed and the other connected to a diaphragm of a dynamic microphone as shown in Fig. 9.12B. To obtain a high order of directivity the line must be at least a wavelength in length.

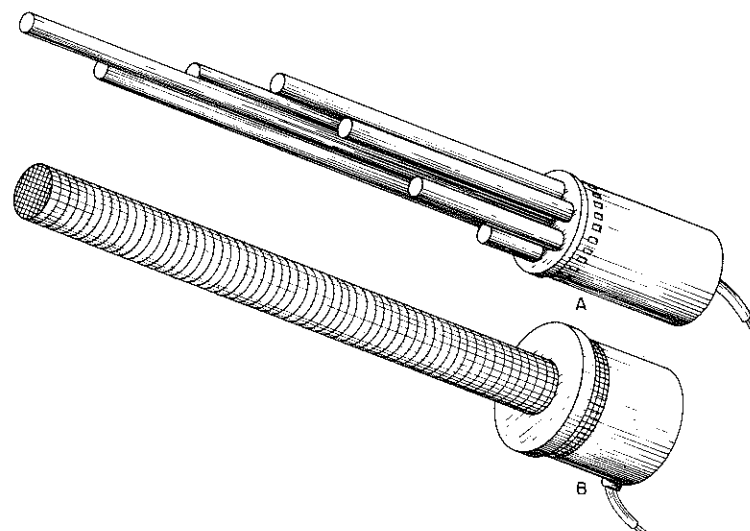


FIG. 9.12. Line microphones. A. A bundle of pipes graduated in length connected to a condenser-unidirectional microphone. B. A pipe with openings along the length and connected to a dynamic-unidirectional microphone.

Therefore, the line becomes long and cumbersome in the low-frequency range. The microphones shown in Fig. 9.12 are cardioid microphones of Figs. 9.11 and 9.10 with a line connected to the diaphragm. The directivity pattern is a cardioid in the low-frequency range. In the mid- and high-frequency ranges the directivity is greater than that of a cardioid microphone.

9.3 LOUDSPEAKERS

A loudspeaker is an electroacoustic transducer actuated by energy in an electrical system and delivering energy to an acoustical system, the waveform in the acoustical system being substantially equivalent to that in the electrical system. There are two general classifications of loudspeakers in

use today, namely, the direct-radiator type and the horn type. In the direct-radiator type a large diaphragm is coupled directly to the air. In the horn type a small diaphragm is coupled to the throat of a horn. Through the years, many types of systems have been used for driving the diaphragm. However, today the dynamic type of drive is almost universally used.

A. Direct-radiator Dynamic Loudspeakers

The almost universal use of the dynamic direct-radiator loudspeaker in radio receivers, phonographs, announce and intercommunicating systems is due to the simplicity of construction, small space requirements, and the relatively uniform response-frequency characteristic. The vibrating system of the dynamic direct-radiator loudspeaker consists of a conical paper diaphragm coupled to a voice coil located in a magnetic field (Fig. 9.13).

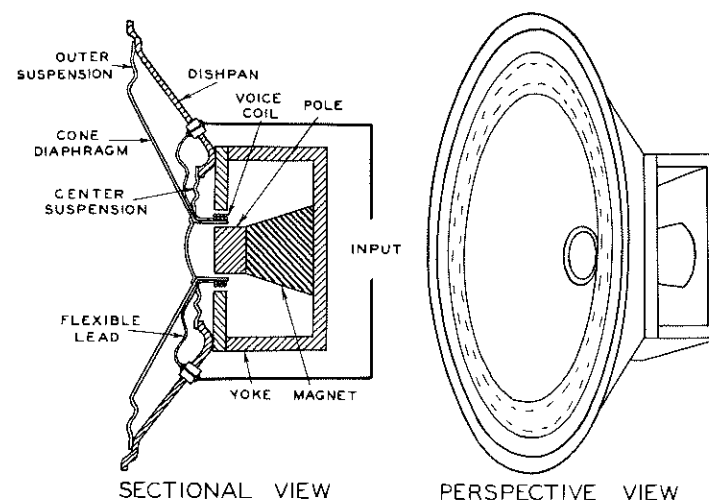


FIG. 9.13. Sectional and perspective views of a dynamic direct-radiator loudspeaker mechanism.

When a current is caused to flow in the voice coil by the application of a voltage, a force is produced by the interaction of the current and the magnetic field which corresponds to the applied voltage. This force is transmitted to the diaphragm and, as result, produces motion of the coil and diaphragm. The motion of the diaphragm produces sound waves. Since the motion of the diaphragm corresponds to the applied voltage, the resultant undulations in the sound wave produced by the motion of the diaphragm corresponds to the variations in the applied alternating voltage.

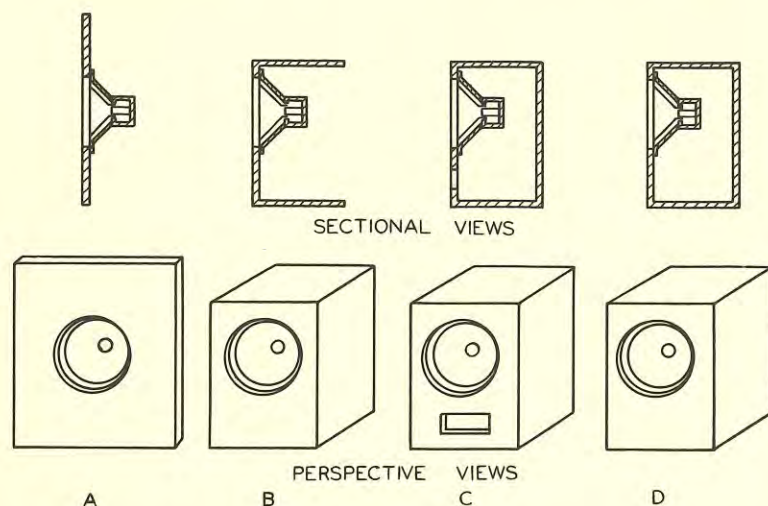


FIG. 9.14. Sectional and perspective views of mounting systems for direct-radiator loudspeaker mechanisms. A. Flat baffle. B. Open-back cabinet. C. Phase inverter or ported cabinet. D. Completely enclosed cabinet.

The direct-radiator loudspeaker is usually mounted in a flat baffle or some type of cabinet enclosure. The most common direct-radiator mounting systems will now be described. A direct-radiator loudspeaker mechanism mounted in a flat baffle is shown in Fig. 9.14A. A relatively large baffle must be used to obtain adequate low-frequency response because the baffle cutoff is determined by the distance from the front to the rear of the mechanism. When this distance is less than a quarter wavelength, the response falls off rapidly below this frequency. To obtain good response to 70 cycles requires a 4-foot baffle. A direct-radiator loudspeaker mechanism housed in a completely enclosed cabinet is shown in Fig. 9.14D. This system will exhibit uniform response to about one-half octave below the fundamental resonant frequency of the combination of the mechanism and cabinet. A direct-radiator loudspeaker mechanism mounted in a completely enclosed cabinet save for a port is shown in Fig. 9.14C. The addition of the port accentuates the low-frequency response but raises the cutoff frequency as compared with a completely enclosed cabinet. A direct-radiator loudspeaker mechanism housed in an open-back cabinet is shown in Fig. 9.14B. This system is used in practically all radio receivers and phonographs both of the table-model and console types. The fundamental resonance of the cabinet as a pipe coupled to the mechanism pro-

duces considerable accentuation of the low-frequency response and is termed cabinet resonance. This accentuation of the low-frequency response has received public acceptance because of the lack of adequate low-frequency components in most music, both original and reproduced.

A single direct-radiator dynamic loudspeaker mechanism is shown in the baffle and cabinets in Fig. 9.14. For wide-frequency range systems, multiple loudspeakers in various arrangements are employed as shown in Fig. 9.15. In Fig. 9.15A, a large direct-radiator dynamic loudspeaker is used to cover the low-frequency range and a small direct-radiator dynamic loudspeaker is used to cover the high-frequency range. In Fig. 9.15B, there are three direct-radiator loudspeakers covering the low-, mid- and high-frequency ranges respectively. In Fig. 9.15C, the direct-radiator

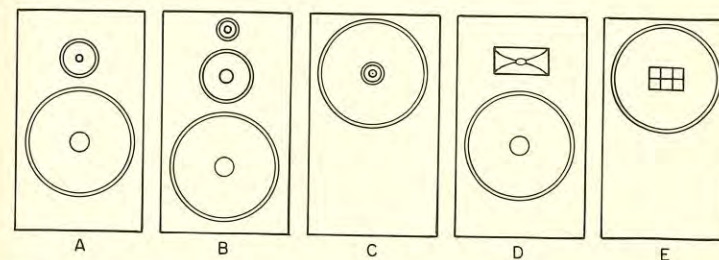


FIG. 9.15. Multiple loudspeaker arrangements. A. Low- and high-frequency direct-radiator loudspeaker mechanisms. B. Low-, mid- and high-frequency direct-radiator loudspeaker mechanisms. C. Low- and high-frequency direct-radiator loudspeaker mechanisms mounted coaxially. D. Direct-radiator low-frequency loudspeaker mechanism and horn high-frequency mechanism. E. Direct-radiator low-frequency mechanism and cellular-horn loudspeaker mechanism mounted coaxially.

high-frequency loudspeaker is coaxially located in the low-frequency direct-radiator loudspeaker. The axial location of the high frequency is such that the surface of the large cone is a continuation of the surface of the small cone. This reduces interference effects in the overlap region and thereby provides a more uniform response and directivity. In Fig. 9.15D, a horn loudspeaker is used to cover the high-frequency range and a direct-radiator loudspeaker is used to cover the low-frequency range. In Fig. 9.15E, a cellular horn loudspeaker, located on the axis of the low-frequency direct-radiator loudspeaker, is used to cover the high-frequency range. In the loudspeaker systems shown in Fig. 9.15, suitable electrical networks are used to divide and allocate the power input to the loudspeaker mechanisms appropriate to the frequency-response ranges of the loudspeaker mechanisms.

B. Horn Loudspeakers

A horn loudspeaker consists of an electrically or mechanically driven diaphragm coupled to a horn. The principal virtue of a horn resides in the possibility of presenting practically any value of acoustical resistance to the driving system. The horn provides a matching means between the large acoustical impedance of the relatively heavy diaphragm and the small acoustical impedance of the relatively light air. The horns used are of the flared type, because this shape provides a more uniform coupling between the diaphragm and the air. The efficiency of horn loudspeakers ranges from 25 to 50 per cent.

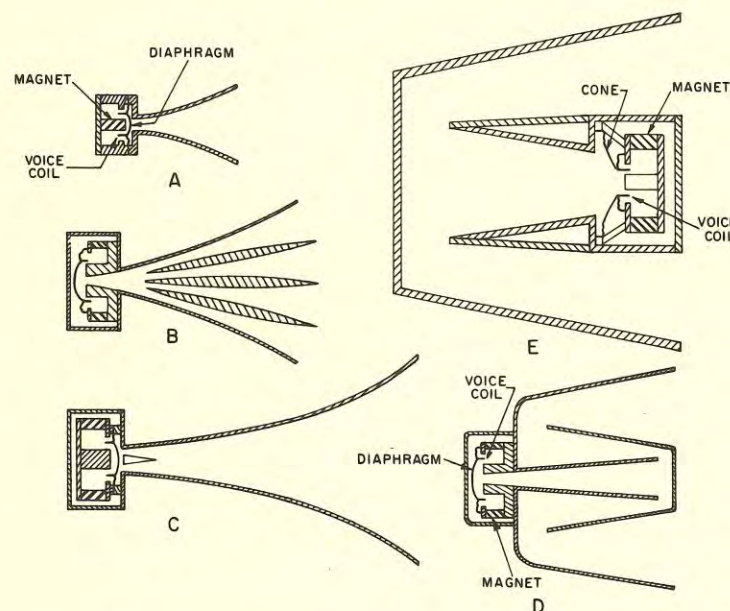


FIG. 9.16. Horn loudspeakers. A. High-frequency horn loudspeaker. B. Cellular high-frequency horn loudspeaker. C. Full-range horn loudspeaker. D. Full-range folded horn loudspeaker. E. Low-frequency folded horn loudspeaker.

Five typical horn loudspeakers are shown in Fig. 9.16. All of the horns are coupled to dynamic-loudspeaker mechanisms. The small horn loudspeaker shown in Fig. 9.16A is used to cover the high-frequency range. An example is the system of Fig. 9.15D. Fig. 9.16B shows a cellular-type horn. The use of a cellular horn maintains essentially uniform directivity

over the response range. The horn loudspeaker of Fig. 9.16B is also used to cover the high-frequency range. In the horn loudspeaker of Fig. 9.16C, a large straight-axis horn is coupled to the dynamic-loudspeaker mechanism. This long and large horn covers the entire frequency range. The horn loudspeaker of Fig. 9.16D is similar to that of 9.16C except that the horn is folded to conserve space in the axial direction. A low-frequency horn loudspeaker is shown in Fig. 9.16E. A large folded horn is coupled to a dynamic-loudspeaker mechanism similar to that of the direct-radiator type.

A horn loudspeaker system consisting of a straight-axis horn loudspeaker for reproducing the high-frequency range and a large folded horn for reproducing the low-frequency range is shown in Fig. 9.17. The loudspeaker system of Fig. 9.17 is used for large-scale applications in sound motion pictures and sound-reinforcing systems.

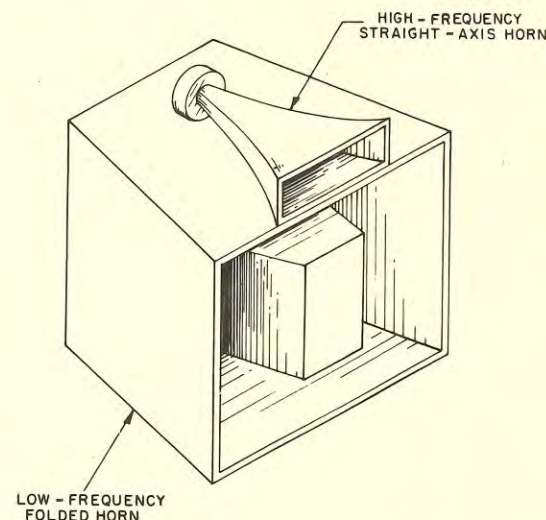


FIG. 9.17. A high-power horn loudspeaker consisting of a low-frequency folded horn loudspeaker and a high-frequency straight-axis horn loudspeaker.

The efficiency of a loudspeaker is the ratio of acoustical-power output to the electrical-power input. The efficiency of the direct-radiator loudspeaker is relatively low, being about 3 to 5 per cent. Therefore, the direct-radiator loudspeaker is not particularly suitable for installations requiring large amounts of sound power as, for example, in sound-motion picture and large-scale high-power sound-reinforcing and public-address

systems because the power amplifiers required to drive the loudspeaker become too large. Under these conditions, it is more economical to employ loudspeakers with higher efficiency. With a horn loudspeaker it is possible to obtain an efficiency of 25 to 50 per cent.

9.4 EARPHONES

An earphone is an electroacoustic transducer actuated by energy in an electrical system and supplying energy to an acoustical system, the waveform in the acoustical system being substantially the same as in the electrical system. The bipolar earphone consists of a steel diaphragm spaced a small distance from the pole pieces which are wound with insulated wire

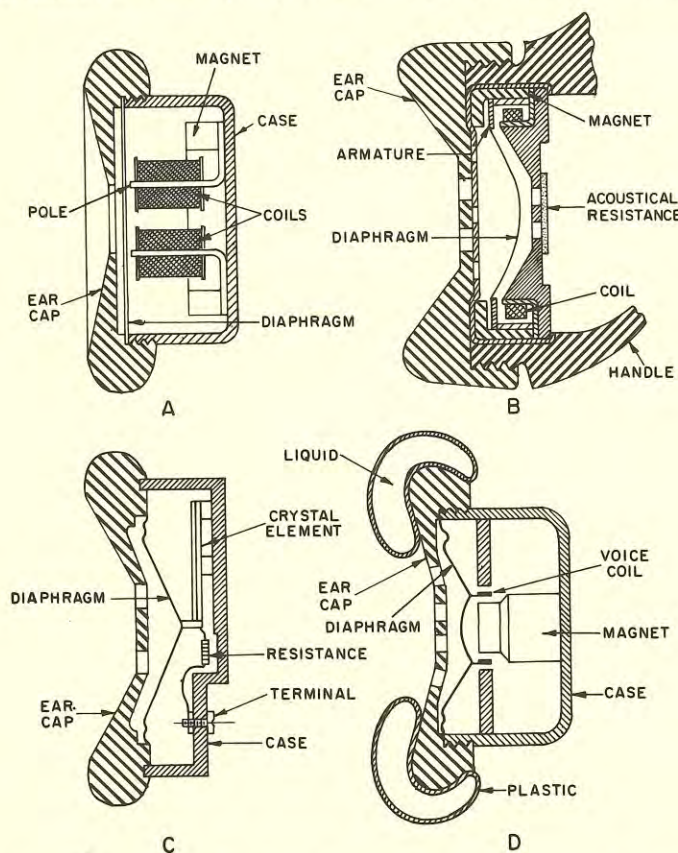


FIG. 9.18. Earphones. A. Bipolar earphone. B. Ring armature earphone. C. Crystal earphone. D. Dynamic earphone.

(Fig. 9.18A). A permanent magnet supplies the steady flux. In the bipolar earphone, the alternating force due to the alternating current in the electromagnet operates directly upon a steel diaphragm. The alternating force is superposed upon the steady force supplied by the permanent magnet. Under these conditions, the motion of the diaphragm and the resultant sound pressure delivered to the ear correspond to the electrical current in the coils.

The frequency response of the bipolar earphone is limited in frequency range and somewhat nonuniform in response with respect to frequency. Therefore, the bipolar earphone is not suitable for high quality monitoring. The use of the bipolar earphone is confined to limited frequency range communication and test applications.

Earphones of the type shown in Fig. 9.18A have been used in telephone systems. However, a more recent development employs an armature of a highly permeable alloy coupled to a diaphragm of a light aluminum alloy as shown in Fig. 9.18B. The diaphragm is damped by means of an acoustical resistance. Relatively smooth response is obtained over a frequency range of 200 to 4,000 cycles. The earphone shown in Fig. 9.18B is used in the modern telephone handset.

In the crystal or ceramic earphone shown in Fig. 9.18C, a small paper cone is coupled to a Rochelle salt or barium titanate element. One of the features of the crystal or ceramic earphone is excellent high-frequency response and light weight.

The vibrating system of the dynamic earphone shown in Fig. 9.18D is similar to that of a small dynamic loudspeaker. Response in the low-frequency part of the audio-frequency range can be maintained by employing a vibrating system with a low-resonant frequency and soft ear caps to eliminate leakage between the cap and the external ear in the low-frequency range. The use of a light-weight vibrating system insures that response will be maintained in the high-frequency part of the audio-frequency range. A properly designed dynamic earphone will reproduce the entire audio-frequency range with good fidelity.

9.5 AMPLIFIERS

An audio amplifier is an electronic transducer in which the output voltage current or power corresponds to and usually exceeds the input voltage current or power. External power is supplied for the transduction process. The most common amplifiers employ vacuum tubes and transistors. The purpose of this section is to describe the action of simple vacuum tube and transistor amplifiers.

The vacuum-tube amplifier revolutionized the reproduction of sound. The advent of the vacuum tube made radio broadcasting, sound-reinforcing

systems, public-address systems, sound motion pictures, long-distance telephony, and other sound-reproducing systems possible. This was due to the fact that the vacuum tube made many electroacoustic transducers practical, whereas before the use had been precluded because of the low efficiency of transduction. The use as an amplifier is the most common application of the vacuum tube. Another application is the oscillator described in Sec. 5.6C.

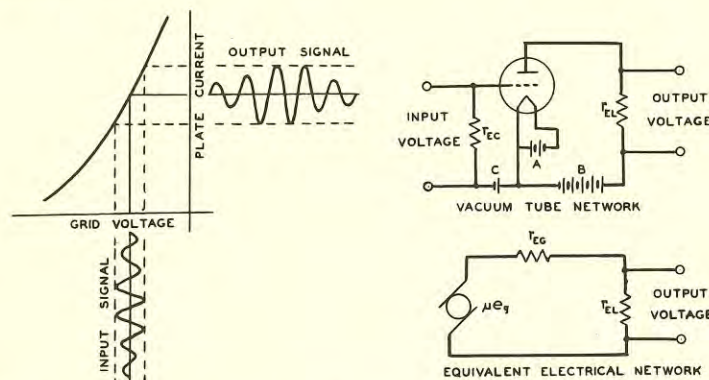


FIG. 9.19. Triode vacuum-tube network, the equivalent electrical network, and the grid-voltage versus plate-current characteristic. e_g = variable grid voltage, μ = amplification factor of the tube, r_{EG} = plate electrical resistance, r_{EL} = load electrical resistance, r_{EC} = grid-bias electrical resistance.

