ELEMENTS OF ACOUSTICAL ENGINEERING

OLSON

SECOND EDITION

ACOUSTICAL

ENGINEERING

HARRY F. OLSON, Ph.D.

Director, Acoustical and Electromechanical
Research Laboratory, RCA Laboratories,
Princeton, New Jersey

D. VAN NOSTRAND COMPANY, INC.
PRINCETON, NEW JERSEY
TORONTO   LONDON
NEW YORK

PREFACE

The first edition of this book, published in 1940, was the subject matter of thirty lectures prepared for presentation at Columbia University. It was an exposition of the fundamental principles used in modern acoustics and a description of existing acoustical instruments and systems.

Many and varied advances were made in acoustical engineering in the seven years following the issuance of the first edition. The second edition of the book, published in 1947, covered the advances in acoustics which were made in the period between the first and second editions. Since the publication of the second edition, the developments in acoustics have been on an ever greater scale than in the period between the first and second editions. Today, the science of acoustics includes the generation, transmission, reception, absorption, conversion, detection, reproduction and control of sound. An important division of acoustical engineering is sound reproduction as exemplified by the telephone, radio, phonograph, sound motion picture and television. These sound reproducing systems are universally employed in all variations of modern living. The impact of the reproduction of sound by these systems upon the dissemination of information, art and culture has been tremendous.

The ultimate useful destination of all informative sound, direct or reproduced, is the human ear. In this connection, great strides have been made in obtaining knowledge on the characteristics and action of the human hearing machine. Measurements play an essential part in the advancement of any scientific field. Instruments have been developed and standards have been established for the measurement of the fundamental quantities in acoustics. The applications of acoustics in the field of music have led to a better understanding of the stuff of which music is made. This knowledge has been applied to the development of new musical instruments employing the latest electronic and acoustical principles.

Accelerated by the requirements in World War II, tremendous advances were made in underwater sound. The developments in underwater sound have resulted in systems for detection and accurate location of underwater craft and obstacles over great distances, depth sounders and other acoustic applications in undersea communication. The industrial applications of ultrasonics have unfolded a new field in acoustics. Some of the important ultrasonic developments include the cleaning of machine parts, drilling and flaw detection. The science of architectural acoustics has advanced to the point where auditoriums, studios and rooms can be designed to obtain excellent acoustics under severe artistic conditions. With ever increasing in-

dustrial expansion comes an increase in noise. Work is now under way actively to control noise by the use of a variety of acoustic countermeasures.

The preceding brief description of the present status of acoustics shows that it plays a very important part in our modern civilization. Furthermore, the fundamentals and applications of the science of acoustics are so well formulated and substantiated that a large area of the field of acoustics has attained an engineering status. In preparing new material and in revising existing material in the third edition, the same principles were followed as in the first and second editions. Particular efforts have been directed towards the development of analogies between electrical, mechanical and acoustical systems because engineers have found that the reduction of a vibrating system to the analogous electrical network is a valuable tool in the analysis of vibrating systems. Each chapter has been brought up to date and amplified. Two new chapters on Complete Sound Reproducing Systems and Means for the Communication of Information have been added. As in the first and second editions most of the illustrations contain several parts so that a complete theme is depicted in a single illustration.

The author wishes to express his appreciation to Miss Patricia Durnan for her work in typing the manuscript and to his wife Lorene E. Olson for assistance in compiling and correcting the manuscript.

HARRY F. OLSON

March, 1957

CONTENTS

v

1

SOUND WAVES

1.1. Introduction.—The term acoustics in its broadest sense is a term used to designate an art and a science involving sound in all its manifold forms and manifestations. Specifically, acoustics includes the generation, transmission, reception, absorption, conversion, detection, reproduction, and control of sound.

An important division of acoustics is the reproduction of sound which is the process of picking up sound at one point, and reproducing it at the same point, or at some other point either at the same time, or at some subsequent time. The most common sound reproducing systems are the telephone, phonograph, radio, sound motion picture, and television.

The radio, phonograph, sound motion picture, and television have made it possible for all the people of the world to hear famous statesmen, artists, actors, and musical aggregations where only a relatively small number had been able to hear them first hand. It is evident that the reproduction of sound has produced in a relatively short time a great change in the education and entertainment of this and other countries. The impact of the telephone, phonograph, radio broadcasting, sound motion pictures, and television upon the dissemination of information, art, and culture has been tremendous. The reproduction of sound in these fields has been as important to the advancement of knowledge as the printing press and the printed page.

The ultimate useful destination of all informative sound, direct or repro-duced, is the human ear. In this connection, great strides have been made in obtaining knowledge on the characteristics and action of the human hearing mechanism.

Measurements play an essential part in the advancement of any scientific field. Instruments have been developed and standards have been established for the measurement of the fundamental quantities in acoustics.

The applications of acoustics in the field of music have led to a better understanding of the stuff of which music is made. This knowledge has been applied to the development of new musical instruments employing the latest electronic and acoustic principles.

Accelerated by the requirements in World War II, tremendous advances were made in underwater sound. The developments in underwater sound have resulted in systems for detection and accurate location of underwater

craft and obstacles over great distances, in depth sounders, and in other acoustic applications in undersea communication.

The industrial applications of ultrasonics have unfolded a new field in acoustics. Some of the important ultrasonic developments include the cleaning of machined parts, drilling, and flaw detection.

The science of architectural acoustics has advanced to the point where auditoriums, studios, and rooms can be designed to obtain excellent acoustics under severe artistic conditions.

With ever-increasing industrial expansion comes an increase in noise. Work is now under way actively to control noise by the use of a variety of acoustic countermeasures.

The preceding brief introduction to the present status of acoustics shows that it plays a very important part in our modern civilization. Furthermore, the fundamentals and applications of the science of acoustics are so well formulated and substantiated that a large area of the field of acoustics has attained an engineering status.

In this book the author has attempted to outline the essentials of modern acoustics from the standpoint of the engineer or applied scientist. It has been the aim and purpose to make the book as complete as possible by covering all the major aspects of modern acoustics as outlined in the preceding text of the introduction. In order to cover a wide range of readers, the book has been written and illustrated so that the derivations may be taken for granted. The concepts of mechanical and acoustical impedance have been introduced and applied so that anyone who is familiar with electrical circuits will be able to analyze the action of vibrating systems.

1.2. Sound Waves.—Sound is an alteration in pressure, particle displacement or particle velocity propagated in an elastic material or the superposition of such propagated alterations.

Sound is also the sensation produced through the ear by the alterations described above.

Sound is produced when air is set into vibration by any means whatsoever, but sound is usually produced by some vibrating object which is in contact with the air. If a string, such as one used in a banjo or similar instrument, is stretched between two solid supports and plucked, sound is produced which dies down in a fairly short time. When the string is plucked it tends to spring back into its rest position, but due to its weight (mass) and speed (velocity) it goes beyond its normal position of rest. Then, in returning it again goes beyond its normal position of rest. The excursions become smaller and smaller and finally the string comes to rest. As the string moves forward it pushes air before it and compresses it, while air rushes in to fill the space left behind the moving string. In this way air is set in motion. Since air is an elastic medium, the disturbed portion transmits its motion to the surrounding air so that the disturbance is propagated in all directions from the source of disturbance.

If the string is connected in some way to a diaphragm such as a stretched drumhead of a banjo, the motion is transmitted to the drum. The drum,

having a large area exposed to the air, sets a greater volume of air in motion and a much louder sound is produced.

If a light piston several inches in diameter, surrounded by a suitable baffle board several feet across, is set in rapid oscillating motion (vibration) by some external means, sound is produced (Fig. 1.1). The air in front of the piston is compressed when it is driven forward, and the surrounding air expands to fill up the space left by the retreating piston when it is drawn back. Thus we have a series of compressions and rarefactions (expansions) of the air as the piston is driven back and forth. Due to the elasticity of air these areas of compression and rarefaction do not remain stationary but move outward in all directions. If a pressure gage were set
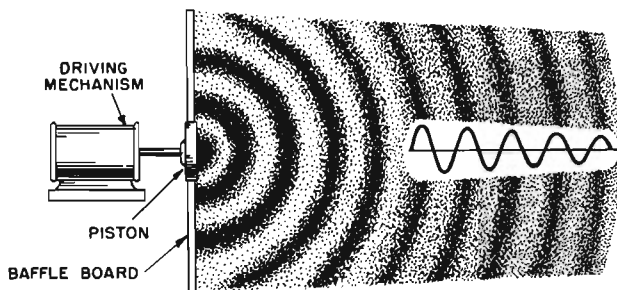
FIG. 1.1. Production of sound waves by a vibrating piston.

up at a fixed point and the variation in pressure noted, it would be found that the pressure varies in regular intervals and in equal amounts above and below the average atmospheric pressure. Of course, the actual variations could not be seen because of the high rate at which they occur. Now, suppose that the instantaneous pressure, along a line in the direction of sound propagation, is measured and plotted with the ordinates representing the pressure; the result would be a wavy line as shown in Fig. 1.1. The points above the straight line represent positive pressures (compressions, condensations); the points below represent negative pressures (expansions, rarefactions) with respect to the normal atmospheric pressure represented by the straight line.

From the above examples a few of the properties of sound waves and vibrations in general may be defined.

Periodic Quantity.—A periodic quantity is an oscillating quantity the values of which recur for equal increments of the independent variable.

Cycle.—One complete set of recurrent values of a periodic quantity comprises a cycle; or, in other words, any one set of variations starting at one condition and returning once to the same condition is a cycle.

Period.—The period is the time required for one cycle of a periodic quantity.

Frequency.—The number of cycles occurring per unit of time, or which would occur per unit of time if all subsequent cycles were identical with

the cycle under consideration, is the frequency. The unit is the cycle per second.

Fundamental Frequency.—A fundamental frequency is the lowest component frequency of a periodic wave or quantity.

Harmonic.—A harmonic is a component of a periodic wave or quantity having a frequency which is an integral multiple of the fundamental frequency. For example, a component, the frequency of which is twice the fundamental frequency, is called the second harmonic.

Subharmonic.—A subharmonic is a component of a complex wave having a frequency which is an integral submultiple of the basic frequency.

Wavelength.—The wavelength of a periodic wave in an isotropic medium is the perpendicular distance between two wave fronts in which the displacements have a phase difference of one complete cycle.

Octave.—An octave is the interval between two frequencies having a ratio of two to one.

Transducer.—A transducer is a device by means of which energy may flow from one or more transmission systems to one or more other transmission systems. The energy transmitted by these systems may be of any form (for example, it may be electrical, mechanical, or acoustical) and it may be the same form or different forms in the various input and output systems.

The example of Fig. 1.1 has shown graphically some of the properties of wave motion. It is the purpose of the next section to derive the fundamental wave equation. It is not necessary that the reader digest all the assumptions and processes involved in order to obtain valuable information concerning the properties of a sound wave.

1.3. Acoustical Wave Equation.—The general case of sound propagation involves three dimensions. The general relation for sound propagation of small amplitudes in three dimensions will be derived and then these relations will be applied to special problems.

A. *Equation of Continuity.*—The fundamental equation of hydrokinetics is the equation of continuity. This equation is merely a mathematical statement of an otherwise obvious fact that matter is neither created nor destroyed in the interior of the medium. That is, the amount of matter which enters the boundaries of a small volume equals the increase of matter inside. Consider the influx and efflux through each pair of faces of the cube of dimensions Δx, Δy, and Δz, the difference between the latter and the former for the whole cube is

$$- \left[\frac{\partial(\rho' u)}{\partial x} + \frac{\partial(\rho' v)}{\partial y} + \frac{\partial(\rho' w)}{\partial z} \right] \Delta x \, \Delta y \, \Delta z \qquad 1.1$$

where x, y, z = coordinates of a particle in the medium,

u, v, w = component velocities of a particle in the medium, and

ρ' = density of the medium.

The rate of growth of mass $\dfrac{\partial \rho'}{\partial t} \Delta x \, \Delta y \, \Delta z$ in the cube must be equal to the expression 1.1. This may be written as

$$\frac{\partial \rho'}{\partial t} + \frac{\partial(\rho'u)}{\partial x} + \frac{\partial(\rho'v)}{\partial y} + \frac{\partial(\rho'w)}{\partial z} = 0 \qquad 1.2$$

where t = time.

This is the equation of continuity which signifies the conservation of matter and the three dimensionality of space.

B. *Equation of Motion.*—Referring again to the space $\Delta x \, \Delta y \, \Delta z$ the acceleration of momentum parallel to x is $\rho' \, \Delta x \, \Delta y \, \Delta z \, \dfrac{\partial u}{\partial t}$. The mean pressures on the faces perpendicular to x are

$$\left(p_0' - \frac{\partial p_0'}{\partial x} \frac{\Delta x}{2}\right) \Delta y \, \Delta z \quad \text{and} \quad \left(p_0' + \frac{\partial p_0'}{\partial x} \frac{\Delta x}{2}\right) \Delta y \, \Delta z$$

where p_0' = pressure in the medium.

The difference is a force $\dfrac{\partial p_0}{\partial x} \Delta x \, \Delta y \, \Delta z$ in the direction of increasing x. Equating this to the acceleration of momentum, the result is the equation of motion,

$$\rho' \frac{\partial u}{\partial t} = - \frac{\partial p_0'}{\partial x}, \; \rho' \frac{\partial v}{\partial t} = - \frac{\partial p_0'}{\partial y}, \; \rho' \frac{\partial w}{\partial t} = - \frac{\partial p_0'}{\partial z} \qquad 1.3$$

The equation of motion may be written

$$\frac{dV_{uvw}}{dt} + \frac{1}{\rho} \operatorname{Grad} p_0' = 0 \qquad 1.4$$

C. *Compressibility of a Gas.*—The next property of a gas which is used to derive the wave equation depends upon the thermodynamic properties of gases. The expansions and contractions in a sound wave are too rapid for the temperature of the gas to remain constant. The changes in pressure and density are so rapid that practically no heat energy has time to flow away from the compressed part of the gas before this part is no longer compressed. When the gas temperature changes, but its heat energy does not, the compression is termed adiabatic.

In the case of an adiabatic process,

$$\frac{p_0'}{p_0} = \left(\frac{\rho'}{\rho}\right)^{\gamma} \qquad 1.5$$

where p_0 = static pressure. The static pressure is the pressure that would exist in the medium with no sound waves present. The unit is the dyne per square centimeter.

ρ = static or original density,

p_0' = total pressure (static + excess),

ρ' = instantaneous density (static + change), and

γ = ratio of specific heat at constant pressure to that at constant volume and has a value of 1.4 for air.

D. *Condensation.*—A new term will now be introduced. Condensation is defined as the ratio of the increment of density change to the original density,

$$s = \frac{\rho' - \rho}{\rho} \qquad 1.6$$

Combining equations 1.5 and 1.6

$$\frac{p_0'}{p_0} = \left(\frac{\rho'}{\rho}\right)^\gamma = (1 + s)^\gamma = 1 + \gamma s \qquad 1.7$$

or

$$p_0' = p_0 + p_0 \gamma s \qquad 1.8$$

The excess pressure, or instantaneous sound pressure p, is $p_0' - p_0$.

$$p = p_0 \gamma s \qquad 1.9$$

The instantaneous sound pressure at a point is the total instantaneous pressure at that point minus the static pressure. The unit is the dyne per square centimeter. This is often called excess pressure.

The effective sound pressure at a point is the root-mean-square value of the instantaneous sound pressure over a complete cycle, at that point. The unit is the dyne per square centimeter. The term "effective sound pressure" is frequently shortened to "sound pressure."

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. In the case of a sinusoidal sound wave this maximum sound pressure is also called the pressure amplitude.

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.

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

E. *D'Alembertian Wave Equation.*—The three equations 1.2, 1.4, and 1.5 characterize disturbances of any amplitude. The first two are non-linear save for small amplitudes. In general, acoustic waves are of infinitesimal amplitudes, the alternating pressure is small compared with the atmospheric pressure and the wavelength is so long that u, v, w, and s change very little with x, y, and z. Substituting equation 1.6 in 1.2 and neglecting high order terms,

$$\frac{\partial s}{\partial t} + \frac{\partial u}{\partial x} + \frac{\partial v}{\partial y} + \frac{\partial w}{\partial z} = 0 \qquad 1.10$$

The type of motion to be considered is irrotational, that is Curl $V_{uvw} = 0$. That is a necessary and sufficient condition for the existence of a scalar velocity potential ϕ which is defined as

$$u = \frac{\partial \phi}{\partial x}, \quad v = \frac{\partial \phi}{\partial y}, \quad w = \frac{\partial \phi}{\partial z} \qquad 1.11$$

or

$$V_{uvw} = \text{Grad } \phi$$

Substitute equations 1.11 in 1.3 and multiply by dx, dy, and dz

$$\frac{\partial}{\partial t}d\phi = -\frac{1}{\rho}dp_0{}''$$

1.12

or integrating

$$\frac{\partial \phi}{\partial t} = -\int \frac{dp_0'}{\rho'}$$

Since the density changes very little, the mean density, ρ, may be used. The $\int dp_0'$ is the excess pressure; then

$$\frac{\partial \phi}{\partial t} = -\frac{p}{\rho}$$

1.13

where p = excess pressure.

From equations 1.9, 1.10, 1.11, and 1.13

$$\frac{\partial^2 \phi}{\partial t^2} - \frac{\gamma p_0}{\rho}\left(\frac{\partial^2 \phi}{\partial x^2} + \frac{\partial^2 \phi}{\partial y^2} + \frac{\partial^2 \phi}{\partial z^2}\right) = 0$$

1.14

or this may be written

$$\frac{\partial^2 \phi}{\partial t^2} = c^2 \nabla^2 \phi$$

which is the standard D'Alembertian wave equation for ϕ. The velocity of propagation is

$$\frac{\gamma p_0}{\rho} = c^2$$

1.15

For the velocity of sound in various mediums see Table 1.1.

TABLE 1.1. YOUNG'S MODULUS Q, IN DYNES PER SQUARE CENTIMETER, POISSON'S RATIO σ, DENSITY ρ, IN GRAMS PER CUBIC CENTIMETER, VELOCITY OF SOUND c, IN METERS PER SECOND, AND THE SPECIFIC ACOUSTICAL RESISTANCE ρc, IN GRAMS PER SECOND PER SQUARE CENTIMETER

METALS

Substance	Q	σ	ρ	c	ρc
Aluminum	7.3×10^{11}	.33	2.7	5200	140×10^4
Antimony	7.8×10^{11}	.33	6.6	3400	220×10^4
Beryllium	26.0×10^{11}	.33	1.8	12000	216×10^4
Bismuth	3.19×10^{11}	.35	9.7	1800	170×10^4
Cadmium	5.3×10^{11}	.30	8.6	2500	215×10^4
Cobalt	19.0×10^{11}	.30	8.7	4700	410×10^4
Copper	11.0×10^{11}	.35	8.9	3500	310×10^4
Gold	8.0×10^{11}	.35	19.3	2000	390×10^4
Iridium	52.0×10^{11}	.33	22.4	4700	1050×10^4
Iron Cast	9.0×10^{11}	.29	7.8	3400	270×10^4
Iron Wrought	20.0×10^{11}	.28	7.9	5100	400×10^4
Lead	1.7×10^{11}	.43	11.3	1200	130×10^4
Magnesium	4.0×10^{11}	.33	1.7	4800	82×10^4
Mercury	...	...	13.5	1400	190×10^4
Nickel	21.0×10^{11}	.31	8.8	4900	430×10^4

METALS (*continued*)

Substance	Q	σ	ρ	c	ρc
Palladium . . .	12.0×10^{11}	.39	12.0	3200	380×10^4
Platinum . . .	17.0×10^{11}	.33	21.4	2800	600×10^4
Rhodium . . .	30.0×10^{11}	.34	12.4	4900	610×10^4
Silver . . .	7.8×10^{11}	.37	10.5	2700	280×10^4
Tantalum . . .	19.0×10^{11}	.31	16.6	3400	560×10^4
Tin	4.5×10^{11}	.33	7.3	2500	180×10^4
Tungsten . . .	35.0×10^{11}	.17	19.0	4300	830×10^4
Zinc	8.2×10^{11}	.17	7.1	3400	240×10^4

ALLOYS

Substance	Q	σ	ρ	c	ρc
Alnico . . .	17.0×10^{11}	.32	7.0	4900	340×10^4
Beryllium Copper . .	12.5×10^{11}	.33	8.2	3900	320×10^4
Brass	9.5×10^{11}	.33	8.4	3400	290×10^4
Bronze Phosphor . .	12.0×10^{11}	.35	8.8	3700	330×10^4
Duraluminum .	7.0×10^{11}	.33	2.8	5000	140×10^4
German Silver . .	11.6×10^{11}	.37	8.1	3800	310×10^4
Monel . . .	18.0×10^{11}	.32	8.8	4500	400×10^4
Steel C.08 . . .	19.0×10^{11}	.27	7.7	5000	390×10^4
Steel C.38 . . .	20.0×10^{11}	.29	7.7	5100	390×10^4

CERAMICS, ROCKS

Substance	Q	σ	ρ	c	ρc
Brick	2.5×10^{11}	. . .	1.8	3700	67×10^4
Clay Rock . . .	2.5×10^{11}	. . .	2.2	3400	75×10^4
Concrete . . .	2.5×10^{11}	. . .	2.6	3100	81×10^4
Glass, Hard . .	8.7×10^{11}	. . .	2.4	6000	144×10^4
Glass, Soft . .	6.0×10^{11}	. . .	2.4	5000	120×10^4
Granite . . .	9.8×10^{11}	. . .	2.7	6000	162×10^4
Isolantite . . .	5.0×10^{11}	. . .	2.4	4600	110×10^4
Limestone . . .	2.9×10^{11}	. . .	2.6	3300	86×10^4
Marble . . .	3.8×10^{11}	. . .	2.6	3800	99×10^4
Porcelain . . .	4.2×10^{11}	. . .	2.4	4200	102×10^4
Quartz, Fused . .	5.2×10^{11}	. . .	2.7	4400	118×10^4
Quartz, 11 Optic . .	10.3×10^{11}	. . .	2.7	6200	168×10^4
Quartz, 1 Optic . .	7.95×10^{11}	. . .	2.7	5400	146×10^4
Slate	5.8×10^{11}	. . .	2.9	4500	131×10^4
Ice	$.94 \times 10^{11}$	. . .	.92	3200	29×10^4

WOODS (WITH THE GRAIN)

Substance	Q	σ	ρ	c	ρc
Ash	1.3×10^{11}	. . .	.64	4500	29×10^4
Beech . . .	1.0×10^{11}	. . .	.65	3900	25×10^4
Cork . . .	$.0062 \times 10^{11}$	. . .	.25	500	1.2×10^4
Elm	1.0×10^{11}	. . .	.54	4300	23×10^4
Fir	1.1×10^{11}	. . .	.51	4700	24×10^4
Mahogany . .	1.1×10^{11}	. . .	.67	4000	27×10^4
Maple . . .	1.3×10^{11}	. . .	.68	4300	29×10^4
Oak, White . .	1.2×10^{11}	. . .	.72	4100	29×10^4
Pine, White . .	$.6 \times 10^{11}$	. . .	.45	3600	16×10^4
Poplar . . .	1.0×10^{11}	. . .	.46	4600	21×10^4
Sycamore . .	1.0×10^{11}	. . .	.54	4300	23×10^4
Walnut . . .	1.2×10^{11}	. . .	.56	4600	26×10^4

Across the grain, $\frac{1}{4}$ to $\frac{1}{2}$ of the above values for c.

PLASTICS

Substance	Q	σ	ρ	c	ρc
Cellulose Acetate, Sheet	1.4×10^{10}	...	1.3	1000	13×10^4
Cellulose Acetate, Molded	2.1×10^{10}	...	1.3	1300	17×10^4
Cellulose Acetate, Butyrate	17.0×10^{10}	...	1.2	3700	44×10^4
Cellulose Acetate, Pyroxylin	21.0×10^{10}	...	1.5	3700	55×10^4
Ethyl Cellulose	2.1×10^{10}	...	1.1	1400	15×10^4
Ivory	9.0×10^{10}	...	1.8	2200	40×10^4
Methyl Metha-Crylate Resin, Cast	3.5×10^{10}	...	1.2	1700	20×10^4
Methyl Metha-Crylate Resin, Molded	2.8×10^{10}	...	1.2	1500	18×10^4
Paper, Parchment	4.8×10^{10}	...	1.0	2200	22×10^4
Paraffin, 16° C.	1.5×10^{10}	...	.9	1300	12×10^4
Phenol-Formaldehyde Wood Filler	8.4×10^{10}	...	1.35	2500	34×10^4
Phenol-Formaldehyde Paper Base	7.0×10^{10}	...	1.3	2300	30×10^4
Phenol-Formaldehyde Fabric Base	8.4×10^{10}	...	1.35	2500	34×10^4
Phenol-Formaldehyde Mineral Filler	10.5×10^{10}	...	1.8	2400	43×10^4
Rubber, Hard	2.3×10^{10}	...	1.1	1400	15×10^4
Rubber, Soft	$.5 \times 10^8$	...	.95	70	$.67 \times 10^4$
Sheepskin	2.0×10^9	...	.9	470	4.2×10^4
Shellac Compound	3.8×10^{10}	...	1.7	1500	26×10^4
Styrene Resin	3.1×10^{10}	...	1.1	1700	19×10^4

LIQUIDS

Substance	Q	σ	ρ	c	ρc
Alcohol, Methyl	....	...	.81	1240	10.0×10^4
Benzine	....	...	.90	1170	10.5×10^4
Chloroform	....	...	1.5	983	14.7×10^4
Ether	....	...	.74	1020	7.6×10^4
Gasoline	....	...	.68	1390	9.4×10^4
Turpentine	....	...	.87	1330	11.6×10^4
Water, 13° C.	....	...	1.0	1441	14.4×10^4
Water, Salt	....	...	1.03	1540	15.5×10^4

GASES

Substance	Q	σ	ρ	c	ρc
Air, 0° C.	....	...	.00129	331	42.7
Air, 20° C.	....	...	.00120	344	41.4
Carbon Monoxide	....	...	.00125	337	42.0
Carbon Dioxide	....	...	.00198	258	51.2
Chlorine	....	...	.00317	205	65.0
Hydrogen	....	...	.00009	1270	11.4
Methane	....	...	.00072	432	31.0
Nitrogen	....	...	.00125	336	42.0
Oxygen	....	...	.00143	317	45.5
Steam	....	...	.00058	405	23.5

1.4. Plane Sound Waves.—Assume that a progressive wave proceeds along the axis of x. Then ϕ is a function of x and t only and the wave equation 1.14 reduces to

$$\frac{\partial^2 \phi}{\partial t^2} = c^2 \frac{\partial^2 \phi}{\partial x^2} \qquad 1.16$$

A solution of this equation for a simple harmonic wave traveling in the positive x direction is

$$\phi = A \cos k(ct - x) \qquad 1.17$$

where A = amplitude of ϕ,
   $k = 2\pi/\lambda$,
   λ = wavelength, in centimeters,
   $c = f\lambda$ = velocity of sound, in centimeters per second, and
   f = frequency, in cycles per second.

A. *Particle Velocity in a Plane Sound Wave.*—The particle velocity, u, employing equations 1.11 and 1.17 is

$$u = \frac{\partial \phi}{\partial x} = kA \sin k(ct - x) \qquad 1.18$$

The particle velocity in a sound wave is the instantaneous velocity of a given infinitesimal part of the medium, with reference to the medium as a whole, due to the passage of the sound wave.

B. *Pressure in a Plane Sound Wave.*—From equations 1.9, 1.13, and 1.15 the following relation may be obtained

$$\frac{\partial \phi}{\partial t} = - c^2 s \qquad 1.19$$

The condensation in a plane wave from equations 1.19 and 1.17 is given by

$$s = \frac{Ak}{c} \sin k(ct - x) \qquad 1.20$$

From equations 1.9 and 1.15 the following relation may be obtained

$$p = c^2 \rho s \qquad 1.21$$

Then, from equations 1.20 and 1.21 the pressure in a plane wave is

$$p = kc\rho A \sin k(ct - x) \qquad 1.22$$

Note: the particle velocity, equation 1.18, and the pressure, equation 1.22, are in phase in a plane wave.

C. *Particle Amplitude in a Plane Sound Wave.*—The particle amplitude of a sound wave is the maximum distance that the vibrating particles of the medium are displaced from the position of equilibrium.

From equation 1.18 the particle velocity is

$$\dot{\xi} = u = kA \sin k(ct - x) \qquad 1.23$$

where ξ = amplitude of the particle from its equilibrium position, in centimeters.

The particle amplitude, in centimeters, is

$$\xi + - \frac{A}{c} \cos k(ct - x) \qquad 1.24$$

From equations 1.20 and 1.24 the condensation is

$$s = - \frac{\partial \xi}{\partial x} \qquad 1.25$$

1.5. Spherical Sound Waves.—Many acoustical problems are concerned with spherical diverging waves. In spherical coordinates $x = r \sin \theta \cos \psi$, $y = r \sin \theta \sin \psi$ and $z = r \cos \theta$ where r is the distance from the center, θ is the angle between r and the oz axis and ψ is the angle between the projection of r on the xy plane and ox. Then $\nabla^2 \theta$ becomes

$$\nabla^2 \phi = \frac{\partial^2 \phi}{\partial r^2} + \frac{2}{r} \frac{\partial \phi}{\partial r} + \frac{1}{r^2 \sin \theta} \frac{\partial}{\partial \theta} (\sin \theta) \frac{\partial \phi}{\partial \theta} + \frac{1}{r^2 \sin^2 \theta} \frac{\partial^2 \phi}{\partial \psi^2} \qquad 1.26$$

For spherical symmetry about the origin

$$\nabla^2 \phi = \frac{\partial^2}{r \partial r^2} (r\phi) \qquad 1.27$$

The general wave equation then becomes,

$$\frac{\partial^2}{\partial t^2} (r\phi) = c^2 \frac{\partial^2}{\partial r^2} (r\phi) \qquad 1.28$$

The wave equation for symmetrical spherical waves can be derived in another way. Consider the flux across the inner and outer surfaces of the spherical shell having radii of $r - \Delta r/2$ and $r + \Delta r/2$, the difference is

$$- 4\pi \frac{\partial}{\partial r} \left(\rho' r^2 \frac{\partial r}{\partial t} \right) \Delta r \qquad 1.29$$

The velocity is

$$\frac{\partial r}{\partial t} = \frac{\partial \phi}{\partial r} \qquad 1.30$$

where ϕ = velocity potential.