The simplest form of amplifying vacuum tube, termed the triode, consists of the following elements: the cathode, the grid, and the plate, as shown in Fig. 9.19. Electrons are emitted by the heated cathode. The grid, located between the cathode and the plate, controls the electron flow from the cathode to the plate. The number of electrons attracted to the plate depends upon the combined effects of the grid and plate polarities. When the tube is used as an amplifier, a steady negative voltage, termed the bias voltage, is applied to the grid. The input signal is applied to the grid. The variations in the resultant output across the load electrical resistance are essentially of the same form as the input signal voltage, but of an increased amplitude. Since the voltage variation obtained in the plate circuit is much larger than that required to swing the grid, amplification of the signal is obtained. A graphical illustration of the amplification process is depicted in Fig. 9.19. The grid-voltage plate-current characteristic gives the relation between the applied grid signal and the resultant plate-current variation. In general, the voltage developed across

the plate electrical resistance load is greater than the voltage applied to the grid.

The plate current flowing through the load resistance r_{EL} , of Fig. 9.19, causes a voltage drop which varies directly with the plate current. The ratio of this voltage variation produced in the load resistance to the input-signal voltage is the voltage amplification, or gain, provided by the tube. The voltage amplification, $V.G.$, due to the tube is expressed by the following equation:

$$V.G. = \frac{\mu r_{EL}}{r_{EL} + r_{EG}} = \frac{g r_{EG} r_{EL}}{(r_{EL} + r_{EG}) 10^6} \quad (9.1)$$

where μ = amplification factor

r_{EG} = plate electrical resistance of the tube, in ohms

r_{EL} = load electrical resistance of the load, in ohms

g = electrical transconductance, in micromhos

From Eq. (9.1) it can be seen that the gain actually obtainable from the tube is less than the tube's amplification factor but that the gain approaches the amplification factor when the load resistance is large compared with the plate resistance of the tube.

The equivalent generator and electrical circuit of a vacuum-tube amplifier are shown in Fig. 9.19.

The transistor is a three-element solid-state device which may be employed for a wide variety of control functions, including amplification, oscillation and frequency conversion.

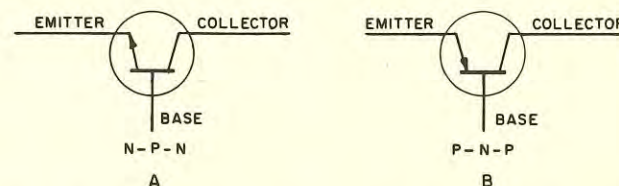


FIG. 9.20. The elements of a transistor. A. n-p-n type. B. p-n-p type.

The transistor is rapidly replacing the vacuum tube in audio amplifiers and oscillators. There are three elements in the common transistor as depicted in Fig. 9.20, namely, the emitter, the base and the collector. There are two varieties of transistors, namely, the n-p-n and the p-n-p as shown in Fig. 9.20.

In the n-p-n transistor shown in Fig. 9.20, electrons flow from the emitter to the collector. In the p-n-p transistor shown in Fig. 9.20, electrons flow from the collector to the emitter. In other words, the direction

of dc electron current is always opposite to that of the arrow on the emitter head. The "conventional current flow" is in the direction of the arrow.

The first two letters of the n-p-n and p-n-p designations indicate the respective polarities of the voltages applied to the emitter and the collector in normal operation. In an n-p-n transistor, the emitter is made negative with respect to both the collector and the base, and the collector is made positive with respect to both the emitter and the base. In a p-n-p transistor, the emitter is made positive with respect to both the collector and the base, and the collector is made negative with respect to both the emitter and the base.

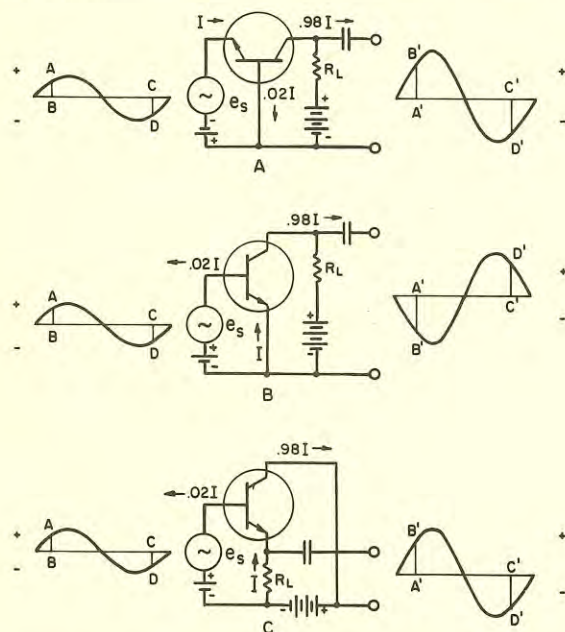


FIG. 9.21. Transistor circuit configurations with input and output waveforms. A. Common base. B. Common emitter. C. Common collector.

There are three basic ways of connecting transistors in a circuit: common-base, common-emitter, and common-collector. In the common-base (or grounded-base) connection shown in Fig. 9.21A the signal is introduced into the emitter-base circuit and extracted from the collector-base circuit. (Thus the base element of the transistor is common to both the input and output circuits.) Because the input or emitter-base circuit has a low impedance, (resistance plus reactance) in the order of 0.5 to 50 ohms, and

the output or collector-base circuit has a high impedance in the order of 1,000 ohms to 1 megohm, the voltage gain is large. The power gain in this type of configuration may be in the order of 1,500.

The direction of the arrows in Fig. 9.21A indicates electron current flow. As stated previously, most of the current from the emitter flows to the collector; the remainder flows through the base. In practical transistors, from 95 to 99.5 per cent of the emitter current reaches the collector. The current gain of this configuration, therefore, is always less than unity, usually in the order of 0.95 to 0.995.

The waveforms in Fig. 9.21A represent the input voltage produced by the signal generator e_s and the output voltage developed across the load resistor R_L . When the input voltage is positive, as shown at AB, it opposes the forward bias produced by the base-emitter battery, and thus reduces current flow through the n-p-n transistor. The reduced electron-current flow through R_L then causes the top point of the resistor to become less negative (or more positive) with respect to the lower point, as shown at A'B' on the output waveform. Conversely, when the input signal is negative, as at CD, the output signal is also negative, as at C'D'. Thus, the phase of the signal remains unchanged in this circuit, that is, there is no voltage phase reversal between the input and the output of a common-base transistor amplifier.

In the common-emitter (or grounded-emitter) connection shown in Fig. 9.21B, the signal is introduced into the base-emitter circuit and extracted from the collector-emitter circuit. This configuration has more moderate input and output impedances than the common-base circuit. The input (base-emitter) impedance is in the range of 20 to 5,000 ohms, and the output (collector-emitter) impedance is about 50 to 50,000 ohms. Power gains in the order of 10,000 (or approximately 40 decibels) can be realized with this circuit because it provides both current gain and voltage gain.

Current gain in the common-emitter configuration is measured between the base and the collector, rather than between the emitter and the collector as in the common-base circuit. Because a very small change in base current produces a relatively large change in collector current, the current gain is always greater than unity in a common-emitter circuit; a typical value is about 50.

The input-signal voltage undergoes a phase reversal of 180 degrees in a common-emitter amplifier, as shown by the waveforms in Fig. 9.21B. When the input voltage is positive, as shown at AB, it increases the forward bias across the base-emitter junction, and thus increases the total current flow through the transistor. The increased electron flow through R_L then causes the output voltage to become negative, as shown at A'B'. During the second half-cycle of the waveform, the process is reversed,

i.e., when the input signal is negative, the output signal is positive (as shown at CD and $C'D'$).

The third type of connection, shown in Fig. 9.21C, is the common-collector (or grounded-collector) circuit. In this configuration, the signal is introduced into the base-collector circuit and extracted from the emitter-collector circuit. Because the input impedance of the transistor is high and the output impedance low in this connection, the voltage gain is less than unity and the power gain is usually lower than that obtained in either a common-base or a common-emitter circuit. The common-collector circuit is used primarily as an impedance-matching device. As in the case of the common-base circuit, there is no phase reversal of the signal between the input and the output.

The circuits shown in Figs. 9.21A, B and C are biased toward n-p-n transistors. When p-n-p transistors are used, the polarities of the batteries must be reversed. The voltage phase relationships, however, remain the same.

9.6 MONAURAL, BINAURAL, MONOPHONIC AND STEREOPHONIC SOUND-REPRODUCING SYSTEMS²

The reproduction of sound is the process of picking up sound at one point and reproducing it either at the same point or some other point or at the same time or some subsequent time. There are many different types of systems employed for the reproduction of sound. In this connection, sound reproducing systems in use today may be classified as follows: monaural, binaural, monophonic, and stereophonic. The purpose of this section is to define and describe the salient features of monaural, binaural, monophonic and stereophonic sound-reproducing systems.

A. Monaural Sound-reproducing System

A monaural sound-reproducing system is a closed circuit type of sound-reproducing system in which one or more microphones are connected to a single transducing channel which in turn is coupled to one or two earphones worn by the listener, as in Fig. 9.22. The most common example of a monaural sound-reproducing system is the telephone in which there is, in general, a single source of sound, one microphone, a transducer, and one earphone coupled to one ear of the listener. In most local applications, the carbon microphone is coupled directly to the earphone. In long-distance telephony vacuum-tube and transistor amplifiers may be used between the microphone and earphone. For other more limited applications, as for example, monitoring purposes, the transducer may be a radio transmitter and receiver, a television sound transmitter and receiver, a disc

² Olson, H. F., *Audio*, Vol. 42, No. 9, p. 28, 1958.

phonograph recorder and reproducer, a sound-motion-picture recorder and reproducer and/or a magnetic tape recorder and reproducer. In some applications, there may be more than one sound source. One or more microphones may be used. In some applications, two earphones may be used transmitting the same program to each of the ears of the listener.

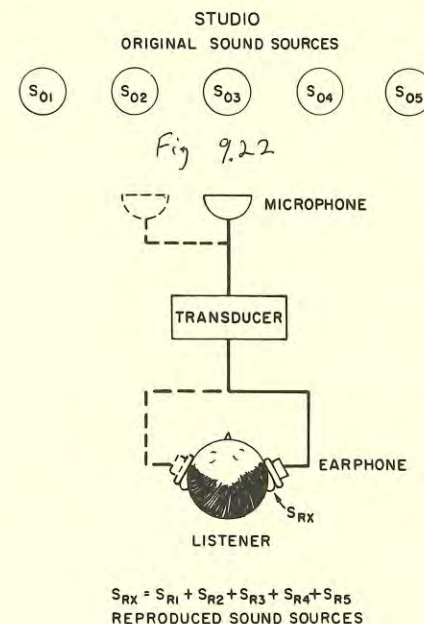


FIG. 9.22. Monaural sound-reproducing system.

The monaural sound-reproducing system is of the closed-circuit type in which the ear of the listener is transferred to a microphone location by means of the microphone, transducer, and earphone combination. The acoustics of a single room are involved in the reproduction of the sound, namely, the studio or room in which the microphone is located.

B. Binaural Sound-reproducing System

A binaural sound-reproducing system is a closed-circuit type of sound-reproducing system in which two microphones, used to pick up the original sound, are each connected to two independent corresponding transducing channels which in turn are coupled to two independent corresponding earphones worn by the listener, as in Fig. 9.23. There is no widespread use of the binaural sound-reproducing system. The use is limited to

specific applications. The binaural sound-reproducing system consists of two separate channels. Each channel consists of a microphone, transducer, and earphone. The microphones are mounted in a dummy simulating the human head in shape and dimensions and at the locations corresponding to the ears of the human head. The transducer may be an

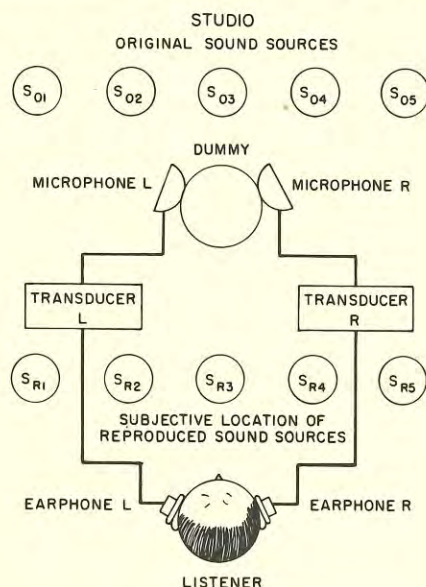


FIG. 9.23. Binaural sound-reproducing system.

amplifier, a radio transmitter and receiver, a phonograph recorder and reproducer, a motion picture recorder and reproducer, or a magnetic tape recorder and reproducer. The binaural sound-reproducing system is of the closed-circuit type. The listener is transferred to the location of the dummy by means of a two-channel sound-reproducing system.

C. Monophonic Sound-reproducing System

A monophonic sound-reproducing system is a field-type sound-reproducing system in which one or more microphones, used to pick up the original sound, are coupled to a single transducing channel which in turn is coupled to one or more loudspeakers in reproduction (Fig. 9.24). The monophonic sound-reproducing system is the most widely employed of all sound-reproducing systems. Examples are the disc phonograph, radio, sound motion picture, television, magnetic tape reproducer and

sound systems. The monophonic sound-reproducing system is of the field type, in which the sound is picked up by a microphone in a field and reproduced by means of a loudspeaker into a field. The sound at the

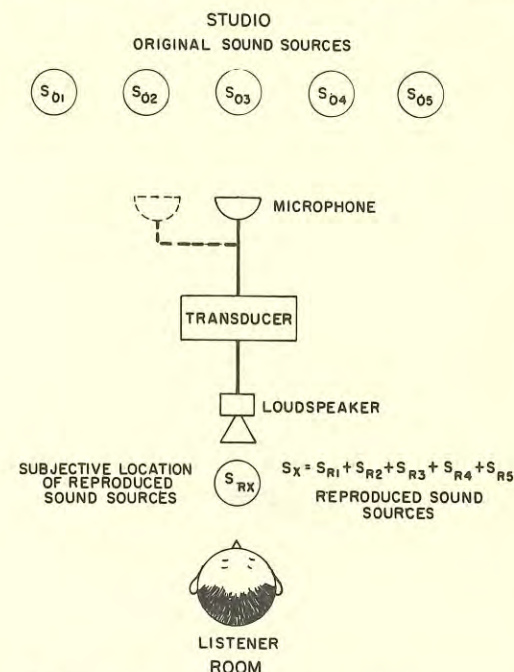


FIG. 9.24. Monophonic sound-reproducing system

microphone is reproduced at the loudspeaker. The transducer may be an amplifier, radio transmitter and receiver, a phonograph recorder and reproducer, a sound-motion-picture recorder and reproducer, a television transmitter and receiver, or a magnetic tape recorder and reproducer. The acoustics of two rooms are involved in monophonic sound reproduction, namely, the studio in which the sound is picked up and the room in which the sound is reproduced.

D. Stereophonic Sound-reproducing System^{3,4,5}

A stereophonic sound-reproducing system is a field-type sound-reproducing system in which two or more microphones, used to pick up the

³ Fletcher, H., *Elec. Eng.*, Vol. 53, No. 1, p. 9, 1934.

⁴ Olson, H. F., *Jour. Audio Eng. Soc.*, Vol. 6, No. 2, p. 80, 1958.

⁵ Olson and Belar, *Jour. Audio Eng. Soc.*, Vol. 8, No. 1, p. 7, 1960.

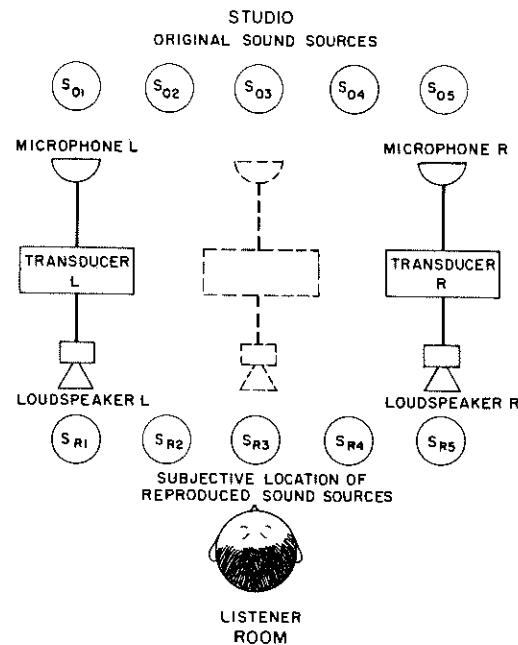


Fig. 9.25. Stereophonic sound-reproducing system.

original sound, are each coupled to a corresponding number of independent transducing channels which in turn are each coupled to a corresponding number of loudspeakers arranged in substantial geometrical correspondence to that of the microphones (Fig. 9.25). The transducer may be an amplifier, radio transmitter and receiver, a phonograph recorder and reproducer, a sound-motion-picture recorder and reproducer, a television transmitter and receiver, or a magnetic tape recorder and reproducer. Two channels are used in the disc phonograph and frequency-modulation radio. Two and three channels are used in the magnetic tape reproducer. Two, three and more channels are used in motion picture reproducers. The acoustics of two rooms are involved in stereophonic sound reproduction, namely, the studio in which the sound is picked up and the room in which the sound is reproduced.

9.7 AUDITORY PERSPECTIVE

Stereophonic sound reproduction as depicted in Fig. 9.25 provides auditory perspective of the reproduced sound. Reproduction of sound in

auditory perspective provides the illusion of the distribution of the reproduced sound in lateral directions as well as in depth in a geometrical configuration and correspondence which approximate the disposition of the original sound sources. This can be illustrated by means of a two-channel stereophonic reproducing system as depicted in Fig. 9.26. The five original sources of sound are S_{O1} , S_{O2} , S_{O3} , S_{O4} and S_{O5} . Under favorable conditions in reproduction, the corresponding reproduced sources of sound appear to originate at S_{R1} , S_{R2} , S_{R3} , S_{R4} and S_{R5} . The two-channel stereophonic sound-reproducing system does provide in actual practice very good reproduction of sound in auditory perspective.

The microphone placement with respect to the sound sources plays an important part in the auditory perspective aspects of the reproduced sound in the living room. The sound sources and the microphone place-

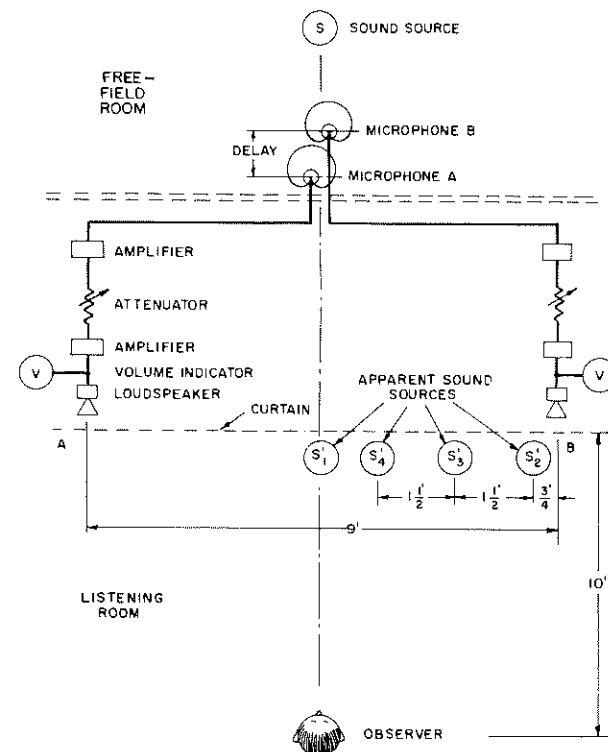


Fig. 9.26. Schematic arrangement of the apparatus which illustrates the effect of the relative delay and amplitude of two spaced sound sources in determining the apparent lateral location of the reproduced sound source.

ment in the studio have been studied by means of listening tests in a typical living room. In this work the most important result has been the observation that the subjective location of a sound source in stereophonic sound reproduction is determined by the phase and amplitude. This can be illustrated by the experiment depicted in Fig. 9.26. In this experiment the difference in phase is obtained by the separation between the microphones. The intensities in the two channels are adjusted by means of the attenuators. When the intensity and phase of the sound emanating from the two loudspeakers are the same, the sound appears to originate from a point midway between the two loudspeakers designated as S'_1 . If the intensity of the sound emanating from loudspeaker B is higher than loudspeaker A the source of sound can be made to appear to originate at S'_4 , S'_3 and S'_2 as the difference in level of the sound emanating from the two loudspeakers is increased. If microphone B is placed closer to the source of sound S than microphone A , the phase of the sound emanating from the loudspeaker in channel B will lead the sound emanating from the loudspeaker in channel A and if the intensity of the sound in the two channels is adjusted to be the same, then the sound can be made to appear at S'_4 , S'_3 and S'_2 as the phase between the two channels is increased. The experiment of Fig. 9.26 can be extended by employing a difference in both the intensity and phase to shift the location of the sound source as perceived by the listener. This experiment shows that the apparent position of the sound source can be shifted in a lateral direction from a point directly in front of either loudspeaker to any point between by varying the relative phases and/or relative intensities of the sound emanating from the loudspeakers of channels A and B .

9.8 TELEPHONE

The telephone is a sound-reproducing system consisting of a carbon microphone (sometimes termed the transmitter), a telephone receiver, and a battery (Fig. 9.27). When sound impinges upon the diaphragm of the carbon microphone, as shown in Fig. 9.27, the diaphragm is moved by the alternating sound pressure. The motion of the diaphragm produces a change in the resistance of the carbon button. The resistance decreases as the motion of the diaphragm compresses the carbon granules and increases as the motion of the diaphragm expands the volume occupied by the carbon granules. Referring to the electrical circuit it will be seen that the current will be an inverse function of the resistance of the carbon button. The change in current produces a corresponding change in the force applied to the diaphragm of the earphone (sometimes termed the receiver) by the electromagnets. This force causes the diaphragm of the earphone to move and thereby produces compressions and rarefactions in

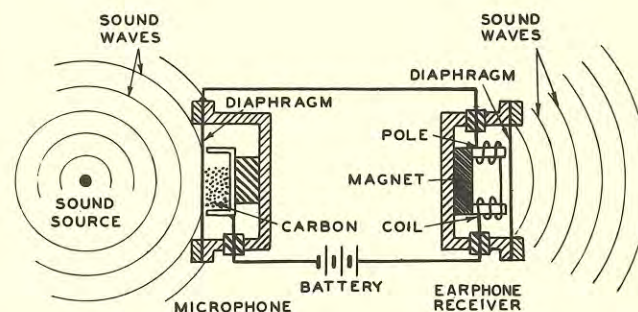


FIG. 9.27. Schematic arrangement of a simple telephone system.

the air which are, of course, the elements of a sound wave. Since the motion of the diaphragm of the microphone, the change in resistance of the carbon button, the current in the electrical circuit, the force applied to the earphone diaphragm, and the motion of the earphone diaphragm all correspond to the undulations in pressure in the impinging sound wave, the sound wave produced by the motion of the earphone diaphragm corresponds to the original sound wave.

The system shown in Fig. 9.27 depicts the action of a telephone system. Much more equipment is needed for a complete telephone system, so that the different telephone subscribers can be connected to each other at will.

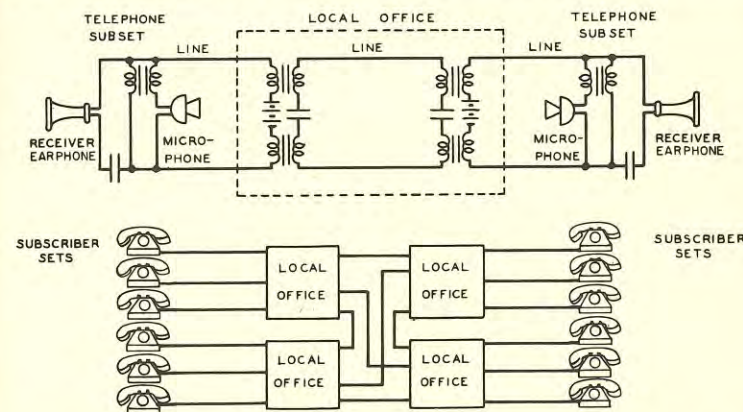


FIG. 9.28. Schematic arrangement of the apparatus in a telephone system. The upper diagram depicts a simple local office. The lower diagram depicts four offices with interconnecting lines.

A schematic diagram of the electroacoustic elements of a modern telephone system is shown in Fig. 9.28. A modern subscriber dial telephone set is shown in Fig. 9.29. The handset contains a carbon microphone as depicted in Fig. 9.1 and an earphone as depicted in Fig. 9.18B. Each telephone station is connected by a line to a central office. The battery supply or other equipment is located at the central office. The function of the central office is to connect any subscriber to any other subscriber. In large cities, there are many central or local offices, because it is not economical or practical for a central office to serve more than about 10,000 stations. The local offices are interconnected by lines, as shown in Fig. 9.28. In local transmission, the electrical output of the microphone is sufficient for the telephone receiver to generate sound of ample loudness

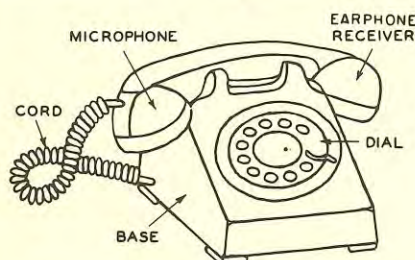


Fig. 9.29. A perspective view of a subscriber dial telephone set.

for intelligent transmission of speech. In long-distance telephony, electronic repeaters are used at regular intervals to restore the level of transmission to normal. The system of Fig. 9.28 depicts the electroacoustic elements of a telephone transmission system. In addition, equipment must be supplied for the subscriber to signal the operator and to permit the operator to send ringing currents to the subscriber. The further consideration of circuits, switchboards, repeaters, manual and automatic exchanges, etc., are outside the scope of this book, and the reader is referred to books on these subjects.

9.9 MAGNETIC TAPE SOUND-REPRODUCING SYSTEM

A magnetic tape sound-recording and -reproducing system consists of a magnetic recording head, magnetic tape, magnetic reproducing head and tape transport mechanism as shown in Fig. 9.30. The additional equipment includes the input and output electronic amplifiers. In recording, the recording magnetic head produces magnetized undulations on the magnetic tape which correspond to the electrical current variations impressed upon the coil in the magnetic recording head. In reproducing, as the tape moves past the reproducing magnetic head, the magnetized elements of the tape produce corresponding magnetic flux variations in the

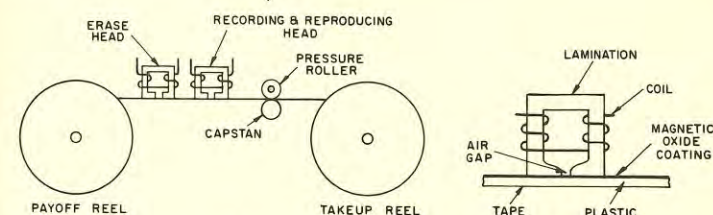


Fig. 9.30. The elements of a magnetic tape sound-recording and -reproducing system showing the reels, magnetic tape recording, reproducing and erase magnetic heads and the capstan drive. The details of the magnetic head and magnetic tape are shown on the right.

magnetic head which lead to an induction of the corresponding electrical voltage in the coil of the reproducing magnetic head.

A complete monophonic-magnetic tape sound-recording and -reproducing system is shown in Fig. 9.31. In recording, switches S_1 , S_2 , S_4 , and S_5 are closed and switch S_3 is placed in position B. The output of the microphone is amplified by an amplifier equipped with suitable equalization to compensate for the response-frequency characteristics of the magnetic tape and the recording and reproducing magnetic heads. A volume control in the form of an attenuator is used to control the recording level. The power amplifier drives the magnetic recording head and loudspeaker. A volume indicator is used in recording so that the appropriate magnetic amplitude will be applied to the magnetic tape. Details of the magnetic-recording system are shown in Fig. 9.30. The recording magnetic head located between the two reels magnetizes the tape in a series of magnetizations which correspond to the variations in the original sound wave. To overcome magnetic nonlinearity in the tape, a high-frequency electrical

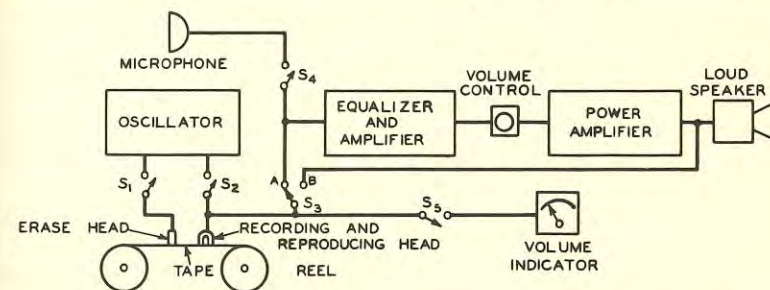


Fig. 9.31. The elements of a complete monophonic-magnetic tape sound-recording and -reproducing system.

signal of 30,000 cycles or higher is fed to the recording head together with the audio signal. The erase head fed from the high-frequency oscillator erases any magnetic modulation which may have been applied to the tape in a previous recording. One of the advantages of a magnetic tape system is that the tape may be used over and over again by erasing the preceding recording. In reproducing, switches S_1 , S_2 , S_4 , and S_5 are opened and switch S_3 is placed in position A. The tape is pulled past the magnetic head at the speed used in recording. As the magnetic variations in the tape travel past the reproducing magnetic head, corresponding variations in flux are induced in the head. These magnetic variations in the magnetic head induce voltages in the coil of the magnetic head which correspond to the magnetic flux variations. The output of the magnetic reproducing head is fed to the equalized amplifier. The output of the amplifier is fed to a volume control which controls the intensity of the reproduced sound in the room. The volume control is followed by the power amplifier which drives the loudspeaker.

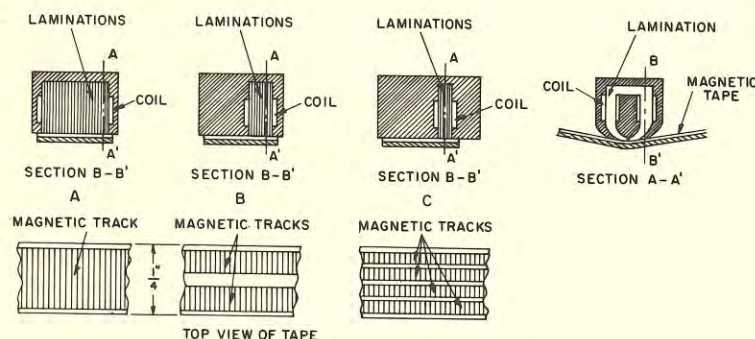


FIG. 9.32. Sectional views of monophonic-magnetic heads and magnetic tracks on the magnetic tape. A. Full-track. B. Half-track. C. Quarter-track.

The monophonic-magnetic tape sound-recording and -reproducing system depicted in Fig. 9.31 is used in practically all consumer-type recorders. The standard width of tape is $\frac{1}{4}$ inch. The consumer type of monophonic-magnetic tape reproducer is usually operated in a small room, or for example, the living room in the home (see Sec. 8.2F and Fig. 8.13).

In the monophonic-magnetic tape system, three different magnetic-track arrangements may be employed as depicted in Fig. 9.32. A full-track magnetic head and the magnetic track on the magnetic tape are shown in Fig. 9.32A. A half-track magnetic head and the magnetic track on the magnetic tape are shown in Fig. 9.32B. A quarter-track magnetic head and the magnetic track on the magnetic tape are shown in Fig. 9.32C.

The tape speeds used in audio-magnetic tape recorders and reproducers are 30, 15, $7\frac{1}{2}$, $3\frac{3}{4}$, $1\frac{7}{8}$, and $\frac{1}{8}$ inches per second.

The magnetic tape is stored on reels of various diameters, as follows, 3, 5, 7 and 10 inches. Magnetic tape is also stored on cartridges of three different designs, namely, two-reel, single-reel and continuous-loop cartridges.

A two-channel stereophonic-magnetic tape recording system is depicted in Fig. 9.33. In addition to the essential elements shown in Fig. 9.33, there may be additional microphones, mixers, compressors, limiters and other electronic compensating elements in each channel. The details of the magnetic head and the magnetic tracks on the tape are shown in Fig. 9.34.

The tape speeds, reels and cartridges are the same as those for the monophonic system described above.

A two-channel stereophonic-magnetic tape reproducing system is shown in Fig. 9.35. In addition to the essential elements shown in Fig. 9.35, there may be various electronic compensating elements. This type of system is usually operated in a room in the home as described in Sec. 8.2F and depicted in Fig. 8.15.

In the case of the master recording, the most common procedure is to record with a three-channel stereophonic system on $\frac{1}{2}$ -inch magnetic tape. The system is similar to Fig. 9.33 except that there are three channels instead of two channels. There may be more than one microphone in each channel with appropriate mixers. The system usually includes compressors and other electronic-compensating elements. The three channels of stereophonic sound are converted to two channels of stereophonic sound as shown in Fig. 9.36. The resultant two-channel master tape is used for the master in the recording of "prerecorded" magnetic tape and the recording of disc records.

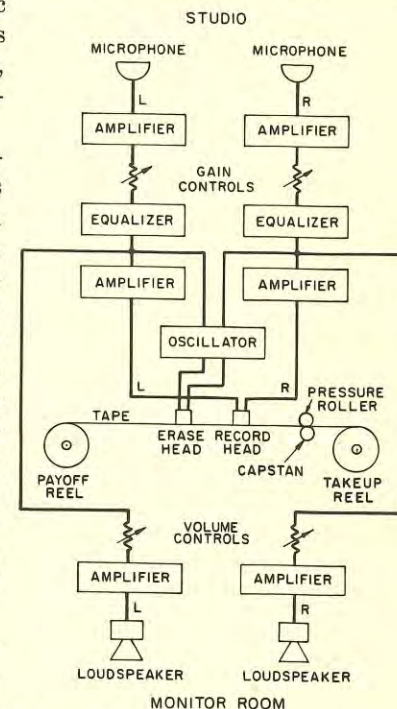


FIG. 9.33. Stereophonic-magnetic tape recording system.

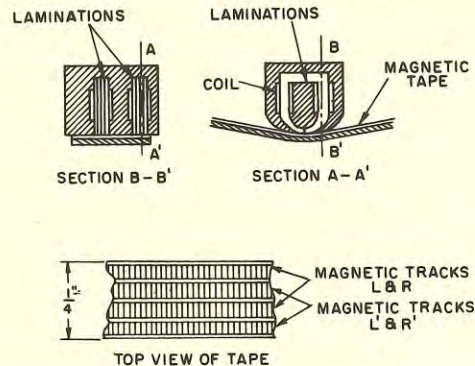


FIG. 9.34. Sectional views of a two-channel stereophonic magnetic head and the location of the four magnetic tracks on the magnetic tape.

There are instances when more than three channels are used. In some cases up to seven channels have been used in master recording. These are then converted to two channels. Regardless of the number of channels, the procedure is essentially the same as for three channels.

The "prerecorded" magnetic tapes contain programs similar to those on disc records. The stereophonic releases are made in various forms, namely, the four-track reels of Fig. 9.35 at a tape speed of $3\frac{3}{4}$ or $7\frac{1}{2}$ inches per second, on the two-reel cartridge at $3\frac{3}{4}$ inches per second, on the single-reel cartridge at $1\frac{1}{2}$ inches per second and on the continuous-reel cartridges at various speeds.

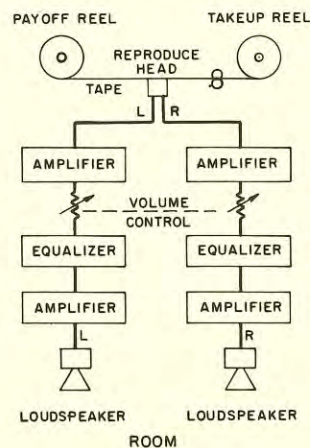


FIG. 9.35. Stereophonic-magnetic tape reproducing system.

The magnetic tape system is also used for recording the master sound record in sound motion pictures. The recording is usually carried out on 35-millimeter magnetic tape (see Fig. 9.60A). The tape is perforated with sprocket holes along the edge for synchronizing the magnetic tape with the picture film in the recording process. The number of magnetic tracks recorded on the 35-millimeter tape may be from two to seven. In general, four magnetic tracks are used.

9.10 DISC PHONOGRAPH

The phonograph was the first sound-reproducing system which made it possible for all the people of the world to hear statesmen, artists, operas, symphonies, bands, and all manner of musical aggregations where only a relatively few had been able to hear these first hand. During the first two decades after the commercialization of the phonograph, the recording and reproducing were made by mechanical means. After the advent of the vacuum tube, recording and reproducing have been made by a combination of electrical and mechanical means.

The mechanical-disc phonograph sound-reproducing system consists of the cutter, the disc record, and the pickup reproducer (Fig. 9.37). The sound wave generated by the sound source travels down the horn and causes the diaphragm located at the throat of the horn to vibrate with a motion corresponding to the variation in the impinging sound wave. This motion in turn is transmitted to the cutting stylus which cuts a wavy groove in the record. Since the motion of the diaphragm and the cutting stylus correspond to the undulations in the pressure of the impinging sound wave, the shape of the wavy groove also corresponds to these varia-

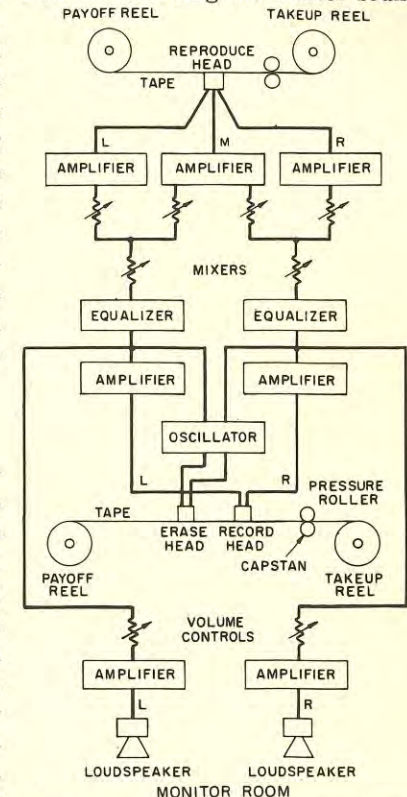


FIG. 9.36. A three-channel stereophonic-magnetic tape reproducer and a two-channel stereophonic-magnetic tape recorder with auxiliary equipment for converting from three-channel stereophonic sound to two-channel stereophonic sound.

tions in sound pressure. In reproduction, when the record is rotated, the wavy groove drives the reproducing stylus, which in turn drives the diaphragm. If the record is rotated at the same speed as that used in recording, the motion of the diaphragm produces a sound wave which corresponds to the original sound wave. The system shown in Fig. 9.37 is the simple original mechanical phonograph. Today, a combination of electrical and mechanical means is used for both recording and reproducing.

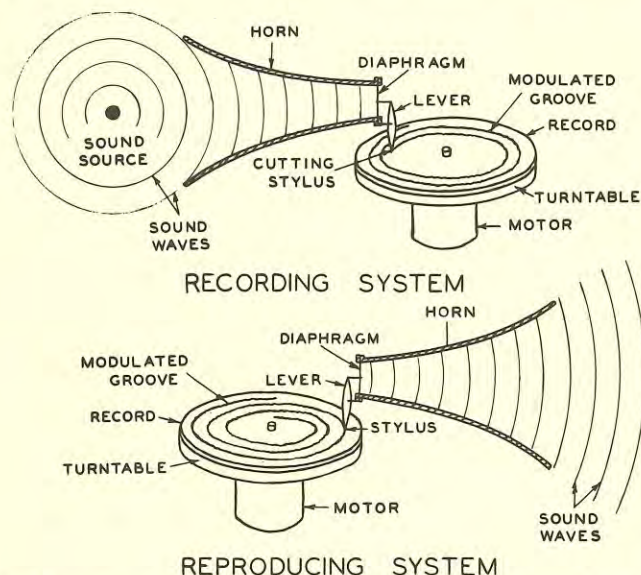


FIG. 9.37. Schematic views of a mechanical-disc phonograph sound-recording and reproducing system.

A complete monophonic-disc phonograph recording system is shown in Fig. 9.38. The sound wave is converted into the corresponding electrical wave by the microphones. The microphones and the source of sound to be recorded are located in the studio (Sec. 8.3C and Fig. 8.27). The output of the microphones is amplified by vacuum-tube amplifiers and applied to the cutter. The mixers make it possible to adjust the outputs from the various microphones and thereby obtain a suitable balance in the recorded sound when several microphones are used to pick up several sound sources. The monitoring system is located in the monitoring room. Compressors, equalizers, and filters are employed to enhance the recorded sound and to compensate for limitations in the system. It will be noted that the output

is also fed to a monitor system which enables the recording engineer to determine the quality of the recording as applied to the cutter.