The expression 1.29 employing equation 1.30 becomes

$$- 4\pi \frac{\partial}{\partial r} \left(\rho' r^2 \frac{\partial \phi}{\partial r} \right) \Delta r \qquad 1.31$$

The rate of growth of mass in the shell is

$$4\pi r^2 \frac{\partial \rho'}{\partial t} \Delta r \qquad 1.32$$

The difference in flux must be equal to the rate of growth of mass, expressions 1.31 and 1.32,

$$r^2 \frac{\partial \rho'}{\partial t} + \frac{\partial}{\partial r} \left(\rho' r^2 \frac{\partial \phi}{\partial r} \right) = 0 \qquad 1.33$$

Using equations 1.6, 1.9, and 1.13, equation 1.33 may be written,

$$r^2 \frac{\partial^2 \phi}{\partial t^2} - c^2 \frac{\partial}{\partial r}\left(r^2 \frac{\partial \phi}{\partial r}\right) = 0 \qquad 1.34$$

Equation 1.34 may be written

$$\frac{\partial^2 (r\phi)}{\partial t^2} - c^2 \frac{\partial^2 (r\phi)}{\partial r^2} = 0 \qquad 1.35$$

which is the same as equation 1.28. The solution of equation 1.35 for diverging waves is

$$\phi = \frac{A}{r} \, \epsilon^{jk(ct-r)} \qquad 1.36$$

From equations 1.19 and 1.36 the condensation is given by

$$s = -\frac{1}{c^2} \frac{\partial \phi}{\partial t} = -\frac{jkA}{cr} \, \epsilon^{jk(ct-r)} \qquad 1.37$$
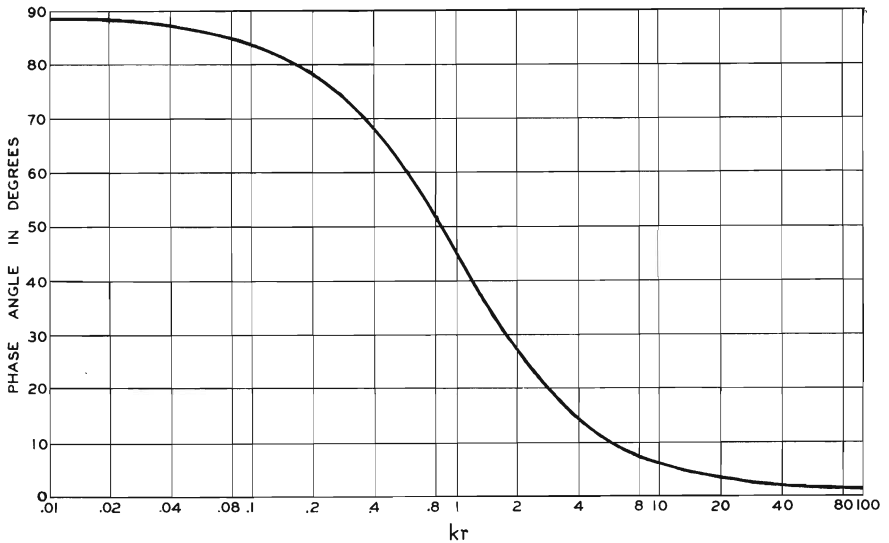
FIG. 1.2.   Phase angle between the pressure and particle velocity in a spherical sound wave in terms of kr. where $k = 2\pi/\lambda$, λ = wavelength and r = distance from the source.

A. *Pressure in a Spherical Sound Wave.*—The pressure from equation 1.21 is

$$p = c^2 \rho s \qquad 1.38$$

The pressure then from equations 1.37 and 1.38 is

$$p = -\frac{jkcA\rho}{r} \, \epsilon^{jk(ct-r)} \qquad 1.39$$

Retaining the real part of equation 1.39 the pressure is

$$p = \rho \frac{kcA}{r} \sin k(ct - r) \qquad 1.40$$

B. *Particle Velocity in a Spherical Sound Wave.*—The particle velocity from equations 1.11 and 1.36 is

$$u = -\left(\frac{1}{r} + jk\right)\frac{A}{r} \epsilon^{jk(ct-r)} \qquad 1.41$$

Retaining the real part of equation 1.41 the particle velocity is

$$u = -\frac{Ak}{r}\left[\frac{1}{kr} \cos k(ct - r) - \sin k(ct - r)\right] \qquad 1.42$$

C. *Phase Angle between the Pressure and the Particle Velocity in a Spherical Sound Wave.*—The particle velocity given by equation 1.42 may be written

$$u = \frac{A}{r} \sqrt{\frac{1}{r^2} + k^2} \sin [k(ct - r) - \theta] \qquad 1.43$$

where $\tan \theta = 1/kr$.
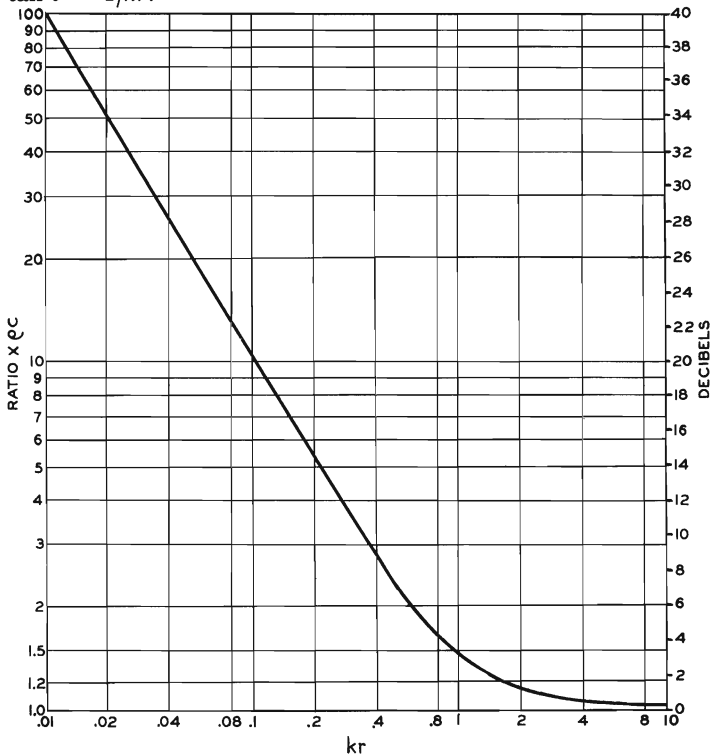
FIG. 1.3. Ratio of the absolute magnitude of the particle velocity to the pressure in a spherical sound wave in terms of kr, where $k = 2\pi/\lambda$, λ = wavelength and r = distance from the source.

Comparing equation 1.43 with equation 1.40 for the pressure it will be seen that the phase angle between the pressure and velocity in a spherical wave is given by

$$\theta = \tan^{-1} \frac{1}{kr} \qquad\qquad 1.44$$

For very large values of kr, that is, plane waves, the pressure and particle velocity are in phase. The phase angle as a function of kr is depicted in Fig. 1.2.

D. *Ratio of the Absolute Magnitudes of the Particle Velocity and the Pressure in a Spherical Sound Wave.*—From equations 1.40 and 1.43 the ratio of the absolute value of the particle velocity to the absolute value of the pressure is given by

$$\text{Ratio} = \frac{\sqrt{1 + k^2 r^2}}{\rho c k r} \qquad\qquad 1.45$$

The ratio in equation 1.45, as a function of kr, is depicted in Fig. 1.3.

1.6. Stationary Sound Waves.—Stationary waves are the wave system resulting from the interference of waves of the same frequencies and are characterized by the existence of nodes or partial nodes.

Consider two plane waves of equal amplitude traveling in opposite directions; the velocity potential may be expressed as

$$\phi = A \left[\cos k(ct - x) + \cos k(ct + x) \right] \qquad\qquad 1.46$$

The pressure in this wave system from equations 1.19 and 1.21 is

$$p = - \rho \frac{\partial \phi}{\partial t} = kc\rho A \left[\sin k(ct - x) + \sin k(ct + x) \right] \qquad 1.47$$

$$p = 2kc\rho A \left[\sin kct \cos kx \right] \qquad\qquad 1.48$$

The particle velocity in this wave system from equations 1.11 and 1.46 is

$$u = \frac{\partial \phi}{\partial x} = kA \left[\sin k(ct - x) - \sin k(ct + x) \right] \qquad 1.49$$

$$u = -2kA \left[\cos kct \sin kx \right] \qquad\qquad 1.50$$

$$u = 2kA \left[\sin \left(kct - \frac{\pi}{2} \right) \cos \left(kx - \frac{\pi}{2} \right) \right] \qquad 1.51$$

Equations 1.48 and 1.51 show that the maxima of the particle velocity and pressure are separated by a quarter wavelength. The maxima of p and u differ by 90° in time phase.

A stationary wave system is produced by the reflection of a plane wave by an infinite wall normal to the direction of propagation. This is the simplest type of standing wave system. Complex stationary wave systems are produced when a sound source operates in a room due to the reflections from the walls, ceiling, and floor.

1.7. Sound Energy Density.—Sound energy density is the sound energy per unit volume. The unit is the erg per cubic centimeter.

The sound energy density in a plane wave is

$$E = \frac{p^2}{\rho c^2} \qquad \qquad 1.52$$

where p = sound pressure, in dynes per square centimeter,
ρ = density, in grams per cubic centimeter, and
c = velocity of sound, in centimeters per second.

The positive radiation pressure in dynes per square centimeter exerted by sound waves upon an infinite wall is

$$p = (\gamma + 1)E \qquad \qquad 1.53$$

where E = energy density of the incident wave train in ergs per cubic centimeter, and
γ = ratio of specific heats, 1.4 for air.

Instruments for measuring the radiation pressure have been built, consisting of a light piston mounted in a large wall with means for measuring the force on the piston. Since the radiation pressure is very small these instruments must be quite delicate.

1.8. Sound Intensity.—The sound intensity of a sound field in a specified direction at a point is the sound energy transmitted per unit of time in the specified direction through a unit area normal to this direction at the point. The unit is the erg per second per square centimeter. It may also be expressed in watts per square centimeter.

The intensity, in ergs per second per square centimeter, of a plane wave is

$$I = \frac{p^2}{\rho c} = pu = \rho c u^2 \qquad \qquad 1.54$$

where p = pressure, in dynes per square centimeter,
u = particle velocity, in centimeters per second,
c = velocity of propagation, in centimeters per second, and
ρ = density of the medium, in grams per cubic centimeter.

The product ρc is termed the specific acoustical resistance of the medium. The specific acoustical resistance of various mediums is shown in Table 1.1.

1.9. Decibels (Bels).—In acoustics the ranges of intensities, pressures, etc., are so large that it is convenient to use a scale of smaller numbers termed decibels. The abbreviation db is used for the term decibel. The bel is the fundamental division of a logarithmic scale for expressing the ratio of two amounts of power, the number of bels denoting such a ratio being the logarithm to the base ten of this ratio. The decibel is one tenth

of a bel. For example, with P_1 and P_2 designating two amounts of power and n the number of decibels denoting their ratio:

$$n = 10 \log_{10} \frac{P_1}{P_2}, \text{ decibels} \qquad 1.55$$

When the conditions are such that ratios of currents or ratios of voltages (or the analogous quantities such as pressures, volume currents, forces, and particle velocities) are the square roots of the corresponding power ratios, the number of decibels by which the corresponding powers differ is expressed by the following formulas:

$$n = 20 \log_{10} \frac{i_1}{i_2}, \text{ decibels} \qquad 1.56$$

$$n = 20 \log_{10} \frac{e_1}{e_2}, \text{ decibels} \qquad 1.57$$

where i_1/i_2 and e_1/e_2 are the given current and voltage ratios, respectively.

For relation between decibels and power and current or voltage ratios, see Table 1.2.

TABLE 1.2. THE RELATION BETWEEN DECIBELS AND POWER AND CURRENT OR VOLTAGE RATIOS

Power Ratio	Decibels	Current or Voltage Ratio	Decibels
1	0	1	0
2	3.0	2	6.0
3	4.8	3	9.5
4	6.0	4	12.0
5	7.0	5	14.0
6	7.8	6	15.6
7	8.5	7	16.9
8	9.0	8	18.1
9	9.5	9	19.1
10	10	10	20
100	20	100	40
1000	30	1000	60
10,000	40	10,000	80
100,000	50	100,000	100
1,000,000	60	1,000,000	120

1.10. Doppler Effect.[1]—The change in pitch of a sound due to the relative motion of the source and observer is termed the Doppler Effect. When the source and observer are approaching each other the pitch observed by the listener is higher than the actual frequency of the sound source. If the source and observer are receding from each other the pitch is lower.

The frequency at the observation point is

$$f_0 = \frac{v - v_0}{v - v_s} f_s \qquad 1.58$$

where v = velocity of sound in the medium,
$\quad v_0$ = velocity of the observer,
$\quad v_s$ = velocity of the source, and
$\quad f_s$ = frequency of the source.

All the velocities must be in the same units.

No account is taken of the effect of wind velocity or motion of the medium in equation 1.58. In order to bring in the effect of the wind, the velocity v in the medium must be replaced by $v + w$ where w is the wind velocity in the direction in which the sound is traveling. Making this substitution in 1.58 the result is

$$f_0 = \frac{v + w - v_0}{v + w - v_s} f_s \qquad 1.59$$

Equation 1.59 shows that the wind does not produce any change in pitch unless there is some relative motion of the sound source and the observer.

1.11. Refraction and Diffraction.—The change in direction of propagation of sound, produced by a change in the nature of the medium which affects the velocity, is termed refraction. Sound is refracted when the density varies over the wave front (see equation 1.15). A sound wave may be bent either downward or upward depending upon the relative temperatures (densities) of the air,[2] Fig. 1.4. The distance over which sound may be heard is greater when the wave is bent downward than when it is bent upward. The first condition usually obtains during the early morning hours while the latter condition prevails during the day.

Structures[3] may be built which refract sound waves. Acoustic lenses and prisms employing these structures may be used for various acoustical applications, as for example, loudspeakers and microphones. See Secs.

---

[1] Perrine, J. O., *Amer. Jour. Phys.*, Vol. 12, No. 1, p. 23, 1944. This paper describes sixteen versions of the Doppler and Doppler Echo Effects. In addition to systems given in the text above are systems involving moving and fixed reflectors.

[2] For other phenomena of atmospheric acoustics such as the effects of wind and temperature upon the propagation of sound waves and the applications to sound ranging and signaling in air, see Stewart and Lindsay, " Acoustics," D. Van Nostrand Company, Princeton, N.J., 1930.

[3] Koch and Harvey, *Jour. Acous. Soc. Amer.*, Vol. 21, No. 5, p. 471, 1949.

COOL AIR — LOW VELOCITY

WAVE FRONTS

WARM AIR — HIGH VELOCITY

SURFACE OF THE EARTH

WARM AIR — HIGH VELOCITY

WAVE FRONTS

COOL AIR — LOW VELOCITY

SURFACE OF THE EARTH
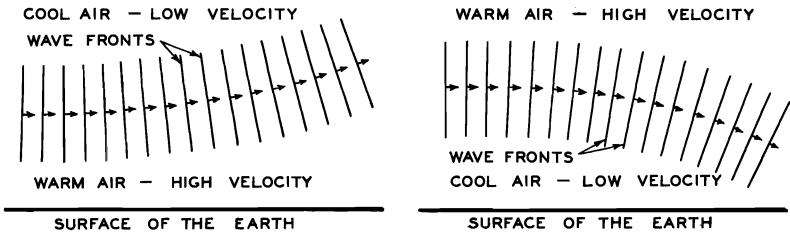
FIG. 1.4. The refraction of a sound wave in air.

WAVE

SPHERES

$2a$

WAVE

$\ell \rightarrow$ WAVE

ℓ_o

PARALLEL PLATES

WAVE

DISKS

$2c$

WAVE

$\ell \rightarrow$ WAVE

ℓ_o

SERPENTINE PLATES

WAVE

STRIPS

$2b$

WAVE

θ

$\ell \rightarrow$ WAVE

ℓ_o

SLANT PLATES
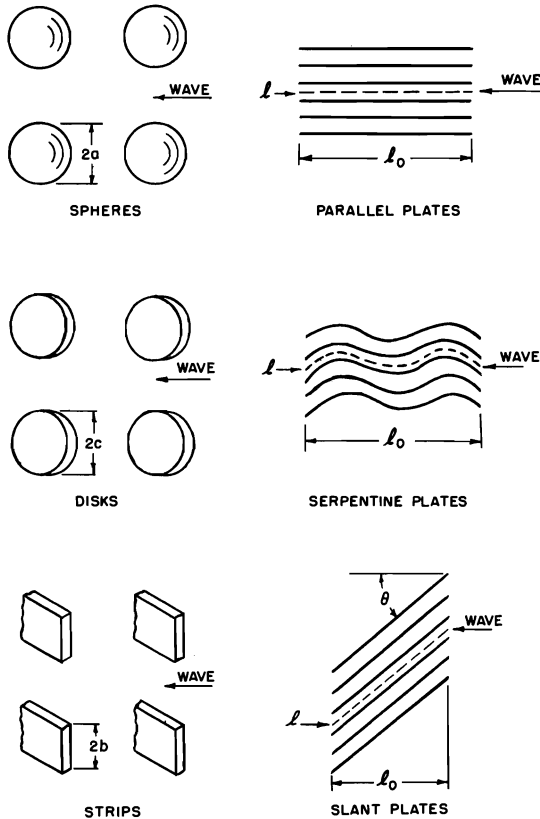
FIG. 1.5. Obstacle and path length structures for refracting sound waves.

6.15 and 8.6B. Practical systems have been developed based upon obstacle arrays and path length devices.

Obstacle arrays increase the effective density of the medium and thus produce a reduced propagation velocity of sound waves passing through the array. Three different obstacle arrays are shown in Fig. 1.5.

The index of refraction n, of a spherical obstacle array as shown in Fig. 1.5, is given by

$$n^2 = 1 + \tfrac{2}{3}\pi a^3 N \qquad\qquad 1.60$$

where a = radius of the sphere, and
N = number of spheres per unit volume.

The index of refraction n, of a disk obstacle array as shown in Fig. 1.5, is given by

$$n^2 = 1 + \tfrac{8}{3}c^3 N \qquad\qquad 1.61$$

where c = radius of the disk, and
N = number of disks per unit volume.

The index of refraction n, of a series of strips as shown in Fig. 1.5, is given by

$$n^2 = 1 + \pi b^2 N \qquad\qquad 1.62$$

where b = half breadth of the strip normal to the direction of propagation of the wave and
N = number of strips per unit area viewed endwise.

Path length devices increase the time of travel of the waves through the path over that in free space.   Three different path length devices are shown in Fig. 1.5.

The index of refraction n, of parallel plates as shown in Fig. 1.5, is unity.

The index of refraction n, of the serpentine plates as shown in Fig. 1.5, is given by

$$n = \frac{l}{l_0} \qquad\qquad 1.63$$

where l = path length through the plates, and
l_0 = path in the absence of the plates.

The index of refraction n, of slant plates as shown in Fig. 1.5, is given by

$$n = \frac{l}{l_0} = \frac{1}{\cos\theta} \qquad\qquad 1.64$$

where θ = angle between the direction of propagation of the wave and the plane of the plates.

In the examples in this book the serpentine system will be used.

An acoustic lens which converges the impinging sound wave is shown in Fig. 1.6A.   The sheet metal is arranged so that the path length through the lens is the greatest at the center of the lens.   The action of the lens is depicted by the ray and wavefront diagram of Fig. 1.6A.   The path lengths of all the pencils of the incident sound wave are all the same at focus.   An acoustic lens which diverges the incident sound wave is shown in Fig. 1.6B. The action of the lens is depicted by the ray diagram and wavefront diagram of Fig. 1.6B.
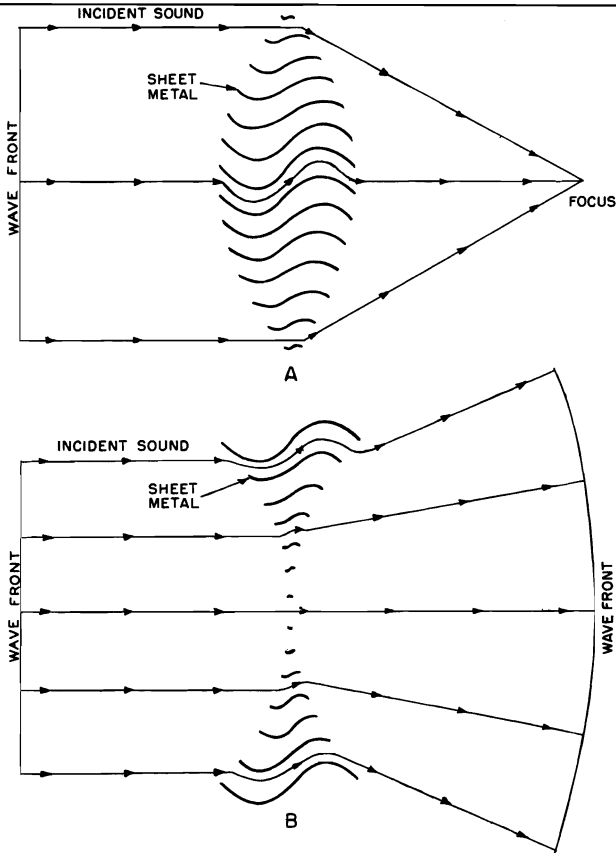
FIG. 1.6.   Acoustic lenses.   A. Converging lens.   B. Diverging lens.

An acoustic prism is shown in Fig. 1.7.   The acoustic prism changes the direction of the impinging sound wave.   The action of the prism is depicted by the ray and wavefront diagrams of Fig. 1.7.

Diffraction is the change in direction of propagation of sound due to the passage of sound around an obstacle.   It is well known that sound will travel around an obstacle.   The larger the ratio of the wavelength to the dimensions of the obstacle the greater the diffraction.   The diffraction around the head is important in both speaking and listening. The diffraction of sound by microphones and loudspeakers is important in the performance of these instruments.   The diffraction[4] of sound by a sphere, a cube, and a cylinder as a function of the dimensions is shown in Fig. 1.8.   These data may be used to predict the diffraction of sound by objects of these general shapes.   As, for example, the sphere may be used to predict the diffraction of sound by the human head.

[4] Muller, Black, and Dunn, *Jour. Acous. Soc. Amer.*, Vol. 10, No. 1, p. 6, 1938.
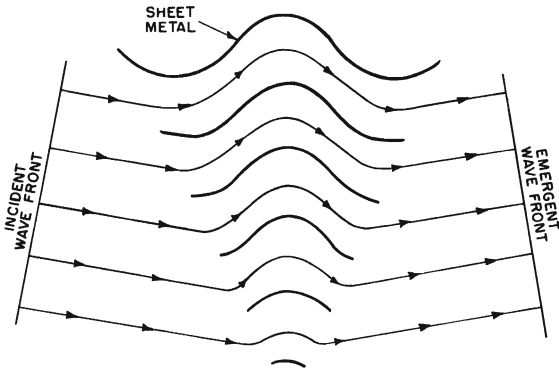
FIG. 1.7.   Acoustic prism.
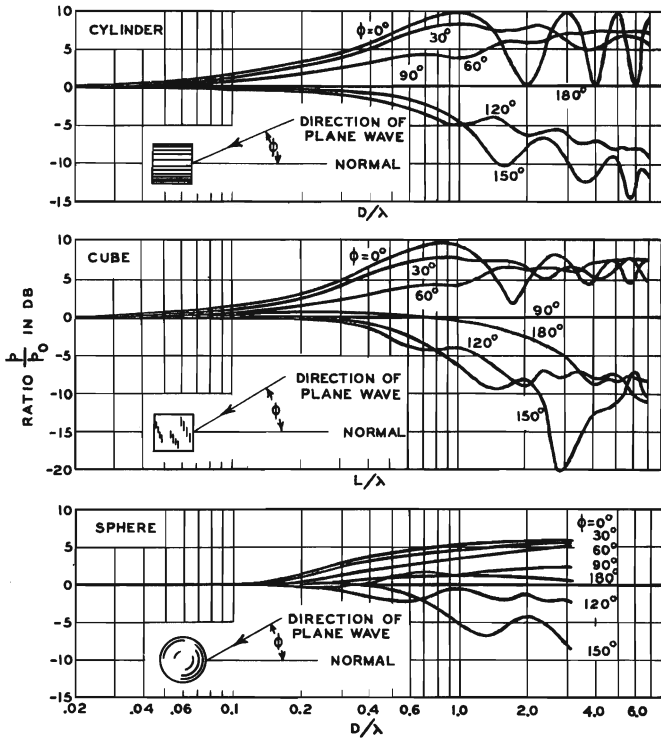
FIG. 1.8.   The diffraction of a sound wave by a cylinder, cube, and sphere.   (After Muller, Black and Dunn.)

There are other shapes[5] besides the cylinder, cube, and sphere that are used for microphone and loudspeaker enclosures. In order to provide additional information, the diffraction of sound by the shapes shown in Fig. 1.9 were obtained experimentally. The dimensions of the ten different
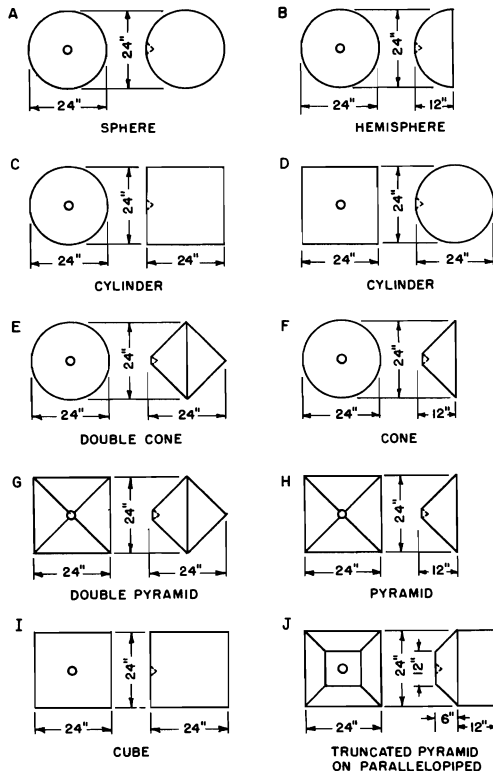
FIG. 1.9. Structures used in sound diffraction studies.

enclosures are shown in Fig. 1.9. The experimentally-determined diffraction of a sound wave by these different enclosures was obtained by comparing the response of a small loudspeaker in free space with the response of the loudspeaker mounted in the enclosures in the position shown in Fig. 1.9. The diameter of the diaphragm of the cone used in the loudspeaker was $\frac{7}{8}$ inch. Since the upper frequency limit of the response was made 4000 cycles, the diameter of the cone is less than one-quarter wavelength. In other words the source is for all practical purposes nondirectional. The diffraction characteristics for the ten shapes are shown in Fig. 1.10. The response frequency characteristics shown in Fig. 1.10 are for the dimensions shown in Fig. 1.9. The response frequency characteristics

[5] Olson, H. F., *Audio Eng.*, Vol. 35, No. 11, p. 34, 1951.
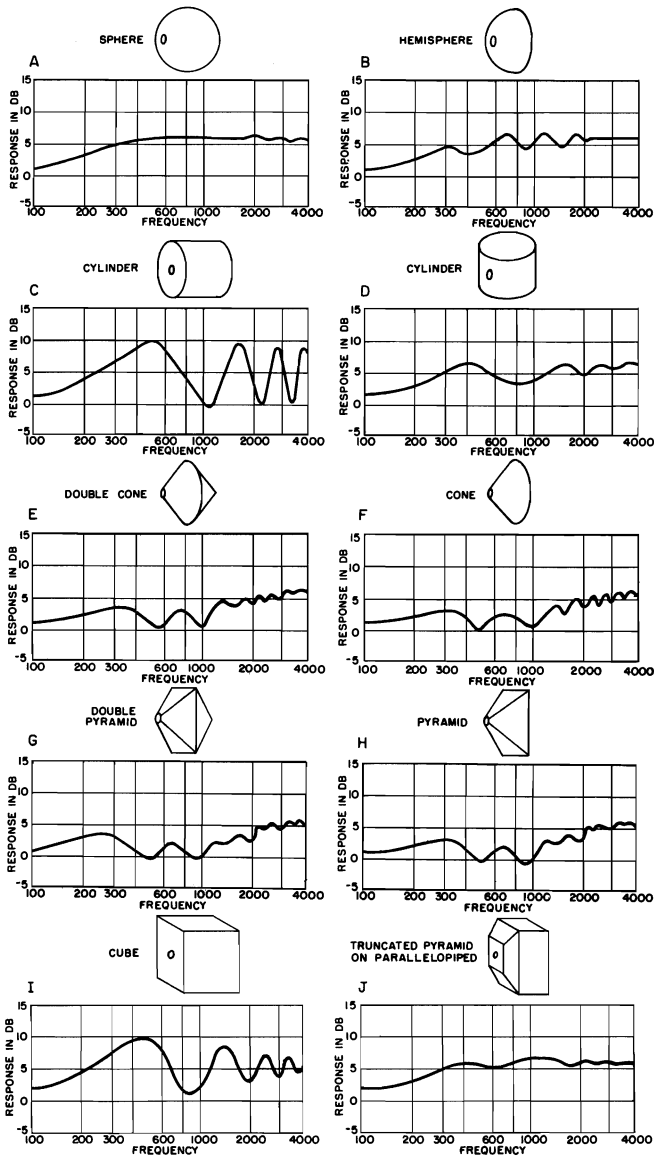
FIG. 1.10. Response frequency characteristics depicting the diffraction of sound by ten objects of different shapes. The dimensions of the objects are given in Fig. 1.9.

for enclosures of other dimensions can be obtained by multiplying the ratio of the linear dimensions of the enclosure of Fig. 1.9 to the linear dimension of the enclosure under consideration by the frequency of Fig. 1.10.

See Sec. 1.13. For example, if the linear dimensions of the new enclosure are two times that of Fig. 1.9, the frequency scale of Fig. 1.10 should be multiplied by one-half.

Another example of diffraction of sound is illustrated by the zone plate shown in Fig. 1.11. The path lengths of the sound from the source to the focus vary by an integral wavelength. As a consequence, all the pencils of sound are in phase at the focus with the result that the sound pressure is considerably greater at this point than any other position behind the zone plate.

FRONT VIEW   CROSS-SECTIONAL VIEW
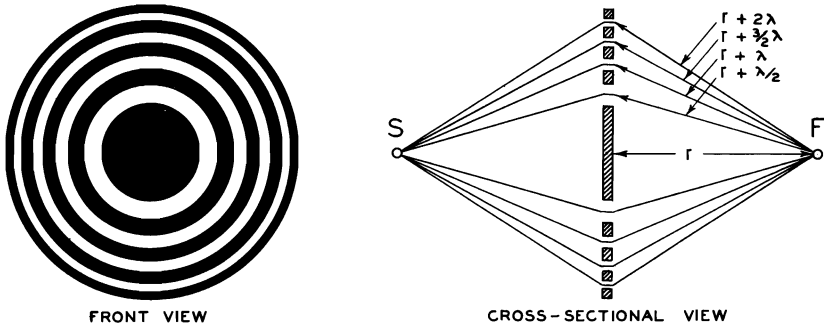
FIG. 1.11. Zone plate. The source S and the focus F are equidistant from the zone plate.

1.12. Acoustical Reciprocity Theorem.[6,7,8,9]—The acoustical reciprocity theorem, as developed by Helmholtz, states: If in a space filled with air which is partly bounded by finitely extended bodies and is partly unbounded, sound waves may be excited at a point A, the resulting velocity potential at a second point B is the same in magnitude and phase as it would have been at A had B been the source of sound. It is the purpose of this section to derive the acoustical reciprocity theorem.

Consider two independent sets of pressures p' and p'' and particle velocities v' and v''. Multiply equation 1.4 by the p and v of the other set.

$$v'' \frac{dv'}{dt} - v' \frac{dv''}{dt} + \frac{1}{\rho} v'' \operatorname{grad} p_0' - \frac{1}{\rho} v' \operatorname{grad} p_0'' = 0 \qquad 1.65$$

If p and v vary as a harmonic of the time, equation 1.65 becomes

$$\frac{1}{\rho} v'' \operatorname{grad} p_0' - \frac{1}{\rho} v' \operatorname{grad} p_0'' = 0 \qquad 1.66$$

There is the following relation:

$$v \operatorname{grad} p = \operatorname{div} vp - p \operatorname{div} v \qquad 1.67$$

[6] Rayleigh, "Theory of Sound," Macmillan and Company, London, 1926.
[7] Ballentine, S., *Proc.*, I.R.E., Vol. 17, No. 6, p. 929, 1929.
[8] Olson, H. F., *RCA Review*, Vol. 6, No. 1, p. 36, 1941.
[9] Olson, "Dynamical Analogies," D. Van Nostrand Company, Princeton, N.J., 1943.

From equations 1.9 and 1.10

$$\frac{1}{\gamma p_0} \frac{\partial p}{\partial t} + \text{div } v = 0 \qquad 1.68$$

From equations 1.66, 1.67, and 1.68,

$$\text{div } (v'' \, p' - v' \, p'') = 0 \qquad 1.69$$

The relation of equation 1.69 is for a point. Integration of equation 1.69 over a region of space gives

$$\iint (v''p' - v'p'')ds = 0 \qquad 1.70$$

If, in an acoustical system comprising a medium of uniform density and propagating irrotational vibrations of small amplitude, a pressure p' produces a particle velocity v' and a pressure p'' produces a particle velocity v'', then

$$\iint (v''p' - v'p'')_n ds = 0 \qquad 1.71$$

where the surface integral is taken over the boundaries of the volume.

In the simple case in which there are only two pressures, as illustrated in the free field acoustical system of Fig. 1.12, equation 1.71 becomes

$$p'v'' = p''v' \qquad 1.72$$

where p', p'' and v', v'' are the pressures and particle velocities depicted in the free field acoustical system of Fig. 1.12.

FIELD   LUMPED CONSTANTS
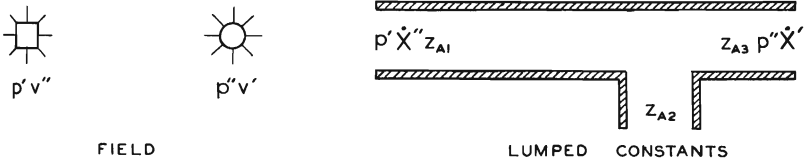
FIG. 1.12. Reciprocity in field and lumped constant acoustical systems.

The above theorem is applicable to all acoustical problems. However, it can be restricted to lumped constants[10] as follows: In an acoustical system composed of inertance, acoustical capacitance, and acoustical resistance, let a set of pressures p_1', p_2', $p_3' \ldots p_n'$, all harmonic of the same frequency acting in n points in the system, produce a volume current distribution $\dot{X}_1' \, \dot{X}_2' \, \dot{X}_3' \ldots \dot{X}_n'$, and let a second set of pressures p_1'', p_2'', $p_3'' \ldots P_n''$, of the same frequency as the first, produce a second volume current distribution $\dot{X}_1''$, $\dot{X}_2''$, $\dot{X}_3'' \ldots \dot{X}_n''$. Then

$$\sum_{j=1}^{n} p_j' \dot{X}_j'' = \sum_{j=1}^{n} p_j'' \dot{X}_j' \qquad 1.73$$

[10] Olson, "Dynamical Analogies," D. Van Nostrand Company, Princeton, N.J., 1943.

This theorem is valid provided the acoustical system is invariable, contains no internal source of energy or unilateral device, linearity in the relations between pressures and volume currents, and complete reversibility in the elements and provided the applied pressures p_1, p_2, $p_3 \ldots p_n$ are all of the same frequency.

In the simple case in which there are only two pressures, as illustrated in the acoustical system of lumped constants in Fig. 1.7, equation 1.73 becomes

$$p\dot{X}''' = p''\dot{X}' \qquad 1.74$$

where p', p'' and $\dot{X}'$, $\dot{X}''$ are the pressures and volume currents depicted in the acoustical system of lumped constants in Fig. 1.12.

1.13. Acoustical Principle of Similarity.[11]—The principle of similarity in acoustics states: For any acoustical system involving diffraction phenomena it is possible to construct a new system on a different scale, which will exhibit similar performance, providing the wavelength of the sound is altered in the same ratio as the linear dimensions of the new system.

The principle of similarity is useful in predicting the performance of similar acoustical systems from a single model. A small model can be built and tested at very high frequencies to predict the performance of similar large systems at lower frequencies. For example: in the diffraction of sound by objects, if the ratio of the linear dimensions of the two objects is $X : 1$, the corresponding configuration of the frequency characteristics will be displaced $1 : X$ in frequency. This is illustrated in Figs. 1.8 and 1.10. Other examples are the directional characteristics of various sound sources Figs. 2.3 to 2.23 inclusive, the air load upon a diaphragm, Fig. 5.2, etc.

1.14. Longitudinal Waves in a Rod.—The preceding considerations have been concerned with sound waves in gases and fluids. In the case of solids, longitudinal waves in rods are of practical interest in many applications. It is the purpose of this section to derive the equations for longitudinal sound waves in a rod of homogeneous material and constant cross section.

The longitudinal axis of the bar will be assumed to coincide with the x axis. Consider an element of the bar δx, determined by two planes perpendicular to x and initially at distances x and $x + \delta x$ from $x = 0$. Assume that the planes are displaced by distances ξ and $\xi + \delta \xi$. The distance between the planes is now

$$\delta x + \delta \xi = \delta x + \frac{\partial \xi}{\partial x}\, \delta x \qquad 1.75$$

The increase in distance between the planes is $\dfrac{\partial \xi}{\partial x}\, \delta x$.

The increase in length of the bar per unit length at this point is $\dfrac{\partial \xi}{\partial x}$.

[11] Olson, H. F., *RCA Review*, Vol. 6, No. 1, p. 36, 1941.

Young's modulus is defined as the ratio of the longitudinal stress to the corresponding extension. At the first face of the element Young's modulus is

$$Q = \frac{\frac{F}{S}}{\frac{\partial \xi}{\partial x}} \tag{1.76}$$

where Q = Young's modulus, in dynes per square centimeter,
    F = force, in dynes,
    S = cross-sectional area of the rod, in square centimeters, and
    $\frac{\partial \xi}{\partial x}$ = extension.

The force acting on the element across the first face is

$$F = QS \frac{\partial \xi}{\partial x} \tag{1.77}$$

The force acting across the second face of the element is

$$F + \delta F = QS \frac{\partial \xi}{\partial x} + \frac{\partial}{\partial x}\left(QS \frac{\partial \xi}{\partial x}\right) \delta x \tag{1.78}$$

$$= QS \frac{\partial \xi}{\partial x} + QS \frac{\partial^2 \xi}{\partial x^2} \delta x \tag{1.79}$$

The resultant force on the element is

$$\delta F = QS \frac{\partial^2 \xi}{\partial x^2} \delta x \tag{1.80}$$

The acceleration of momentum of the element is

$$S \rho \delta x \frac{\partial^2 \xi}{\partial t^2} \tag{1.81}$$

where ρ = density, in grams per cubic centimeter.

Equating the resultant force on the element to the acceleration of momentum, the result is

$$\frac{\partial^2 \xi}{\partial t^2} = \frac{Q}{\rho} \frac{\partial^2 \xi}{\partial x^2} \tag{1.82}$$

This is the wave equation for ξ. Equation 1.82 is analogous to equation 1.16 for plane waves in a gas and the solution of the differential equation is similar. The velocity of propagation, in centimeters per second, of longitudinal waves is in a rod

$$c = \sqrt{\frac{Q}{\rho}} \tag{1.83}$$

where Q = Young's modulus, in dynes per square centimeter (see Table 1.1), and
    ρ = density, in grams per cubic centimeter (see Table 1.1).

The velocity of sound, Young's modulus and the density for various solids are given in Table 1.1.

1.15. Torsional Waves in a Rod.—A rod may be twisted about an axis of the rod in such a manner that each transverse section remains in its own plane. If the section is not circular there will be motion parallel to the axis of the bar. For a circular cross section and a homogeneous bar the equations of motion are analogous to those of longitudinal waves in the rod. The velocity of propagation, in centimeters per second, of torsional waves in a rod, is

$$c = \sqrt{\frac{Q}{2\rho\,(\sigma + 1)}} \qquad 1.84$$

where Q = Young's modulus, in dynes per square centimeter (see Table 1.1),

ρ = density, in grams per cubic centimeter (see Table 1.1), and

σ = Poisson's ratio (see Table 1.1).

1.16. Cylindrical Sound Waves.[12]—From a practical standpoint, the important waves in acoustics are plane and spherical waves. However, it may be interesting as an addition in the chapter on sound waves to indicate some of the characteristics of cylindrical sound waves.

The consideration will be the sound pressure and particle velocity produced by a long cylinder expanding and contracting radially with a velocity u_0, in centimeters per second, given by

$$u_0 = U_0 \epsilon^{jwt} \qquad 1.85$$

where U_0 = maximum velocity of the vibration in centimeters per second,

$\omega = 2\pi f$,

f = frequency, in cycles per second, and

t = time, in seconds.

When the radius of the cylinder is small compared to the wavelength, the sound pressure p, in dynes per square centimeter, at a distance large compared to the radius of the cylinder may be expressed as

$$p = \pi\rho a U_0 \sqrt{\frac{cf}{r}}\, \epsilon^{jk(r-ct)-j\pi/4} \qquad 1.86$$

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

c = velocity of sound in the medium, in centimeters per second,

$k = 2\pi/\lambda$

λ = wavelength, in centimeters,

a = radius of the cylinder, in centimeters, and

r = distance from the axis of the cylinder.

[12] Morse, " Vibration and Sound," McGraw-Hill Book Company, New York, N.Y., 1948.

The particle velocity u, in centimeters per second, under the same conditions is given by

$$u = \pi a U_0 \sqrt{\frac{f}{cr}} \; \epsilon^{jk(r-ct)-j\pi/4} \qquad\qquad 1.87$$

It will be seen that the pressure and particle velocity decrease inversely as the square root of the distance from the cylinder.

The product of the pressure and the particle velocity gives the flow of energy per square centimeter as follows,

$$P = \tfrac{1}{2}\pi \rho a^2 U_0^2 \frac{f}{r} \qquad\qquad 1.88$$

It will be seen that the intensity falls off inversely as the distance.

2

ACOUSTICAL RADIATING SYSTEMS

2.1. Introduction.—There are almost an infinite number of different types of sound sources. The most common of these are the human voice, musical instruments, machinery noises, and loudspeakers. The most important factors which characterize a sound source are the directional pattern, the radiation efficiency, and the output as a function of the frequency. In the case of some sound sources as, for example, musical instruments, it is almost impossible to analyze the action. However, in the case of most sound reproducers the action may be predicted with amazing accuracy. It is the purpose of this chapter to consider some of the simple sound sources that are applicable to the problems of sound reproduction.

2.2. Simple Point Source.—A point source is a small source which alternately injects fluid into a medium and withdraws it.

A. *Point Source Radiating into an Infinite Medium. Solid Angle of 4π Steradians.*—Consider a point source having a maximum rate of fluid emission of $4\pi A$ cubic centimeters per second. The momentary rate at a time t is $4\pi A \cos \omega t$. The maximum rate of fluid emission may be written

$$- 4\pi A = S\dot{\xi}_0 \qquad 2.1$$

where S = area of the surface of the source, in square centimeters, and
    ξ_0 = maximum velocity, in centimeters per second over the surface S.

The velocity potential of a point source from equation 1.36 is

$$\phi_r = \frac{A}{r} \, \epsilon^{jk(ct-r)} \qquad 2.2$$

The particle velocity at a distance r from equation 1.42 is

$$u = - \frac{Ak}{r} \left[\frac{1}{kr} \cos k \, (ct - r) - \sin k(ct - r) \right] \qquad 2.3$$

The pressure at a distance r from equation 1.40 is

$$p = \frac{\rho k c A}{r} \sin k(ct - r) \qquad 2.4$$

30

The intensity or average power, in ergs per second, transmitted through a unit area at a distance r, in centimeters, is the product of p and u and is given by

$$P = \frac{\rho c k^2 A^2}{2r^2} \qquad 2.5$$

The total average power in ergs per second emitted by the source is

$$P_T = 2\pi \rho c k^2 A^2 \qquad 2.6$$

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

c = velocity of sound, in centimeters per second,

$k = 2\pi/\lambda$,

λ = wavelength, in centimeters, and

A is defined by equation 2.1.

B. *Point Source Radiating into a Semi-Infinite Medium. Solid Angle of 2π Steradians.*—The above example considered a point source operating in an infinite medium. The next problem of interest is that of a point source operating in a semi-infinite medium, for example, a point source near an infinite wall.

In this case we can employ the principle of images as shown in Fig. 2.1. The pressure, assuming the same distance from the source, is two times that of the infinite medium. The particle velocity is also two times that of the infinite medium. The average power transmitted through a unit area is four times that of the infinite medium. The average power output of the source, however, is two times that of a simple source operating in an infinite medium.

C. *Point Source Radiating into a Solid Angle of π Steradians.*—Employing the method of images Fig. 2.1 the pressure is four times, the particle velocity is four times, and the average power transmitted through a unit area is sixteen times that of an infinite medium for the same distance. The average power output of the source is four times that of a simple source operating in an infinite medium.

D. *Point Source Radiating into a Solid Angle of $\pi/2$ Steradians.*—Employing the method of images, Fig. 2.1, the pressure is eight times, the particle velocity eight times, and the average power transmitted through a unit area is sixty-four times that of the same source operating in an infinite medium at the same distance. The average power output is eight times that of the same simple source operating in an infinite medium.

E. *Application of the Simple Source.*—The above data may be applied to acoustic radiators in which the dimensions are small compared to the wavelength and located close to the boundaries indicated above. For example, A would correspond to a loudspeaker, which acts as a simple source, suspended in space at a large distance from any walls or boundaries. B would correspond to a loudspeaker placed on the floor in the center of the room. C would correspond to a loudspeaker placed on the floor along a wall, and D would correspond to a loudspeaker placed in the corner of

the room. Of course, as pointed out above, these examples hold only when the dimensions of the radiator and the distance from the wall are small compared to the wavelength.

	SOLID ANGLE OF SOUND EMISSION	PRESSURE AT A DISTANCE r	POWER OUTPUT	ENERGY DENSITY DISTANCE r
SOURCE	4π	p	W	I
IMAGE SOURCE	2π	2p	2W	4I
SOURCE / IMAGES	π	4p	4W	16I
SOURCE / IMAGES	π/2	8p	8W	64I
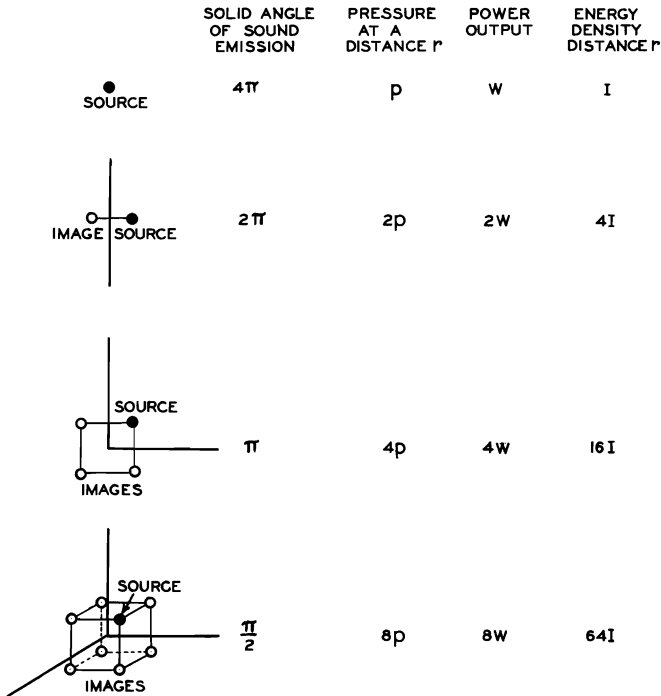
Fig. 2.1. The sound pressure, total power output, and energy density delivered by a point source operating in solid angles of 4π, 2π, π, and $\pi/2$ steradians.

2.3. Double Source (Doublet Source).[1,2,3,4]—A double source consists of two point sources equal in strength $\pm 4\pi A'$, but opposite in phase separated by a vanishingly small distance δr. The strength of the doublet is $4\pi A'\delta r$. Let $A'\delta r = A$. In these considerations A' corresponds to A of equation 2.1, that is $4\pi A' = S\dot{\xi}_0$.

At a distance r in a direction inclined at an angle α to the axis of the doublet the velocity potential is

$$\phi = \frac{\left(\dfrac{1}{r} + jk\right)A}{r} \, \epsilon^{jk(ct-r)} \cos \alpha \qquad 2.7$$

[1] Lamb, " Dynamical Theory of Sound," E. Arnold, London, 1931.
[2] Davis, " Modern Acoustics," The Macmillan Co., New York, N.Y., 1934.
[3] Wood, " A Textbook of Sound," Bell and Sons, London, 1930.
[4] Crandall, " Theory of Vibrating Systems and Sound," D. Van Nostrand Company, Princeton, N.J., 1926.

The pressure from equation 2.7 is

$$p = -\rho \frac{\partial \phi}{\partial t} = -j \frac{\rho c k A}{r} \left(\frac{1}{r} + jk\right) \epsilon^{jk(ct-r)} \cos \alpha \qquad 2.8$$

Retaining the real parts of equation 2.8

$$p = \frac{\rho c k A}{r} \left[\frac{1}{r} \sin k(ct - r) + k \cos k(ct - r)\right] \cos \alpha \qquad 2.9$$

At a very large distance

$$p \propto \frac{k^2 A}{r} \cos \alpha \qquad 2.10$$

At a very small distance

$$p \propto \frac{kA}{r^2} \cos \alpha \qquad 2.11$$

The particle velocity has two components, the radial $\frac{\partial \phi}{\partial r}$ and the transverse $\frac{1}{r} \frac{\partial \phi}{\partial \alpha}$. The radial component of the particle velocity from equation 2.7 is,

$$u = \frac{\partial \phi}{\partial r} = -\left[\left(\frac{2}{r^3} + \frac{jk}{r^2}\right) + jk\left(\frac{1}{r^2} + \frac{jk}{r}\right)\right] A \epsilon^{jk(ct-r)} \cos \alpha \qquad 2.12$$

Retaining the real parts of equation 2.12

$$u = -A \left[\left(\frac{2}{r^3} - \frac{k^2}{r}\right) \cos k(ct - r) - \frac{2k}{r^2} \sin k(ct - r)\right] \cos \alpha \qquad 2.13$$

At a very large distance

$$u \propto \frac{Ak^2}{r} \cos \alpha \qquad 2.14$$

At a very small distance

$$u \propto \frac{A}{r^3} \cos \alpha \qquad 2.15$$

The transverse component of the particle velocity is

$$u = \frac{1}{r} \frac{\partial \phi}{\partial \alpha} = -\left(\frac{\frac{1}{r} + jk}{r^2}\right) A \epsilon^{jk(ct-r)} \sin \alpha \qquad 2.16$$

Retaining the real parts of equation 2.16

$$u = -A \left[\frac{1}{r^3} \cos k(ct - r) - \frac{k}{r^2} \sin k(ct - r)\right] \sin \alpha \qquad 2.17$$

At a very large distance

$$u \propto \frac{Ak}{r^2} \sin \alpha \qquad 2.18$$

At a very small distance

$$u \propto \frac{A}{r^3} \sin \alpha \qquad 2.19$$

Fig. 2.2 shows the velocity components and the pressure for various points around a doublet source. A common example of a doublet source is a direct radiator loudspeaker mounted in a small baffle. (Dimensions of the baffle are small compared to the wavelength.) If the response of such a loudspeaker is measured with a pressure microphone for various angles at a constant distance, the result will be a cosine characteristic. If the response is measured with a velocity microphone keeping the axis pointed toward the loudspeaker, the result will be a cosine directional characteristic. If the same is repeated keeping the axis of the velocity microphone normal to the line joining the microphone and the loudspeaker, the result will be a sine directional characteristic.

The total power, in ergs, emitted by a doublet source is

$$P = \iint \frac{p^2}{\rho c} \, dS \qquad 2.20$$

where p = pressure, in dynes per square centimeter,
ρ = density, in grams per cubic centimeter,
c = velocity of sound, in centimeters per second, and
dS = area, in square centimeters, over which the pressure is p.

FIG. 2.2. The sound pressure and particle velocity at a constant distance from a doublet source. The magnitude of the pressure is indicated by the circle. The particle velocity has two components, a radial and a transverse component. The direction and magnitude of these two components are indicated by vectors.

Taking the value of p from equation 2.9 (for r very large), the total average power in ergs per second emitted by a doublet source is

$$P_T = 2\pi r^2 \int_0^\pi \frac{\rho c k^4 A^2}{2r^2} \cos^2 \alpha \sin \alpha \, d\alpha \qquad 2.21$$

$$P_T = \tfrac{2}{3}\pi \rho c k^4 A^2 \qquad 2.22$$

where ρ = density, in grams per cubic centimeter,
$k = 2\pi/\lambda$,
λ = wavelength, in centimeters,
c = velocity of sound, in centimeters per second, and
A is defined in the first paragraph of this section.

The power output from a simple source (equation 2.6) is proportional to the square of the frequency, while the power output from a doublet source (equation 2.22) is proportional to the fourth power of the frequency. For this reason the power output of a direct radiator loudspeaker falls off rapidly with frequency when the dimensions of the baffle are small compared to the wavelength (see Sec. 6.8).
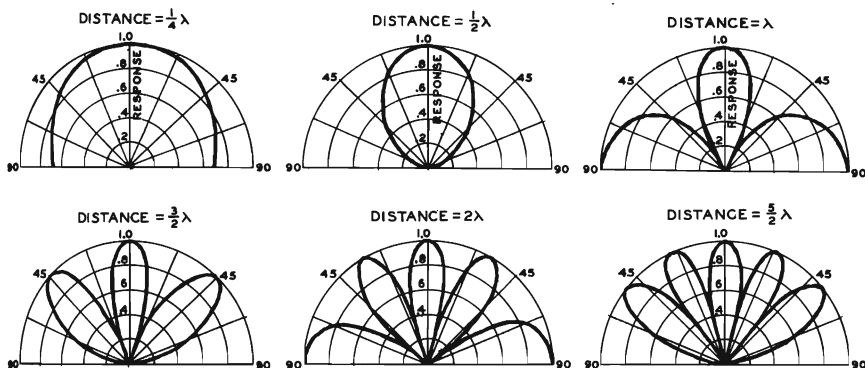
FIG. 2.3.   Directional characteristics of two separated equal small sources vibrating in phase as a function of the distance between the sources and the wavelength.   The polar graph depicts the sound pressure, at a fixed distance, as a function of the angle.   The sound pressure for the angle 0° is arbitrarily chosen as unity.   The direction corresponding to the angle 0° 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.

2.4. Series of Point Sources.—The directional characteristic[5,6,7] 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

$$R_\alpha = \frac{\sin\left(\dfrac{n\pi d}{\lambda}\sin\alpha\right)}{n\sin\left(\dfrac{\pi d}{\lambda}\sin\alpha\right)}$$ 2.23

where R_α = 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, and
    λ = wavelength, in centimeters.

The directional characteristics of a two-point source are shown in Fig. 2.3.   It will be noted that the secondary lobes are equal to the main lobe.

[5] Wolff, I., and Malter, L., *Jour. Acous. Soc. Amer.*, Vol. 2, No. 2., p. 201, 1930.
[6] Stenzel, H., *Elek. Nach. Tech.*, Vol. 4, No. 6, p. 239, 1927.
[7] Stenzel, H., *Elek. Nach. Tech.*, Vol. 6, No. 5, p. 165, 1929.

2.5. Straight-Line Source.—A straight-line source may be made up of a large number of points of equal strength and phase on a line separated by equal and very small distances. If the number of sources n approach infinity and d, the distance between the sources, approaches zero in such a way that

$$nd = l$$

the limiting case is the line source. If this is carried out, equation 2.23 becomes

$$R_\alpha = \frac{\sin\left(\frac{\pi l}{\lambda}\sin\alpha\right)}{\frac{\pi l}{\lambda}\sin\alpha} \qquad\qquad 2.24$$

The directional characteristics of a continuous line source are shown in Fig. 2.4. The directional characteristics are symmetrical about the line
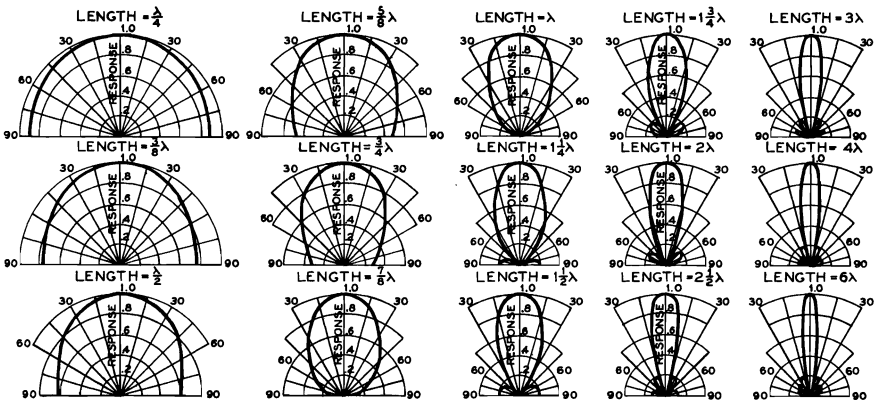
FIG. 2.4. Directional characteristics of a line source as a function of the length and the wavelength. The polar graph depicts the sound pressure, at a large fixed distance, as a function of the angle. The sound pressure for the angle 0° is arbitrarily chosen as unity. The direction corresponding to the angle 0° is perpendicular to the line. The directional characteristics in three dimensions are surfaces of revolution about the line as an axis.

as an axis. Referring to Fig. 2.4, it will be seen that there is practically no directivity when the length of the line is small compared to the wavelength. On the other hand, the directional characteristics are sharp when the length of the line is several wavelengths.

2.6. Beam Tilting by Phase Shifting.—The direction and shape of the wavefront produced by a series of sound sources may be altered by the introduction of a delay pattern in the excitation of the sources. An example of a series of point sources of sound equally spaced along a straight line in

combination with a delay system is shown in Fig. 2.5. In the system shown in Fig. 2.5 the distance x, in centimeters, is given by

$$x = ct \qquad 2.25$$

where c = velocity of sound, in centimeters, and
t = time delay, in seconds.

The angle θ, the angle by which the wavefront is shifted by the delay system, is given by

$$\theta = \sin^{-1} \frac{x}{d} \qquad 2.26$$

where d = distance between the units, in centimeters.

Phase shifting can be used in many other ways besides beam tilting. For example, practically any wavefront shape can be obtained by introducing the appropriate phase shift in the sound sources.
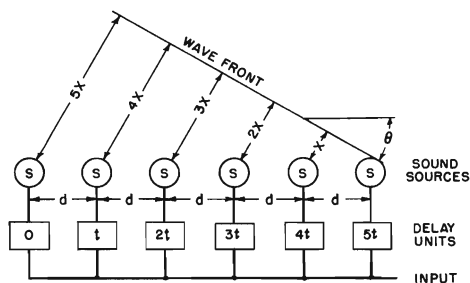
FIG. 2.5. A delay system for tilting the directional characteristic of a line of sound sources.

2.7. Tapered Straight-Line Source.—The directional characteristic[8] of a line source, all parts vibrating in phase, in which the strength varies linearly from its value at the center to zero at either end, is given by

$$R_\alpha = \frac{\sin^2\left(\frac{\pi l}{2\lambda} \sin \alpha\right)}{\left(\frac{\pi l}{2\lambda} \sin \alpha\right)^2} \qquad 2.27$$

where R_α = 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,
l = total length of the line in centimeters, and
λ = wavelength, in centimeters.

The directional characteristics of a tapered line source are shown in Fig. 2.6. Comparing the directional characteristics of Fig. 2.6 with those of the uniform line of Fig. 2.4, it will be seen that the main lobe is broader and the secondary lobes are reduced in amplitude.

[8] Menges, Karl, *Akus Zeit.*, Vol. 6, No. 2, p. 90, 1941.
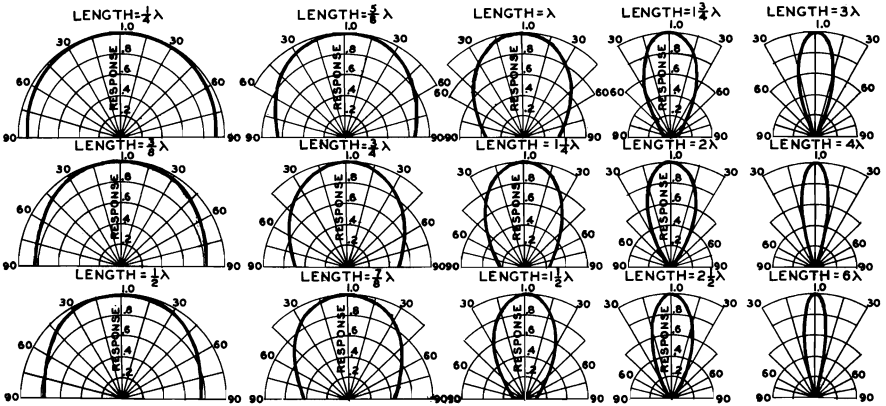
FIG. 2.6.   Directional characteristics of a tapered line source as a function of the length and the wavelength.   The volume current output along the line varies linearly from a maximum at the center to zero at the two ends.   The polar graph depicts the sound pressure, at a fixed distance, as a function of the angle.   The sound pressure for the angle 0° is arbitrarily chosen as unity.   The direction corresponding to the angle 0° is perpendicular to the line.   The directional characteristics in three dimensions are surfaces of revolution about the line as an axis.

2.8. Nonuniform Straight-Line Source.—The directional characteristic of a line, all parts vibrating in phase, in which the strength varies as a function of the distance x along a line is given by

$$R_\alpha = \frac{\displaystyle\int_{-\frac{d}{2}}^{+\frac{d}{2}} f(x)\,\epsilon^{-j(2\pi x/\lambda)\,\sin\alpha}\,dx}{\displaystyle\int_{-\frac{d}{d}}^{+\frac{d}{2}} f(x)\,dx} \qquad\qquad 2.28$$

where x = distance from the center of the line, in centimeters,
         d = total length of the line, in centimeters,
      $f(x)$ = strength distribution function and the other quantities are
             the same as those in equation 2.27.

2.9. End Fired Line Source.—An end fired line source is one in which there is progressive phase delay between the elements of the line.   In the case in which the time delay of excitation between the elements corresponds to the time of wave propagation in space for this distance the maximum directivity occurs in direction corresponding to the line joining the elements. The directional characteristics of an end fired line of this type and of uniform strength is given by

$$R_\alpha = \frac{\sin\dfrac{\pi}{\lambda}\,(l - l\cos\alpha)}{\dfrac{\pi}{\lambda}\,(l - l\cos)\,\alpha} \qquad\qquad 2.29$$

where R_a = ratio of the pressure for an angle α to the pressure for the
            angle $\alpha = 0$.   The direction $\alpha = 0$ is along the line,
   l = length of the line, and
   λ = wavelength.