The cutter used for recording monophonic-lateral phonograph records is shown in Fig. 9.39. The driving system is of the dynamic type with two wire coils located in a magnetic field. One coil, used as the driving coil, is connected to the output of the power amplifier. The other coil, used as a sensing coil, is fed in an out of phase relationship to the input of the amplifier as shown in Fig. 9.38. The use of a feedback system provides uniform response and low nonlinear distortion. A current in the driving coil causes motion of the cutting stylus in a lateral direction which corresponds to the electrical input. The motion is transferred to the stylus which cuts a wavy groove in the lacquer disc. The undulations in the wavy groove correspond to the variations in the original sound wave.

The general practice is to record the master on magnetic tape instead of on the lacquer original. The lacquer original is cut from the magnetic tape master as shown in Fig. 9.40. The process of cutting the lacquer disc is the same as that described for Figs. 9.38 and 9.39.

The steps for the mass production of discs are shown in Fig. 9.41. In the record plant the lacquer disc recording termed the "original" of Fig. 9.41A is metalized and then electroplated. The plating is separated from the lacquer and reinforced by backing with a solid metal plate. The assembly is termed the "master" (Fig. 9.41B). The master is electro-

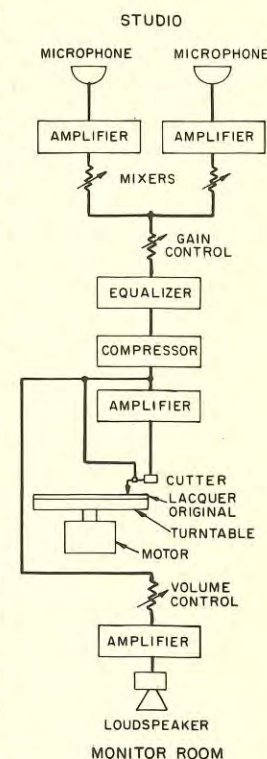


FIG. 9.38. Monophonic-disc recording system.

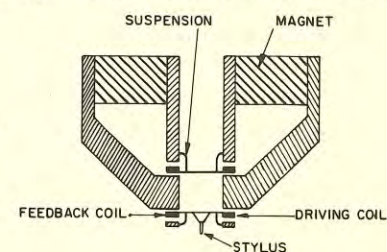


FIG. 9.39. A monophonic lateral-disc cutter.

plated. This plating is separated from the master and reinforced by backing with a solid metal plate. The assembly is termed the "mother" (Fig. 9.41C). Several mothers may be made from the master. The mother is electroplated. This plating is separated from the mother and reinforced by a solid metal plate. The assembly is termed the "stamper" (Fig. 9.41D). Several stampers may be made from each mother. One

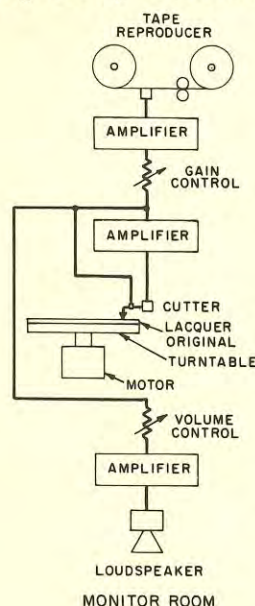


FIG. 9.40. A monophonic-magnetic tape reproducer and a monophonic disc recorder and auxiliary equipment for the production of disc masters from the master magnetic tape.

The record is rotated by the reproducing turntable at the same angular speed as that used in the original recording. The stylus or needle of the pickup follows the wavy spiral groove and generates a voltage corresponding to the undulations in the wavy groove of the record. The phonograph pickup shown in Fig. 9.42 is of the crystal type. Many other methods of transduction are used for phonograph pickups as, for example, magnetic, dynamic, electrostatic, ceramic, and electronic. In the crystal pickup, the stylus is connected to a Rochelle salt crystal by means of a lever system. When the crystal is twisted by the motion of the stylus, a voltage is developed which corresponds to the deformation.

stamper containing a sound selection to be placed on one side of the final record is mounted in the upper jaw, and another stamper containing a sound selection to be placed on the other side of the record is placed in the lower jaw of a hydraulic press equipped with means for heating and cooling the stampers (Fig. 9.41E). A preform or biscuit of thermoplastic material such as a shellac compound or vinylite is placed between the two stampers. The stampers are heated, and the jaws of the press are closed which presses the two stampers against the thermoplastic material. When an impression of the stampers has been obtained in the thermoplastic material, the stampers are cooled which cools and sets the plastic record. The jaws of the hydraulic press are opened, and the record is removed from the press. The modulated grooves in the record correspond to those in the original lacquer disc (Fig. 9.41F). The stamping procedure is repeated again and again until sufficient records are obtained. The above process constitutes the "mass-production system" for the production of phonograph records.

The monophonic system for reproducing the disc phonograph record is shown in Fig. 9.42.

A barium titanate ceramic element may be used instead of the Rochelle salt element. The action and general performance is the same except that a phonograph pickup with a ceramic element exhibits slightly lower sensi-

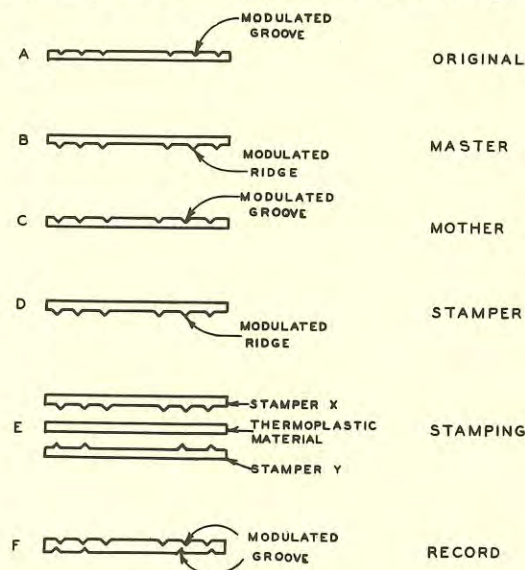


FIG. 9.41. The steps in the process for the mass production of disc phonograph records.

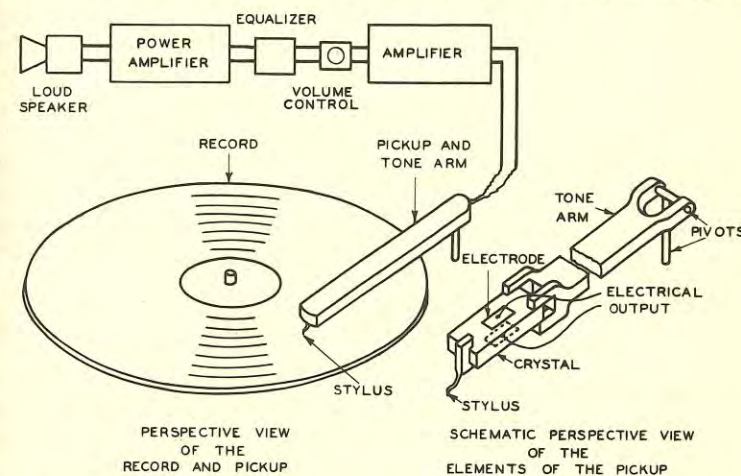


FIG. 9.42. A monophonic-disc reproducing system. Perspective view, schematic diagram of the elements and a schematic perspective view of the elements of a monophonic phonograph pickup.

tivity. The barium titanate element will operate at higher temperatures than the Rochelle salt element without suffering permanent damage.

The output of the pickup is amplified by means of an electronic amplifier. Suitable electrical equalizers are used to compensate for the response of

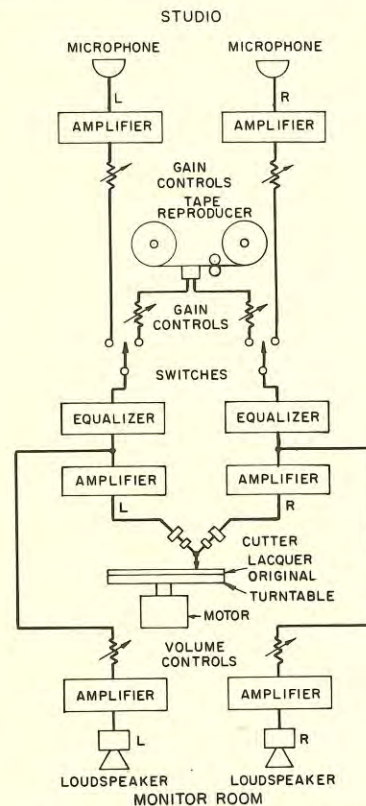


FIG. 9.43. Stereophonic-disc recording system. The source may be microphones in a studio or a two-channel stereophonic master magnetic tape.

original from a master tape as shown in Fig. 9.43. The two-channel stereophonic-master magnetic tape is produced from the three-channel stereophonic magnetic tape as shown in Fig. 9.36. A schematic-sectional view of a stereophonic cutter is shown in Fig. 9.44. The elements in the two channels are disposed at 90° with respect to each other. The driving

⁶ Frayne and Davis, *Jour. Audio Eng. Soc.*, Vol. 7, No. 4, p. 146, 1959.

the pickup and the recording characteristics. A volume control in the form of an attenuator is employed to control the intensity of the reproduced sound. The volume control is followed by an electronic power amplifier which drives the loudspeaker. The phonograph reproducing instrument is usually operated in a small room as, for example, the living room in the home (Sec. 8.2F and Fig. 8.13).

Dynamic and magnetic pickups are also employed for the reproduction of monophonic disc records. The construction of stereophonic dynamic and magnetic pickups will be described later in the text. Therefore, monophonic dynamic and magnetic pickups will not be described.

The essential elements of a stereophonic-disc recording system⁶ are shown in Fig. 9.43. Two means of recording the lacquer original are shown in Fig. 9.43. The lacquer original may be recorded directly from the sound pickup in the studio. This procedure is employed in very limited applications. The general practice is to record the lacquer

coil is connected to the output of the amplifier. The feedback coil is connected to the input of the amplifier.

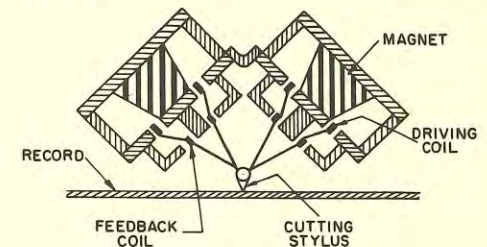


FIG. 9.44. Sectional view of a stereophonic disc cutter.

Four modulation possibilities of the 45° - 45° stereophonic-disc recording system are shown in Fig. 9.45 as follows: signal on the left channel only,

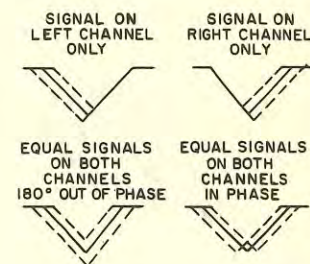


FIG. 9.45. Groove modulations for various signal conditions in the 45° - 45° stereophonic disc.

signal on the right channel only, equal signals on both channels 180° out of phase and equal signals on both channels in phase. The signal recorded

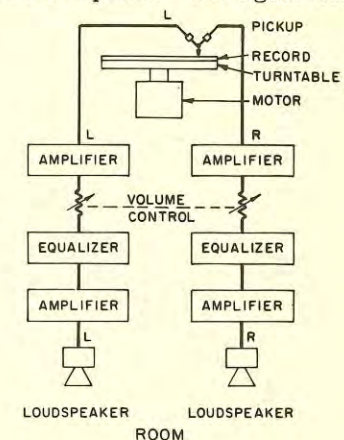


FIG. 9.46. Stereophonic-disc reproducing system.

in the right and left channels are independent in both recording and reproducing.

The mass production of stereophonic disc records is the same as for the monophonic disc records depicted in Fig. 9.41.

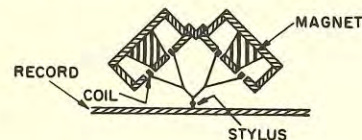


FIG. 9.47. Sectional view of a stereophonic-dynamic pickup:

The elements of stereophonic-disc reproducing systems are shown in Fig. 9.46. The stereophonic-disc reproducing system shown in Fig. 9.46 is usually operated in a room in the home as described in Sec. 8.2*F* and depicted in Fig. 8.15.

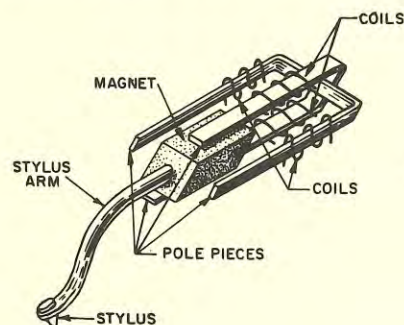


FIG. 9.48. Perspective view depicting the elements of a stereophonic-magnetic pickup.

A schematic-sectional view of a stereophonic-dynamic pickup for the reproduction of a stereophonic disc record is shown in Fig. 9.47. The electrical output in the right and left channels of the pickup correspond to

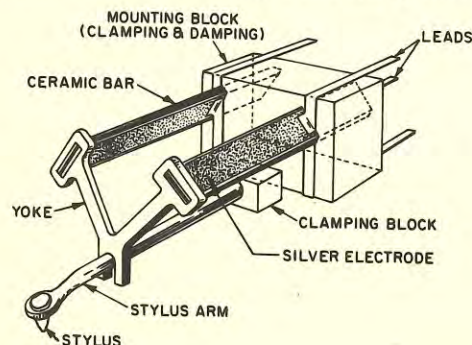


FIG. 9.49. Perspective view depicting the elements of a stereophonic-ceramic pickup.

the modulation in the right and left walls of the record which in turn correspond to the inputs to the right and left channels of the cutter.

A large number of magnetic and stereophonic pickups have been developed for the reproduction of stereophonic disc records. The magnetic pickup depicted in Fig. 9.48 exhibits superior performance from the standpoint of uniform response with respect to frequency and a low mechanical impedance presented to record groove by the stylus. Considerable electrical compensation is required to obtain uniform output from the magnetic pickup. However, the ceramic pickup is most widely used because the output is high and practically no compensation with respect to frequency is required. The essential elements of a ceramic pickup are shown in Fig. 9.49. In general, the frequency response of the ceramic pickup is not as smooth as the magnetic pickup. Furthermore, a relatively high mechanical impedance is presented to the record at certain portions of the frequency range. The electrical impedance of a magnetic pickup is relatively low, whereas the electrical impedance of the ceramic pickup is relatively high.

9.11 RADIO SOUND-REPRODUCING SYSTEM

A monophonic amplitude-modulation radio transmitter is shown in Fig. 9.50. In the radio transmitter of Fig. 9.50, the sound wave is converted into corresponding electrical waves by the microphone. The electrical outputs of the microphones are amplified by electronic amplifiers and fed to mixers. The microphones and source of sound to be picked up are located in the studio (Sec. 8.3*B* and Fig. 8.26). The mixers, volume control, amplifiers, and monitoring system are located in the control room. A soundproof glass wall partition which separates the studio and control room gives the engineer full view of the action in the studio (Sec. 8.3*B*). The output of the compressor amplifier is coupled to an equalizer followed by an amplifier. The output of the amplifier is fed to the monitor-

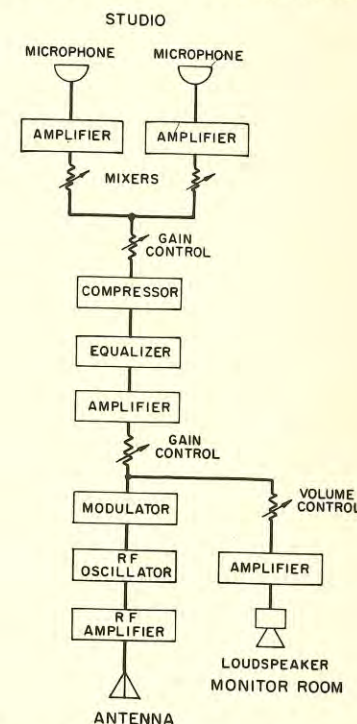


FIG. 9.50. Monophonic amplitude-modulation radio transmitter.

ing system and the modulator. The monitoring system is located in the control room. In the standard broadcast band, the frequency of the oscillator lies somewhere between 550 to 1,700 kilocycles. The modulator varies the amplitude of the oscillator. The amplitude of the variations corresponds to the variations in the original sound wave. The output of the oscillator is coupled to the radio-frequency power amplifier. The

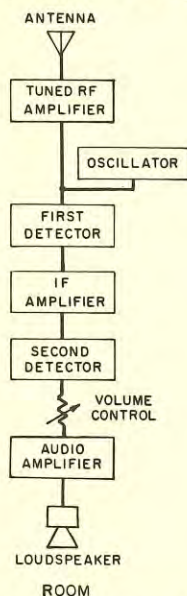


Fig. 9.51. Monophonic amplitude-modulation radio receiver.

to the undulations in the original sound wave. The second detector is followed by a volume control which controls the intensity of the reproduced sound in the room. The volume control is followed by a power amplifier which drives the loudspeaker. The loudspeaker converts the electrical variations into the corresponding sound vibrations. The latter corresponds to the original variations in the studio. The radio receiver is usually operated in a small room, as for example, the living room in the home (Sec. 8.2F and Fig. 8.13).

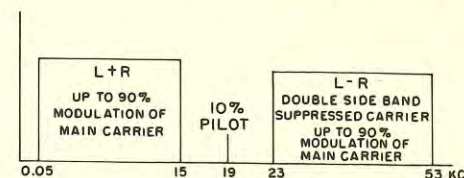
The radio sound-reproducing system shown in Figs. 9.50 and 9.51 employs amplitude modulation. Frequency modulation is also used for standard sound broadcasting and for television sound broadcasting. The difference between amplitude-modulation (AM) and frequency-modulation (FM) systems resides in the modulator in the transmitter and in the detector in the receiver. In the amplitude-modulation system shown in

the radio-frequency power amplifier is coupled to the transmitting antenna. The modulated radio-frequency wave is transmitted in the ether in all directions. The input to the AM transmitter shown in Fig. 9.50 is provided by sound pickup by microphones. The input to the transmitter may also be disc records or prerecorded magnetic tape.

In the monophonic amplitude-modulation radio-receiving system of Fig. 9.51 a very small portion of the radio frequency radiated by the transmitter antenna is picked up by the receiving antenna. The output of the receiving antenna is amplified by a radio-frequency vacuum-tube amplifier. The output is combined with that of a higher radio-frequency oscillator and fed to the first detector. The resultant intermediate-frequency carrier, with the original modulation, is amplified by the intermediate-frequency amplifier. The second detector converts the modulated intermediate-frequency carrier to an audio-frequency electrical wave in which the variations correspond

Fig. 9.50 the amplitude of the oscillator is varied to correspond to the audio-frequency signal. In the frequency-modulation system, the frequency of the oscillator is varied to correspond to the audio-frequency signal. Some form of frequency discriminator is used in the receiver to convert the frequency modulations into the corresponding audio variations.

Fig. 9.52. Spectrum of the signals of stereophonic frequency-modulation radio.



Standards for stereophonic radio transmission in the frequency-modulation band in the United States was established by the U. S. Federal Com-

munications Commission in 1962. The standards provide for the transmission of the sum of the left and right stereophonic audio signals as the direct-frequency modulation of the radiated carrier as depicted by the spectrum diagram of Fig. 9.52. The difference between the left and right stereophonic audio signals is used to amplitude-modulate a 38-kilocycle subcarrier in suppressed-carrier fashion, with the result impressed as frequency modulation on the radiated carrier as shown in Fig. 9.52. In addition, a pilot subcarrier of 19 kilocycles, shown in Fig. 9.52, is transmitted for synchronizing the FM receiver.