The directional characteristics [9] of a uniform end fired line with progressive time delay between elements corresponding to the time of wave propagation over this distance in free space are shown in Fig. 2.7.   The maximum directivity occurs along the direction corresponding to the line.   The directional characteristics are symmetrical about the line as an axis.
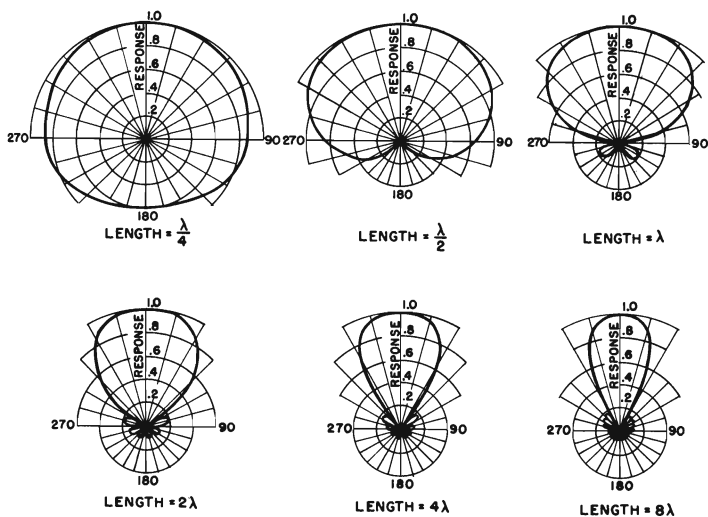
FIG. 2.7.   Directional characteristics of an end fired line source as a function of the length and wavelength.   The polar graph depicts the sound pressure, at a large fixed distance, as a function of the angle. The sound pressure for the angle 0° is arbitrarily chosen as unity.   The direction   corresponding   to angle 0° coincides with the line.   The directional characteristics in three dimensions are surfaces of revolution about the line as an axis.

2.10. Super Directivity Source.—From the preceding examples of directional systems it will be seen that in order to obtain some order of directivity the dimensions of the radiator must be greater than a wavelength. It is possible to obtain a high order of directivity from a source which is smaller in dimension than wavelength.   These systems have been termed super directional sources.[10]   A super directional system may be considered to be the difference between two patterns each of which is due to a conventional linear array employing in phase excitation.   A super directional array

[9] Olson, H. F., *Jour. Inst. Rad. Eng.*, Vol. 27, No. 7, p. 438, 1939.
[10] Pritchard, R. L., *Jour. Acous. Soc. Amer.*, Vol. 25, No. 5, p. 879, 1953.

is shown in Fig. 2.8. It will be seen that alternate elements are oppositely phased. Comparing the directional pattern of the super directivity source with the directivity pattern of the simple line source of Fig. 2.4, it will be
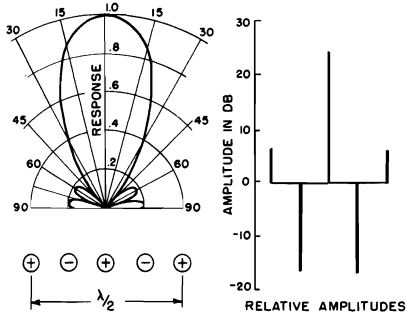
FIG. 2.8. Directional characteristics of a super directivity source consisting of five sources spaced ½ wavelength apart. The relative amplitudes and the phases of the sources are shown in the diagrams above. The polar graph depicts the sound pressure at a large fixed distance from the array. The sound pressure for the angle 0° is arbitrarily chosen as unity. The direction corresponding to the angle 0° is perpendicular to the line joining the sources. The directional characteristics in three dimensions are surfaces of revolution about the line as an axis.

seen that approximately the same directivity is obtained with a line of one-third the length of the simple line source. This added directivity is obtained at the expense of some other factors. The reversed phase excitation results in a loss in efficiency. Close tolerances must be maintained upon the strength of the elements and the operating wavelength, otherwise the directivity pattern will not be maintained. Thus a super directional system is sensitive to frequency changes and is, therefore, not suitable for broad band operation. Therefore, the applications for a super directional source are where a narrow frequency band width and low efficiency can be tolerated in exchange for smaller space requirements.

2.11. Curved-Line Source (Arc of a Circle).—A curved-line source may be made up of a large number of point sources vibrating in phase on the arc of a circle separated by very small distances. The directional characteristics of such a line in the plane of the arc are,

$$R_\alpha = \frac{1}{2m+1} \left| \sum_{k=-m}^{k=m} \cos \left[\frac{2\pi R}{\lambda} \cos (\alpha + k\theta) \right] \right.$$
$$\left. + j \sum_{k=-m}^{k=m} \sin \left[\frac{2\pi R}{\lambda} \cos (\alpha + k\theta) \right] \right| \quad 2.30$$

where R_α = 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, and

k = variable.

Another method[11] is to break up the arc into a large number of equal chords. The strength is assumed to be uniform over each chord. Also the phase of all the chords is the same. In this case the result takes the form,

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

$$\left. + j\sum_{k=-m}^{k=m} \sin\left\{\frac{2\pi R}{\lambda}\cos(\alpha+k\theta)\right\} \frac{\sin\left[\frac{\pi d}{\lambda}\sin(\alpha+k\theta)\right]}{\frac{\pi d}{\lambda}\sin(\alpha+k\theta)} \right| \quad 2.31$$

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

λ = wavelength, in centimeters,

k = variable,

R = radius of the arc, in centimeters,

$2m + 1$ = number of chords,

θ = angle subtended by any of the chords at the center of circumscribing circle, and

d = length of one of the chords, in centimeters.
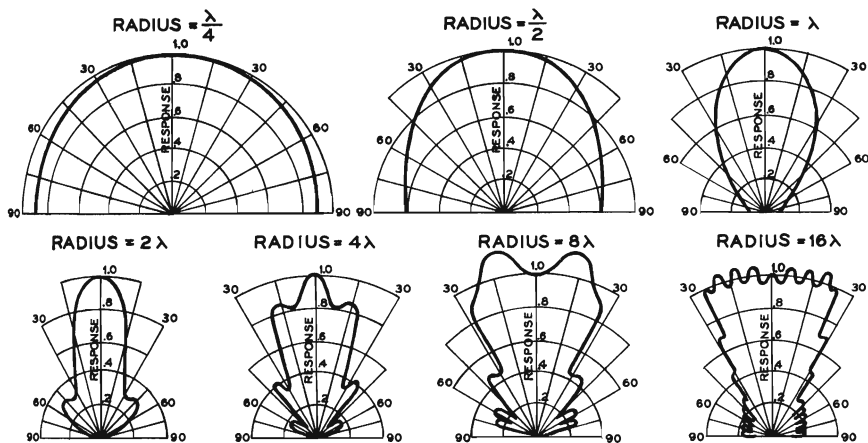
Fig. 2.9. Directional characteristics of a 60° arc as a function of the radius and the wavelength. The polar graph depicts the sound pressure, at a large fixed distance, as a function of the angle in the plane of the arc. The sound pressure for the angle 0° is arbitrarily chosen as unity.

The directional characteristics for an arc of 60°, 90°, and 120° are shown in Figs. 2.9, 2.10, and 2.11. The interesting feature of the directional

[11] Wolff, I., and Malter, L., *Jour. Acous. Soc. Amer.*, Vol. 2, No. 2, p. 201, 1930.

characteristics of an arc is that the directional characteristics are very broad for wavelengths large compared to the dimensions, and are narrow for wavelengths comparable to the dimensions and are broad again for wavelengths small compared to the dimensions of the arc. The arc must be several wavelengths in length in order to yield a "wedge-shaped" directional characteristic.
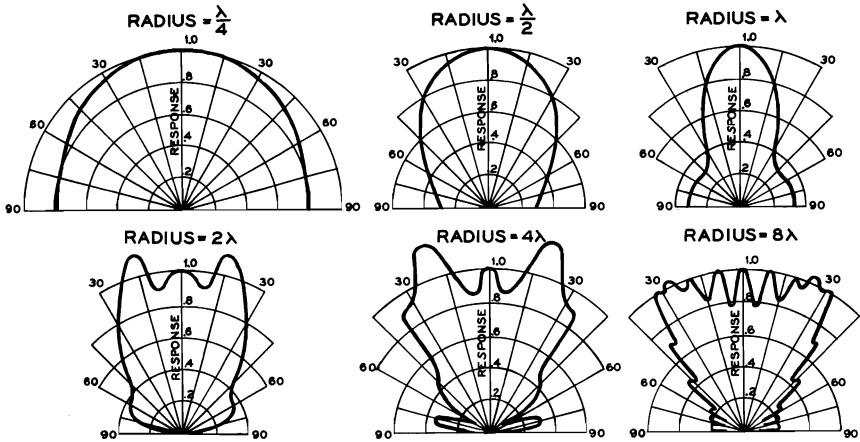
FIG. 2.10. Directional characteristics of a 90° arc as a function of the radius and the wavelength. The polar graph depicts the sound pressure at a large fixed distance, as a function of the angle in the plane of the arc. The sound pressure for the angle 0° is arbitrarily chosen as unity.
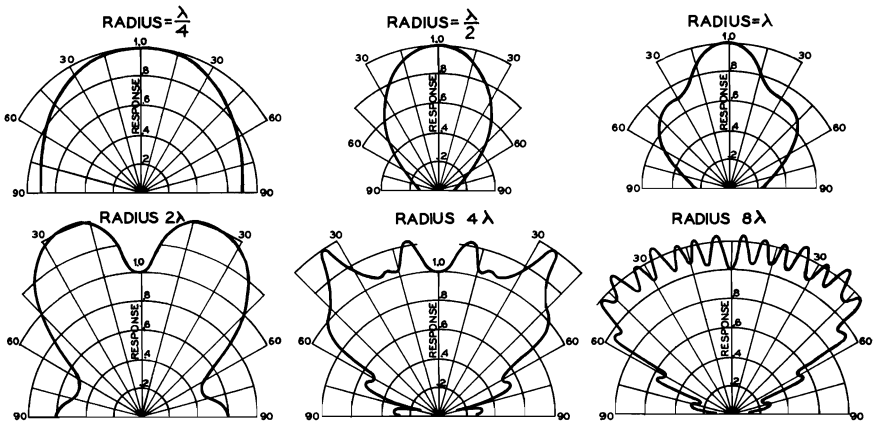
FIG. 2.11. Directional characteristics of a 120° arc as a function of the radius and the wavelength. The polar graph depicts the sound pressure, at a large fixed distance, as a function of the angle in the plane of the arc. The sound pressure for the angle 0° is arbitrarily chosen as unity.

2.12. Circular-Ring Source.—The directional characteristics[12,13] of a circular-ring source of uniform strength and the same phase at all points on the ring is

$$R_\alpha = J_0 \left[\left(\frac{2\pi R}{\lambda} \right) \sin \alpha \right] \qquad 2.32$$

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

J_0 = Bessel function of zero order,

R = radius of the circle, in centimeters, and

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

The directional characteristics of a circular-ring source as a function of the diameter and the wavelength are shown in Fig. 2.12. The shapes are
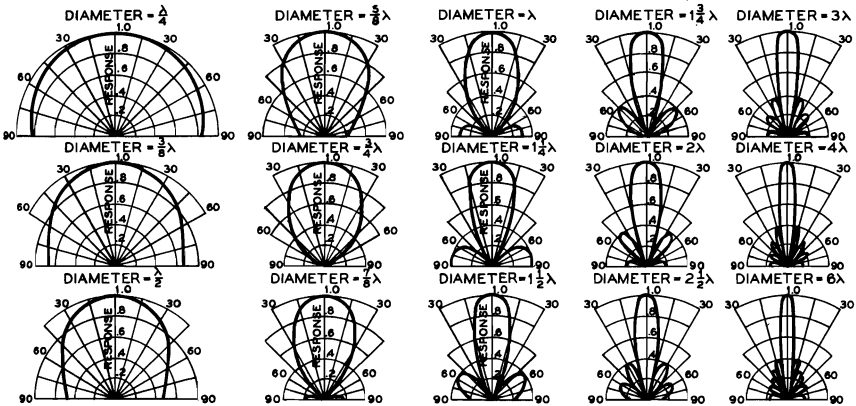
FIG. 2.12. Directional characteristics of a circular-line or ring source as a function of the diameter and wavelength. The polar graph depicts the sound pressure, at a large fixed distance, as a function of the angle. The sound pressure for the angle 0° is arbitrarily chosen as unity. The direction corresponding to the angle 0° is the axis. The axis is the center line perpendicular to the plane of the circle. The directional characteristics in three dimensions are surfaces of revolution about the axis.

quite similar to those of a straight line. The characteristic is somewhat sharper than that of a uniform line of length equal to the diameter of the circle, but has almost the same form. The amplitudes of the secondary lobes are greater than those of the uniform line.

2.13. Plane Circular-Piston Source.—The directional characteristics[14, 15] of a circular-piston source mounted in an infinite baffle with all

[12] Stenzel, H., *Elek. Nach. Tech.*, Vol. 4, No. 6, p. 1, 1927.

[13] Wolff, I., and Malter, L., *Jour. Acous. Soc. Amer.*, Vol. 2, No. 2, p. 201, 1930.

[14] Stenzel, H., *Elek. Nach. Tech.*, Vol. 4, No. 6, p. 1, 1927.

[15] Wolff, I., and Malter, L., *Jour. Acous. Soc. Amer.*, Vol. 2, No. 2, p. 201, 1930.

parts of the surface of the piston vibrating with the same strength and phase
are

$$R_\alpha = \frac{2J_1\left(\dfrac{2\pi R}{\lambda}\sin\alpha\right)}{\dfrac{2\pi R}{\lambda}\sin\alpha} \qquad\qquad 2.33$$

where R_α = 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 circular piston, in centimeters,
  α = angle between the axis of the circle and the line joining the
    point of observation and the centre of the circle, and
  λ = wavelength, in centimeters.
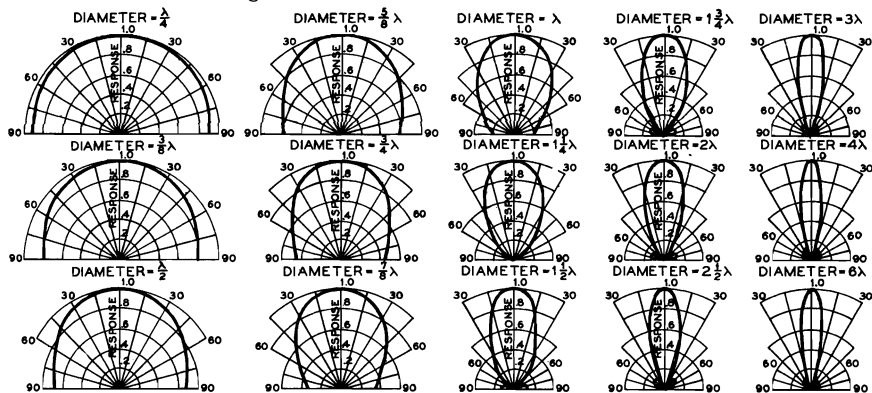
FIG. 2.13.   Directional characteristics of a circular-piston source mounted in an infinite
baffle as a function of the diameter and wavelength.   The polar graph depicts the sound
pressure, at a large fixed distance, as a function of the angle.   The sound pressure for
the angle 0° is arbitrarily chosen as unity.   The direction corresponding to the angle 0°
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.

The directional characteristics of a plane circular-piston source mounted
in an infinite baffle as a function of the diameter and wavelength are shown
in Fig. 2.13.   The characteristic is somewhat broader than that of the
uniform line of length equal to the diameter of the circle, but has approxi-
mately the same form.   The amplitudes of the secondary lobes are smaller
than those of the uniform line.

2.14. Nonuniform Plane Circular-Surface Source.[16]—The integration
of the expression for a plane circular-surface source in which the strength
varies as a function of the distance from the center cannot be obtained in

[16] Jones, R. Clark, *Jour. Acous. Soc. Amer.*, Vol. 16, No. 3, p. 147, 1945.   This is a
comprehensive paper on the study of directional patterns of plane surface sources with
specified normal velocities.   A number of directional patterns and tables are given.

simple terms. An approximate method may be employed in which the plane circular surface with nonuniform strength is divided into a number of rings with the proper strength assigned to each ring. An alternative method may be employed in which the strength distribution is obtained by superposing a number of plane circular-surface sources of different radii with the proper strength assigned to each surface.

2.15. Plane Circular-Piston Source Set in the End of an Infinite Pipe.[17,18]—The directional characteristics of a plane circular-piston set in the end of an infinite pipe with all parts of the piston vibrating with the same amplitude and phase as a function of the diameter and wavelength are shown in Fig. 2.14.
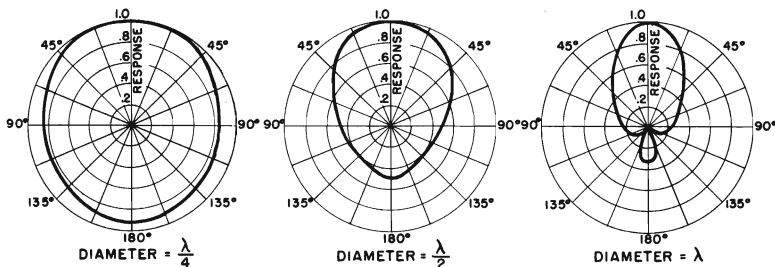
FIG. 2.14. Directional characteristics of a circular-piston source located in the end of an infinite pipe as a function of the diameter and wavelength. The polar graph depicts the sound pressure, at a large fixed distance, as a function of the angle. The sound pressure for the angle 0° is arbitrarily chosen as unity. The direction corresponding to the angle 0° 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.

An example of a vibrating piston set in a tube is that of a loudspeaker mechanism set in a completely enclosed cabinet having a face area not appreciably larger than the loudspeaker mechanism.

2.16. Plane Circular-Piston Source in Free Space.[19]—The directional characteristics of a plane circular piston in free space with all parts of the piston vibrating with the same amplitude and phase as a function of the diameter and wavelength are shown in Fig. 2.15. In the low-frequency range the directional pattern is the same as that of a doublet source because it is doublet in this frequency range.

An example of a vibrating piston in free space is a loudspeaker mechanism operating in free space without a baffle, cabinet, etc.

2.17. Plane Square-Surface Source.—The directional characteristics of a plane square-surface source, with all parts of the surface vibrating with the same intensity and phase, in a normal plane parallel to one side, is the

[17] Levine and Schwinger, *Phys. Rev.*, Vol. 73, No. 4, p. 383, 1948.
[18] Beranek, " Acoustics," McGraw-Hill Book Company, New York, N.Y., 1954.
[19] Wiener, F. M., *Jour. Acous. Soc. Amer.*, Vol. 23, No. 6, p. 697, 1951.

same as that of a uniform line source having a length equal to one side of the square (equation 2.24).

The directional characteristics of a plane square-surface source, with all parts of the surface vibrating with the same strength and phase, in a normal plane containing the diagonal is the same as that of the tapered line source having a length equal to the diagonal (equation 2.28).
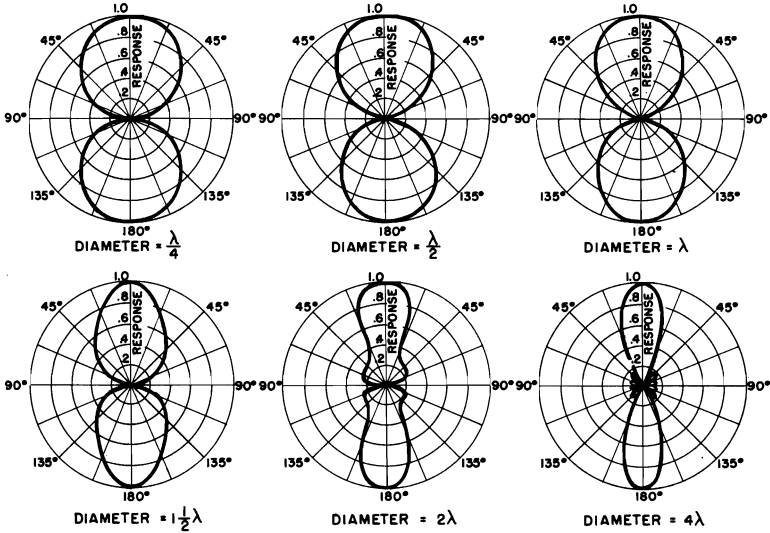
FIG. 2.15.   Directional characteristics of a circular piston located in free space as a function of the diameter and wavelength.   The polar graph depicts the sound pressure, at a large fixed distance, as a function of the angle.   The sound pressure for the angle 0° is arbitrarily chosen as unity.   The direction corresponding to the angle 0° 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.

2.18. Plane Rectangular-Surface Source.—The directional characteristics of a rectangular-surface source with all parts of the surface vibrating with the same strength and phase are

$$R_a = \frac{\sin\left(\dfrac{\pi l_a}{\lambda}\sin\alpha\right)}{\dfrac{\pi l_a}{\lambda}\sin\alpha} \cdot \frac{\sin\left(\dfrac{\pi l_b}{\lambda}\sin\beta\right)}{\dfrac{\pi l_b}{\lambda}\sin\beta} \qquad\qquad 2.34$$

where l_a = length of the rectangle,

$\quad\ l_b$ = width of the rectangle,

$\quad\ \alpha$ = 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, and

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_b.

The directional characteristic of a plane rectangular-surface source with uniform strength and phase is the same as the product of the characteristic of two line sources at right angles to each other and on each of which the strength and phase are uniform.

2.19. Horn Source.—The directional characteristics of a horn depend upon the shape, mouth opening, and the frequency. It is 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 determines the directional characteristics of the horn. The particular complexion of the velocities and phases of these areas is governed by the flare and dimensions and shape of the mouth. In these considerations the mouth will be of circular cross section and mounted in a large flat baffle. 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.

A. *Exponential Horns.*—The effect of the diameter of the mouth for a constant flare upon the directional characteristics[20,21] of an exponential horn is depicted in Fig. 2.16. At the side of each polar diagram is the diameter of a vibrating piston which will yield approximately the same directional characteristic. It will be seen that up to the frequency at which the wavelength becomes comparable to the mouth diameter, the directional characteristics are practically the same as those of a piston of the size of the mouth. Above this frequency the directional characteristics are practically independent of the mouth size and appear to be governed primarily by the flare.

To further illustrate the relative effects of the mouth and flare, Fig. 2.17 shows the effect of different rates of flare, for a constant mouth diameter, upon the directional characteristics of an exponential horn. These results also show that, for the wavelengths larger than the mouth diameter, the directional characteristics are approximately the same as those of a vibrating piston of the same size as the mouth. Above this frequency the directional

---

[20] Olson, H. F., *RCA Review*, Vol. 1, No. 4, p. 68, 1937.

[21] Goldman, S., *Jour. Acous. Soc. Amer.*, Vol. 5, p. 181, 1934, reports the results of an investigation upon the directional characteristics of exponential horns at 15,000 and 25,000 cycles. A comparison can be made with the results shown in Figs. 2.16 and 2.17 by increasing the dimensions of the horns used by him to conform with those shown here and decreasing the frequency by the factor of increase in dimensions. Such a comparison shows remarkable agreement between the two sets of data.

characteristics are broader than those obtained from a piston the size of the mouth. From another point of view, the diameter of the piston which will yield the same directional characteristic is smaller than the mouth. These results also show that the directional characteristics vary very slowly with frequency at these smaller wavelengths. Referring to Fig. 2.17 it will be seen that for any particular high frequency, 4000, 7000, or 10,000 cycles per second, the directional characteristics become progressively sharper as the rate of flare decreases.
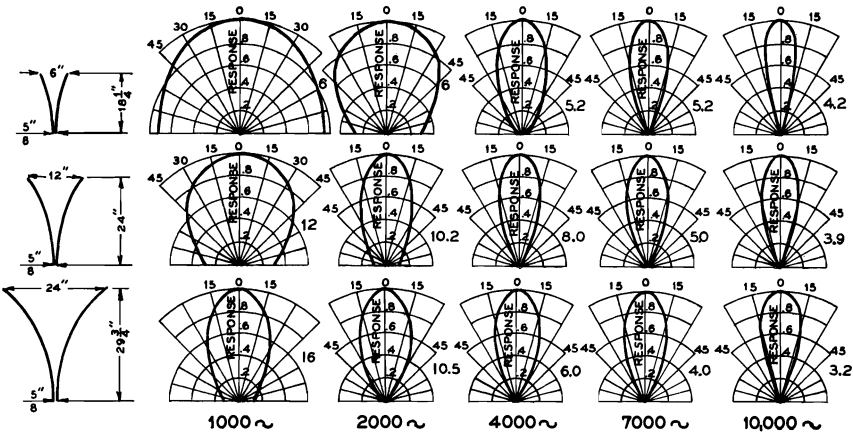
FIG. 2.16. The directional characteristics of a group of exponential horns, with a constant flare and throat diameter of $\frac{5}{8}$ inch as a function of the mouth diameter. The number at the right of each polar diagram indicates the diameter of a circular piston which will yield the same directional characteristic. The polar graph depicts the sound pressure, at a fixed distance, as a function of the angle. The sound pressure for the angle 0° is arbitrarily chosen as unity. The direction corresponding to 0° is the axis of the horn. The directional characteristics in three dimensions are surfaces of revolution about the horn axis.

B. *Conical Horns.*—In the case of the circular conical horn the directional pattern should be the same as that of a circular, spherical surface source. The radius of the spherical surface is the distance along the side of the horn from the apex to the mouth. The directional characteristics of two conical horns are shown in Fig. 2.18. At the lower frequencies the directional pattern is approximately the same as that of a piston of the same size as the mouth. The directional pattern becomes sharper with an increase of the frequency. However, at the higher frequencies where the diameter of the mouth is several wavelengths, the pattern becomes broader as would be expected from a spherical surface source. The directional characteristics of a conical horn as depicted in Fig. 2.18 are practically the same as those of a spherical surface source.

C. *Parabolic Horns.*—In the parabolic horn the sectional area is proportional to the distance from the apex. This horn may be constructed
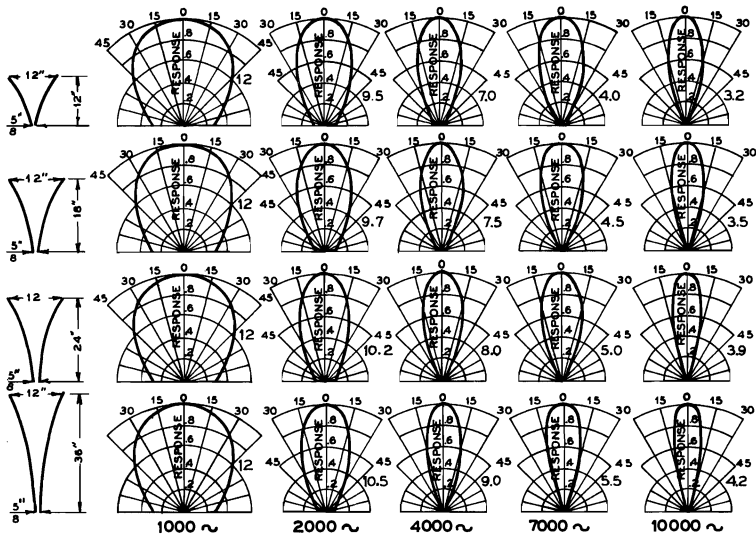
FIG. 2.17. The directional characteristics of a group of exponential horns, with a mouth diameter of 12 inches and a throat diameter of ⅝ inch, as a function of the flare. The number at the right of each polar diagram indicates the diameter of a circular piston which will yield the same directional characteristic. The polar graph depicts the sound pressure, at a fixed distance, as a function of the angle. The sound pressure for the angle 0° is arbitrarily chosen as unity. The direction corresponding to 0° is the axis of the horn. The directional characteristics in three dimensions are surfaces of revolution about the horn axis.
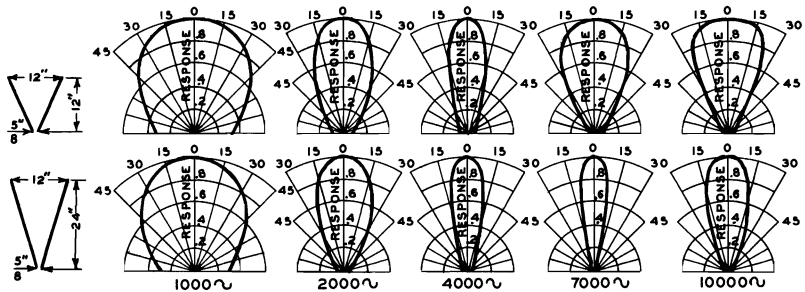
FIG. 2.18. The directional characteristics of two conical horns with mouth diameters of 12 inches and throat diameters of ⅝ inch and lengths of 12 inches and 24 inches. The polar graph depicts the sound pressure, at a fixed distance, as a function of the angle. The sound pressure for the angle 0° is arbitrarily chosen as unity. The direction corresponding to 0° is the axis of the horn. The directional characteristics in three dimensions are surfaces of revolution about the horn axis.

as shown in Fig. 2.19 in which two opposite horn walls are parallel and the other two are inclined at an angle with respect to each other. The directional characteristics of a 90° parabolic horn are shown in Fig. 2.19. The

source at the mouth is essentially a curved-line source described in Sec. 2.11. Therefore, the directional characteristics in a plane parallel to the two parallel sides of the horn should be essentially the same as that of a 90° arc. Comparing Fig. 2.19 with the 90° arc source of Fig. 2.10 it will be seen that the two directional patterns are quite similar.

From the directional patterns of horn-type radiators described in the preceding sections, it is evident that a wide range of directional patterns is possible in simple horns by variations in the shape of the horn and the mouth opening.

The results of Figs. 2.16, 2.17, 2.18, and 2.19 are applicable to other geometrically similar horns by changing the wavelength (or the reciprocal of the frequency) in the same ratio as the linear dimensions in accordance with the principle of similarity of Sec. 1.13.

A commercial application[22] of the principles of the parabolic horn is shown in Fig. 2.20. The horn is of the exponential rate of flare with straight sides on two boundaries and curved sides on the other two boundaries. The directivity patterns in the plane normal to the straight sides are the same as those of the parabolic horn of Fig. 2.19. The coverage in the vertical plane can be obtained by using the proper number of horn units as shown in Fig. 7.18.

2.20. Curved-Surface Source.—A sphere vibrating radially radiates sound uniformly outward in all directions. A portion of a spherical surface, large compared to the wavelength and vibrating radially, emits uniform sound radiation over a solid angle subtended by the surface at the center of curvature. To obtain uniform sound distribution over a certain solid angle, the radial air motion must have the same phase and amplitude over the spherical surface intercepted by the angle having its center of curvature at the vertex and the dimensions of the surface must be large compared to the wavelength. When these conditions are satisfied for all frequencies, the response characteristic will be independent of the position within the solid angle.

A loudspeaker[23,24,25] consisting of a large number of small horns with the axis passing through a common point will satisfy, for all practical purposes, the requirement of uniform phase and amplitude over the spherical surface formed by the mouths of the horns. A cellular or multihorn of this type is shown in Fig. 2.21A. This particular horn system consists of fifteen horns arranged in five vertical rows and three horizontal rows. The mouth opening of each horn is 8 × 8 inches. The horizontal and vertical angle between the axis of the individual horn is 17°.

The directional characteristics of a multihorn loudspeaker may be predicted theoretically[25] from the directional characteristics of an individual horn and the geometrical configuration of the assembly of horns.

[22] Volkmann, J. E., Unpublished Report.
[23] Wente, E. C., and Thuras, A. L., *Jour. A. I. E. E.*, Vol. 53, No. 1, p. 17, 1934.
[24] Hilliard, J. K., *Tech. Bull. Acad. Res. Council*, March, 1936.
[25] Olson, H. F., *RCA Review*, Vol. 1, No. 4, p. 68, 1937.
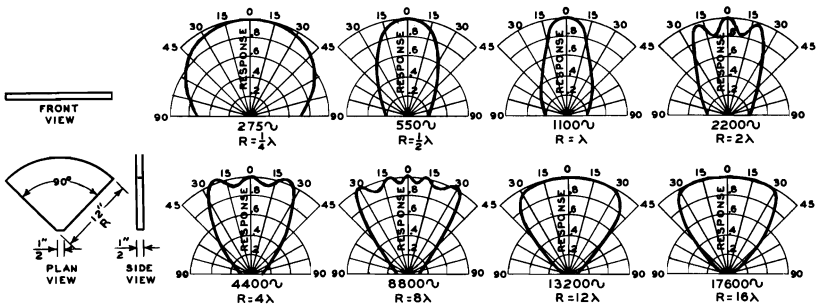
FIG. 2.19. The directional characteristics of a parabolic horn of the shape and the dimensions shown in the sketches on the left. The patterns were obtained in the plane midway between and parallel to the two parallel sides. The polar graph depicts the sound pressure, at a fixed distance, as a function of the angle. The sound pressure for the angle 0° is arbitrarily chosen as unity. The direction corresponding to 0° is spaced midway between the two nonparallel sides of the horn. $R = 12$ inches. The ratio of R/λ is also given for comparison with Fig. 2.10.
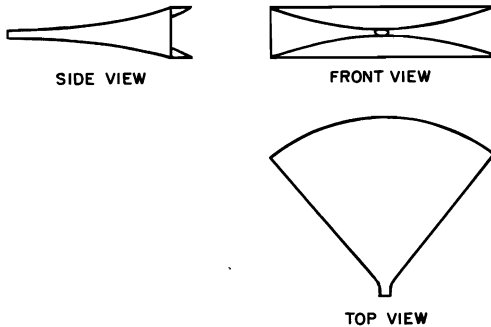
FIG. 2.20. A horn of exponential flare with two straight sides.
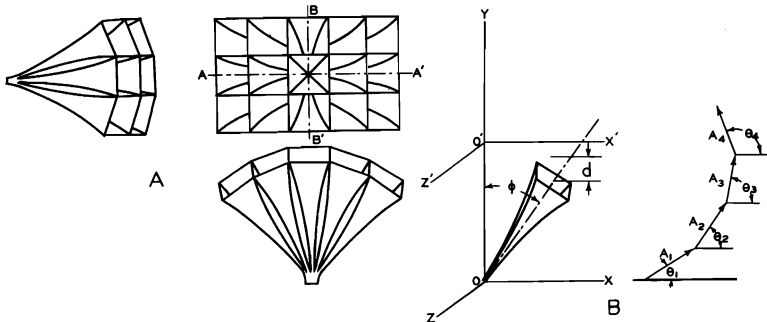
FIG. 2.21. A. A spherical radiating surface consisting of 15 individual exponential horns. B. Geometry for predicting the directional characteristics of a cluster of small horns.

Assume that the point of observation is located on the OY axis, Fig. 2.21B, at a distance several times the length of the horn. The amplitude of the vector contributed by an individual horn for the angle ϕ can be determined from its individual directional characteristic. In this illustration, the plane XOZ is chosen as reference plane for the phase of the vector. The phase angle of the vector associated with an individual horn is

$$\theta = \frac{d}{\lambda} 360° \qquad\qquad 2.35$$

where $d =$ the distance between the center of the mouth of the horn and the reference plane $X'O'Z'$, in centimeters, and
$\lambda =$ wavelength, in centimeters.

The vectors, having amplitudes A_1, A_2, A_3, A_4, etc., determined from the directional characteristics and having phase angles θ_1, θ_2, θ_3, θ_4, etc., determined from equation 2.35, are added vectorially as shown in Fig. 2.21B. This method of predicting the directional characteristics assumes that there is no interaction between individual horns which changes the complexion of the velocities at the mouth from that which obtains when operating an individual horn. Obviously, this condition is not absolutely satisfied. Apparently, the discrepancy has no practical significance because it has been found that this method of analysis agrees quite well with experimental results.

The directional characteristics of the cellular horn of Fig. 2.21A are shown in Figs. 2.22 and 2.23. Above 2000 cycles the dimensions of the total mouth surface are several wavelengths and the directional characteristics are fairly uniform and defined by the total angular spread. Where the dimensions are comparable to the wavelength the directional characteristics become very sharp, as shown by the polar curves for 500 and 1000 cycles. Then, as the dimensions of the surface become smaller than the wavelength, 250 cycles, the angular spread broadens, as is illustrated by the larger spread for the smaller vertical dimension when compared to the smaller spread for the larger horizontal dimension.

The directional characteristics of a cellular horn show a striking resemblance to those of an arc of the same angular spread. For example, the angular spread of the horn of Fig. 2.21 in the plane containing the line AA' and the axis is $87\frac{1}{2}°$. This may be compared to the arc of Fig. 2.10. In this case $\lambda/4$, $\lambda/2$, λ, 2λ, 4λ, and 8λ will correspond to 145, 290, 580, 1160, 2320, and 4640 cycles. The angular spread in the plane containing the line BB' and the axis is $52\frac{1}{2}°$. This may be compared to the 60° arc of Fig. 2.9 with the same relation between the wavelengths and frequencies, as noted above. It will be seen that there is a marked resemblance between corresponding frequencies. Of course, there is some variation due to the fact that the frequencies do not correspond exactly. Further, there is some difference in the angular spread. For most spherical surfaces of this type the directional characteristics in various planes correspond very closely to the directional characteristics of the corresponding arc.
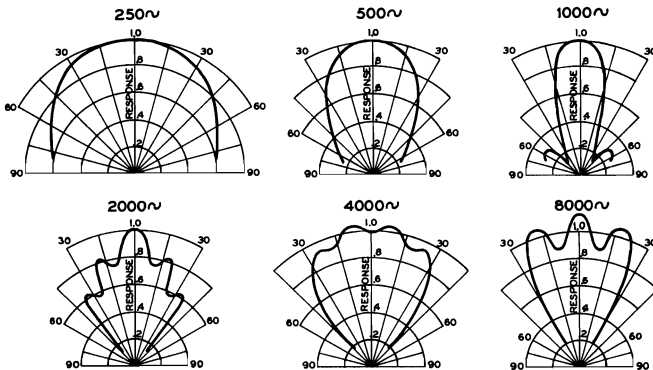
FIG. 2.22. Directional characteristics of the 15-cell cellular horn shown in Fig. 2.21A in a plane containing the line B-B' and the axis of the center horn. The polar graph depicts the sound pressure, at a fixed distance, as a function of the angle. The sound pressure for the angle 0° is arbitrarily chosen.
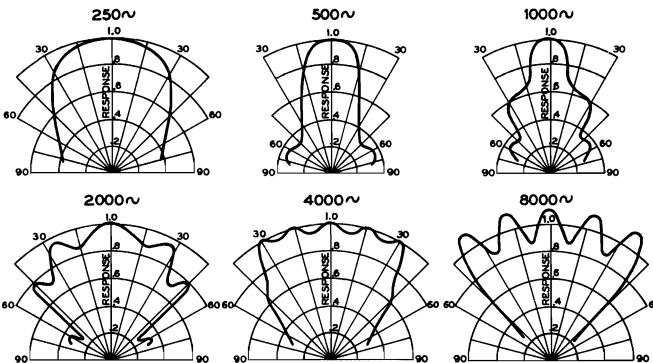
FIG. 2.23. Directional characteristics of the 15-cell cellular horn shown in Fig. 2.21A in a plane containing the line A-A' and the axis of the center horn. The polar graph depicts the sound pressure, at a fixed distance, as a function of the angle. The sound pressure for the angle 0° is arbitrarily chosen.

2.21. Cone-Surface Source.[26]—The directional characteristics[27] of a paper or felted paper cone used in the direct radiator-type loudspeaker may be predicted theoretically from the dimensions and shape of the cone and the velocity of sound propagation in the material. For this type of analysis the cone is divided into a number of ring-type radiators as shown in Fig. 2.24. The dimension of the ring along the cone should be a small

[26] Carlisle, R. W., *Jour. Acous. Soc. Amer.*, Vol. 15, No. 1, p. 44, 1943.
[27] The analysis in this section assumes that there is no reflected wave at the outer boundary. In order to obtain a uniform response frequency characteristic the reflected wave must be small. If the reflected wave is small, the effect upon the directional pattern may be neglected.

fraction of the wavelength of sound in the paper. The output of the cone at any angle is the vector sum of the vectors A_0, A_1, $A_2 \ldots A_n$ where the A's are the amplitudes of the individual rings.

The phase angle of the amplitude of the first ring is

$$\theta_0 = 0 \qquad\qquad 2.36$$

The phase angle of the amplitude of the second ring is

$$\theta_1 = 2\pi\left(\frac{d_1}{\lambda_A} - \frac{D_1}{\lambda_P}\right)\cos\alpha \qquad\qquad 2.37$$

The phase angle of the amplitude of the third ring is

$$\theta_2 = 2\pi\left(\frac{d_1 + d_2}{\lambda_A} - \frac{D_1 + D_2}{\lambda_P}\right)\cos\alpha \qquad\qquad 2.38$$
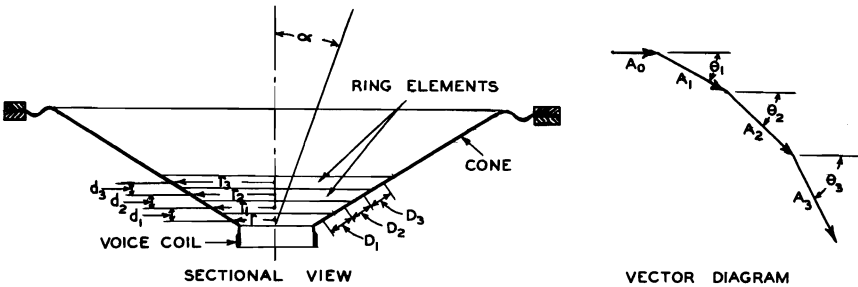
FIG. 2.24. Geometry for obtaining the directional pattern of a cone-type radiator.

The phase angle of the amplitude of the nth ring is

$$\theta_n = 2\pi\left(\frac{d_1 + d_2 \ldots d_n}{\lambda_A} - \frac{D_1 + D_2 \ldots D_n}{\lambda_P}\right)\cos\alpha \qquad\qquad 2.39$$

where $d_1, d_2, \ldots$ = axial distances shown in Fig. 2.24 in centimeters, and
$D_1, D_2, \ldots$ = distances along the cone shown in Fig. 2.24 in centimeters,
λ_A = wavelength of sound in air, in centimeters,
λ_P = wavelength of the sound in the paper cone, in centimeters, and
α = angle between the axis of the cone and the line joining the observation point and the center of the first ring.

The relative amplitude of the vector A_n is given by

$$A_n = 2\pi r_n D_n J_0\left(\frac{2\pi r_n}{\lambda_A}\sin\alpha\right) \qquad\qquad 2.40$$

where r_n = radius of the nth ring, in centimeters,
D_n = width of the nth ring along the cone, in centimeters,
λ_A = wavelength of sound in air, in centimeters,
α = angle between the axis of the cone and the line joining the observation point and the center of the cone, and
J_0 = Bessel function of zero order.

The directional characteristic of the cone is

$$R_a = \frac{\displaystyle\sum_{K=0}^{K=n} A_K \cos \theta_K - j \sum_{K=0}^{K=n} A_K \sin \theta_K}{\displaystyle\sum_{K=0}^{K=n} A_K} \qquad 2.41$$

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

A consideration of equation 2.41 shows that the directional pattern is a function of the frequency and becomes sharper as the frequency increases. For a particular frequency, cone angle, and material the directional patterns are practically similar for the same ratio of cone diameter to wavelength. For a particular frequency and the same cone material the directional pattern becomes broader as the cone angle is made larger. For a particular frequency and cone angle the directional pattern becomes broader as the velocity of propagation in the material decreases (see Sec. 6.2).

3

MECHANICAL VIBRATING SYSTEMS

3.1. Introduction.—The preceding chapters have been confined to the considerations of simple systems, point sources, homogeneous mediums, and simple harmonic motion. Sources of sound such as strings, bars, membranes, and plates are particularly liable to vibrate in more than one mode. In addition, there may be higher frequencies which may or may not be harmonics. The vibrations in solid bodies are usually termed as longitudinal, transverse, or torsional. In most cases it is possible to confine the motion to one of these types of vibrations. For example, the vibrations of a stretched string are usually considered as transverse. It is also possible to excite longitudinal vibrations which will be higher in frequency. If the string is of a fairly large diameter torsional vibrations may be excited. The vibrations of a body are also affected by the medium in which it is immersed. Usually, in the consideration of a particular example it is necessary to make certain assumptions which will simplify the problem. The mathematical analysis of vibrating bodies is extremely complex and it is beyond the scope of this book to give a detailed analysis of the various systems. For complete theoretical considerations, the reader is referred to the treatises which have been written on this subject. It is the purpose of this chapter to describe the most common vibrators in use today, to illustrate the form of the vibrations, and to indicate the resonant frequencies.

3.2. Strings.—In all string instruments the transverse and not the longitudinal vibrations are used. In the transverse vibrations all parts of the string vibrate in a plane perpendicular to the line of the string. For the case to be described 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 nonyielding supports, Fig. 3.1. Since the string is fixed at the ends, nodes will occur at these points. The fundamental frequency of the string is given by

$$f = \frac{1}{2l}\sqrt{\frac{T}{m}} \qquad\qquad 3.1$$

where T = tension, in dynes,
m = mass per unit length, in grams,
l = length of the string, in centimeters.

56

The shape of the vibration of a string is sinusoidal. In addition to the fundamental, other modes of vibration may occur, the frequencies being 2, 3, 4, 5, etc., times the fundamental. The first few modes of vibration of a string are shown in Fig. 3.1. 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.
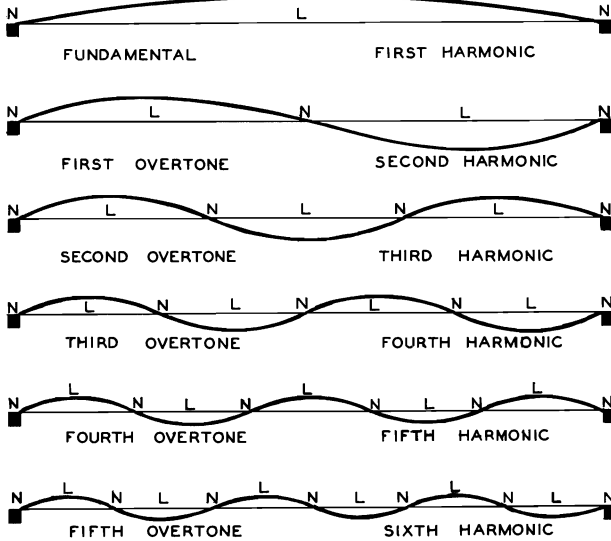
FIG. 3.1. Modes of vibration of a stretched string. The nodes and loops are indicated by N and L.

The above example is the simplest form of vibration of a string. A few of the problems which have been considered by different investigators[1,2,3,4,5] are as follows: nonuniform strings, loaded strings, stiff strings, nonrigid supports, the effect of damping, and the effect of different types of excitation. These factors of course alter the form of vibration and the overtones.

3.3. Transverse Vibration of Bars.[1,3,4,5]—In the preceding section the perfectly flexible string was considered where the restoring force due to stiffness is negligible compared to that due to 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

[1] Rayleigh, " Theory of Sound," Macmillan and Company, London, 1926.

[2] Crandall, " Theory of Vibrating Systems and Sound," D. Van Nostrand Company, Princeton, N.J., 1926.

[3] Wood, " A Text Book of Sound," Bell and Sons, London, 1930.

[4] Morse, " Vibration and Sound," McGraw-Hill Book Company, New York, N.Y., 1936.

[5] Lamb, " Dynamical Theory of Sound," E. Arnold, London, 1931.

plane and, as in the case of the string, only the transverse vibrations will be considered.

A. *Bar Clamped at One End.*—Consider a bar clamped in a rigid support at one end with the other end free (Fig. 3.2A). The fundamental frequency is given by

$$f_1 = \frac{.5596}{l^2} \sqrt{\frac{QK^2}{\rho}} \qquad 3.2$$

where l = length of the bar, in centimeters,

ρ = density, in grams per cubic centimeter, see Table 1.1,

Q = Young's modulus, in dynes per square centimeter, see Table 1.1: and

K = radius of gyration.
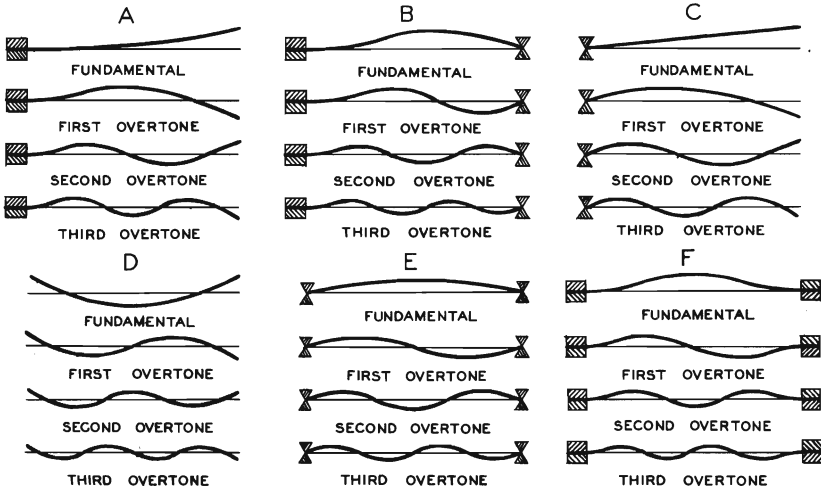
Fig. 3.2. Modes of transverse vibrations of bars. A. A bar clamped at one end and free at the other. B. A bar clamped at one end and supported at the other. C. A bar supported at one end and free at the other. D. A bar free at both ends. E. A bar supported at both ends. F. A bar clamped at both ends.

For a rectangular cross section the radius of gyration is

$$K = \frac{a}{\sqrt{12}}$$

where a = thickness of the bar, in centimeters, in the direction of vibration. For a circular cross section,

$$K = \frac{a}{2}$$

where a = radius of the bar, in centimeters.

For a hollow circular cross section,

$$K = \frac{\sqrt{a^2 + a_1^2}}{2}$$

where a = outside radius of the pipe, in centimeters, and
$\quad a_1$ = inside radius of the pipe, in centimeters.

The modes of vibration of a bar clamped at one end are shown in Fig. 3.2A. The table below gives the position of the nodes and the frequencies of the overtones.

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	.2165	$6.267f_1$
3	2	.1321, .4999	$17.55f_1$
4	3	.0944, .3558, .6439	$34.39f_1$

It will be seen that the overtones are not harmonics. The first overtone of a bar or reed has a higher frequency than the sixth harmonic of a string. The tuning fork is the most common example of a bar clamped at one end, because it can be considered to be two vibrating bars clamped at the lower ends. The overtone or the high-frequency sound of a tuning fork is quickly damped out leaving almost a pure sound.

B. *Bar Free at Both Ends.*—Consider a perfectly free bar (Fig. 3.2D). The fundamental frequency is given by

$$f_1 = \frac{1.133\pi}{l^2} \sqrt{\frac{QK^2}{\rho}} \qquad\qquad 3.3$$

where l = length of the bar, in centimeters. All the other quantities are the same as in equation 3.2.

The modes of vibration of a perfectly free bar are shown in Fig. 3.2D. The table which follows gives the position of the nodes and the frequencies of the overtones.

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	.2242, .7758	f_1
2	3	.1321, .50, .8679	$2.756f_1$
3	4	.0944, .3558, .6442, .9056	$5.404f_1$
4	5	.0734, .277, .05, .723, .9266	$9.933f_1$

C. *Bar Clamped at Both Ends.*—Consider a bar rigidly clamped at both ends (Fig. 3.2F). The same tones are obtained as in the case of the perfectly free bar.

D. *Bar Supported at Both Ends.*—Consider a bar supported on knife edges at the two edges at the two ends (Fig. 3.2E). The fundamental frequency is given by

$$f_1 = \frac{\pi}{2l^2}\sqrt{\frac{QK^2}{\rho}} \qquad 3.4$$

where l = length of the bar, in centimeters. All the other quantities are the same as in equation 3.2.

The overtones are

$$f_2 = 4f_1$$
$$f_3 = 9f_1$$
$$f_4 = 16f_1 \text{ etc.}$$

The nodes are equidistant as in case of the string.

E. *Bar Clamped at One End and Supported at the Other.*—Consider a bar clamped at one end and supported at the other end (Fig. 3.2B). The fundamental frequency is given by

$$f_1 = \frac{2.45}{l^2}\sqrt{\frac{QK^2}{\rho}} \qquad 3.5$$

The overtones are

$$f_2 = 3.25f_1$$
$$f_3 = 6.75f_1$$
$$f_4 = 11.5f_1,$$

and

$$f_5 = 17.7f_1$$

The modes of vibration are shown in Fig. 3.2B.

F. *Bar Supported at One End and Free at the Other.*—Consider a bar supported at one end and free at the other (Fig. 3.2C). The fundamental frequency is zero. The first overtone is given by

$$f_2 = \frac{2.45}{l^2}\sqrt{\frac{QK^2}{\rho}} \qquad 3.6$$

The overtones are

$$f_1 = 0$$
$$f_3 = 3.25f_2$$
$$f_4 = 6.75f_2$$
$$f_5 = 11.5f_2,$$

and

$$f_6 = 17.7f_2$$

The modes of vibration are shown in Fig. 3.2C.

G. *Tapered Cantilever Bars.*—In the preceding, considerations have been concerned with bars of uniform cross section. It is the purpose of

this section to give the formulas for the resonant frequencies of tapered cantilever bars.

The resonant frequency of a wedge-shaped bar vibrating normal to the two parallel sides of the wedge, Fig. 3.3A, is

$$f = \frac{1.14}{l^2}\sqrt{\frac{Qb^2}{12\rho}} \qquad 3.7$$

where b = thickness of the bar in the direction of vibration, in centimeters.
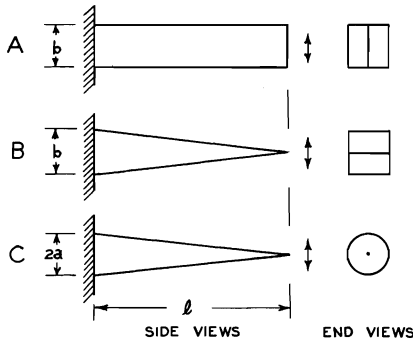
FIG. 3.3. Tapered cantilever bars, that is, bars clamped at one end and free at the other. A. A wedge-shaped bar vibrating in a direction normal to the two parallel sides. B. A wedge-shaped bar vibrating in a direction parallel to the two parallel sides. C. A conical bar.

The resonant frequency of a wedge-shaped bar vibrating parallel to the two parallel sides of the wedge, Fig. 3.3B, is

$$f = \frac{.85}{l^2}\sqrt{\frac{Qb^2}{12\rho}} \qquad 3.8$$

The resonant frequency of a conical bar, Fig. 3.3C, is

$$f = \frac{1.39}{l^2}\sqrt{\frac{Qa^2}{4\rho}} \qquad 3.9$$

where a = radius of the cone at the base, in centimeters.

3.4 Stretched Membranes.[6,7,8,9,10]—The ideal membrane is assumed to be flexible and very thin in cross section, and stretched in all directions by a

[6] Lamb, " Dynamical Theory of Sound," E. Arnold, London, 1931.

[7] Rayleigh, " Theory of Sound," Macmillan and Company, London, 1926.

[8] Morse, " Vibration and Sound," McGraw-Hill Book Company, New York, N.Y., 1936.

[9] Wood, " A Text Book of Sound," Bell and Sons, London, 1930.

[10] Crandall, " Theory of Vibrating Systems and Sound," D. Van Nostrand Company, Princeton, N.J., 1926.

force which is not affected by the motion of the membrane. Complete theoretical analyses have been made of circular, square, and rectangular membranes. For cases of practical interest the membrane is assumed to be rigidly clamped and stretched by a massive surround. It is the purpose of this section to consider circular, square, and rectangular stretched membranes.

A. *Circular Membrane.*—The fundamental frequency of a circular stretched membrane is given by

$$f_{01} = \frac{.382}{R} \sqrt{\frac{T}{m}} \qquad\qquad 3.10$$

where m = mass, in grams per square centimeter of area,
R = radius of the membrane, in centimeters, and
T = tension, in dynes per centimeter.
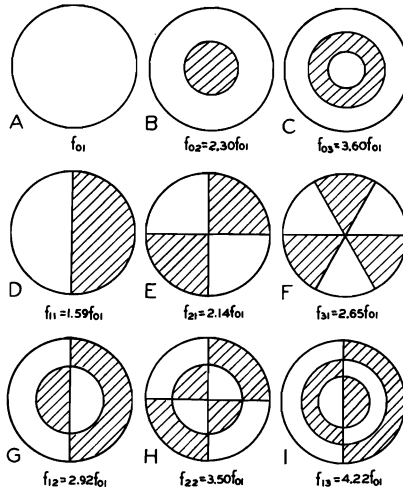
FIG. 3.4. Modes of vibration of a stretched circular membrane. Shaded segments are displaced in opposite phase to unshaded.

The fundamental vibration is with the circumference as a node and a maximum displacement at the center of the circle (Fig. 3.4A). The frequencies of the next two overtones with nodal circles are

$$f_{02} = 2.30 f_{01}$$
$$f_{03} = 3.60 f_{01}$$

and are shown in Figs. 3.4B and 3.4C. The frequencies of the first, second and third overtones with nodal diameters are

$$f_{11} = 1.59 f_{01}$$
$$f_{21} = 2.14 f_{01}$$
$$f_{31} = 2.65 f_{01}$$

These nodes are shown in Figs. 3.4D, 3.4E, and 3.4F. 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. 3.4G, is

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

The frequency of one nodal circle and two nodal diameters, Fig. 3.4H, is

$$f_{22} = 3.50f$$

The frequency of two nodal circles and one nodal diameter, Fig. 3.4I, is

$$f_{13} = 4.22f_{01}$$

The stretched circular membrane is used in the condenser microphone (see Sec. 8.2B). The fundamental resonance frequency is placed at the upper limit of the frequency range. A resistive load is coupled to the diaphragm for damping the response in the neighborhood of the fundamental resonance frequency. This resistance is incorporated in the back plate which serves as the stationary electrode.

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

B. *Square Membrane.*—The fundamental frequency of a square stretched membrane is given by

$$f = \frac{.705}{a} \sqrt{\frac{T}{m}} \qquad 3.11$$

where m = mass, in grams per square centimeter of area,
a = length of a side, in centimeters, and
T = tension, in dynes per centimeter.

C. *Rectangular Membrane.*—The fundamental frequency of a rectangular stretched membrane with the sides in the ratio of 1 to 2 is given by

$$f = \frac{.792}{\sqrt{ab}} \sqrt{\frac{T}{m}} \qquad 3.12$$

where m = mass, in grams per square centimeter,
$a = 2b$ = length of the long side, in centimeters,
b = length of the short side in centimeters, and
T = tension, in dynes per centimeter.

3.5. Circular Plates.[11,12,13,14,15]—The circular plates shown in Fig. 3.5 are assumed to be of uniform cross section and under no tension. It is

[11] Rayleigh, " Theory of Sound," Macmillan and Company, London, 1926.
[12] Morse, " Vibration of Sound," McGraw-Hill Book Company, New York, N.Y., 1936.
[13] Wood, " A Text Book of Sound," Bell and Sons, London, 1930.
[14] Crandall, " Theory of Vibrating Systems and Sound," D. Van Nostrand Company, Princeton, N.J., 1926.
[15] Lamb, " Dynamical Theory of Sound," E. Arnold, London, 1931.

the purpose of this section to consider the vibration of circular plates for the various support means of Fig. 3.5.

A. *Circular Clamped Plate.*—Consider a circular clamped plate as shown in Fig. 3.5A. The fundamental frequency is given by

$$f_{01} = \frac{.467t}{R^2} \sqrt{\frac{Q}{\rho(1 - \sigma^2)}} \qquad 3.13$$

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 (see Table 1.1),

σ = Poisson's ratio (see Table 1.1), and

Q = Young's modulus, in dynes per square centimeter (see Table 1.1).

The fundamental frequency is with the circumference as a node and a maximum displacement at the center (Fig. 3.6A).

The frequency of the next two overtones with nodal circles, Fig. 3.6B and 3.6C, are,

$$f_{02} = 3.91 f_{01}$$
$$f_{03} = 8.75 f_{01}$$

The frequencies of the first, second, and third overtones with nodal diameters are

$$f_{11} = 2.09 f_{01}$$
$$f_{21} = 3.43 f_{01}$$
$$f_{31} = 4.95 f_{01}$$

These nodes are shown in Figs. 3.6D, 3.6E, and 3.6F.

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. 3.6G, is

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

The frequency of one nodal circle and two nodal diameters, Fig. 3.6H, is

$$f_{22} = 8.74 f_{01}$$

The frequency of two nodal circles and one nodal diameter, Fig. 3.6I, is

$$f_{13} = 11.9 f_{01}$$

The clamped plate is used in electromagnetic telephone receivers in which the steel diaphragm serves as the armature (see Sec. 9.2A). It is used in carbon microphones (see Sec. 8.2A). It is used in the subaqueous condenser microphone (Sec. 15.4) and the magnetic subaqueous loud speaker (sec. 15.6). Clamped plate diaphragms have been used in miniature condenser microphones. The disadvantage of a plate is the difficulty of mounting a thin plate to give a small mass per unit area for high sensitivity and still have sufficient stiffness to yield a high fundamental frequency.
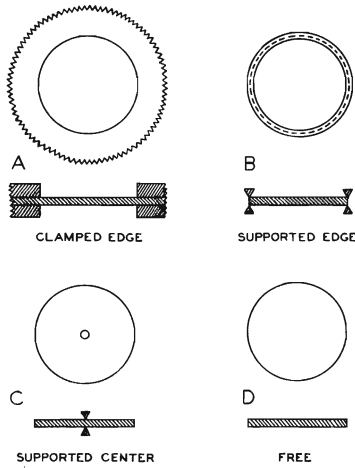
FIG. 3.5. Circular plates. A. A circular plate clamped at the edge. B. A circular plate supported at the edge. C. A circular plate supported at the center. D. A free circular plate.
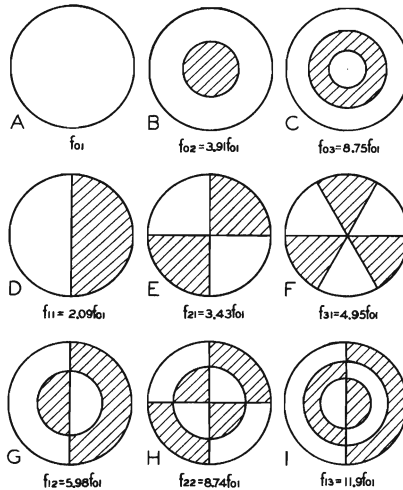
FIG. 3.6. Modes of vibration of a clamped circular plate. Shaded segments are displaced in opposite phase to unshaded.