A typical FM stereophonic radio transmitter⁷ is shown in Fig. 9.53. The outputs of the right, R, and

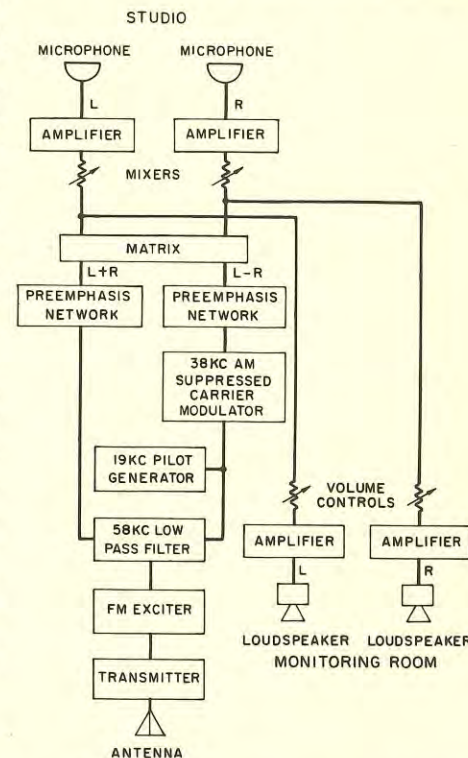


Fig. 9.53. Stereophonic frequency-modulation radio transmitter.

⁷ Csicak and Lima, *Jour. Audio Eng. Soc.*, Vol. 10, No. 1, p. 2, 1962.

left, L, microphones are amplified and fed to a matrix. In general, more than one microphone is used in each channel. The output of the matrix contains two audio signals, the $L + R$ and the $L - R$. Both are passed through preemphasis networks which accentuate the high-frequency response with respect to frequency. The $L + R$ preemphasized audio signal is fed directly to the FM exciter. The $L - R$ preemphasized audio signal is applied to a 38-kilocycle AM double-sideband modulator in which the 38-kilocycle carrier is suppressed. A 19-kilocycle signal is produced by a pilot generator. The suppressed-carrier modulation signal of $L - R$ and pilot signal are combined and passed through a 58-kilocycle filter and on to an FM exciter. The output of the FM exciter is coupled to the FM transmitter, which in turn is coupled to the FM antenna. The input to the FM radio transmitter shown in Fig. 9.53 is provided by sound pickup by microphones. The input to the FM radio transmitter may also be disc records or prerecorded magnetic tape.

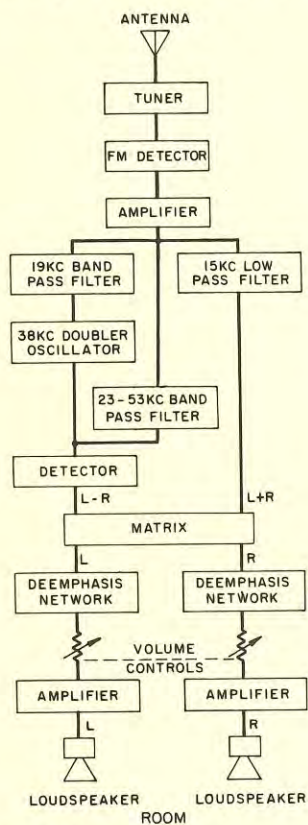


FIG. 9.54. Stereophonic frequency-modulation radio receiver.

A typical FM stereophonic radio receiver is shown in Fig. 9.54. The antenna, tuner, and detector are standard FM units. The output of the detector is amplified and fed to the following: A low-pass filter, a 23- to 54-kilocycle bandpass filter, and a 19-kilocycle narrow-bandpass filter. The output of the 19-kilocycle filter is coupled to a 38-kilocycle doubler oscillator which generates the 38-kilocycle carrier. The carrier is combined with the two sideband signals in the detector. The resultant detector output is the $L - R$ audio signal. The $L + R$ and $L - R$ audio signals are combined in the matrix, the output of which yields the separate L and R audio signals. The deemphasis network normalizes the response with respect to frequency of the audio signals. The audio signals are amplified and converted to left and right sound signals in air by the loudspeakers. The stereophonic frequency-modulation radio receiver shown in Fig. 9.54 is usually operated in a room in the home as described in Sec. 8.2F and depicted in Fig. 8.15.

The FM stereophonic system provides high-quality stereophonic sound reproduction. The essential characteristics are a frequency range of 30 to 15,000 cycles, a separation between left and right signals of 30 decibels and nonlinear distortion in accordance with existing FCC requirements.

9.12 SOUND-MOTION-PICTURE REPRODUCING SYSTEM

A complete monophonic sound-motion-picture recording system is shown in Fig. 9.55. The outputs of the microphones are amplified and fed to attenuators termed mixers. The microphones and source of sound to be recorded are located on the sound stage (Sec. 8.3I and Fig. 8.36). If more than one microphone is used as, for example, a soloist accompanying an

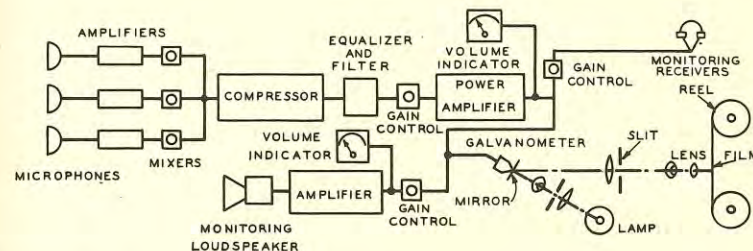


FIG. 9.55. The elements of a monophonic sound-motion-picture film recording system.

orchestra, one microphone for the soloist and one for the orchestra, the output of the two may be adjusted for the proper balance by the mixer system. The mixer system is followed by a compressor which reduces the volume range. A low-pass filter is usually used to reduce ground noise above the upper limits of reproduction. A high-pass filter is used on speech with the lower limit placed below the speech range. This latter expediency reduces low-frequency noises without impairing the speech quality. An equalizer is used to accentuate the high frequencies and thereby compensate for the film-transfer loss in the high-frequency range. A gain control in the form of an attenuator is used to control the over-all volume. The output of the power amplifier is coupled to the light modulator and the monitoring system. By means of the optical system and light modulator the electrical variations are recorded on the film into the corresponding variations in density (termed variable-density recording) or in area (termed variable-area recording). The recording system depicted in Fig. 9.55 is of the variable-density type. The optical and modulating system for both variable-density and variable-area recording will be subsequently described in greater detail in this section. The monitoring system is also connected to the output of the recording amplifier. In general, the monitoring system is equalized so that the reproduced sound

heard in the monitoring system simulates that of the ultimate reproduced sound heard in the theater. An amplifier and loudspeaker are used if the mixing and monitoring are carried out in a booth on the sound stage or in a monitoring room next to the sound stage. Headphones are used if the monitoring is carried out on the stage (Sec. 8.37 and Fig. 8.36). Volume indicators are provided to ensure that the proper amplitude is obtained in the modulation system.

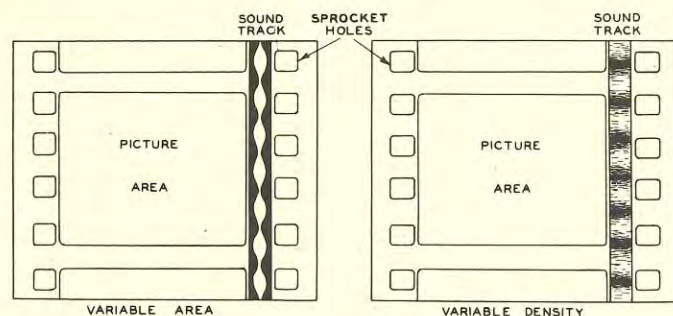


FIG. 9.56. The position of the picture and sound track in 35-millimeter sound-motion-picture film. Two types of track are shown, namely, variable area and variable density. (After Olson, *Acoustical Engineering*, D. Van Nostrand Company, Inc., Princeton, 1957.)

The sound track on 35-millimeter film occupies a space about 0.1 inch wide. There are two types of sound track in general use today, namely, variable area and variable density (Fig. 9.56). The type of variable-area sound track shown in Fig. 9.56 is termed bilateral variable area.

In the variable-area system the transmitted light amplitude is a function of the amount of unexposed area in the positive print. This type of sound track is produced by means of a mirror galvanometer which varies the width of the light slit under which the film passes. The elements of a variable-area recording system are shown in Fig. 9.57. The triangular aperture is uniformly illuminated by means of a lamp and lens system. The image of the triangular aperture is reflected by the galvanometer mirror focused on the mechanical slit. The mechanical slit in turn is focused on the film. The galvanometer mirror swings about an axis parallel to the plane of the paper. The triangular light image on the mechanical slit moves up and down on the mechanical slit. The result is that the width of the exposed portion of the negative sound track corresponds to the rotational vibrations of the galvanometer. In the positive record the width of the unexposed portion corresponds to the signal.

The amount of ground noise produced is proportional to the exposed

portion of the positive sound track. For this reason it is desirable to make the unexposed portion of the record just wide enough to accommodate the modulation. This is accomplished by applying a bias signal to the galvanometer. In the absence of a signal a very narrow exposed portion is produced on the negative record, which means a correspondingly narrow

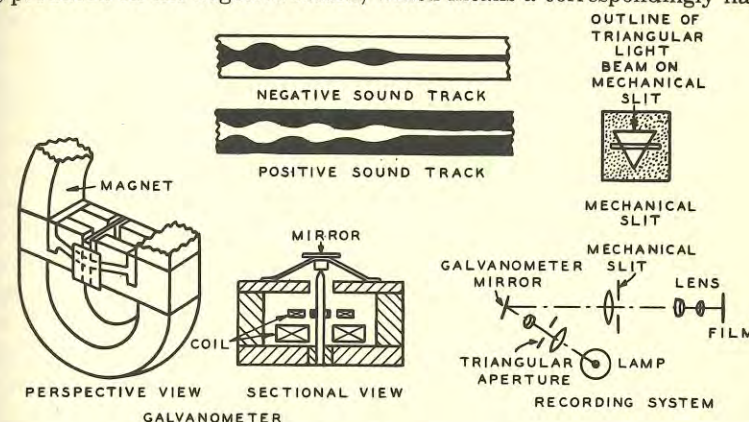


FIG. 9.57. The elements of a variable-area sound-motion-picture film recording system. The negative and positive sound tracks. Perspective and sectional views of the galvanometer.

unexposed portion on the positive record. When a signal appears, the triangular spot on the mechanical slit moves down just enough to accommodate the signal. The initial bias is accomplished within a millisecond. However, the return to normal bias after a large signal followed by a small signal is about 1 second. Faster return action produces thumping in the reproduced record.

A film sound-reproducing system is an amplitude system, that is, the voltage output is proportional to the amplitude on the film. Therefore, in order to obtain a uniform response-frequency characteristic, neglecting the frequency discrimination due to finite recording and reproducing slits, the amplitude of the galvanometer should be independent of the frequency.

In the variable-density system the transmitted light amplitude is an inverse function of the amount of exposure in the positive print. This type of sound track is produced by means of a light valve which varies the amount of light which falls on the moving film. The elements of a variable-density recording system are shown in Fig. 9.58. The ribbons of the valve are illuminated by means of a lamp and lens system. The image of the illuminated slit produced by the ribbons of the light valve is focused on the film. The amount of exposure on the negative film varies with the

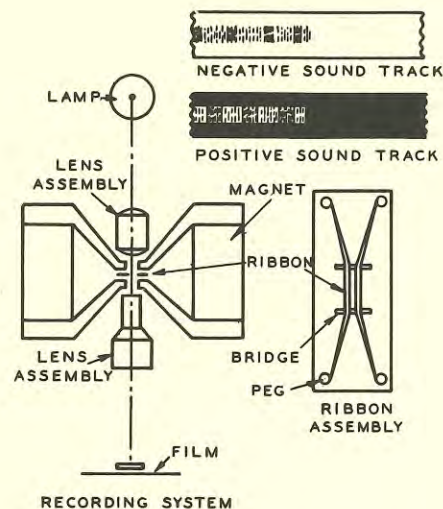


Fig. 9.58. The elements of a variable-density sound-motion-picture film recording system. The negative and positive sound tracks. Sectional and ribbon assembly views.

aperture at the ribbons. In the positive record the amount of exposure is an inverse function of the input to the light valve. Ground-noise reduction can also be obtained with a light valve. In the absence of a signal the light valve is biased so that the aperture between the ribbons is almost closed. When a signal appears, the ribbons open just enough to accommodate the signal. The noise-reducing action is similar to that in the variable-area system.

The elements of a moving-picture-film monophonic sound-reproducing system are shown in Fig. 9.59. The reproducing system save for the loudspeakers is located in the projection booth (Sec. 8.2E and Fig. 8.11). The light source, in the form of an incandescent lamp, is focused upon a

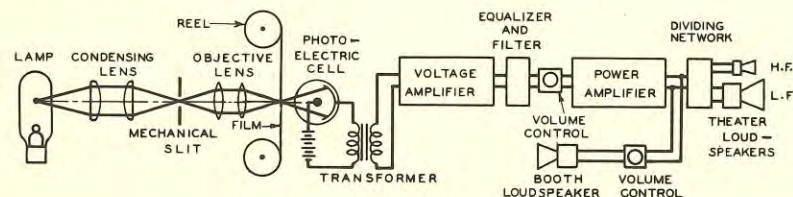


Fig. 9.59. The elements of a monophonic sound-motion-picture film reproducing system.

mechanical slit by means of a condensing lens. The mechanical slit in turn is focused on the negative film. The height of the image on the film is usually about 0.00075 inch. Under these conditions the amount of light which impinges upon the photocell is proportional to the unexposed portion of the sound track in variable-area recording or to the inverse function of the density in variable-density recording. When the film is in motion, the photographic variations in the film produce corresponding undulations in the light which pass through the film and fall upon the photocell. Electrons are emitted when light falls upon the cathode of a photocell. The number of electrons which are emitted is proportional to the intensity of the light. The electrons constitute an electrical current. Thus it will be seen that the undulations in light which fall upon the photocell produce corresponding variations in the current output of the photocell. The voltage output developed by the combination of the voltage polarizing supply and transformer corresponds to the current developed by the photocell and polarizing supply. Therefore, the voltage output at the transformer corresponds to the undulations in density in the film, which in turn correspond to the variations in the original sound which impinged upon the microphone. The lamp, optical system, mechanical slit, photocell, and transformer are located in the motion picture projector. The output of the photocell is fed to a voltage amplifier. The voltage amplifier is followed by an equalizer which compensates for some of the recording characteristics. In some cases a low-pass filter is used to reduce film ground noise. A volume control follows the voltage amplifier, equalizers, and filters. The volume control is followed by the power amplifier which feeds the loudspeakers. The voltage amplifier, equalizers, filters, volume control, and power amplifier are mounted in a rack located in the projection booth. The theater loudspeakers are located behind the screen (Sec. 8.2E and Fig. 8.11). Theater loudspeakers are of the dual type, one set for covering the low-frequency range and another set for covering the high-frequency range (see Sec. 9.3B and Fig. 9.17). A dividing network allocates the power with respect to frequency in the theater loud-speaker system. A monitoring loudspeaker is located in the projection booth.

Some recording of sound motion pictures is still carried out by the direct recording process depicted in Fig. 9.55. However, the general practice is to record the sound on magnetic tape. In this case, the magnetic tape recorder is substituted for the film recorder of Fig. 9.55. The magnetic tape used for recording the sound is in general 35 millimeters in width with sprocket holes on each side as shown in Fig. 9.60. The magnetic tape with sprocket holes makes it possible to synchronize the picture and sound in recording. The magnetic tape master is converted to the film

by feeding the output of the tape reproducer to the sound film recorder of Fig. 9.55.

Stereophonic sound was introduced in sound motion pictures several years ago. The original master sound record is recorded on the sprocket hole 35-millimeter magnetic tape shown in Fig. 9.60A. Any number of magnetic tracks up to seven have been used. However, in general, four magnetic tracks are employed. Three channels of stereophonic sound are recorded. The fourth channel may be used for control, sound effects, etc.

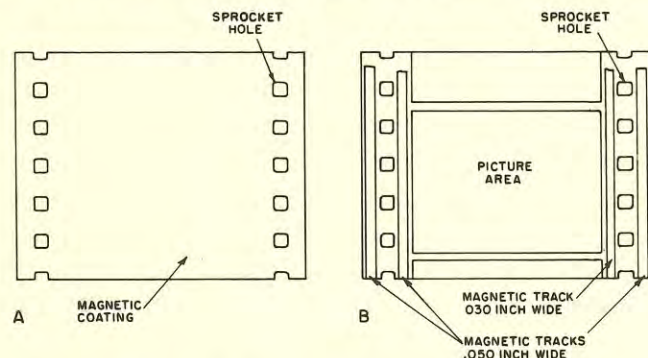


FIG. 9.60. A. Magnetic tape for original sound recording in sound motion pictures. B. Magnetic strips on a motion picture positive film.

The 35-millimeter release print with magnetic tracks for use in projection in the theater is shown in Fig. 9.60B. The three tracks carry the stereophonic sound for reproduction in audio perspective by means of the system shown in Fig. 8.11. The fourth track is used for the reproduction of sound effects, or for control, etc.

9.13 TELEVISION SOUND-REPRODUCING SYSTEM

The elements of a sound channel for a television broadcasting system are essentially the same as those of a monophonic frequency-modulation radio sound-broadcasting system described in Sec. 9.11. The picture and sound electrical variations are transmitted and reproduced simultaneously. In the case of plays, dramas, operas, the sound pickup techniques are a combination of those employed in radio and sound motion pictures (see Sec. 8.3I). In the audience type show, the technique is described in Sec. 8.3H. In some cases, the television show is recorded on sound motion picture film or video magnetic tape for broadcast at a later time. The sound portion of the receiving system in a television receiver is the same as that of a monophonic frequency-modulation radio receiver described in

Sec. 9.11. The reproduction of sound by the television receiver in a room is similar to that of a radio receiver or phonograph (see Sec. 8.2F and Fig. 8.13).

9.14 SOUND-REPRODUCING SYSTEM

A typical sound-reproducing system is shown in Fig. 9.61. The system consists of the following sources: one or more microphones with preamplifiers, a phonograph reproducer, a magnetic tape reproducer, and a radio receiver. Mixers are provided so that any of the sources may be coupled to the reproducing system. The mixers are followed by an amplifier. A volume control in the form of an attenuator is used to control the intensity of the reproduced sound. A volume indicator is used so that the appropriate sound level will be obtained at the reproducing points. The system or parts of the system shown in Fig. 9.61 are used for sound reinforcement, public-address, announce, and paging systems, music reproduction, and

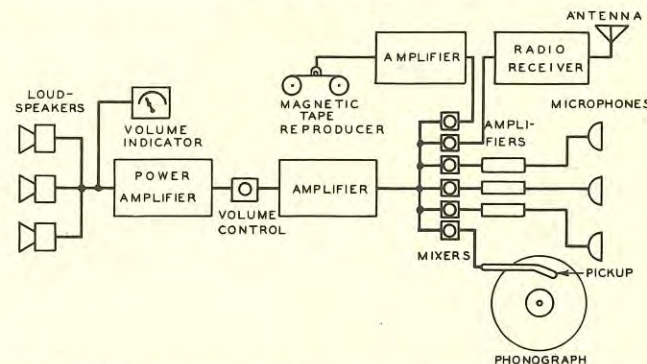


FIG. 9.61. The arrangement of the elements of a sound-reproducing system.

numerous other applications in theaters, churches, auditoriums, outdoor theaters, mass meetings, athletic events, factories, offices, railroad stations, airports, hotels, hospitals, etc. (Chap. 8).

9.15 HEARING AIDS

Tests made upon representative cross sections of the people in this country show a very large percentage to be hard of hearing. Practically all these people may obtain satisfaction from the use of a hearing aid. A hearing aid is a complete sound-reproducing system which increases the sound pressure over that normally received by the ear.

During the past few years, hearing aids employing transistor amplifiers have been developed with a high order of quality combined with small size.

The advent of transistors as described in Sec. 9.5 and miniature batteries have made this possible. A schematic diagram of the elements of a transistor hearing aid is shown in Fig. 9.62. The microphone employed in the transistor hearing aid is the magnetic type shown in Fig. 9.4.

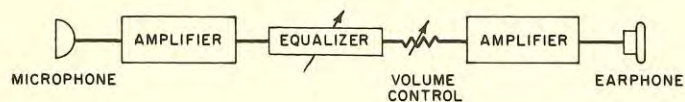


Fig. 9.62. Elements of a transistor hearing aid.

Two types of earphones are used, namely, the air-conduction type and the bone-conduction type. A perspective and sectional view of an air-conduction insert-type hearing-aid earphone is shown in Fig. 9.63A. A molded plug fits the ear cavity and holds the receiver in place. Under these conditions, the leak at the ear is very small. Therefore, good response is obtained at the low frequencies. The action of the system is essentially the same as the bipolar telephone receiver described in Sec. 9.4.