In telephone receivers, microphones, and loudspeakers employing a clamped diaphragm, the effective mass and effective area of the diaphragm, in terms of the velocity at the center, are needed when the system is reduced to a lumped element representation. The effective mass or effective area for this condition is one third of the total mass or total area of the diaphragm.

The air or water load on the diaphragm can be determined by assuming the effective radius of the equivalent piston to be .55 times the radius of the diaphragm (see Sec. 5.8).

B. *Circular Free Plate.*—Consider a circular plate under no tension, uniform in cross section and perfectly free (Fig. 3.5D). For a vibration with nodal circle, as depicted in Fig. 3.4B, the frequency is

$$f = \frac{.412t}{R^2} \sqrt{\frac{Q}{\rho(1 - \sigma^2)}} \qquad 3.14$$

where t = thickness of the plate, in centimeters,
R = radius of the plate, in centimeters,
ρ = density, in grams per cubic centimeter (see Table 1.1),
σ = Poisson's ratio (see Table 1.1), and
Q = Young's modulus, in dynes per square centimeter (see Table 1.1).

For a vibration with two nodal diameters, as depicted in Fig. 3.4E, the frequency is

$$f = \frac{.193t}{R^2} \sqrt{\frac{Q}{(\rho 1 - \sigma^2)}} \qquad 3.15$$

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. 3.5C). The frequency, for the umbrella mode, is

$$f = \frac{.172t}{R^2} \sqrt{\frac{Q}{\rho(1 - \sigma^2)}} \qquad 3.16$$

D. *Circular Plate Supported at the Outside.*—Consider a plate under no tension, uniform in cross section, edges simply supported at the periphery (Fig. 3.5B). The fundamental frequency is

$$f = \frac{.233t}{R^2} \sqrt{\frac{Q}{\rho(1 - \sigma^2)}} \qquad 3.17$$

3.6. Longitudinal Vibration of Bars.[16,17,18,19]—Consider an entirely free rod of homogeneous material and constant cross section (see Sec. 1.14). 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. 3.7, may be obtained from equation 1.78 as follows,

$$f_1 = \frac{c}{\lambda} = \frac{c}{2l} = \frac{1}{2l} \sqrt{\frac{Q}{\rho}} \qquad 3.18$$

[16] Rayleigh, " Theory of Sound," Macmillan and Company, London, 1926.
[17] Morse, " Vibration and Sound," McGraw-Hill Book Company, New York, N.Y., 1936.
[18] Wood, " A Text Book of Sound," Bell and Sons, London, 1930.
[19] Lamb, " Dynamical Theory of Sound," E. Arnold, London, 1931.

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 (see Table 1.1),

   c = velocity of sound, in centimeters per second (see Table 1.1, and equation 1.78), and

   λ = wavelength of the sound wave, in centimeters.

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. 3.7.
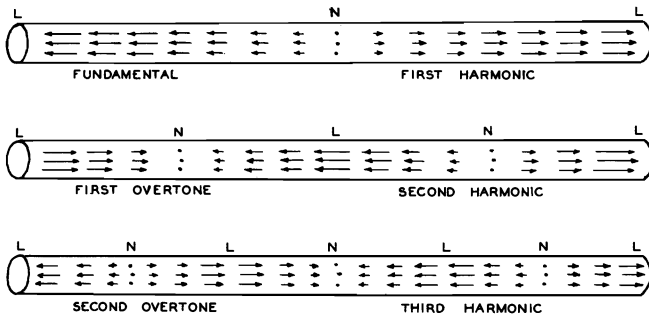
FIG. 3.7. Modes of longitudinal vibrations of a free rod. The nodes and loopes are indicated by N and L.

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, 5000 cycles and above, where a tuning fork is not very satisfactory.

Longitudinal waves in a rod may be set up by electromagnetic, electrostatic, or magnetostriction means. In the first case, if the rod is of magnetic material and is held near an electromagnet in which an alternating current is flowing a longitudinal force will be set up in the rod. If the frequency of the driving current is continuously variable, the rod will be set into violent vibrations at the fundamental resonant frequency. If the plane end of a rod is placed near a metallic disk, the two plane surfaces may be used to serve as plates of a condenser. An alternating current sent through the condenser will cause an alternating force to be exerted upon the end of the rod. The rod will be sent into violent vibrations when the frequency of the impressed alternating current corresponds to the fundamental frequency or one of the overtones. Magnetization of magnetic materials produces small changes in the dimensions of these materials.

A rod of magnetic material placed in a coil of wire will experience a change in length corresponding to the alterations in the actuating current. If the coil is part of the circuit of a vacuum tube oscillator the rod will vibrate and the vacuum tube will oscillate at the fundamental frequency of the rod. Such a system is termed a magnetostriction sonic, ultrasonic, or supersonic generator[20] and may be used to produce sound waves in air or any other medium (see Secs. 15.7 and 15.8).

3.7. Torsional Vibration of Bars.[21,22]—A solid bar or tube may be twisted about the axis of the rod in such a manner that each transverse section remains in its own plane (see Sec. 1.15). If the section is not circular
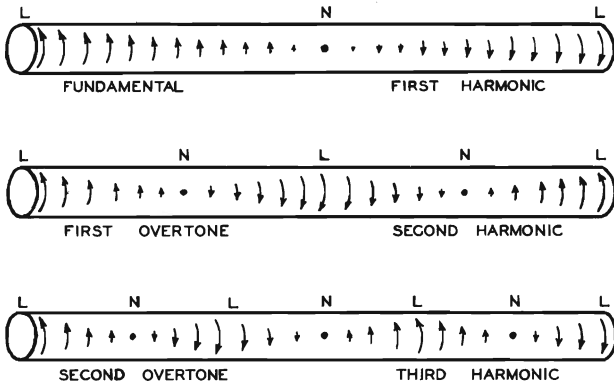
FIG. 3.8. Modes of torsional vibration of a free rod. The nodes and loops are indicated by N and L.

there will be motion parallel to the axis of the bar. Consider an entirely free rod of homogeneous material and circular cross section. The simplest or fundamental mode of torsional vibration occurs when there is a node in the middle and a loop at each end, that is, when the length of the rod is one-half wavelength. The fundamental resonant frequency, Fig. 3.8, may be obtained from equation 1.79, as follows

$$f_1 = \frac{c}{\lambda} = \frac{c}{2l} = \frac{1}{2l}\sqrt{\frac{Q}{2\rho(\sigma + 1)}} \qquad 3.19$$

where l = length of the rod, in centimeters,

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

Q = Young's modulus, in dynes per square centimeter (see Table 1.1),

σ = Poisson's ratio (see Table 1.1),

c = velocity of propagation of torsional waves, in centimeters per second, see equation 1.79, and

λ = wavelength of the torsional wave, in centimeters.

[20] Pierce, G. W., *Proc. Am. Acad. Arts and Sci.*, Vol. 63, p. 1, 1928.
[21] Wood, " A Text Book of Sound," Bell and Sons, London, 1930.
[22] Rayleigh, " Theory of Sound," Macmillan and Company, London, 1926.

The overtones, as in the case of longitudinal vibrations, are harmonics of the fundamental. That is, $f_2 = 2f_1$, $f_3 = 3f_1$, $f_4 = 4f_1$, etc. The nodes and antinodes for the various harmonics are formed as in the case of longitudinal vibrations.

Torsional vibrations may be set up in bars by any means which applies tangential forces to the free end. From a comparison of the longitudinal and torsional vibrations in the same bar, Poisson's ratio may be determined.

3.8. Open and Closed Pipes.—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. 3.9, is

$$f = \frac{c}{\lambda} = \frac{c}{2l} \qquad\qquad 3.20$$

where l = length of the pipe, in centimeters,

$\quad c$ = velocity of sound, in centimeters per second (see Table 1.1), and

$\quad \lambda$ = wavelength, in centimeters.
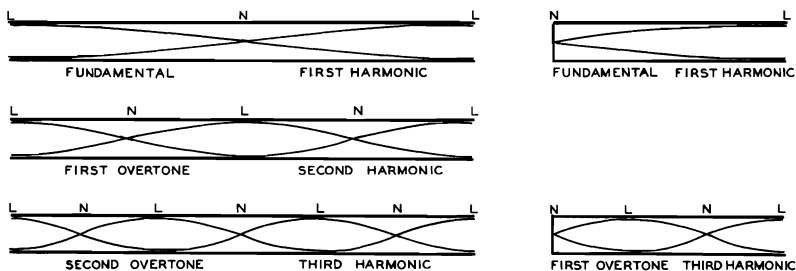
FIG. 3.9. Modes of vibration of the air column in a pipe open at both ends and in a pipe closed at one end and open at the other end. The velocity nodes and loops are indicated by N and L.

The 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.

The fundamental resonant frequency of a pipe closed at one end and open at the other end, Fig. 3.9, is

$$f = \frac{c}{\lambda} = \frac{c}{4l} \qquad\qquad 3.21$$

The overtones of the pipe closed at one end are the odd harmonics. That is $f_2 = 3f_1$, $f_3 = 5f_1$, etc.

In the above examples the end connection has been omitted. Rayleigh[23]

[23] Rayleigh, " Theory of Sound," Macmillan and Company, London, 1926.

shows the added length at the open end to be .82R where R is the radius of the pipe.   If the pipe is terminated in a large flange the end connection will be that given in Sec. 5.12.

Organ pipes and whistles have been built to cover the range from 16 cycles to 30,000 cycles.   The frequency of open and closed pipes may be computed from the above equations.   The sound vibrations in the pipe are set up by the stream of air which is controlled by the vibration in the pipe.   It is an oscillatory system fed by a direct current of air or gas.

4

DYNAMICAL ANALOGIES

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

A large part of engineering analysis 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 equations of electrical circuit theory may be based on Maxwell's dynamical theory in which the currents play the role of velocities. Expressions for the kinetic energy, potential energy, and dissipation show that network equations are deducible from general dynamic equations. In other words, an electrical circuit may be considered to be a vibrating system. This immediately suggests analogies between electrical circuits and other dynamical systems as, for example, mechanical and acoustical vibrating systems.

The equations of motion of mechanical systems were developed a long time before any attention was given to equations for electrical circuits. For this reason, in the early days of electrical circuit theory, it was natural to explain the action in terms of mechanical phenomena. However, at the present time, electrical circuit theory has been developed to a much higher state than the corresponding theory of mechanical systems. The number of engineers and scientists versed in electrical circuit theory is many times the number equally familiar with mechanical systems.

Almost any work involving mechanical or acoustical systems also includes electrical systems and electrical circuit theory. The acoustical engineer is interested in sound reproduction or the conversion of electrical or mechanical energy into acoustical energy, the development of vibrating systems, and the control of sound vibrations. This involves acoustical, electroacoustical, mechanoacoustical, or electromechanoacoustical systems. The mechanical engineer is interested in the development of various mechanisms or vibrating systems involving masses, springs, and friction.

71

Electrical circuit theory is the branch of electromagnetic theory which deals with electrical oscillations in linear electrical networks.[1] An electrical network is a connected set of separate circuits termed branches or meshes. A circuit may be defined as a physical entity in which varying magnitudes may be specified in terms of time and a single dimension.[2] The branches or meshes are composed of elements. Elements are the constituent parts of a circuit. Electrical elements are resistance, inductance, and capacitance. Vibrations in one dimension occur in mechanical systems made up of mechanical elements, as, for example, various assemblies of masses, springs, and brakes. Confined acoustical systems in which the dimensions are small compared to the wavelength are vibrations in a single dimension.

The number of independent variables required to completely specify the motion of every part of a vibrating system is a measure of the number of degrees of freedom of the system. If only a single variable is needed the system is said to have a single degree of freedom. In an electrical circuit the number of degrees of freedom is equal to the number of independent closed meshes or circuits.

The use of complex notation has been applied extensively to electrical circuits. Of course, this operational method can be applied to any analytically similar system.

Mathematically the elements in an electrical network are the coefficients in the differential equations describing the network. When the electric circuit theory is based upon Maxwell's dynamics, the network forms a dynamical system in which the currents play the role of velocities. In the same way the coefficients in the differential equations of a mechanical or acoustical system may be looked upon as mechanical or acoustical elements. Kirchhoff's electromotive force law plays the same role in setting up the electrical equations as D'Alembert's principle does in setting up the mechanical and acoustical equations. That is to say, every electrical, mechanical, or acoustical system may be considered as a combination of electrical, mechanical, or acoustical elements. Therefore, any mechanical or acoustical system may be reduced to an electrical network and the problem may be solved by electrical circuit theory.

In view of the tremendous amount of study which has been directed toward the solution of circuits, particularly electrical circuits, and the engineer's familiarity with electrical circuits, it is logical to apply this knowledge to the solution of vibration problems in other fields by the same theory as that used in the solution of electrical circuits.

It is the purpose of this chapter to develop the analogies between elements[3] in electrical, mechanical, and acoustical systems.

---

[1] The use of the terms " circuit " and " network " in the literature is not established. The term " circuit " is often used to designate a network with several branches.

[2] The term " single dimension " implies that the movement or variation occurs along a path. In a field problem there is a variation in two or three dimensions.

[3] For further considerations of analogies see Olson, " Dynamical Analogies," D. Van Nostrand Company, Princeton, N.J., 1943.

4.2. Definitions.—A few of the terms used in dynamical analogies will be defined in this section. Terms not listed below will be defined in subsequent sections.

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 cycle. The unit is the dyne.

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

Dyne Centimeter—A dyne centimeter is the unit of torque or rotatomotive force.

Instantaneous Torque (Instantaneous Rotatomotive Force)—The instantaneous torque at a point is the total instantaneous torque. The unit is the dyne centimeter.

Effective Torque (Effective Rotatomotive Force)—The effective torque is the root mean square of the instantaneous torque over a complete cycle. The unit is the dyne centimeter.

Maximum Torque (Maximum Rotatomotive Force)—The maximum torque for any given cycle is the maximum absolute value of the instantaneous torque during that cycle. The unit is the dyne centimeter.

Peak Torque (Peak Rotatomotive Force)—The peak torque for a specified interval is the maximum absolute value of the instantaneous torque during that interval. The unit is the dyne centimeter.

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.

Radian per Second—A radian per second is the unit of angular velocity.

Instantaneous Angular Velocity—The instantaneous angular velocity at a point is the total instantaneous angular velocity at that point. The unit is the radian per second.

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

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

Peak Angular Velocity—The peak angular velocity for any specified time interval is the maximum absolute value of the instantaneous angular velocity in that interval. The unit is the radian 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 Rectilineal Impedance[4] (Mechanical Impedance)—Mechanical rectilineal impedance is the complex quotient of the alternating force

---

[4] The word " mechanical " is ordinarily used as a modifier to designate a mechanical system with rectilineal displacements and the word " rotational " is ordinarily used as a modifier to designate a mechanical system with rotational displacements. To avoid ambiguity in this book, where both systems are considered concurrently, the words " mechanical rectilineal " are used as modifiers to designate a mechanical system with rectilineal displacements and the words " mechanical rotational " are used as modifiers to designate a mechanical system with rotational displacements.

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 Rectilineal Resistance (Mechanical Resistance)—Mechanical rectilineal resistance is the real part of the mechanical rectilineal impedance. This is the part responsible for the dissipation of energy.   The unit is the mechanical ohm.

Mechanical Rectilineal Reactance (Mechanical Reactance)—Mechanical rectilineal reactance is the imaginary part of the mechanical rectilineal 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 rectilineal 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 rectilineal impedance.   The unit is the centimeter per dyne.

Mechanical Rotational Impedance (Rotational Impedance)—Mechanical rotational impedance is the complex quotient of the alternating torque applied to the system by the resulting angular velocity in the direction of the torque at its point of application.   The unit is the rotational ohm.

Mechanical Rotational Resistance (Rotational Resistance)—Mechanical rotational resistance is the real part of the mechanical rotational impedance. This is the part responsible for the dissipation of energy.   The unit is the rotational ohm.

Mechanical Rotational Reactance (Rotational Reactance)—Mechanical rotational reactance is the imaginary part of the mechanical rotational impedance.   The unit is the rotational ohm.

Moment of Inertia—Moment of inertia in a mechanical rotational system is that coefficient which, when multiplied by 2π times the frequency, gives the positive imaginary part of the mechanical rotational impedance.   The unit is the gram centimeter to the second power.

Rotational Compliance—Rotational compliance in a mechanical rotational system is that coefficient which, when multiplied by 2π times the frequency, is the reciprocal of the negative imaginary part of the mechanical rotational impedance.   The unit is the radian per 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 rectilineal, mechanical rotational, 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 rectilineal system are mechanical rectilineal resistance, mass, and compliance. The elements in a mechanical rotational system are mechanical rotational resistance, moment of inertia, and rotational 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 electrical elements: electrical resistance, inductance, and electrical capacitance.

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

Mechanical Rotational System—A mechanical rotational system is a system adapted for the transmission of rotational vibrations consisting of one or all of the following mechanical rotational elements: mechanical rotational resistance, moment of inertia, and rotational 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 rectilineal 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.

Rotational Ohm—A mechanical rotational resistance, reactance, or impedance is said to have a magnitude of one rotational ohm when a torque of one dyne centimeter produces an angular velocity of one radian 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.

4.3. Elements.[5]—An element or circuit parameter in an electrical system

[5] The symbols used in this book conform with American Standards Association, "Letter Symbols for Acoustics," Y10.11–1953.

defines a distinctive activity in its part of the circuit. In an electrical system these elements are resistance, inductance, and capacitance. They are distinguished from the devices: resistor, inductor, and capacitor. A resistor, inductor, or capacitor idealized to have only resistance, inductance, and capacitance is a circuit element. As indicated in the preceding chapter, the study of mechanical and acoustical systems is facilitated by the introduction of elements analogous to the elements of an electric circuit. In this procedure, the first step is to develop the elements in these vibrating systems. It is the purpose of this chapter to define and describe electrical, mechanical rectilineal, mechanical rotational, and acoustical elements.

4.4. Resistance.—A. *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

$$r_E = \frac{e}{i} \qquad\qquad 4.1$$

where e = voltage across the electrical resistance, in abvolts, and
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.

B. *Mechanical Rectilineal Resistance.*—Mechanical rectilineal 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 rectilineal resistance is displaced a distance x by a force f_M.

Mechanical rectilineal resistance (termed mechanical resistance) r_M, in mechanical ohms, is defined as

$$r_M = \frac{f_M}{u} \qquad\qquad 4.2$$

where f_M = applied mechanical force, in dynes, and
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 rectilineal resistance is proportional to the mechanical rectilineal resistance and the linear velocity.

C. *Mechanical Rotational Resistance.*—Mechanical rotational energy is changed into heat by a rotational motion which is opposed by a rotational resistance (rotational friction). Energy is lost by the system when a mechanical rotational resistance is displaced by an angle ϕ by a torque f_R.

Mechanical rotational resistance (termed rotational resistance) r_R, in rotational ohms, is defined as

$$r_R = \frac{f_R}{\theta}$$ 4.3

where f_R = applied torque, in dyne centimeters, and
θ = angular velocity at the point of application about the axis, in radians per second.

Equation 4.3 states that the driving torque applied to a mechanical rotational resistance is proportional to the mechanical rotational resistance and the angular velocity.

D. *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 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

$$r_A = \frac{p}{U}$$ 4.4

where p = pressure, in dynes per square centimeter, and
U = volume current, in cubic centimeters per second.

Equation 4.4 states that the driving pressure applied to an acoustical resistance is proportional to the acoustical resistance and the volume current.

4.5. Inductance, Mass, Moment of Inertia, Inertance.—A. *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 abhenries, is defined as

$$e = L \frac{di}{dt}$$ 4.5

where e = electromotive or driving force, in abvolts, and
di/dt = rate of change of current, in abamperes per second.

Equation 4.5 states that the electromotive force across an inductance is proportional to the inductance and the rate of change of current.

B. *Mass.*—Mechanical rectilineal inertial energy is associated with mass in the mechanical rectilineal system. Mechanical rectilineal 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

$$f_M = m\frac{du}{dt} \qquad\qquad 4.6$$

where du/dt = acceleration, in centimeters per second per second, and
   f_M = driving force, in dynes.

Equation 4.6 states that the driving force applied to the mass is proportional to the mass and the rate of change of velocity.

C. *Moment of Inertia.*—Mechanical rotational inertial energy is associated with moment of inertia in the mechanical rotational system. Mechanical rotational energy increases as the angular velocity of a moment of inertia increases, that is, during angular acceleration. It decreases when the angular velocity decreases. It remains a constant when the angular velocity is a constant. Moment of inertia is the rotational element which opposes a change in angular velocity. Moment of inertia I, in gram (centimeter)^4, is given by

$$f_R = I\frac{d\theta}{dt} \qquad\qquad 4.7$$

where $d\theta/dt$ = angular acceleration, in radians per second per second, and
   f_R = torque, in dyne centimeters.

Equation 4.7 states that the driving torque applied to the moment of inertia is proportional to the moment of inertia and the rate of change of angular velocity.

D. *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

$$p = M\frac{dU}{dt} \qquad\qquad 4.8$$

where M = inertance, in grams per (centimeter)^4,
   dU/dt = rate of change of volume current, in cubic centimeters per second per second, and
   p = driving pressure, in dynes per square centimeter.

Equation 4.8 states that the driving pressure applied to an inertance is proportional to the inertance and the rate of change of volume current.

Inertance[6] may be expressed as

$$M = \frac{m}{S^2} \qquad\qquad 4.9$$

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

[6] See Sec. 5.6.

The inertance of a circular tube is

$$M = \frac{\rho l}{\pi R^2} \qquad\qquad 4.10$$

where R = radius of the tube, in centimeters,

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

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

4.6. Electrical Capacitance, Rectilineal Compliance, Rotation Compliance, Acoustical Capacitance.—A. *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 charges 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 released. Electrical capacitance is the electrical circuit element which opposes a change in voltage. Electrical capacitance C_E, in abfarads, is defined as

$$i = C_E \frac{de}{dt} \qquad\qquad 4.11$$

Equation 4.11 may be written

$$e = \frac{1}{C_E} \int i\, dt = \frac{q}{C_E} \qquad\qquad 4.12$$

where q = charge on electrical capacitance, in abcoulombs, and

e = electromotive force, in abvolts.

Equation 4.12 states that the charge on an electrical capacitance is proportional to the electrical capacitance and the applied electromotive force.

B. *Rectilineal Compliance.*—Mechanical rectilineal 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 a constant, and is stored, when the spring remains immovably compressed. Rectilineal compliance is the mechanical element which opposes a change in the applied force. Rectilineal compliance C_M (termed compliance) is defined as

$$f_M = \frac{x}{C_M} \qquad\qquad 4.13$$

where x = displacement, in centimeters, and

f_M = applied force, in dynes.

Equation 4.13 states that the displacement of a compliance is proportional to the compliance and the applied force.

Stiffness is the reciprocal of compliance.

C. *Rotational Compliance.*—Mechanical rotational potential energy is associated with the twisting of a spring or compliant element. Mechanical energy increases as the spring is twisted. It decreases as the spring is allowed to unwind. It is constant, and is stored when the spring remains immovably twisted. Rotational compliance is the mechanical element which opposes a change in the applied torque. Rotational compliance C_R, or moment of compliance, is defined as

$$f_R = \frac{\phi}{C_R} \qquad 4.14$$

where ϕ = angular displacement, in radians, and
  f_R = applied torque, in dyne centimeters.

Equation 4.14 states that the rotational displacement of the rotational compliance is proportional to the rotational compliance and the applied torque.

D. *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, in dynes per square centimeter, in terms of the condensation, is from equation 1.21

$$p = c^2 \rho s \qquad 4.15$$

where c = velocity, in centimeters per second,
  ρ = density, in grams per cubic centimeter, and
  s = condensation, defined in equation 4.16.

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

$$s = \frac{V - V'}{V} \qquad 4.16$$

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

$$V - V' = X \qquad 4.17$$

where X = volume displacement, in cubic centimeters.
From equations 4.15, 4.16, and 4.17, the pressure is

$$p = \frac{\rho c^2}{V} X \qquad 4.18$$

Acoustical capacitance C_A is defined as

$$p = \frac{X}{C_A} \qquad 4.19$$

where p = sound pressure in dynes per square centimeter, and
X = volume displacement, in cubic centimeters.

Equation 4.19 states the volume displacement in an acoustical capacitance is proportional to the pressure and the acoustical capacitance.

From equations 4.18 and 4.19 the acoustical capacitance of a volume is

$$C_A = \frac{V}{\rho c^2} \qquad\qquad 4.20$$

where V = volume, in cubic centimeters.
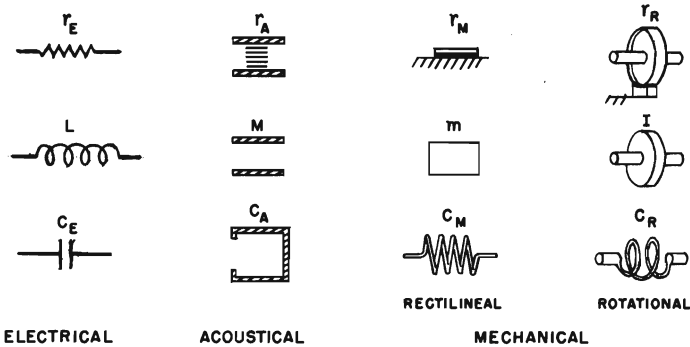
FIG. 4.1. Graphical representation of the three basic elements in electrical, mechanical rectilineal, mechanical rotational, and acoustical systems.

r_E = electrical resistance r_A = acoustical resistance r_M = mechanical rectilineal resistance

r_R .. mechanical rotational resistance L = inductance M = inertance

m .. mass I = moment of inertia C_E = electrical capacitance

C_A = acoustical capacitance C_M = compliance C_R = rotational compliance

4.7. Representation of Electrical, Mechanical Rectilineal, Mechanical Rotational, and Acoustical Elements.—Electrical, mechanical rectilineal, mechanical rotational, and acoustical elements have been defined in the preceding sections. Fig. 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.

Mechanical rectilineal resistance is represented by sliding friction which causes dissipation. Mechanical rotational resistance is represented by a

wheel with a sliding friction brake 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 rectilineal system is represented by a mass. Moment of inertia in the mechanical rotational system is represented by a flywheel. 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 rectilineal system is represented as a spring. Rotational compliance in the mechanical rotational 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.

In the preceding discussion of electrical, mechanical rectilineal, mechanical rotational, and acoustical systems it was observed that the four systems are analogous. As pointed out in the introduction, using the dynamical concept for flow of electrical currents in electrical circuits the fundamental laws are of the same nature as those which govern the dynamics of a moving body. In general, the three fundamental dimensions are mass, length, and time. These quantities are directly connected to the mechanical rectilineal system. Other quantities in the mechanical rectilineal system may be derived in terms of these dimensions. In terms of analogies the dimensions in the electrical circuit corresponding to length, mass, and time in the mechanical rectilineal system are charge, self-inductance, and time. The corresponding analogous dimensions in the rotational mechanical system are angular displacement, moment of inertia, and time. The corresponding analogous dimensions in the acoustical system are volume displacement, inertance, and time. The above-mentioned fundamental dimensions in each of the four systems are shown in tabular form in Table 4.1. Other quantities in each of the four systems may be expressed in terms of the dimensions of Table 4.1. A few of the most important quantities have been tabulated in Table 4.2. Tables 4.1 and 4.2 depict analogous quantities in each of the four systems. Further, they show that the four systems are dynamically analogous.

The dimensions given in Table 4.1 should not be confused with the classical dimensions of electrical, mechanical, and acoustical systems given in Table 4.3. Table 4.3 uses mass M, length L, and time T. In the case of electrical units dielectric and permeability constants are assumed to be dimensionless.

For further considerations of dynamical analogies, as, for example, electrical, mechanical rectilineal, mechanical rotational, and acoustical systems of one, two, and three degrees of freedom, corrective networks, wave filters, transients, driving systems, generating systems, theorems and applications, the reader is referred to Olson, "Dynamical Analogies," D. Van Nostrand Company, Princeton, N.J., 1943.

TABLE 4.1

Electrical		Mechanical Rectilineal		Mechanical Rotational		Acoustical	
Quantity	Symbol	Quantity	Symbol	Quantity	Symbol	Quantity	Symbol
Self-Inductance	L	Mass	m	Moment of Inertia	I	Inertance	M
Electrical Charge	q	Linear Displacement	x	Angular Displacement	ϕ	Volume Displacement	X
Time	t	Time	t	Time	t	Time	t

TABLE 4.2

Electrical			Mechanical Rectilineal			Mechanical Rotational			Acoustical		
Quantity	Symbol	Dimension	Quantity	Symbol	Dimension	Quantity	Symbol	Dimension	Quantity	Symbol	Dimension
Current	i	qt^{-1}	Linear Velocity	$\dot{x}$ or v	xt^{-1}	Angular Velocity	$\dot{\phi}$ or θ	ϕt^{-1}	Volume Current	$\dot{X}$ or U	Xt^{-1}
Electromotive Force	e	Lqt^{-2}	Force	f_M	mxt^{-2}	Torque	f_R	$I\phi t^{-2}$	Pressure	p	MXt^{-2}
Electrical Resistance	r_E	Lt^{-1}	Mechanical Resistance	r_M	mt^{-1}	Rotational Resistance	r_R	It^{-1}	Acoustical Resistance	r_A	Mt^{-1}
Electrical Capacitance	C_E	$L^{-1}t^2$	Compliance	C_M	$m^{-1}t^2$	Rotational Compliance	C_R	$I^{-1}t^2$	Acoustical Capacitance	C_A	$M^{-1}t^2$
Energy	W_E	Lq^2t^{-2}	Energy	W_M	mx^2t^{-2}	Energy	W_R	$I\phi^2t^{-2}$	Energy	W_A	MX^2t^{-2}
Power	P_E	Lq^2t^{-3}	Power	P_M	mx^2t^{-3}	Power	P_R	$I\phi^2t^{-3}$	Power	P_A	MX^2t^{-3}

TABLE 4.3

Electrical				Mechanical Rectilineal			
Quantity	Unit	Symbol	Dimension	Quantity	Unit	Symbol	Dimension
Electromotive Force	Volts $\times$ 10^8	e	$M^{1/2}L^{3/2}T^{-2}$	Force	Dynes	f_M	MLT^{-2}
Charge or Quantity	Coulombs $\times$ 10^{-1}	q	$M^{1/2}L^{1/2}$	Linear Displacement	Centimeters	x	L
Current	Amperes $\times$ 10^{-1}	i	$M^{1/2}L^{1/2}T^{-1}$	Linear Velocity	Centimeters per Second	$\dot{x}$ or v	LT^{-1}
Electrical Impedance	Ohms $\times$ 10^9	z_E	LT^{-1}	Mechanical Impedance	Mechanical Ohms	z_M	MT^{-1}
Electrical Resistance	Ohms $\times$ 10^9	r_E	LT^{-1}	Mechanical Resistance	Mechanical Ohms	r_M	MT^{-1}
Electrical Reactance	Ohms $\times$ 10^9	x_E	LT^{-1}	Mechanical Reactance	Mechanical Ohms	x_M	MT^{-1}
Inductance	Henries $\times$ 10^9	L	L	Mass	Grams	m	M
Electrical Capacitance	Farads $\times$ 10^{-9}	C_E	$L^{-1}T^2$	Compliance	Centimeters per Dyne	C_M	$M^{-1}T^2$
Power	Ergs per Second	P_E	ML^2T^{-3}	Power	Ergs per Second	P_M	ML^2T^{-3}

TABLE 4.3—*Continued*.