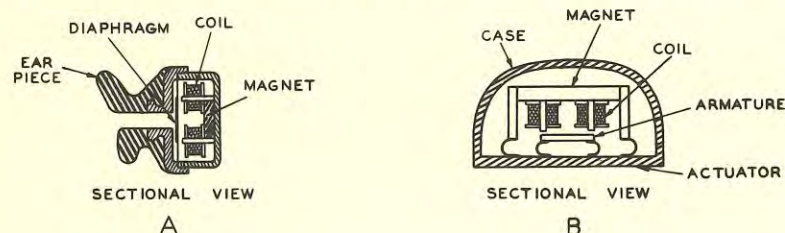


Fig. 9.63. Sectional views of hearing aid earphones. A. Insert-type earphone. B. Bone-conduction earphone.

In certain types of deafness, the middle ear, which consists of a series of bones that conduct sound to the inner ear, is damaged while the inner ear, which consists of the cochlea, frequency discriminating means, and nerves, is normal. Under these conditions, sound may be transmitted through the bones of the head to the inner ear by means of the bone-conduction earphone shown in Fig. 9.63B. The term bone-conduction earphone is not universally accepted to designate the electroacoustic transducer shown in Fig. 9.63B. Other designations are bone-conduction receiver, bone-conduction oscillator and bone-conduction vibrator. The face of the bone-conduction earphone is placed against the mastoid back of the external ear. The multiple-resonant system of the bone-conduction earphone delivers a large signal to the mastoid bone.

Complete hearing aids are shown in Fig. 9.64. The most common hearing aid consists of a case housing the microphone, amplifier, battery and controls and insert earphone, shown in Fig. 9.64A. The case is worn at a convenient location on the clothing. A cord connects the output of the amplifier to the earphone. A bone-conduction earphone may also be used with the hearing aid shown in Fig. 9.64A. For this condition, the bone-conduction receiver may be held against the skin and flesh covering the mastoid bone with a headband or by means of double-sided pressure adhesive tape with one side cemented to the earphone and the other side cemented to the skin covering the mastoid bone. An insert-type earphone

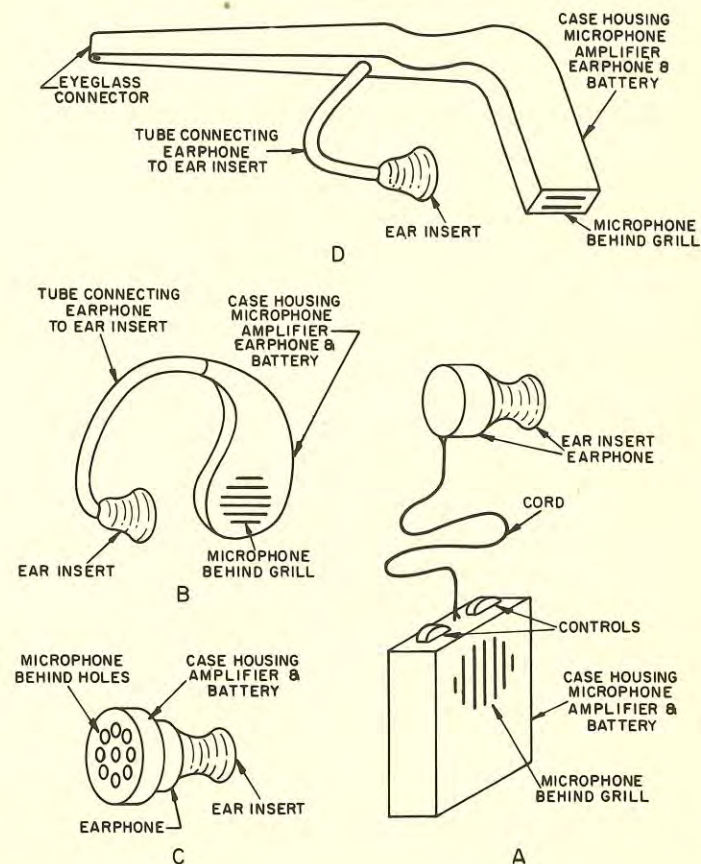


Fig. 9.64. Perspective views of typical hearing aids. A. Body-type. B. Behind the ear-type. C. In the ear-type. D. Eyeglass-type.

is shown in Fig. 9.64B. The case is worn behind the ear. The tube connecting the earphone in the case to the ear insert is carried over the top of the ear and serves as a support. The microphone, amplifier, controls and batteries are housed in the case behind the ear. A very compact hearing aid is shown in Fig. 9.64C. The entire system consisting of the microphone, amplifier, controls, battery and earphone is housed in a small case which fits in the ear. In another form of hearing aid, the entire system consisting of the microphone, amplifier, controls, battery and earphone is housed in one side of the side pieces of the eye spectacles as shown in Fig. 9.64D. The air-type earphone is depicted in Fig. 9.64D. The bone-conduction earphone may also be used in which case the earphone forms the curved-end portion of the side piece and is worn in intimate contact with the skin and flesh covering the mastoid bone.

9.16 FREQUENCY AND VOLUME RANGES OF SPEECH AND MUSIC

The frequency range of the average normal ear is 20 to 20,000 cycles. The frequency range of most sound-producing channels such as the radio, telephone, phonograph, sound motion picture, and television is considerably less than that of the hearing range of the human ear. Therefore, it is interesting to observe the effect of the frequency range upon the intelligibility of speech and the quality of music.

The frequency and volume ranges⁸ of speech and orchestral music for no

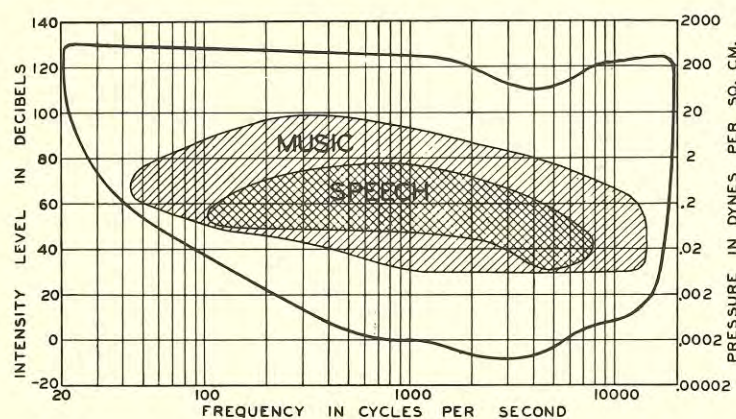


FIG. 9.65. Frequency and volume ranges of speech and music. The solid line depicts the boundaries of normal hearing, that is, the upper and lower limits of intensity and frequency. (From *Bell Laboratories Record*, June, 1934.)

⁸ *Bell Labs. Record*, Vol. 12, No. 10, p. 314, 1934.

perceptible change in quality are shown in Fig. 9.65. The reproduction of speech with subjectively perfect fidelity requires a frequency range of 100 to 8,000 cycles and a volume range of 40 decibels. The reproduction of orchestral music with subjectively perfect fidelity requires a frequency range of 40 to 15,000 cycles and a volume range of 70 decibels.

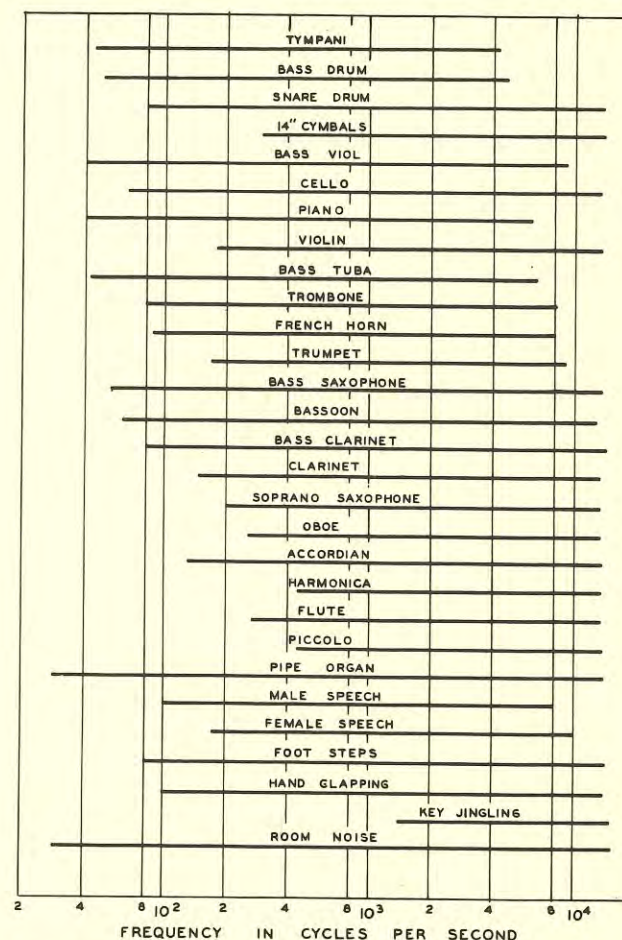


FIG. 9.66. The frequency ranges for the reproduction of speech, musical instruments, and noises without any noticeable distortion. (After Snow.)

The frequency ranges required for the reproduction of speech, musical instruments, and noises without any noticeable frequency discrimination

or distortion⁹ are shown in Fig. 9.66. It will be seen that, owing to the extensive high-frequency range of the overtones of musical instruments, the upper limit of sound reproduction must extend to 15,000 cycles in order to obviate any perceptible change in quality. In order to reproduce the fundamentals of musical instruments, the lower limit of reproduction should extend to at least 40 cycles.

9.17 EFFECT OF FREQUENCY DISCRIMINATION UPON THE ARTICULATION OF REPRODUCED SPEECH¹⁰

The most important aspect of the perception of speech is the process involving the ability of a person to recognize correctly the speech sounds which are spoken. The method for measuring speech perception is to have a speaker read aloud certain speech sounds to a listener who writes what he thinks he hears. A comparison of sounds, syllables or words recorded by the listener with those spoken by the speaker shows the fraction that is interpreted correctly. This fraction is termed articulation.

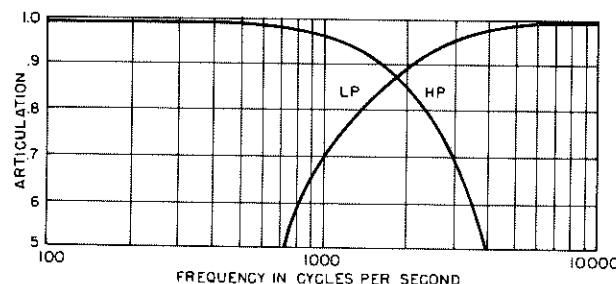


FIG. 9.67. The effect of the frequency range upon the sound articulation of speech. *HP.* High-pass filter: all frequencies below the frequency given by the abscissa are removed. *LP.* Low-pass filter: all frequencies above the frequency given by the abscissa are removed. (After Fletcher, *Speech and Hearing in Communication*, D. Van Nostrand Company, Inc., Princeton, 1953.)

Sound articulation refers to the use of speech sounds such as "p," "a," "t," etc. Syllable articulation refers to the use of syllables such as pat, run, etc. Word articulation refers to the use of a complete word.

The medium between the speaker and the listener may be the direct path through the air or through a sound-reproducing system.

The effect of reducing the high- and low-frequency range upon the sound articulation of speech at a normal conversational level is shown in Fig.

⁹ Snow, W. B., *Jour. Acoust. Soc. Amer.*, Vol. 3, No. 1, Part 1, p. 155, 1931.

¹⁰ Fletcher, *Speech and Hearing in Communication*, D. Van Nostrand Company, Inc., Princeton, 1953.

9.67. It will be seen that a relatively high articulation can be obtained with a very narrow transmission band. However, the quality of the reproduced speech is very much impaired by transmission over a narrow frequency band. From the standpoint of articulation, a limited frequency range may be actually superior to a wider frequency band, owing to the introduction of additional noises and distortions in a wider band, unless particular precautions are observed. In the case of speeches, plays, and songs, a limited frequency range impairs the quality and artistic value of the reproduced sound.

9.18 EFFECT OF FREQUENCY DISCRIMINATION UPON THE QUALITY OF REPRODUCED MUSIC

The effect of the frequency range¹¹ upon the quality of reproduction of orchestral music is shown in Fig. 9.68. It will be seen that the frequency

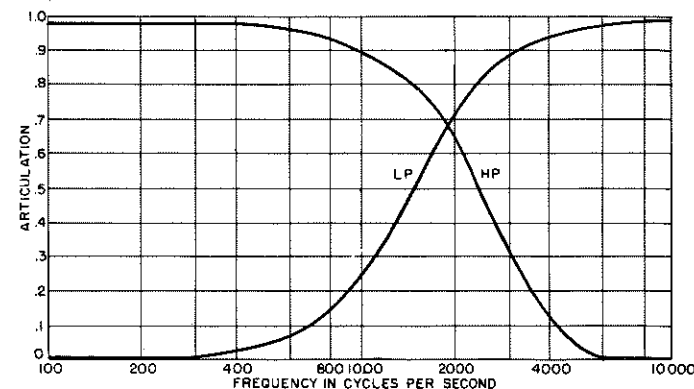


FIG. 9.68. The effect of the frequency range upon the quality of orchestral music. *HP.* High-pass filter: all frequencies below the frequency given by the abscissa are removed. *LP.* Low-pass filter: all frequencies above the frequency given by the abscissa are removed. (After Snow.)

range required for no appreciable loss in quality is 40 to 15,000. As the next section will show, the frequency ranges of high-quality magnetic tape recorders and reproducers, disc recorders and reproducers, radio and television transmitters and receivers, sound-motion-picture recorders and reproducers and sound systems cover the frequency range depicted in Fig. 9.68. In some cases, the volume range of the music exceeds the capabilities of the reproducing equipment. Under these conditions, some form of volume compression must be used to cover the amplitude range (see Sec. 8.3L).

¹¹ Snow, W. B., *Jour. Acoust. Soc. Amer.*, Vol. 3, No. 1, Part 1, p. 155, 1931.

9.19 FREQUENCY RANGES OF SOUND-REPRODUCING SYSTEMS

The frequency ranges of the most common sound-reproducing systems are shown in Fig. 9.69. The frequency ranges shown are averages of existing systems. In specific cases the frequency ranges may be greater or less than those shown in Fig. 9.69.

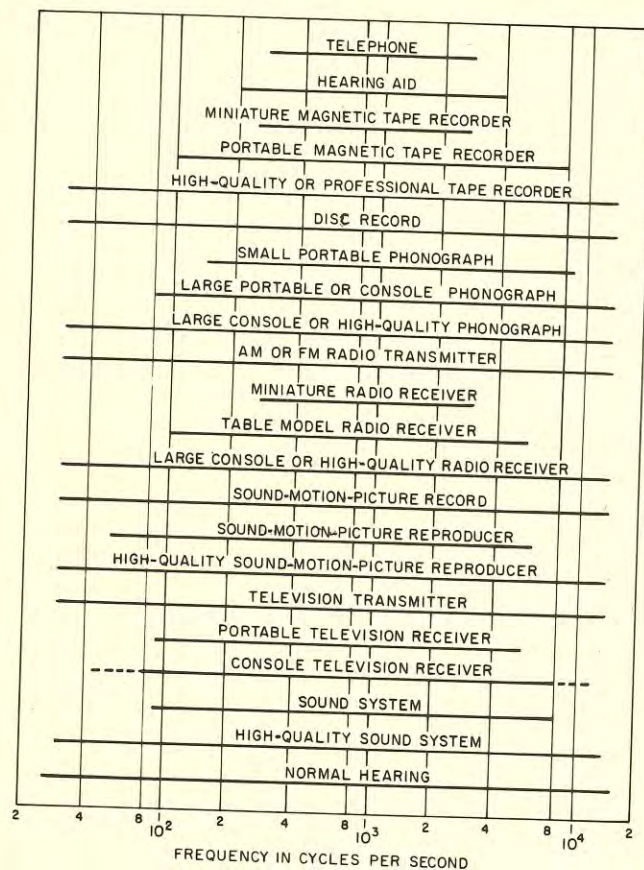


Fig. 9.69. The frequency ranges of sound-reproducing systems.

The frequency ranges of telephones vary over wide limits, depending upon the type of instrument, the central offices, and the interconnecting lines. The frequency range depicted is for instruments made in the last decade. Extending the frequency range would probably result in reduced

articulation due to ambient room noise and noises produced by electrical interferences.

The frequency range of the hearing aid shown in Fig. 9.69 represents the average response of high-quality transistor hearing aids in use today. The low-frequency range may be somewhat greater but in general, this added range cannot be used because of "rumble" and other low-frequency noises.

The frequency ranges of magnetic tape recorders and reproducers are shown in Fig. 9.69. The low tape speeds used in miniature and portable tape recorders limits the high-frequency range. The small size of the loudspeaker and cabinet limits the low-frequency response of the miniature and portable magnetic tape recorders. The frequency range of high-quality and professional tape recorders covers the entire audio-frequency range.

The commercial disc record covers the entire audio-frequency range as shown in Fig. 9.69.

The frequency ranges of phonographs are shown in Fig. 9.69. The size of the loudspeaker and cabinet determines the low-frequency response of the portable and small console phonographs. The high-frequency range is limited to provide proper balance. The high-quality phonograph covers the entire audio-frequency range.

AM and FM radio transmitters cover the entire audio-frequency range. The frequency ranges of radio receivers are shown in Fig. 9.69. As in all sound-reproducing systems the low-frequency range is determined by the size of the loudspeaker and cabinet. The frequency ranges increase with the size of the radio receiver as shown in Fig. 9.69. The high-quality AM receiver covers the entire frequency range if there is no cochannel interference; the high-quality FM receiver covers the entire audio-frequency range.

The frequency range of a commercial sound-film record covers the entire audio-frequency range as shown in Fig. 9.69. The frequency range of a sound-motion-picture system shown in Fig. 9.69 refers to the average of systems used in theaters. A small number of high-quality sound-motion-picture systems are in use with a frequency range varying from the commercial system up to the high-quality system shown in Fig. 9.69. The frequency ranges of 16-millimeter and 8-millimeter film record and reproducers are restricted as compared to the commercial 35-millimeter film record and reproducers.

The frequency range of a television sound transmitter is shown in Fig. 9.69. In general, the low-frequency range is limited in transmission due to the high-ambient low-frequency noise level in the studio. The high-frequency range is also limited by electrical noise and the frequency range of the audio-transmission networks.

The low-frequency range of the table model television receiver is limited

by the size of the loudspeaker and cabinet. The frequency range of the large console television receiver is limited by the signal input.

The frequency range of the common low-cost sound system is shown in Fig. 9.69. For theaters, auditoriums, and other high-quality installations the frequency range of the sound system covers the entire audio-frequency range. In some cases, the frequency range must be tailored to compensate for the poor acoustics of the auditorium or theater.

The hearing frequency range of a person with excellent hearing is shown in Fig. 9.69.

An examination of the characteristics noted in Fig. 8.69 shows that the consumer can obtain radio receivers, magnetic tape records and reproducers and disc records and phonographs that cover the entire audio-frequency range without discrimination.

9.20 EFFECT OF NONLINEAR DISTORTION UPON THE QUALITY OF REPRODUCED SPEECH AND MUSIC¹²

The various elements in an ideal sound-reproducing system are absolutely invariant with respect to time. However, in actual sound-reproducing systems in use today, all the elements exhibit nonlinear characteristics.

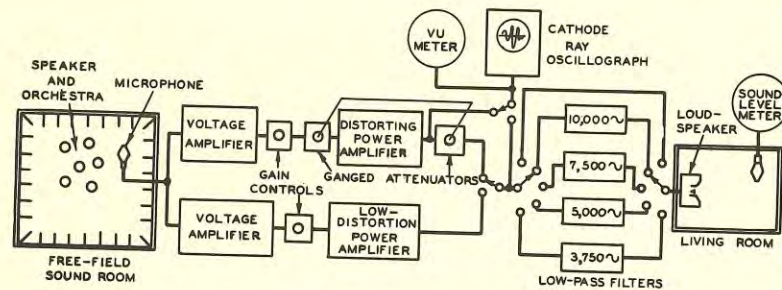


FIG. 9.70. Schematic arrangement of the apparatus for the subjective determination of the relation between nonlinear distortion and the frequency range of the reproduced sound. (After Olson, *Acoustical Engineering*, D. Van Nostrand Company, Inc., Princeton, 1957.)

A nonlinear element introduces nonlinear distortion. The effect of nonlinear distortion is the introduction of harmonic components in the reproduced sound which were not present in the original sound. In order to obtain a better understanding of some of the effects of nonlinear distortion upon the reproduction of sound, a subjective test has been performed to determine the perceptible, tolerable, and objectionable nonlinear distortion

¹² Olson, H. F., *Acoustical Engineering*, D. Van Nostrand Company, Inc., Princeton, 1957.

in a sound-reproducing system as a function of the frequency for different types of distortion.

The effect of various types of nonlinear distortion upon the reproduction of speech and music has been determined by means of the sound-reproducing system shown in Fig. 9.70. The basic system is a high-quality sound-reproducing system. The over-all response frequency of the com-

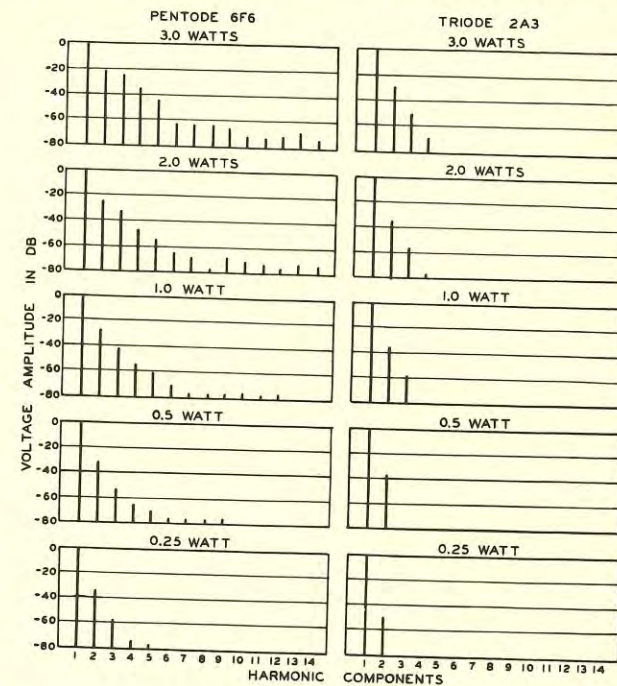


FIG. 9.71. Distortion characteristics of single-ended pentode and triode amplifiers. (After Olson, *Acoustical Engineering*, D. Van Nostrand Company, Inc., Princeton, 1957.)

bination of the microphone, amplifier, and loudspeaker is flat to within ± 2 decibels from 40 to 15,000 cycles. The over-all nonlinear distortion in the system is less than 0.25 per cent. Means are provided so that nonlinear distortion of a predetermined amount could be introduced. Two types of distorting amplifiers were used to introduce distortion of a predetermined amount, namely, a single-ended triode amplifier and a single-ended pentode amplifier. The distortion power outputs for various power levels are shown in Fig. 9.71. Referring to Fig. 9.70, it will be seen that by means of the gain control preceding and the attenuator following the distorting

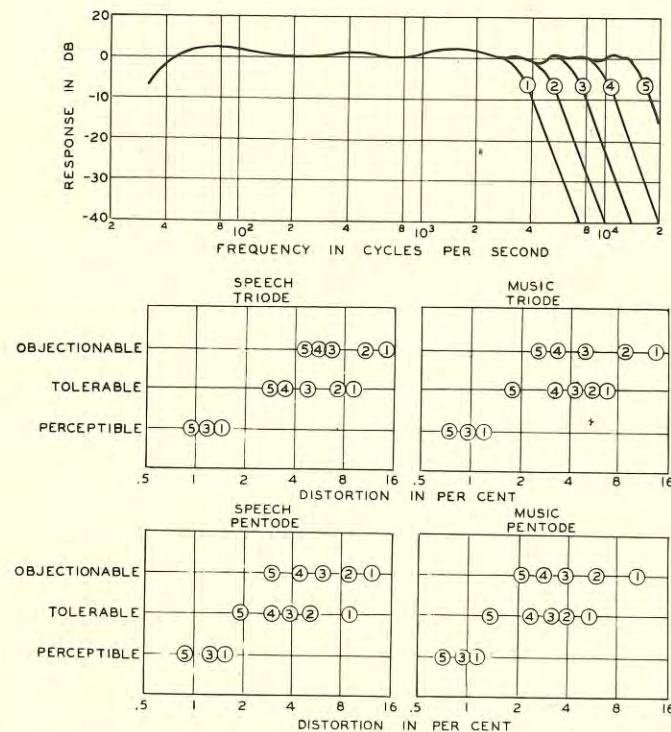


FIG. 9.72. Experimental results of subjective tests of reproduced speech and music depicting objectionable, tolerable, and perceptible nonlinear distortion for various high-frequency cutoffs. The numbers in the distortion data points correspond to the numbers which label the response frequency characteristics. The distortion, in per cent, is the ratio, multiplied by 100, of the total root mean square of the second, third, fourth, etc., components of the distortion to the root mean square of the fundamental. (After Olson, *Acoustical Engineering*, D. Van Nostrand Company, Inc., Princeton, 1957.)

amplifier it is possible to operate the distorting amplifier at practically any distorting level and still retain the same level in the reproduced sound. The pickup studio for these tests was a free field or anechoic sound room, that is, a room devoid of echoes and reverberations as well as having a low noise level. The sound was reproduced in a room with acoustics similar to that of a typical living room (Sec. 8.2F). The noise level at the pickup point was 0 decibels in the absence of any performers. The noise level in the listening room was about 20 decibels. The reproducing level was about 70 decibels.

These tests were limited to three subjective gradations of nonlinear distortion, namely, perceptible, tolerable, and objectionable. Perceptible is the amount of distortion in the distorting system required to be just discernible when compared with the reference system. Tolerable and objectionable are not as definite and are a matter of opinion. By tolerable distortion is meant the amount of distortion which could be allowed in low-grade commercial sound reproduction. By objectionable distortion is meant the amount of distortion which would be definitely unsatisfactory for the reproduction of sound in phonograph and radio systems.

Both speech and music were used in making these tests. In the case of music, a six-piece orchestra was employed.

The average results of a few of these tests, with a limited number of critical observers, are shown in Fig. 9.72. As would be expected from the frequency ranges of speech and music together with the masking curves, a distorting system with high-order components is more objectionable than one with low-order components. The amount of tolerable distortion is greater for speech than for music. Referring to Fig. 9.72, it will be seen that the amount of perceptible, tolerable, or objectionable distortion decreases as the high-frequency cutoff increases. This means that as the high-frequency range is decreased the amount of distortion which can be tolerated is increased. For example, these tests show that the tolerable nonlinear distortion in a sound-reproducing system with a 5,000-cycle high-frequency cutoff is of the order of 6 to 10 per cent, whereas the tolerable nonlinear distortion in a sound-reproducing system with a 15,000-cycle high-frequency cutoff is of the order of 1 to 2 per cent. These tests show that unless the nonlinear distortion is a wide-frequency-range system is kept to this relatively low value it will not be acceptable. While it is a comparatively simple matter to design and relatively inexpensive to build a sound-reproducing system with 6 to 10 per cent nonlinear distortion and a limited frequency range, it requires painstaking designs coupled with expensive components to achieve the relatively low order of nonlinear distortion of 1 to 2 per cent in a system of wide frequency range.

9.21 FREQUENCY-RANGE PREFERENCE FOR REPRODUCED SPEECH AND MUSIC¹³

The effect of frequency discrimination upon the quality of reproduced sound has been considered in Sec. 9.18. The effect of frequency discrimination upon the articulation of reproduced speech has been considered in Sec. 9.17. The effect of nonlinear distortion upon the quality of reproduced speech and music has been considered in Sec. 9.20. The frequency and volume ranges of speech and music have been considered in Sec. 9.16.

¹³ Chinn and Eisenberg, *Proc. Inst. Radio Engrs.*, Vol. 33, No. 9, p. 571, 1945.

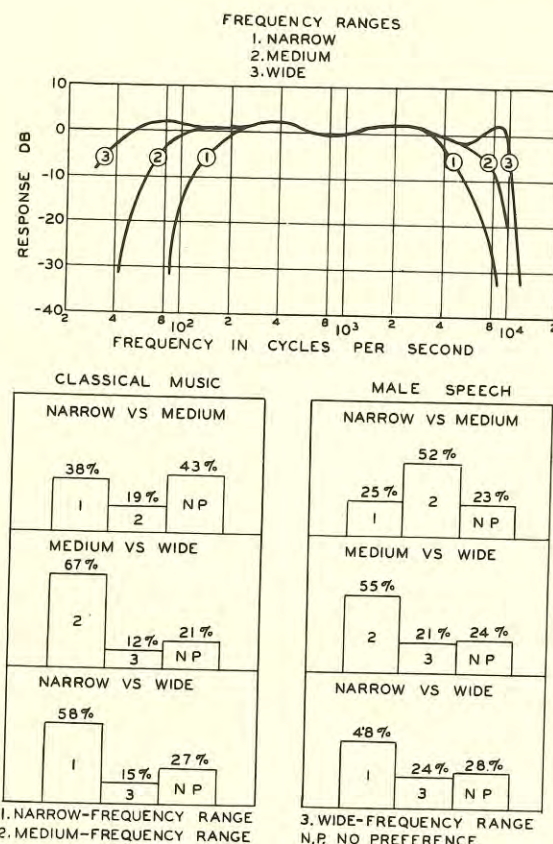


Fig. 9.73. The frequency-range preferences of a cross section of listeners for classical music and speech. The graph depicts the response-frequency characteristics of the following: 1 = narrow-frequency range, 2 = medium-frequency range, 3 = wide-frequency range. The block diagram depicts the preference. NP means no preference. When the narrow range was compared with the medium range, there was a preference for narrow for music and medium for speech. However, in the comparison between medium and wide, the preference was markedly for the narrow bands for both classical music and male speech. (After Chinn and Eisenberg.)

The frequency ranges of sound-reproducing systems have been considered in Sec. 9.19. A consideration of all the above data indicates that the subject of sound reproduction is exceedingly complex. In order to add to the general understanding, a study has been made of the frequency-range preference of a representative cross section of broadcast listeners.

As contrasted with the other data presented, the purpose of this investigation was the determination of the frequency range of reproduced speech and music that is most pleasant to the average listener. The investigation was made with a variety of musical and voice passages. The tests were made in a room with acoustics similar to those of a large living room. Both high-quality records and direct-wire transmission from the studio were used with very little difference in the results.

The frequency ranges employed for the tests are shown in Fig. 9.73 and were designated as wide, medium, and narrow frequency ranges.

The results of the tests are shown in Fig. 9.73. The general conclusion of these tests is that listeners prefer either a narrow or medium frequency range to a wide one. However, the exact choice of band width varies to some extent within these limits, for different types of program content. Listeners prefer a narrow to a wide tonal range even when informed that one condition is low fidelity and the other high fidelity. Listeners prefer a slightly wider band for female speech, piano, and popular orchestra selections than for male speech, mixed dramatic speech, and classical orchestra selections.

The reasons for the variance of these subjective tests with other subjective tests will be delineated in the sections which follow.

9.22 FREQUENCY-RANGE PREFERENCE FOR SPEECH AND MUSIC¹⁴

The frequency-range preference for reproduced speech and music has been considered in the preceding section. These tests indicate that listeners prefer a restricted frequency range in monaural reproduced speech and music. There are three possible reasons for the results of these tests, as follows: (1) The average listener, after years of listening to the radio and the phonograph, has become conditioned to a restricted frequency range and feels that this is the natural state of affairs. (2) Musical instruments are not properly designed and would be more pleasing and acceptable if the production of fundamentals and overtones in the high-frequency range were suppressed. (3) The distortions and deviations from true reproduction of the original sound are less objectionable with a restricted frequency range. The distortions and deviations from true reproduction of the original sound are as follows:

1. Frequency discrimination.
2. Nonlinear distortion.
3. Spatial distribution.
 - a. Relatively small source.
 - b. Separated sources in two-way loudspeaker systems.

¹⁴ Olson, H. F., *Jour. Acoust. Soc. Amer.*, Vol. 19, No. 4, p. 549, 1947.

- c. Nonuniform directional pattern with respect to frequency.
4. Single-channel system.
5. Phase distortion.
6. Transient distortion.
7. Microphone placement and balance.
8. Acoustics of two rooms, the pickup studio and the listening room.
9. Limited dynamic range.
10. Difference in level of the original and reproduced sound.
11. Noise.

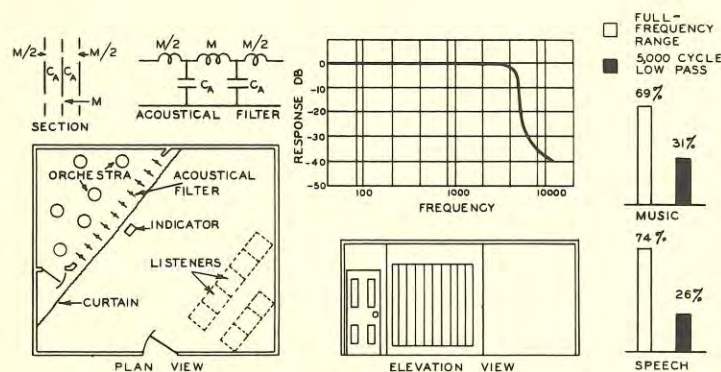


FIG. 9.74. Plan and elevation views of the schematic arrangement of the apparatus for direct testing of frequency-range preference for speech and music. A sectional view, acoustical network, and response-frequency characteristic of the acoustical filter used in the test. The results for speech and music are depicted on the right. (After Olson, *Acoustical Engineering*, D. Van Nostrand Company, Inc., Princeton, 1957.)

In order to obtain a better understanding of the reason for the preference of a restricted frequency range in reproduced sound, a fundamental all-acoustic test of frequency-range preference was made. The general arrangement of the test is shown in Fig. 9.74. An acoustical filter is placed between the orchestra and the listeners and is arranged so that it can be turned in or out. It is composed of three sheets of perforated metal to form a two-section acoustical filter, as shown in Fig. 9.74. The response-frequency characteristic of the acoustical filter shown in Fig. 9.74 approximated commercial good radio or phonograph reproduction in the high-frequency range at the time of the test in 1947. The acoustical filter is composed of 10 units with each unit pivoted at the top and bottom. The 10 units are coupled together and rotated by means of a lever. In this way the acoustical filters can be put in or out by merely turning the

units through 90 degrees. The acoustical filters are shown in the full frequency-range position in Fig. 9.74. A sheer cloth curtain which transmits sound with no appreciable attenuation over the frequency range up to 10,000 cycles and less than 2-decibel attenuation from 10,000 to 15,000 cycles is placed between the acoustical filter and the listeners. The curtain is illuminated so that the listeners cannot see what transpires behind the curtain. The particular condition, that is, the full frequency range or 5,000 cycles low-pass transmission, is shown on an AB indicator.

The tests made up to the present time have been conducted in a small room which simulates an average living room in dimensions and acoustics. The orchestra was a six-piece dance band playing popular music. The average sound level in a room was about 70 decibels. The changes from wide open to low pass to wide open, etc., were made every 30 seconds. Two selections were played, and the listeners were asked to indicate a preference. The results of these tests, as shown in Fig. 9.74, indicated a preference for full-frequency range. Similar tests have been made for speech. The preference in the case of speech is also for the full-frequency range. There is a distinct lack of presence in speech with the limited-frequency range.

The results of the all-acoustic frequency-range preference are at variance with similar tests employing reproduced sound, as described in Sec. 9.21. The reason for the difference between the results of the two tests is, without doubt, due to the distortions listed in the first paragraph of this section. The subjective tests of nonlinear distortion, described in Sec. 9.20, indicated that the amount of tolerable distortion decreases as the frequency range is increased. These tests also indicated that a very small amount of nonlinear distortion can be detected when employing the full-frequency range.

9.23 FREQUENCY-RANGE PREFERENCE FOR STEREOPHONICALLY REPRODUCED SPEECH AND MUSIC¹⁵

Subjective tests of frequency-range preference of live speech and music comparing a restricted frequency range with the full frequency range have shown that the average listener prefers the full frequency range (see Sec. 9.22). These tests were all acoustic. The sound was not reproduced. Therefore, there were no electroacoustic transducers in the form of microphones, amplifiers, modulators, transmitters, records, receivers, pickups, demodulations, loudspeakers, etc. used in these tests. The frequency discrimination was accomplished by means of acoustical filters. The question arises whether similar results can be obtained with reproduced sound. The only reason that the same results could not be obtained with reproduced

¹⁵ Olson, Preston, Woodward, May, Morgan, and Bleazey, Unpublished Report, 1955.

sound would be due to distortions and deviations from true reproduction of the original sound. The principal distortions and deviations from true reproduction of the original sound have been described in Sec. 9.22.

Following the all-acoustic frequency-range preference tests, it appeared logical to follow these tests with frequency-range preference tests employing

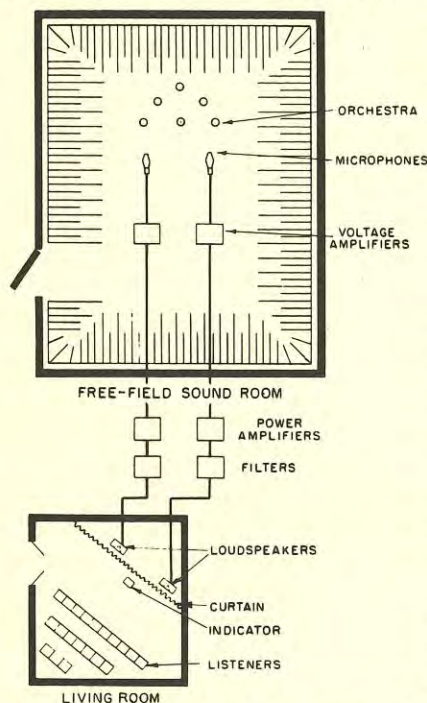


FIG. 9.75. Plan view of the schematic arrangement of the apparatus for frequency-range preference for stereophonically reproduced speech and music. (After Olson, *Acoustical Engineering*, D. Van Nostrand Company, Inc., Princeton, 1957.)

reproduced sound. In the tests involving reproduced sound it was felt desirable to simulate the conditions of the all-acoustic tests as closely as possible. To attain this objective the orchestra was reproduced in perspective. The acoustics of the studio were eliminated by using a free-field room for the studio housing the orchestra. These expedients eliminated some of the distortions above. The remaining distortions listed, referred to above, were reduced to the lowest possible level.

The floor plans of the free-field room used as a studio and listening room and the general arrangement of the tests are shown in Fig. 9.75. The idea of the test is to "transfer" the orchestra to the listening room by means of microphones, amplifiers, and loudspeakers. In order to simulate the all-acoustic tests in this transfer of the orchestra, it is obviously necessary that the studio be devoid of acoustics, that is, reverberation. In order to obtain

these conditions the anechoic or free-field sound room is used as the studio. With the orchestra operating in the free-field sound room the level of the reflected sound is far below the level of the direct sound. The level of the reflected sound at the microphones for steady-state sound conditions is about 50 decibels below the direct sound. Therefore, it is impossible to detect any acoustics of the studio in the reproduced sound.

The listening room is designed to be the acoustical equivalent of an average living room.

The reproducing system used in these tests employs two channels. Each channel consists of an RCA 44BX Velocity Microphone, an RCA OP-6 Amplifier, a laboratory developed triode-type push-pull power amplifier, laboratory developed high- and low-pass electrical filters, and an RCA LC1A Loudspeaker.

The over-all response-frequency characteristics depicting the ratio of the sound-pressure output from the loudspeaker in free space to the sound pressure at the microphone in free space, with and without the electrical filters, are shown in Fig. 9.76. In the restricted range condition there is

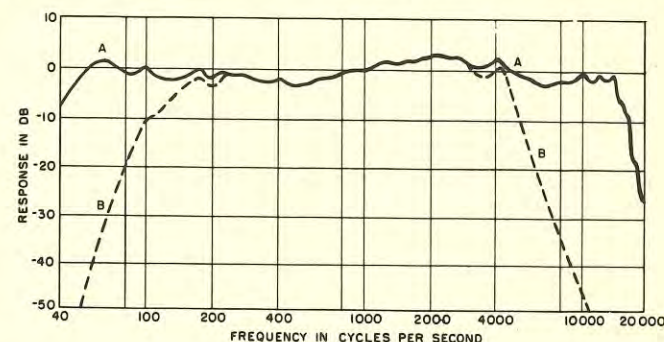


FIG. 9.76. Over-all response frequency characteristic of the two conditions used in the frequency-range preference for stereophonically reproduced speech and music. A. Wide frequency range. B. Restricted frequency range. (After Olson, *Acoustical Engineering*, D. Van Nostrand Company, Inc., Princeton, 1957.)

attenuation in both the low- and high-frequency ranges. The product of the low and high cutoff frequencies is 500,000 (cycles)². In this we have deviated from the original all-acoustic frequency-preference test in which a high-frequency cutoff alone was used. An argument in favor of the combination of high- and low-frequency cutoffs is that it approximates conventional radio and phonograph response-frequency characteristics.