Mechanical Rotational				Acoustical			
Quantity	Unit	Symbol	Dimension	Quantity	Unit	Symbol	Dimension
Torque	Dyne Centimeter	f_R	ML^2T^{-2}	Pressure	Dynes per Square Centimeter	p	$ML^{-1}T^{-2}$
Angular Displacement	Radians	ϕ	1	Volume Displacement	Cubic Centimeters	X	L^3
Angular Velocity	Radians per Second	$\dot{\phi}$ or θ	T^{-1}	Volume Current	Cubic Centimeter per Second	$\dot{X}$ or U	L^3T^{-1}
Rotational Impedance	Rotational Ohms	z_R	ML^2T^{-1}	Acoustical Impedance	Acoustical Ohms	z_A	$ML^{-4}T^{-1}$
Rotational Resistance	Rotational Ohms	r_R	ML^2T^{-1}	Acoustical Resistance	Acoustical Ohms	r_A	$ML^{-4}T^{-1}$
Rotational Reactance	Rotational Ohms	x_R	ML^2T^{-1}	Acoustical Reactance	Acoustical Ohms	x_A	$ML^{-4}T^{-1}$
Moment of Inertia	(Gram) (Centimeter)^2	I	ML^2	Inertance	Grams per (Centimeter)^4	M	ML^{-4}
Rotational Compliance	Radians per Dyne per Centimeter	C_R	$M^{-1}L^{-2}T^2$	Acoustical Capacitance	(Centimeter)^5 per Dyne	C_A	$M^{-1}L^4T^2$
Power	Ergs per Second	P_R	ML^2T^{-3}	Power	Ergs per Second	P_A	ML^2T^{-3}

5

ACOUSTICAL ELEMENTS

5.1. Introduction.—The preceding chapter is devoted to analogies between electrical, mechanical, and acoustical systems. The purpose of drawing these analogies is to facilitate the solution of problems in mechanical and acoustical vibrating systems by converting these problems into the corresponding electrical analogies and solving the resultant electrical circuits by conventional electrical circuit theory. An electrical circuit is composed of electrical elements. In the same way the acoustical system is composed of acoustical elements. The type of element, that is, acoustical resistance, inertance, or acoustical capacitance, will depend upon the characteristic manner in which the medium behaves for different sources of sound and in the different ways of confining the medium. It is the purpose of this chapter to consider acoustical elements and combination of elements.

5.2. Acoustical Resistance.—Acoustical resistance may be obtained by forcing air through a small hole. The resistance is due to viscosity which may be considered as friction between adjacent layers of air. In the ordinary transmission of sound in a large tube the motion of all the particles in a plane normal to the axis is the same, therefore the frictional losses are small. When sound travels in a small tube the particle velocity varies from zero at the boundary to a maximum at the center. The same is true when a steady stream of air is forced through a small hole or tube, the velocity of adjacent layers varies from zero at the boundary to a maximum at the center. The smaller the hole the higher will be the resistance because of the greater effect of the sides.

A small tube also has inertance. Therefore, the reactive component increases with frequency. The inertive reactance increases as the size of the hole decreases as does the acoustical resistance, but at a slower rate. Therefore, the inertive reactance may be made negligible compared to the acoustical resistance if the hole is made sufficiently small.

Acoustical resistance employing viscosity may be made in various forms as, for example, a large number of small holes or a large number of slits. The acoustical impedance of fine holes and slits will be considered in the next two sections.

5.3. Acoustical Impedance of a Tube of Small Diameter.—The transmission of sound waves or direct currents of air in a small tube is influenced

by acoustical resistance due to viscosity. The diameter is assumed to be small compared to the length so that the end correction may be neglected. The length is assumed to be small compared to the wavelength.

The acoustical impedance, in acoustical ohms, of a small-diameter tube[1,2,3] is given by

$$z_A = \frac{l}{\pi R^2}\left(\frac{8\mu}{R^2} + \frac{4}{3}j\omega\rho\right) \qquad 5.1$$

where R = radius of the tube, in centimeters,

  μ = viscosity coefficient, 1.86×10^{-4} for air,

  $\omega = 2\pi f, f$ = frequency, in cycles per second,

  l = length of the tube, in centimeters, and

  ρ = density, in grams per cubic centimeter.

The effect of viscosity is to introduce acoustical resistance in the form of dissipation as well as to add to the acoustical reactance.

The acoustical resistance of a single hole is ordinarily much too high. The desired acoustical resistance may be obtained by using a sufficient number of holes.

5.4. Acoustical Impedance of a Narrow Slit.—A narrow slit acts in a manner quite similar to the narrow tube. The length is assumed to be small compared to the wavelength. The thickness is assumed to be small compared to the length.

The acoustical impedance, in acoustical ohms, of a narrow slit[4,5,6] is given by

$$z_A = \frac{12\mu w}{d^3 l} + j\frac{6\rho w \omega}{5ld} \qquad 5.2$$

where μ = viscosity coefficient, 1.86×10^{-4} for air,

  ρ = density, in grams per cubic centimeter,

  d = thickness of the slit normal to the direction of flow, in centimeters,

  l = width of the slit normal to the direction of flow, in centimeters,

  w = length of the slit in the direction of the flow, in centimeters,

  $\omega = 2\pi f$, and

  f = frequency, in cycles per second.

[1] Crandall, " Vibrating Systems and Sound," D. Van Nostrand Company, Princeton, N.J., 1926.

[2] Lamb, " Dynamical Theory of Sound," E. Arnold, London, 1931.

[3] Rayleigh, " Theory of Sound," Macmillan and Company, London, 1926.

[4] Crandall, " Vibrating Systems and Sound," D. Van Nostrand Company, Princeton, N.J., 1926.

[5] Lamb, " Dynamical Theory of Sound," E. Arnold, London, 1931.

[6] Rayleigh, " Theory of Sound," Macmillan and Company, London, 1926.

In equation 5.2 the acoustical resistance varies inversely as the cube of d and the inertance inversely as d. Therefore, practically any ratio of inertance to acoustical resistance may be obtained. The magnitude may be obtained by a suitable choice of w and l. A slit type of acoustical resistance may be formed by using a pile of washers spaced by small shims. Another form consists of a spiral of tape with adjacent turns very close together.

5.5. Acoustical Resistance of Silk Cloth.—Silk cloth provides a simple means of obtaining an acoustical resistance. The magnitude of the acoustical resistance is governed by the size and nature of the holes in the material
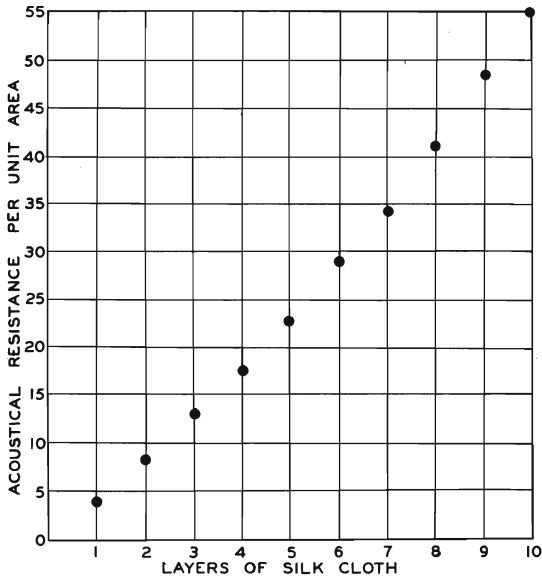
FIG. 5.1. The acoustical resistance, per square centimeter, of sheer silk cloth as a function of the number of layers.

and the number of layers of the cloth. The acoustical resistance of sheer silk cloth as a function of the number of layers of the material is shown in Fig. 5.1. As in the case of the small tube and narrow slit, the ratio of acoustical resistance to inertance is governed by the size of the holes (see equations 5.1 and 5.2).

Silk cloth has been used as an acoustical resistance element in microphones, telephone receivers, and loudspeakers for many years. The structural simplicity and the high ratio of acoustical resistance to inertance make it particularly desirable as an acoustical resistance.

5.6. Inertance.—Inertance is defined, in Sec. 4.5D, as

$$M = \frac{\text{mass}}{S^2} \qquad 5.3$$

where S = area, in square centimeters, over which the driving pressure acts to drive the mass, in grams.

The acoustical impedance of various types of systems will be considered in Secs. 5.8, 5.9, 5.10, 5.11, 5.12, 5.13 and 5.14. The imaginary part of these expressions is due to the inertance of the systems.

For closed systems the acoustical resistance term of Secs. 5.8, 5.9, 5.10, 5.11, 5.12, 5.13 and 5.14 should be omitted because there is no radiation. In this case the entire acoustical impedance is positive acoustical reactance. The acoustical reactance term of equations 5.1 and 5.2 is due to inertance.

5.7. Acoustical Capacitance.—The most common type of acoustical capacitance used in acoustical systems consists of a cavity or volume with rigid boundaries. The linear dimensions of the enclosure are assumed to be small compared to the wavelength.

For equation 1.21 the sound pressure is

$$p = \rho c^2 s \qquad 5.4$$

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

c = velocity of sound, in centimeters per second, and

s = condensation.

The condensation, from Sec. 1.3D, is

$$s = \frac{dV}{V} \qquad 5.5$$

where dV is the change in the original volume V.

$$dV = Sx = X \qquad 5.6$$

where x = displacement, in centimeters, over the area S, in square centimeters, and

X = volume displacement, in cubic centimeters.

From equations 5.4, 5.5, and 5.6.

$$\frac{X}{p} = \frac{V}{\rho c^2} \qquad 5.7$$

The ratio X/p is termed the acoustical capacitance by definition (see Sec. 4.6D). Therefore the acoustical capacitance of a volume is

$$C_A = \frac{V}{\rho c^2} \qquad 5.8$$

The next consideration will be an acoustical capacitance combined with an acoustical resistance. The acoustical impedance of a cavity in which

the boundaries or a portion of the boundary is terminated in an acoustical resistance is

$$z_A = \frac{r_A}{1 + j\omega r_A C_A} \qquad 5.9$$

where r_A = acoustical resistance of the boundary, in acoustical ohms,

$\quad C_A$ = acoustical capacitance of the volume, in cubic centimeters per second,

$\quad \omega = 2\pi f$, and

$\quad f$ = frequency, in cycles per second.

5.8. Mechanical and Acoustical Impedance Load upon a Vibrating Piston.[7,8,9]—The mechanical impedance, in mechanical ohms, of the air load upon one side of a vibrating piston set in an infinite baffle is

$$z_M = \pi R^2 \rho c \left[1 - \frac{J_1(2kR)}{kR} \right] + j \frac{\pi \omega \rho}{2k^3} K_1(2kR) \qquad 5.10$$

where R = radius of piston, in centimeters,

$\quad \rho$ = density, in grams per cubic centimeter,

$\quad c$ = velocity of sound, in centimeters per second,

$\quad k = 2\pi/\lambda$,

$\quad \lambda$ = wavelength, in centimeters,

$\quad \omega = 2\pi f$, and

$\quad f$ = frequency, in cycles per second.

J_1 and K_1 may be found in treatises[10,11] on Bessel functions. They are also defined by the series,

$$1 - \frac{J_1(2kR)}{kR} = \frac{k^2 R^2}{2} - \frac{k^4 R^4}{2^2 \cdot 3} + \frac{k^6 R^6}{2^2 \cdot 3^2 \cdot 4} \cdots$$

$$K_1(2kR) = \frac{2}{\pi} \left[\frac{(2kR)^3}{3} - \frac{(2kR)^5}{3^2 \cdot 5} + \frac{(2kR)^7}{3^2 \cdot 5^2 \cdot 7} \cdots \right] \qquad 5.11$$

The acoustical impedance, in acoustical ohms, of the air load upon one side of a vibrating piston in an infinite baffle is

$$z_A = \frac{\rho c}{\pi R^2} \left[1 - \frac{J_1(2kR)}{kR} \right] + \frac{j\omega \rho}{2\pi R^4 k^3} K_1(2kR) \qquad 5.12$$

The acoustical impedance per unit area of the piston is

$$z_1 = \rho c \left[1 - \frac{J_1(2kR)}{kR} \right] + \frac{j\omega \rho}{2R^2 k^3} K_1(2kR) \qquad 5.13$$

[7] Rayleigh, " Theory of Sound," Macmillan and Company, London, 1926.

[8] Crandall, " Vibrating Systems and Sound," D. Van Nostrand Company, Princeton, N.J., 1926.

[9] Stewart and Lindsay, " Acoustics," D. Van Nostrand Company, Princeton, N.J., 1930.

[10] Watson, " Theory of Bessel Functions," Cambridge Press, London, 1922.

[11] Jahnke and Emde, " Tables of Function," Teubner, Berlin, 1928.

The resistive and reactive acoustical impedance components of the air load per unit area on one side of a vibrating piston set in an infinite baffle are shown in Fig. 5.2. These characteristics are useful in determining the radiation resistance and reactive component of the air load on the cone in a

FIG. 5.2. The acoustical resistance, r_1, and the acoustical reactance, x_1, load per unit area divided by ρc, as a function of kR for the following radiators: 1. a vibrating piston of radius R set in an infinite baffle; 2. a pulsating sphere of radius R; 3. an oscillating sphere of radius R. Note: The ordinate scale of the characteristics labeled 3 must be multiplied by one-third. (See Sec. 5.10.)

direct radiator loudspeaker. It is also customary to use these characteristics for the acoustical impedance at the mouth of a finite horn in computing the throat acoustical impedance.

5.9. Mechanical and Acoustical Impedance Load Upon a Pulsating Sphere.—The pulsating sphere is a sphere whose radius increases and decreases with time. The motion of the air around the sphere will, like the

motion of the sphere itself, take place only in radial directions and will have the same velocity in all directions, but will depend upon the distance from the center of the sphere.

The mechanical impedance, in mechanical ohms, of a pulsating sphere is

$$z_M = 4\pi R^2 \rho c \left[\frac{(kR)^2 + jkR}{1 + (kR)^2} \right] . \qquad 5.14$$

where R = radius of the sphere, in centimeters,

ρ = density, in grams per cubic centimeter,

$k = 2\pi/\lambda$,

λ = wavelength, in centimeters, and

c = velocity of sound, in centimeters per second.

The acoustical impedance, in acoustical ohms, of the air load upon pulsating sphere is

$$z_A = \frac{\rho c}{4\pi R^2} \left[\frac{(kR)^2 + j(kR)}{1 + (kR)^2} \right] \qquad 5.15$$

The acoustical impedance per unit area is

$$z_1 = \rho c \left[\frac{(kR)^2 + jkR}{1 + (kR)^2} \right] \qquad 5.16$$

The resistive and reactive acoustical impedance components of the air load per unit area of a pulsating sphere are shown in Fig. 5.2. It will be noticed that the load upon a pulsating sphere is practically the same as that of a vibrating piston.

5.10. Mechanical and Acoustical Impedance Load upon an Oscillating Sphere.—An oscillating sphere is a sphere whose radius remains constant while the sphere executes a movement of translation as a function of the time. The mechanical impedance, in mechanical ohms, of the air load upon an oscillating sphere is

$$z_M = \frac{4\pi R^2 \rho c}{3} \left[\frac{k^4 R^4 + j(2kR + k^3 R^3)}{4 + k^4 R^4} \right] \qquad 5.17$$

where R = radius of the sphere, in centimeters,

ρ = density, in grams per cubic centimeter,

$k = 2\pi/\lambda$,

λ = wavelength, in centimeters, and

c = velocity of sound, in centimeters per second.

The acoustical impedance, in acoustical ohms, of the air load upon an oscillating sphere is

$$z_A = \frac{\rho c}{12\pi R^2} \left[\frac{k^4 R^4 + j(2kR + k^3 R^3)}{4 + k^4 R^4} \right] \qquad 5.18$$

The acoustical impedance per unit area of an oscillating sphere is

$$z_1 = \frac{\rho c}{3}\left[\frac{k^4R^4 + j(2kR + k^3R^3)}{4 + k^4R^4}\right] \qquad 5.19$$

The average reactive and resistive acoustical impedance components of the air load upon an oscillating sphere are shown in Fig. 5.2. The load on an oscillating sphere is not uniform. In order to compare the radiation characteristics with those of a piston and a pulsating sphere, the ultimate acoustical resistance has been made the same. However, the average acoustical impedance per unit area of a vibrating sphere is one-third that of characteristics 3 shown in Fig. 5.2.

The oscillating sphere is an acoustical doublet (see Sec. 2.3). Therefore, the acoustical resistance component is proportional to the fourth power of the frequency when the dimensions are small compared to the wavelength. The oscillating sphere represents the direct radiator loudspeaker without a baffle.

5.11. Mechanical and Acoustical Impedance Load upon a Pulsating Cylinder.[12]—The pulsating cylinder is a cylinder whose radius increases and decreases with time. The motion of the air around the cylinder will, like the motion of the cylinder itself, take place in radial directions in planes normal to the axis of the cylinder and will have the same radial velocity in all directions but will depend upon the distance from the center of the cylinder.

The mechanical impedance, in mechanical ohms, of the air load, per unit length, upon a pulsating cylinder is

$$z_M = 2\pi R\rho c\left[\frac{(2kR)^2 + j2kR}{1 + (2kR)^2}\right] \qquad 5.20$$

where R = radius of the cylinder, in centimeters,

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

$k = 2\pi/\lambda$

λ = wavelength, in centimeters, and

c = velocity of sound in centimeters per second.

The acoustical impedance, in acoustical ohms, of the air load per unit length upon a pulsating cylinder is given by

$$z_A = \frac{\rho c}{2\pi R}\left[\frac{(2kR)^2 + j2kR}{1 + (2kR)^2}\right] \qquad 5.21$$

The acoustical impedance per unit area is

$$z_1 = \rho c\left[\frac{(2kR)^2 + j2kR}{1 + (2kR)^2}\right] \qquad 5.22$$

The resistive and reactive impedance components of the air load per unit area of a pulsating cylinder are shown in Fig. 5.3.

[12] Hueter and Bolt, " Sonics," John Wiley and Sons, New York, N.Y., 1955.
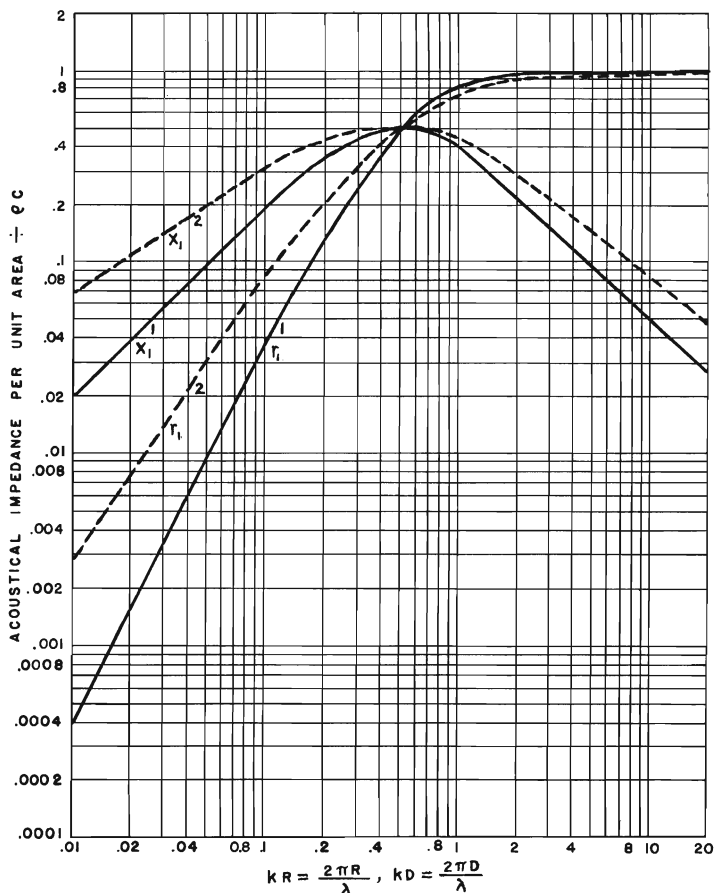
FIG. 5.3. The acoustical resistance, r_1, and acoustical reactance, x_1, load per unit area divided by ρc as a function of kR or kD, for two radiators as follows: 1. A pulsating cylinder of infinite length and of radius R. 2. A vibrating strip of infinite length and of width $2D$, set in an infinite baffle. (1. After Hueter and Bolt.)

5.12. Mechanical and Acoustical Impedance Load upon a Vibrating Strip.[13]—The mechanical impedance,[14] in mechanical ohms, of the air load, per unit length, upon one side of an infinitely long vibrating strip set in an infinite baffle is

$$z_M = 2\rho cD\left[\frac{(2kD)^{3/2} + (2kD)^{2/3}}{1 + (2kD)^{3/2}}\right] \qquad 5.23$$

[13] Hueter and Bolt, " Sonics," John Wiley and Sons, New York, N.Y., 1955.
[14] The expressions given in equations 5.23, 5.24, and 5.25 are not the result of an analytical derivation, but are approximations obtained from numerical integration.

where $2D$ = width of the vibrating strip, in centimeters,

ρ = density, air grams per cubic centimeter,

c = velocity of sound, in centimeters per second,

$k = 2\pi/\lambda$, and

λ = wavelength, in centimeters.

The acoustical impedance, in acoustical ohms, of the air load per unit length, upon a vibrating strip set in an infinite baffle is

$$z_A = \frac{\rho c}{2D}\left[\frac{(2kD)^{3/2} + (2kD)^{2/3}}{1 + (2kD)^{3/2}}\right] \qquad 5.24$$

The acoustical impedance per unit area of the strip is

$$z_1 = \rho c\left[\frac{(2kD)^{3/2} + (2kD)^{2/3}}{1 + (2kD)^{3/2}}\right] \qquad 5.25$$

The resistive and reactive acoustical inpedance components of the air load per unit area of one side of an infinitely long vibrating strip set in an infinite baffle is shown in Fig. 5.3.

5.13. Mechanical and Acoustical Impedance upon a Vibrating Piston in the End of an Infinite Tube.[15,16]—The resistive and reactive acoustical impedance components of the air load per unit area on the free space side of a vibrating piston set in the end of an infinite tube is shown in Fig. 5.4.

The mechanical impedance, z_M, in mechanical ohms, of the air load on the free space side of a vibrating piston set in the end of an infinite tube is given by

$$z_M = \pi R^2 z_1 = \pi R^2(r_1 + x_1) \qquad 5.26$$

where z_1 = acoustical impedance per unit area,

r_1 = acoustical resistance per unit area, Fig. 5.4,

x_1 = acoustical reactance per unit area, Fig. 5.4, and

R = radius of the piston, in centimeters.

The acoustical impedance, z_A, in acoustical ohms, of the air load on the free space side of a vibrating piston set in the end of an infinite tube is given by

$$z_A = \frac{z_1}{\pi R^2} = \frac{r_1 + x_1}{\pi R^2} \qquad 5.27$$

Where the quantities are the same as in equation 5.26.

An example of a vibrating piston set in a tube is that of a loudspeaker mechanism set in a completely-enclosed cabinet having front face area not appreciably larger than the loudspeaker mechanism.

[15] Levine and Schwinger, *Phys. Rev.*, Vol. 73, No. 4, p. 383, 1948.

[16] Beranek, " Acoustics," McGraw-Hill Book Company, New York, N.Y., 1954.

Referring to Figs. 5.2 and 5.4, it will be seen that in the region below $k = 1$ the radiation resistance for a vibrating piston in an infinite baffle is two times the radiation resistance of a vibrating piston located in the end of an infinite tube. This agrees with the conclusions of Sec. 2.2 for a point source radiating
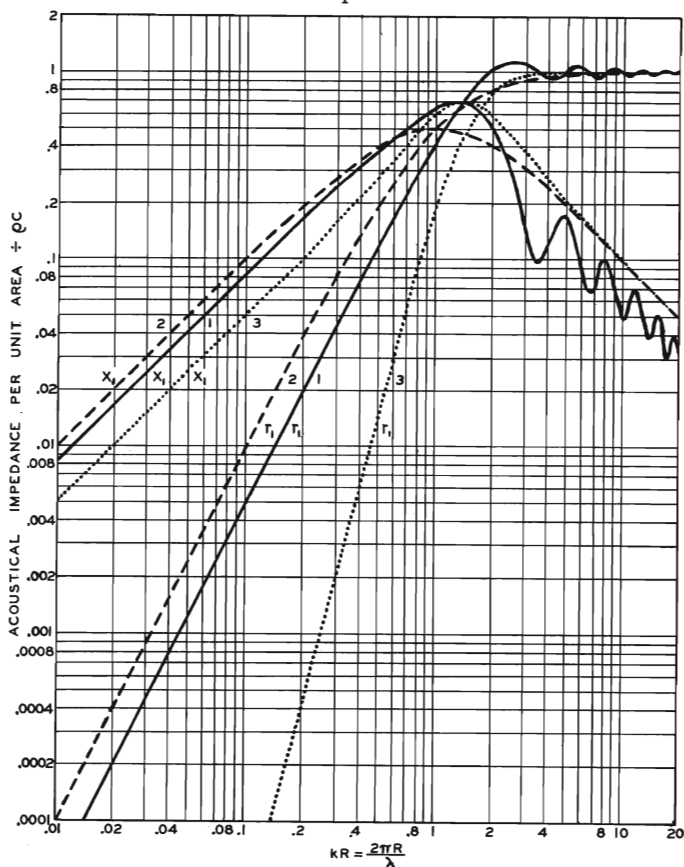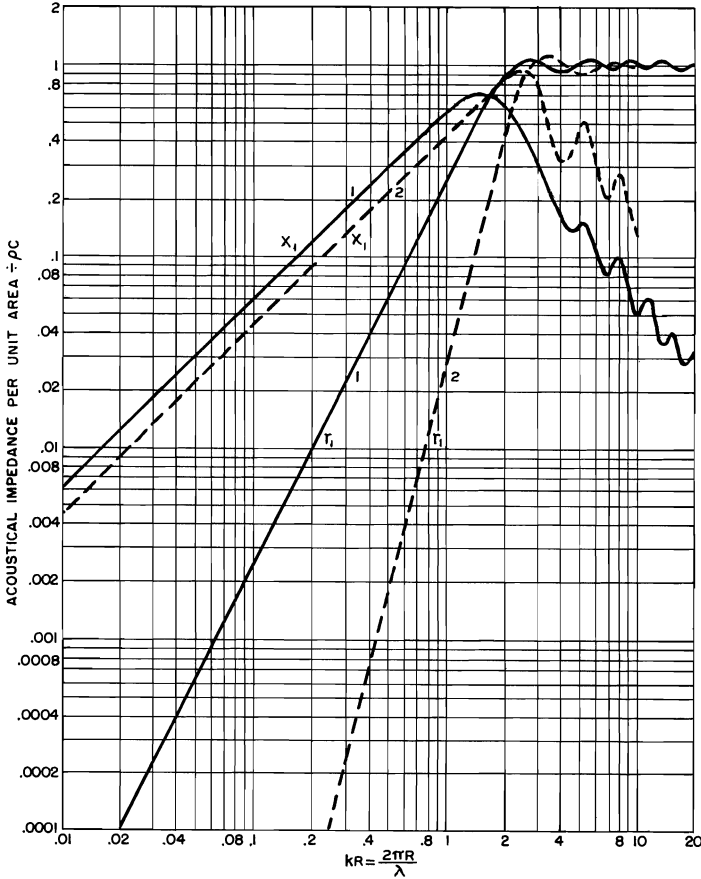
FIG. 5.4. The acoustical resistance, r_1, and the acoustical reactance, x_1, load per unit area divided by ρc as a function of kR for the following radiators: 1. a vibrating piston of radius R set in the end of an infinite pipe. 2. a vibrating piston of radius R in free space.

into 2π and 4π solid angles. This is to be expected because in the region below $kR = 1$ the piston is essentially a point source.

Computing the end correction[17] for a flanged and an unflanged pipe from the mass reactance in the region below $kR = 1$ from Figs. 5.2 and 5.4, the end corrections are $.82R$ and $.61R$, respectively.

[17] Levine and Schwinger, *Phys. Rev.*, Vol. 73, No. 4, p. 383, 1948.

5.14. Mechanical and Acoustical Impedance upon a Vibrating Piston in Free Space.[18,19]—The resistive and reactive acoustical impedance components of the air load on one side of a vibrating piston in free space is shown in Fig. 5.4.

The mechanical impedance, z_M, in mechanical ohms, of the air load on one side of a vibrating piston in free space is given by

$$z_M = \pi R^2 z_1 = \pi R^2 (r_1 + x_1) \qquad 5.28$$

where z_1 = acoustical impedance per unit area,

   r_1 = acoustical resistance per unit area, Fig. 5.4,

   x_1 = acoustical reactance per unit area, Fig. 5.4, and

   R = radius of the piston, in centimeters.

The acoustical impedance, z_A, in acoustical ohms, of the air load on one side of a vibrating piston in free space is given by

$$z_A = \frac{z_1}{\pi R^2} = \frac{(r_1 + x_1)}{\pi R^2} \qquad 5.29$$

Where the quantities are the same as in equation 5.28.

An example of a vibrating piston in free space is a loudspeaker mechanism operating in free space without a baffle, cabinet, etc.

5.15. Acoustical Impedance of a Circular Orifice in a Wall of Infinitesimal Thickness.—The acoustical impedance of a circular orifice in a wall of infinitesimal thickness may be considered to be the same as that of the air load upon a piston of infinitesimal thickness and zero mass set in the opening.   Then the acoustical impedance of a circular aperture in a thin wall is obtained from equation 5.12 by multiplying by 2.

5.16. Acoustical Impedance of an Open Pipe with Large Flanges.— In this case it will be assumed that the mouths of the pipe are fitted with freely moving massless pistons and that the length of the pipe is small compared to the wavelength.   The acoustical impedance is the sum of the mass reactance of the air between the pistons and the acoustical impedance of the air load upon the pistons.

The acoustical reactance, in acoustical ohms, of the column of air between the two pistons, from equation 5.3, is

$$x_A = \frac{\rho l}{\pi R^2} \omega \qquad 5.30$$

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

   l = length of the pipe, in centimeters,

   R = radius of the pipe, in centimeters,

   $\omega = 2\pi f$, and

   f = frequency, in cycles per second.

---

[18] Wiener, F. M., *Jour. Acous. Soc. Amer.*, Vol. 23, No. 6, p. 697, 1951.
[19] Beranek, " Acoustics," McGraw-Hill Book Company, New York, N.Y., 1954.