The directivity pattern of the loudspeakers is very important where the listeners are located at relatively large angles with respect to the loudspeaker. In the loudspeakers used in these tests the variation in response at any frequency over a total angle of 90° is less than ± 2 decibels.

The nonlinear distortion is another important factor in reproduced sound. The over-all nonlinear distortion was measured by supplying a distortionless signal to the input of the chain consisting of the voltage

amplifier, power amplifier, and loudspeaker. The sound output of the loudspeaker was picked up by the microphone and fed to a harmonic analyzer. This method of measurement provides an over-all distortion characteristic from sound input to the microphone to sound output of the loudspeaker. The total nonlinear distortion measured at the peak level of the reproduced sound was less than 0.3 per cent. From the results reported in Sec. 9.20, it will be seen that this value of nonlinear distortion is sufficiently low to be practically imperceptible.

The level of the reproduced sound in the listening room is important in any subjective test. Tests have shown that a peak level of about 70 decibels to 80 decibels is most pleasing for serious listening in a small room. The average peak sound intensity level on a standard-level indicator was 75 decibels.

The same six-piece band was used in these tests as in the case of the all-acoustic frequency-range tests. The change from full-frequency range to restricted-frequency range was made every 30 seconds. The results of these tests indicate a preference for the full-frequency range. Similar tests were made for speech. The frequency-preference tests for speech also indicate a preference for the full-frequency range.

9.24 THE OBJECTIVE AND SUBJECTIVE ASPECTS OF SOUND REPRODUCTION¹⁶

The definition and connotation of the term sound reproduction imply that a facsimile of the original sound is achieved in the reproduction. Sound reproduction involves both objective and subjective considerations in the mechanism and phenomena required to achieve a high order of fidelity of performance. From an objective viewpoint, sound reproduction is a complex process because a large number of acoustic, mechanical and electronic elements and combinations thereof are involved. These elements must exhibit a high order of excellence of performance in order to provide a resemblance to a perfect transfer characteristic. From a subjective viewpoint, sound reproduction is a complex process because a large number of psychoacoustical effects are involved. These effects must be used in an appropriate manner in order to produce an ideal transfer characteristic which supplies a close artistic resemblance of the original live rendition.

The main purpose of sound reproduction is to provide the listener with the highest order of artistic and subjective resemblance to the condition of a live rendition. To achieve this objective requires the topmost degree of excellence of the physical performance of the equipment as directed by the psychological factors involved in the process.

¹⁶ Olson, H. F., *Audio Engineering Society Preprint No. 348*, October 12-16, 1964.

In general, the state of the art in sound reproduction has advanced to a stage where a high order of physical performance can be obtained. Up until very recent times, the application of the psychological characteristics as related to sound reproduction has lagged behind the physical considerations. The commercialization of stereophonic sound in the consumer complex has hastened the work in the subjective aspects of sound reproduction.

Recent work in the field of psychological acoustics as related to the performance of a sound-reproducing system involves the following specific subjects: tolerable sound-pressure levels; sound-pressure level and ear-frequency response; loudness and dynamics; dynamics, noise and masking; auditory perspective; reverberation; quality and timbre.

A. Tolerable Sound-pressure Levels

The subjectively tolerable loudness of the sound reproduced in a small room in a home is an important factor involved in the reproduction of sound. Extensive subjective tests have been conducted on the stereophonic reproduction of sound at various sound-pressure levels under the acoustical conditions and environments of the average living room in the home. Studies have also been made of the reproduction level employed by consumers in their homes. These tests have shown that peak sound-pressure levels¹⁷ of sound reproduction in the homes of consumers run from 70 to 90 decibels for 90 per cent of the listeners. The average listener in the home operates a sound-reproducing system at a peak sound-pressure level of 80 decibels.

The peak sound-pressure level of sound by a live performance in a large acoustic enclosure is about 100 decibels. Thus, it will be seen that the peak level of sound reproduction in the home is much lower than the level of a live performance.

The main reason why the average listener prefers a lower level of sound reproduction in the home as contrasted to the sound level in the concert hall, is that the tolerable peak sound-pressure level¹⁸ in a small room is lower than the tolerable peak sound-pressure level in a large hall. The shorter mean free path and resultant faster growth and decay of sound in a small room appears to lead to a lower tolerable peak level in the small room. Subjective tests have indicated that the same results are obtained regardless of whether the sound program is live or reproduced.

¹⁷ Peak sound-pressure level in these considerations is used to designate a level at which 95 per cent of the program lies below the peak level.

¹⁸ The term tolerable peak sound-pressure level is used to designate the peak sound-pressure level of sound reproduction which the listener feels is acceptable and agreeable.

B. Sound-pressure Level and Ear-frequency Response

The frequency-response characteristic of the ear and the sound-pressure level of the reproduced program in a small room are two factors involved in the relatively low level of sound reproduction in the home.

The equal loudness frequency relations of hearing or the response-frequency characteristics of the human hearing mechanism have been determined by several investigators. The characteristics shown in Fig. 9.77 are an average of the investigations carried out by Fletcher and Munson,¹⁹ Churcher and King,²⁰ and Robinson and Dadson.²¹ As far as the compensation is concerned in going from one level to another for a change of 20 decibels or less—which is the point of interest—the individual data from each of the three different investigations yield very nearly the same results. Essentially the same results are obtained from the response-frequency characteristics of the hearing mechanism from the draft recommendation of the I.S.O.²² Therefore, it seems logical to use the average data of the three investigations listed above.

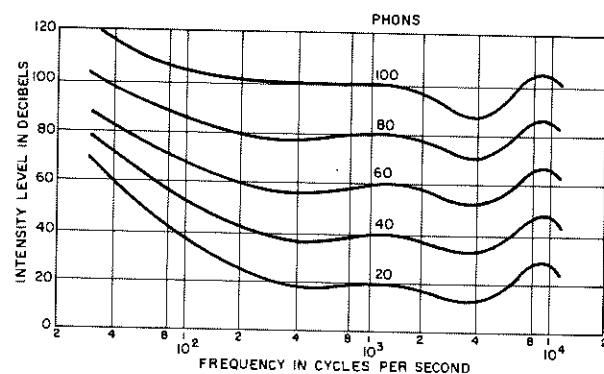


Fig. 9.77. Free-field contour lines of equal loudness for normal ears. Numbers on curves indicate loudness level. 0 dB = 10^{-6} watts per square centimeter. 0 dB = 0.0002 dyne per square centimeter.

The response-frequency characteristics of the human hearing mechanism as depicted in Fig. 9.77 indicate that certain of the frequency ranges must be increased or decreased in amplitude in order to maintain the quality balance of music when it is reproduced at a lower level than the original.

¹⁹ Fletcher and Munson, *Jour. Acoust. Soc. Amer.*, Vol. 5, No. 1, p. 82, 1933.

²⁰ Churcher and King, *Jour. Inst. Elec. Eng. (London)*, Vol. 81, No. 1, p. 57, 1937.

²¹ Robinson and Dadson, *Brit. Jour. App. Phys.*, Vol. 7, No. 5, p. 116, 1956.

²² International Organization for Standardization, Draft Recommendation, R 226, 1961.

Therefore, means must be provided to correct dynamically for the change in response of the ear with respect to frequency for a level drop of 20 decibels. There must be a continuous variation in the response-frequency characteristic as the level changes.

The frequency-response characteristics of the ear shown in Fig. 9.77 are essentially for free-field conditions. Therefore, the acoustic characteristics of the average room in a home and the loudspeaker operating in the room must be considered in establishing over-all frequency-response characteristics of the ear, the loudspeaker, and the room.

In the reproduction of sound in a room there are two sources of sound with respect to the listener, namely, the direct sound from the loudspeaker and the generally reflected sound. The acoustical characteristics of the average room in a residence accentuate the low-frequency response as can

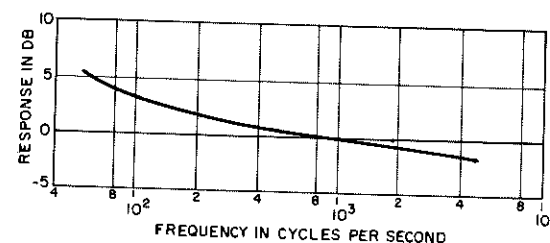


Fig. 9.78. Relative response derived from the dynamic average of the direct and reflected sound reproduced in a room in a residence.

be deduced from the reverberation characteristic of a typical living room in a home as shown in Fig. 8.14. The general run of direct-radiator loudspeakers exhibit increased directivity with increase of the frequency. The combination of the acoustical characteristics of the room and loudspeaker conspires to produce an accentuation in the low-frequency response as perceived by the listener. The relative response²³ at normal listening distances derived from the dynamic average of the direct and generally reflected sound for the case of music reproduced in a room in a residence is shown in Fig. 9.78.

C. Loudness and Dynamics

The loudness versus loudness level^{24,25} of a performance must also be considered in the transition from the concert hall, auditorium, studio or any large room to the small room in the home. The loudness in sones is

²³ H. F. Olson, *Acoustical Engineering*, D. Van Nostrand Company, Princeton, 1957.

²⁴ Stephens, S. S., *Jour. Acoust. Soc. Amer.*, Vol. 27, No. 5, p. 815, 1955.

²⁵ Lochner and Burges, *Jour. Acoust. Soc. Amer.*, Vol. 34, No. 5, p. 576, 1962.

the average person's estimate of the loudness sensation. The loudness in sones is given by

$$S = 2^{(P-40)/10} \quad (9.2)$$

where S = loudness in sones

P = loudness level, in phons

The loudness versus loudness level is another factor which must be considered in the transition from the relatively high-sound level in the large acoustic enclosure to the relatively low-sound level in a small room. Dynamics in the framework of sound reproduction is used to designate the relation between the subjective and objective aspects of the amplitude of the sound.

D. Dynamics, Noise and Masking

In view of the fact that the level of sound reproduction in the home is relatively low, some consideration must be given to the amplitude range between the lower level established by the ambient noise and the average upper level of 80 decibels.

The ambient noise in a room masks and renders inaudible the reproduced sounds below a certain level. Therefore, the ambient noise level in the average residence is a factor that must be considered in the reproduction of sound.

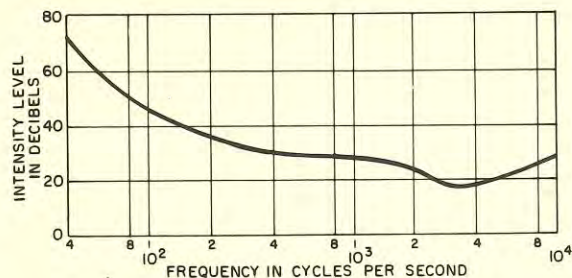


FIG. 9.79. Hearing limit for pure tones for the average listener in a typical residence.

The ambient noise level²⁶ for the entire audible range in 90 per cent of the residences falls somewhere between 33 and 52 decibels. The noise level in a room in an average residence is 43 decibels.

The spectrum of room noise is another factor that determines the level at which sounds disappear in the ambient as a function of frequency. The spectrum of typical room noise²⁷ decreases 5 decibels per octave with in-

²⁶ Seacord, D. F., *Jour. Acoust. Soc. Amer.*, Vol. 12, No. 1, p. 183, 1940.

²⁷ Hoth, D. F., *Jour. Acoust. Soc. Amer.*, Vol. 12, No. 4, p. 499, 1941.

crease in frequency as shown in Fig. 8.41. This type of characteristic appears to be the nature of practically all types of ambient noise.

From the average sound level and spectrum of the ambient noise it is possible to determine the lower hearing limit for pure tones.²⁸ The graph of Fig. 9.79 depicts the level below which a pure tone cannot be heard for a room in which the total sound level is 43 decibels. A large number of direct listening tests have been carried out to determine the level below which a pure tone disappears in the ambient. These data substantiate the characteristics of Fig. 9.79 within the usual limits of subjective tests. The threshold characteristic of Fig. 9.79 establishes the lower level of hearing in a room in an average residence.

E. Auditory Perspective²⁹

Stereophonic sound reproduction as depicted in Fig. 9.25 provides auditory perspective of the reproduced sound. Reproduction of sound in auditory perspective provides the illusion of the distribution of the reproduced sound in lateral directions as well as in depth in a geometrical configuration and correspondence which approximate the disposition of the original sound sources.

The microphone placement³⁰ with respect to the sound sources plays an important part in the auditory perspective aspects of the reproduced sound in the living room. The sound sources and the microphone placement in the studio have been studied by means of listening tests in a typical living room. In this work the most important result has been the observation that the subjective location of a sound source in stereophonic sound reproduction is determined by the phase and amplitude. This has been illustrated by the experiment depicted in Fig. 9.26. This experiment shows that the apparent position of the sound source can be shifted in a lateral direction from a point directly in front of either loudspeaker to any point between by varying the relative phases and/or relative intensities of the sound emanating from the loudspeakers of channels *A* and *B*.

F. Reverberation³¹

The growth, duration, decay and reverberation characteristics of the sound as perceived by the listener of Fig. 9.25 involve the combination of two rooms. Here the objective is to provide an over-all growth, duration, decay and reverberation characteristic which provides the listener with a subjective impression of these properties which match those of the

²⁸ Fletcher and Munson, *Jour. Acoust. Soc. Amer.*, Vol. 9, No. 1, p. 1, 1937.

²⁹ Fletcher, H., *Jour. Soc. Mot. Pic. Tel. Eng.*, Vol. 61, No. 3, p. 415, 1953.

³⁰ Olson, H. F., *Jour. Audio Eng. Soc.*, Vol. 6, No. 2, p. 80, 1958.

³¹ Olson, H. F., *Jour. Audio Eng. Soc.*, Vol. 12, No. 2, p. 98, 1964.

live rendition in the acoustic enclosure. In view of the fact that these characteristics in the living room play a minor part in the impression of the reverberant sound, the design of the acoustics of the studio³² and the placement of microphones become the important considerations. In this connection, the growth characteristic plays the most important part in providing the desired artistic effects. The relative importance of the reflected sounds decreases with each reflection due to absorption at each encounter with boundaries. Therefore, the first reflections should be spaced in time so as to produce a smooth growth characteristic. In view of the fact that two rooms are involved in sound reproduction, the envelope of the transfer characteristic of the studio should be smooth with respect to frequency. The general reverberation of the sound which provides the blending of the reflected sounds with the direct sound imparts the artistic aspects associated with concert halls, auditoriums and theaters. The reverberation time of the acoustic enclosure in combination with the placement of the direct and reverberant microphones should be coordinated so that the over-all effect produced at the listener corresponds to the live condition. If the above general procedures are followed the general impression of the reverberant sound will be a subjective resemblance of the reverberant sound in the live condition.³³

G. Quality and Timbre

Studies³⁴ have been carried out on the characteristics of musical instruments. These studies have included the frequency range, frequency spectrum, the directional pattern and the growth, duration and decay of the tones produced by the instruments.

The frequency range of musical instruments covers the entire audible frequency range. Therefore, to achieve faithful sound reproduction the reproducing system must cover the audible frequency range without frequency discrimination. The frequency response of the system must be free of variations otherwise the frequency spectrum of the instrument will be altered. The effect of nonuniform response with respect to frequency can be illustrated by referring to typical spectrums of musical instruments as shown in Sec. 6.3C. Obviously, if there are variations in the response the spectrum will be changed and as a result the character of the musical instrument will be changed.

³² Bolle, Voldner, Pulley and Volkmann, *Jour. Audio Eng. Soc.*, Vol. 11, No. 1, p. 80, 1963.

³³ The reverberation of the live condition to be simulated in the room in a home may be that of a cathedral, church, concert hall, auditorium, theater, ballroom, restaurant, etc. In some popular music for example, the reverberation is almost nonexistent.

³⁴ See Chap. 6.

The directional pattern of voiced and musical instruments with respect to frequency and orientation are exceedingly complex as shown in Sec. 6.5. Therefore, the timbre which plays such an important part in the identification of the voice or instrument varies with the orientation. As a consequence great care must be taken in providing the orientation with respect to the microphone which will give the most realistic reproduced sound identified with the particular voice or instrument.

H. Implementation of Objectives

The main purpose of sound reproduction as manifested by the performance and experienced by the listener is to provide the listener with the highest order of artistic and subjective resemblance to the condition of the live rendition.

The psychological and psychophysical effects involved in achieving the objective have been described in the preceding sections. The purpose of this section is to delineate in abbreviated form the processes that must be implemented to attain the objective.

The growth, duration, decay, and reverberation characteristics as perceived by the listener involves the combination of two rooms. Here the objective is to provide an over-all growth, duration, decay and reverberation characteristic which provides the listener with a subjective impression of these properties which match those of the live rendition. In view of the fact that these characteristics in the average room in the home play a minor part in the impression of the reverberant sound, the design of the studio and the placement of the microphones are the important considerations.

The auditory perspective of the reproduced sound should approximate that of the best location in the standard live condition. To accomplish this result requires the correct placement of the microphones.

The quality of the reproduced sound must be preserved in reproduction. This means that all the characteristics of the musical instruments must be faithfully reproduced. The technical data relating to the characteristics of musical instruments must be employed as outlined in preceding sections in order to provide faithful reproduction of voices and instruments.

The dynamics of the sound which involves the relation between the subjective and objective aspects of the amplitude of the sound should simulate the live condition in the reproduced sound. One of the most important considerations relating to the dynamics is the subjective effect of the drop in sound level of approximately 20 decibels between the sound level of the live program and the tolerable sound level of the reproduced sound, the reason being that the frequency response of the human hearing mechanism

is a function of the sound level. Therefore, suitable dynamic frequency-response compensation as a function of the amplitude must be introduced by appropriate electronic means.

I. Dynamic Spectrum Equalizer

The preceding sections on the objective and subjective aspects of sound reproduction have been concerned with the relations involved in obtaining a simulation of the live condition in the reproduction of sound. One of the main considerations relating to the dynamics is compensation for the 20 decibels difference in level between the live and reproduced sound. An electronic system termed a Dynamic Spectrum Equalizer has been developed to provide the compensation for the difference in sound levels and other minor differences in the subjective aspects.

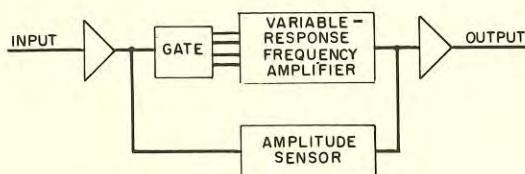


FIG. 9.80. Schematic block diagram of the Dynamic Spectrum Equalizer.

A block diagram of the Dynamic Spectrum Equalizer is shown in Fig. 9.80. The system operates in a continuous manner to change the response-frequency characteristic as a function of the amplitude. Typical response-frequency characteristics for various levels are shown in Fig. 9.81. There is a continuous variation in response from one level to another. The response-frequency characteristics differ for different types of musical selections; in effect, when the levels are low the low-frequency components are accentuated. For medium levels there are slight accentuations in the low-frequency region and the presence region of 2,000 to 6,000 cycles per second and a reduction in response in the region from 400 to 1,000 cycles per second. For high sound levels there is accentuation in response in the presence region, and a reduction in the frequency range below 1,000 cycles per second. When the sound level of the program is low, the objective is to raise the sound level of the appropriate frequency regions so that the music can be appreciated under the ambient noise and surrounding conditions of the average residence. When the sound level of the program is high, the level of the presence region is raised and the level of the low-frequency range is lowered. This procedure does not upset the dynamic

balance but rather enhances this aspect of sound reproduction in a small room.

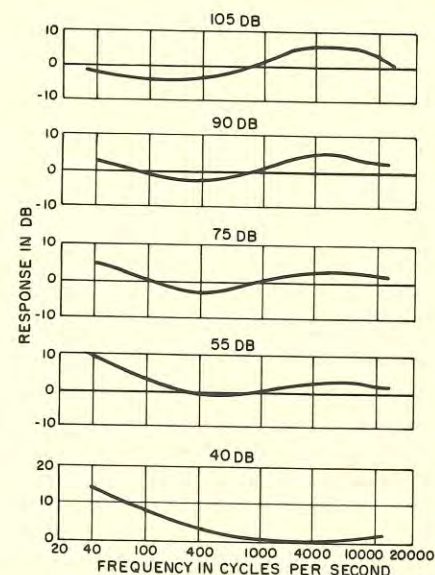


FIG. 9.81. Response-frequency characteristics of the Dynamic Spectrum Equalizer for various sound levels of the program.

The Dynamic Spectrum Equalizer was designed to provide a dynamic alteration of the projection qualities of sound so that under conditions of playback, which differ from those in which the music was performed, the best perception of the qualities of the original performance would be obtained.