The acoustical impedance, in acoustical ohms, of the entire system is

$$z_A = \frac{2\rho c}{\pi R^2}\left[1 - \frac{J_1(kR)}{kR}\right] + j\frac{\omega\rho}{\pi R^4 k^3}K_1(2kR) + \frac{j\rho l}{\pi R^2}\omega \qquad 5.31$$

5.17. Horns.—A horn is an acoustical transducer consisting of a tube of varying sectional area. Horns have been used widely for centuries for increasing the radiation from a sound source. The principal virtue of a horn resides in the possibility of presenting practically any value of acoustical impedance to the sound generator. This feature is extremely valuable for obtaining maximum overall efficiency in the design of an acoustical system. As an example, in a horn loudspeaker high efficiency is obtained by designing the system so that the driving force works against resistance instead of inertia of the diaphragm. Employing suitable combination of horns, directional characteristics which are independent of frequency, as well as practically any type of directional pattern, may be obtained. The combination of high efficiency and the possibility of any directional pattern makes the horn loudspeaker particularly suitable for larger scale sound reproduction. It is the purpose of this section to consider some of the factors which influence the characteristics of a horn.

5.18. Fundamental Horn Equation.[20,21,22,23,24,25,26,27,28,29,30,31]—Consider a tube with a certain rate of flare and with the diameter small compared to the wavelength of the sound passing through it. Let the axis of the tube coincide with the x axis. Take an element of volume of the tube defined as

$$S\Delta x \qquad 5.32$$

where S = cross-sectional area of the tube at x, and

Δx = length of the element of volume.

The growth of matter in this volume is the difference between the influx and efflux of fluid through the faces and may be expressed as

$$\Delta x \frac{\partial(S\rho'u)}{\partial x} \qquad 5.33$$

where u = component of the particle velocity along the axis, and

ρ' = density of the medium.

[20] Webster, A. G., *Jour. Natl. Acad. Sci.*, Vol. 5, p. 275, 1919.
[21] Stewart, G. W., *Phys. Rev.*, Vol. 16, No. 4, p. 313, 1920.
[22] Goldsmith and Minton, *Proc. Inst. Rad. Eng.*, Vol. 12, No. 4, p. 423, 1924.
[23] Slepian and Hanna, *Jour. Amer. Inst. Elec. Eng.*, Vol. 43, p. 393, 1924.
[24] Ballantine, G., *Jour. Frank. Inst.*, Vol. 203, No. 1, p. 85, 1927.
[25] Crandall, " Vibrating Systems and Sound," D. Van Nostrand Company, Princeton, N.J., 1926.
[26] Stewart and Lindsay, " Acoustics," D. Van Nostrand Company, Princeton, N.J., 1930.
[27] Olson and Massa, " Applied Acoustics," P. Blakiston's Son and Company, Philadelphia, Pa., 1934.
[28] Mawardi, Osman K., *Jour. Acous. Soc. Amer.*, Vol. 21, No. 4, p. 323, 1949.
[29] Lambert, Robert F., *Jour. Acous. Soc. Amer.*, Vol. 26, No. 4, p. 1024, 1954.
[30] Jensen and Lambert, *Jour. Acous. Soc. Amer.*, Vol. 26, No. 4, p. 1029, 1954.
[31] Scibor-Marchoki, R. I., *Jour. Acous. Soc. Amer.*, Vol. 27, No. 5, p. 939, 1955.

The principle of continuity was expressed in Sec. 1.3. Applying the principle, the difference between the influx and efflux of the fluid into the element of volume must be equal to the time rate of growth of mass.

$$\frac{\partial \rho'}{\partial t} S \Delta x = - \Delta x \frac{\partial (S \rho' u)}{\partial x} \qquad 5.34$$

or

$$S \frac{\partial \rho'}{\partial t} + \frac{\partial (S \rho' u)}{\partial x} = 0 \qquad 5.35$$

From equations 1.19 and 1.6

$$- \rho \dot{\phi} = c^2 \cdot \dot{\rho}' \qquad 5.36$$

From equation 1.11

$$u = \frac{\partial \phi}{\partial x} \qquad 5.37$$

Substituting equations 5.36 and 5.37 in 5.35 the result may be written as

$$\ddot{\phi} - c^2 \frac{\partial \phi}{\partial x} \frac{\partial}{\partial x} (\log S) - c^2 \frac{\partial^2 \phi}{\partial x^2} = 0 \qquad 5.38$$

Equation 5.38 is the wave equation for the axial motion in a tube of varying section.

5.19. Infinite Cylindrical Horn (Infinite Pipe).—The equation expressing the cross-sectional area as a function of the distance along the axis is

$$S = S_1 \qquad 5.39$$

where S_1 = cross section of the pipe, in square centimeters.

The general horn equation for the infinite pipe from equations 5.38 and 5.39 is

$$\ddot{\phi} - c^2 \frac{\partial^2 \phi}{\partial x^2} = 0 \qquad 5.40$$

The velocity potential, pressure, and volume current are

$$\phi = A \epsilon^{jk(ct-x)} \qquad 5.41$$

$$p = kc\rho A \epsilon^{jk(ct-x)} \qquad 5.42$$

$$U = S_1 kA \epsilon^{jk(ct-x)} \qquad 5.43$$

where $k = 2\pi/\lambda$,

λ = wavelength, in centimeters, and

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

The real and imaginary components of the acoustical impedance, in acoustical ohms, at the throat or input end of the pipe are

$$r_A = \frac{\rho c}{S_1} \qquad 5.44$$

$$x_A = 0 \qquad 5.45$$

5.20. Infinite Parabolic Horn.[32]—The equation expressing the cross-sectional area as a function of the distance along the axis is

$$S = S_1 x \qquad 5.46$$

The general horn equation for the parabolic horn is

$$\ddot{\phi} - \frac{c^2}{x}\frac{\partial \phi}{\partial x} - c^2\frac{\partial^2 \phi}{\partial x^2} = 0 \qquad 5.47$$

The velocity potential, pressure, and volume current are

$$\phi = A\left[J_0(kx) - jY_0(kx)\right]\epsilon^{j\omega t} \qquad 5.48$$

$$p = -j\omega\rho A\left[J_0(kx) - jY_0(kx)\right]\epsilon^{j\omega t} \qquad 5.49$$

$$U = ASk\left[-J_0'(kx) + jY_0'(kx)\right]\epsilon^{j\omega t} \qquad 5.50$$

The real and imaginary components of the acoustical impedance, in acoustical ohms, at the throat are

$$r_A = \frac{\rho c}{S_1}\frac{2}{\pi k x_1[J_1^2(kx_1) + Y_1^2(kx_1)]} \qquad 5.51$$

$$x_A = \frac{\rho c}{S_1}\frac{J_0(kx_1)J_1(kx_1) + Y_0(kx_1)Y_1(kx_1)}{J_1^2(kx_1) + Y_1^2(kx_1)} \qquad 5.52$$

where J_0, J_1 = Bessel functions of the first kind of order zero and one,

    Y_0, Y_1 = Bessel functions[33] of the second kind of order zero and one,

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

    c = velocity of sound, in centimeters,

    S_1 = area at x_1, in square centimeters,

    x_1 = distance of the throat from $x = 0$, in centimeters,

    $k = 2\pi/\lambda$, and

    λ = wavelength, in centimeters.

5.21. Infinite Conical Horn.—The equation expressing the cross-sectional area as a function of the distance along the axis is,

$$S = S_1 x^2 \qquad 5.53$$

The general horn equation for the conical horn is

$$\ddot{\phi} - \frac{2c^2}{x}\frac{\partial \phi}{\partial x} - c^2\frac{\partial^2 \phi}{\partial x^2} = 0 \qquad 5.54$$

The velocity potential, pressure, and volume current are

$$\phi = \frac{A}{x}\epsilon^{j(\omega t - kx)} \qquad 5.55$$

$$p = -\frac{j\omega\rho A}{x}\epsilon^{j(\omega t - kx)} \qquad 5.56$$

$$U = -\frac{AS(1 + jkx)\epsilon^{j(\omega t - kx)}}{x^2} \qquad 5.57$$

---

[32] Olson and Wolff, *Jour. Acous. Soc. Amer.*, Vol. 1, No. 3, p. 410, 1930.
[33] Jahnke and Emde, " Tables of Functions," Tuebner, Berlin, 1928.

The real and imaginary components of the acoustical impedance, in acoustical ohms, at the throat are

$$r_A = \frac{\rho c}{S_1} \frac{k^2 x_1^2}{1 + k^2 x_1^2} \qquad 5.58$$

$$x_A = \frac{\rho c}{S_1} \frac{k x_1}{1 + k^2 x_1^2} \qquad 5.59$$

where S_1 = area at x_1, in square centimeters,

x_1 = distance of throat from $x = 0$, in centimeters,

$k = 2\pi/\lambda$, and

λ = wavelength, in centimeters.

5.22. Infinite Exponential Horn.—The equation expressing the cross-sectional area as a function of the distance along the axis

$$S = S_1 \epsilon^{mx} \qquad 5.60$$

where S_1 = area at the throat, that is, at $x = 0$, and

m = flaring constant.

The general horn equation for the exponential horn is

$$\ddot{\phi} - c^2 m \frac{\partial \phi}{\partial x} - c^2 \frac{\partial^2 \phi}{\partial x^2} = 0 \qquad 5.61$$

The velocity potential, pressure, and volume current are

$$\phi = \epsilon^{-(m/2)x} \left[A \epsilon^{-j \frac{\sqrt{4k^2 - m^2}}{2} x} \right] \epsilon^{j\omega t} \qquad 5.62$$

$$p = -j\omega\rho \epsilon^{-(m/2)x} \left[A \epsilon^{-j \frac{\sqrt{4k^2 - m^2}}{2} x} \right] \epsilon^{j\omega t} \qquad 5.63$$

$$U = -AS \left[\frac{m}{2} + j \frac{\sqrt{4k^2 - m^2}}{2} \right] \epsilon^{-\frac{m}{2}x - j \frac{\sqrt{4k^2 - m^2}}{2} x + j\omega t} \qquad 5.64$$

The real and imaginary components of the acoustical impedance, in acoustical ohms, at the throat are

$$r_A = \frac{\rho c}{S_1} \sqrt{1 - \frac{m^2}{4k^2}} \qquad 5.65$$

$$x_A = \frac{\rho c}{S_1} \frac{m}{2k} \qquad 5.66$$

When $m = 2k$ or $4\pi f = mc$ the acoustical resistance is zero. This is termed the cutoff frequency of the exponential horn.

Below the cutoff frequency the acoustical impedance is entirely reactive and

$$x_A = \frac{\rho c}{S_1} \left(\frac{m}{2k} - \sqrt{1 - \frac{m^2}{4k^2}} \right) \qquad 5.67$$

5.23. Infinite Hyperbolic Horn.[34]—The equation expressing the cross-sectional area along the axis is

$$S = S_1 (\cosh \alpha + T \sinh \alpha)^2 \qquad 5.68$$

where $T =$ family parameter, in the hyperbolic horn $T < 1$,

$\alpha = x/x_0$, dimensionless axial distance,

$x =$ axial distance from the throat, in centimeters,

$x_0 =$ reference axial distance, in centimeters, and

$S_1 =$ area at the throat, in square centimeters, that is, at $x = 0$.

The expressions for the velocity potential, pressure, and volume current are quite complex and will not be considered.

The real and imaginary components of the acoustical impedance, in acoustical ohms, at the throat are

$$r_A = \frac{\rho c}{S_1} \frac{\sqrt{1 - \dfrac{1}{\mu}}}{1 - \dfrac{1 - T^2}{\mu^2}} \qquad 5.69$$

$$x_A = \frac{\rho c}{S_1} \frac{\dfrac{T}{\mu}}{1 - \dfrac{1 - T^2}{\mu^2}} \qquad 5.70$$

where $\mu = kx_0 = f/f_0$,

$k = 2\pi/\lambda$,

$f_0 =$ cutoff frequency, and

$f =$ frequency under consideration.

Below the cutoff frequency, $\mu = 1$, the acoustical impedance is entirely reactive and

$$x_A = \frac{\rho c}{S_1} \frac{\dfrac{T}{\mu} - \sqrt{\dfrac{1}{\mu^2} - 1}}{1 - \dfrac{1 - T^2}{\mu^2}} \qquad 5.71$$

5.24. Throat Acoustical Impedance Characteristic of Infinite Parabolic, Conical, Exponential, Hyperbolic, and Cylindrical Horns.—The throat acoustical impedance of infinite horns may be computed from the equations of Secs. 5.18. 5.19, 5.20, 5.21, 5.22, and 5.23. In order to compare the throat acoustical impedance characteristics of infinite parabolic, conical, exponential, hyperbolic, and cylindrical horns, a specific example has been selected in which the throat area is the same in all horns. In

[34] Salmon, V., *Jour. Acous. Soc. Amer.*, Vol. 17, No. 3, p. 212, 1946.

addition, the area at a distance of 100 centimeters from the throat is the same for the four horns with flare, as shown in Fig. 5.5. The value of T for the hyperbolic horn is .5. The acoustical resistance and acoustical reactance frequency characteristics for the five horns are shown in Fig. 5.5.
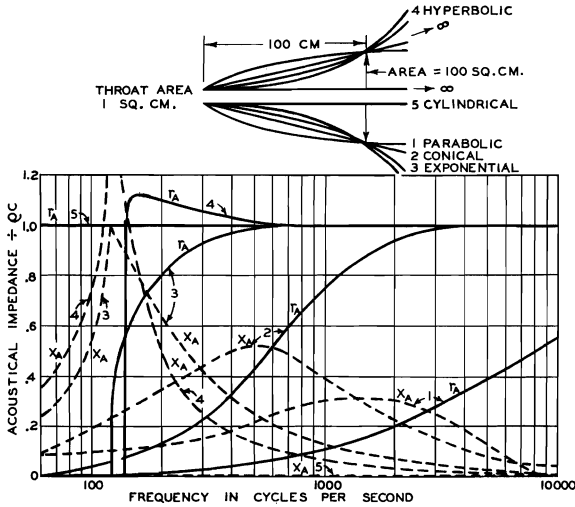
FIG. 5.5. Throat acoustical resistance r_A, and acoustical reactance x_A, frequency characteristics of infinite parabolic, conical, exponential, hyperbolic, and cylindrical horns having a throat area of 1 square centimeter. The cross-sectional area of the parabolic, conical, exponential, and hyperbolic horns is 100 square centimeters at a distance of 100 centimeters from the throat.

5.25. Finite Cylindrical Horn.—The acoustical impedance, in acoustical ohms, at the throat of the finite cylindrical horn of Fig. 5.6 is

$$z_{A1} = \frac{p_1}{U_1} \qquad 5.72$$

where p_1 = pressure at the throat, in dynes per square centimeter, and
U_1 = volume current, in cubic centimeters per second.

The acoustical impedance, in acoustical ohms, at the mouth of a cylindrical horn is

$$z_{A2} = \frac{p_2}{U_2} \qquad 5.73$$

where p_2 = pressure at the mouth, in dynes per square centimeter, and
U_2 = volume current, in cubic centimeters per second.

From equations 5.52 and 5.53 the expressions for the pressures and volume currents at the throat and mouth are given by

FIG. 5.6. 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. x_{A2} = terminating acoustical impedance at the throat. S_2 = cross-sectional area at the mouth, in square centimeters. l = length, in centimeters.
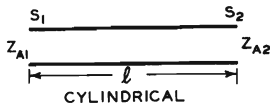
At $x = 0$, $\qquad p_1 = kc\rho A\,\epsilon^{jkct}$ $\qquad$ 5.74

$\qquad\qquad\qquad U_1 = S_1 kA\,\epsilon^{jkct}$ $\qquad$ 5.75

At $x = l$, $\qquad p_2 = kc\rho A\,\epsilon^{jk(ct-l)}$ $\qquad$ 5.76

$\qquad\qquad\qquad U_2 = S_1 kA\,\epsilon^{jk(ct-l)}$ $\qquad$ 5.77
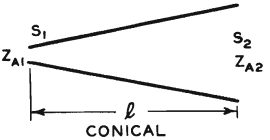
From equations 5.72, 5.73, 5.74, 5.75, 5.76, and 5.77 the expression for the acoustical impedance, z_{A1}, at the throat in terms of the length and cross-sectional area of the horn and the acoustical impedance, z_{A2}, at the mouth is

$$z_{A1} = \frac{\rho c}{S_1}\left(\frac{S_1 z_{A2}\cos{(kl)} + j\rho c\sin{(kl)}}{jS_1 z_{A2}\sin{(kl} + \rho c\cos{(kl)}}\right) \qquad 5.78$$
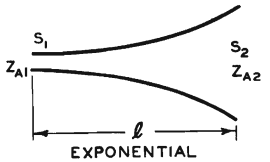
where ρ = density of the medium, in grams per cubic centimeter,

$k = 2\pi/\lambda$,

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, and

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

The throat acoustical impedance characteristics of a finite cylindrical horn or pipe are shown in Fig. 5.7. The mouth acoustical impedance is assumed to be the same as that of a piston in an infinite baffle. In this case the mouth acoustical impedance, z_{A2}, is given by equation 5.12. It will be seen that the variations in the acoustical resistance and acoustical reactance components are quite large at the low frequencies where the mouth acoustical resistance is small.

5.26. Finite Conical Horn.—The acoustical impedance at the throat of a finite conical horn of Fig. 5.8 may be obtained in a manner similar to the procedure for the finite cylindrical horn in the preceding section by employing the equations for the pressure and velocity in an infinite conical horn and applying the proper boundary conditions. The expression for the acoustical impedance, z_{A1}, at the throat in terms of the dimensions of the horn and the acoustical impedance, z_{A2}, at the mouth is

$$z_{A1} = \frac{\rho c}{S_1}\left[\frac{jz_{A2}\dfrac{\sin k(l - \theta_2)}{\sin k\theta_2} + \dfrac{\rho c}{S_2}\sin kl}{z_{A2}\dfrac{\sin k(l + \theta_1 - \theta_2)}{\sin k\theta_1 \sin k\theta_2} - \dfrac{j\rho c}{S_2}\dfrac{\sin k(l + \theta_1)}{\sin k\theta_1}}\right] \qquad 5.79$$
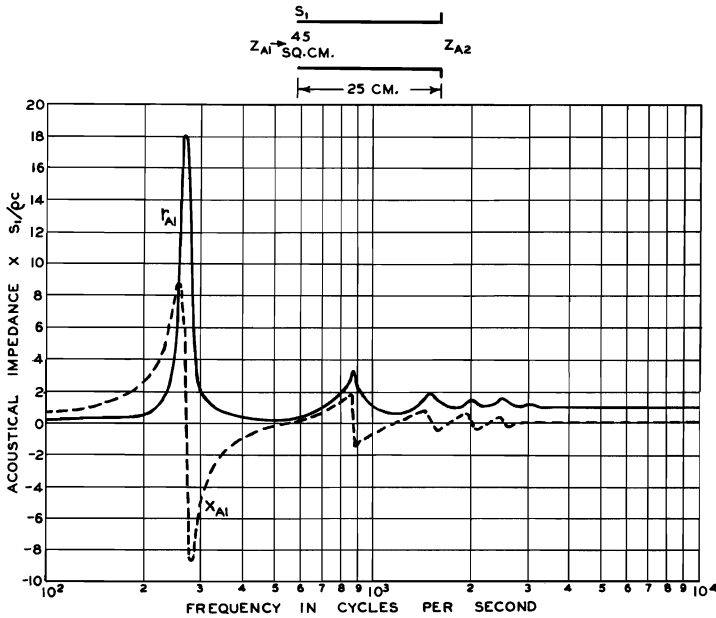
FIG. 5.7. The throat acoustical resistance and acoustical reactance frequency characteristics of a finite cylindrical horn. r_{A1} = acoustical resistance. x_{A1} = acoustical reactance. Note: The characteristics shown are the throat acoustical resistance and acoustical reactance multiplied by S_1 and divided by ρc.

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,

$k = 2\pi/\lambda$,

λ = wavelength, in centimeters,

c = velocity of sound, in centimeters per second,

ρ = density of air in grams per cubic centimeter,

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

The throat acoustical impedance characteristics of a finite conical horn are shown in Fig. 5.8. The acoustical impedance at the mouth of the horn is usually assumed to be the same as that of a piston in an infinite baffle. In this case the mouth acoustical impedance, z_{A2}, is given by equation 5.12.
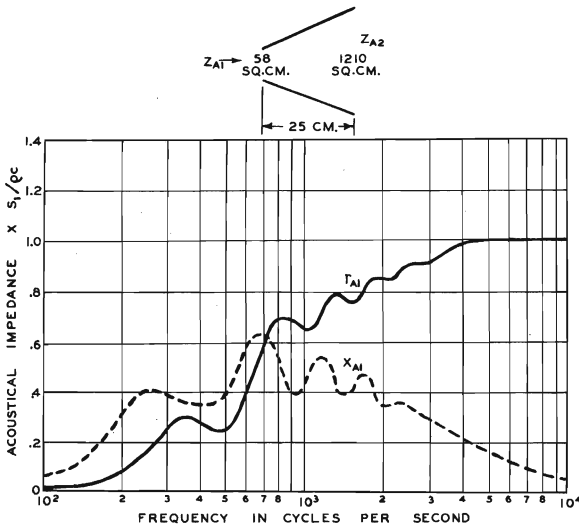
FIG. 5.8. The throat resistance and acoustical reactance frequency characteristics of a finite conical horn. r_{A1} = acoustical resistance. x_{A1} = acoustical reactance. Note: The characteristics shown are the throat acoustical resistance and acoustical reactance multiplied by S_1 and divided by ρc.

5.27. Finite Exponential Horn.[35]—The acoustical impedance at the throat of a finite exponential horn of Fig. 5.6 may be obtained in a manner similar to the procedure for the finite cylindrical horn in the preceding section by employing the equations for the pressure and velocity in an infinite exponential horn and applying the proper boundary conditions. The expression for the acoustical impedance, z_{A1}, at the throat in terms of the length and flare constant of the horn and the acoustical impedance, z_{A2}, at the mouth is

$$z_{A1} = \frac{\rho c}{S_1}\left[\frac{S_2 z_{A2}\left[\cos{(bl + \theta)}\right] + j\rho c\left[\sin{(bl)}\right]}{jS_2 z_{A2}\left[\sin{(bl)}\right] + \rho c\left[\cos{(bl - \theta)}\right]}\right] \qquad 5.80$$

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,

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

$a = m/2$, and

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

[35] Olson, H. F., *RCA Review*, Vol. 1, p. 68, 1937.

For $b = 0$, equation 5.80 is indeterminate. To evaluate, take the derivative of the numerator and denominator with respect to b and set $b = 0$. Then the expression for the throat acoustical impedance becomes

$$z_{A1} = \frac{\rho c}{S_1} \left[\frac{z_{A2}\left(1 - \dfrac{ml}{2}\right) + j\,\dfrac{\rho c}{S_2}\dfrac{lm}{2}}{jz_{A2}\dfrac{lm}{2} + \dfrac{\rho c}{S_2}\left(1 + \dfrac{ml}{2}\right)} \right] \qquad 5.81$$

Below the frequency range corresponding to $b_1 = 0$, b_1 is imaginary. For evaluating this portion of the frequency range the following relations are useful:

$$\tan^{-1} jA = j\tanh^{-1} A = \tfrac{1}{2}j[\log_\epsilon (1 + A) - \log_\epsilon (1 - A)] \qquad 5.82$$

$$\log_\epsilon (-1) = \pm j\pi (2K + 1), \quad K = \text{any integer} \qquad 5.83$$

$$\cos (A \pm jB) = \cos A \cosh B \mp j \sin A \sinh B \qquad 5.84$$

$$\sin jA = j \sinh A \qquad 5.85$$

The resistive and reactive components of the acoustical impedance of a finite exponential horn are shown in Fig. 5.9. The acoustical impedance, z_{A2}, at the mouth was assumed to be that of a piston in an infinite baffle as given by equation 5.12. An examination of the acoustical resistance characteristic of Fig. 5.9 shows that there is a sudden change in acoustical
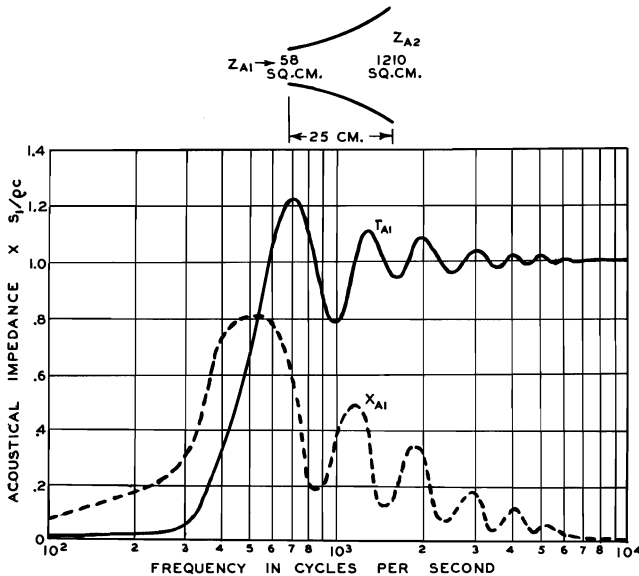
FIG. 5.9. The throat acoustical resistance and acoustical reactance frequency characteristics of a finite exponential horn. r_{A1} = acoustical resistance. x_{A1} = acoustical reactance. Note: The characteristics shown are the throat acoustical resistance or acoustical reactance multiplied by S_1 and divided by ρc.

impedance in the frequency region, $f = mc/4\pi$. Above this frequency the acoustical resistance multiplied by $S_1/\rho c$ approaches unity, below this region the acoustical resistance is relatively small. In the finite exponential horn the acoustical resistance is not zero below the frequency, $f = mc/4\pi$, the flare cutoff frequency, which means that the horn will transmit below this frequency. In the case of the finite conical horn, Fig. 5.8, there is no sudden change in the acoustical resistance. On the other hand, the exponential horn shows a larger ratio of acoustical resistance to acoustical reactance. This, coupled with the more uniform acoustical resistance characteristic, makes the exponential horn more desirable and accounts for its almost universal use. In view of its widespread use it is interesting to examine some of the other characteristics of exponential horns.

5.28. Throat Acoustical Impedance Characteristics of Finite Exponential Horns.[36]—The throat acoustical impedance characteristic as a function of the mouth area, with the flare and throat kept constant, is of interest in determining the optimum dimensions for a particular application. The acoustical impedance characteristics of four finite horns having a cutoff of 100 cycles, throat diameter of 1 inch and mouth diameters of 10, 20, 30, and 40 inches and the corresponding infinite horn are shown in Fig. 5.10. These results may be applied to horns of a different flare by multiplying all the dimensions by the ratio of 100 to the new flare cutoff frequency (see Sec. 1.13). The flare cutoff frequency of an exponential horn is given by

$$2\omega = mc \qquad\qquad 5.86$$

where $\omega = 2\pi f$,

$\quad f$ = frequency, in cycles per second, and

$\quad c$ = velocity of sound, in centimeters per second.

The acoustical radiation resistance of a mouth 10 inches in diameter is relatively small below 500 cycles. The large change in acoustical impedance in passing from the mouth to the free atmosphere introduces reflections at the mouth and as a result wide variations in the acoustical impedance characteristic as shown in Fig. 5.10A. For example, the first maximum in the acoustical resistance characteristic is 150 times the acoustical resistance of the succeeding minimum.

By doubling the diameter of the mouth the maximum variation in the acoustical resistance characteristic is 7.5, Fig. 5.10B.

Fig. 5.10C shows the acoustical impedance characteristic of a horn with a mouth diameter of 30 inches. The maximum variation in the acoustical resistance characteristic of this horn is 2.

The acoustical impedance characteristic of a horn with a mouth diameter of 40 inches, Fig. 5.10D, shows a deviation in acoustical resistance of only a few per cent from that of the infinite horn of Fig. 5.10E.

These results show that as the change in acoustical impedance in passing from the mouth to the free atmosphere becomes smaller by employing a

[36] Olson, H. F., *RCA Review*, Vol. 1, No. 4, p. 68, 1937.

mouth diameter comparable to the wavelength, the reflection becomes correspondingly less and the variations in the acoustical impedance characteristic are reduced.

The throat acoustical impedance characteristic as a function of the
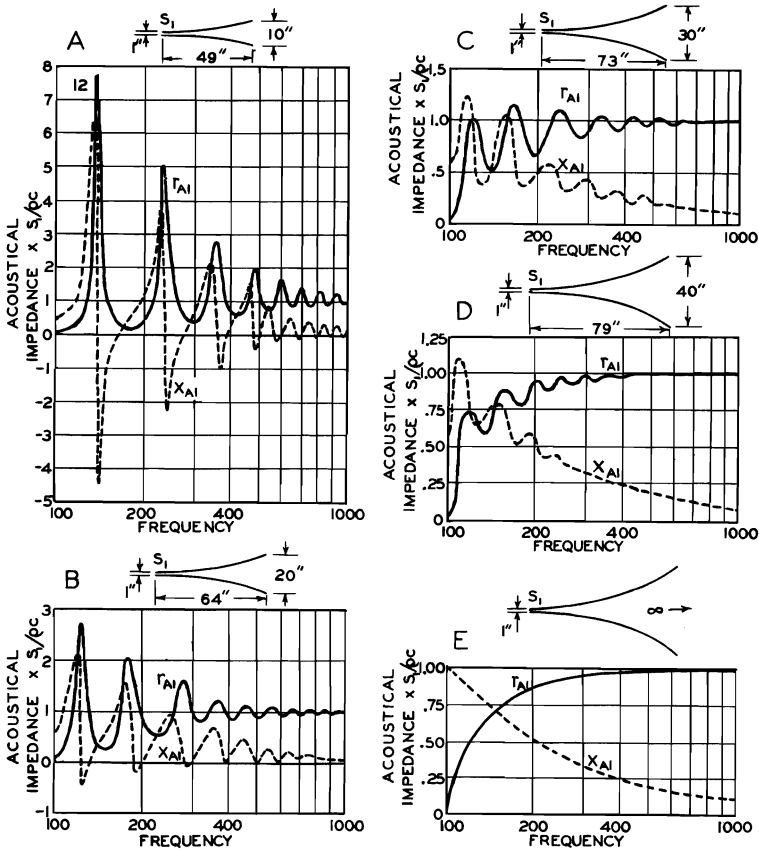
FIG. 5.10. The throat acoustical resistance and acoustical reactance frequency characteristics of a group of exponential horns, with a flare cutoff of 100 cycles and a throat diameter of 1 inch, as a function of the mouth diameter. S_1 = the throat diameter in square centimeters. r_{A1} = acoustical resistance. x_{A1} = acoustical reactance. Note: The characteristics shown are the throat acoustical resistance or acoustical reactance multiplied by S_1 and divided by ρc.

throat size with the mouth and flare held constant is of interest in determining the optimum length and a suitable matching impedance for the driving mechanism. The acoustical impedance characteristics of four horns having a cutoff of 100 cycles, mouth diameter of 20 inches, and throat diameter of 1, 2, 4, and 8 inches are shown in Fig. 5.11. A consideration of these

characteristics shows that the throat size has no appreciable effect upon the amplitude of the variations in the acoustical impedance characteristics. However, the separation in frequency between successive maxima is increased, as the throat becomes larger, due to the decreased length of the horn. The frequency at which the first maximum in the acoustical resistance characteristic occurs becomes progressively higher as the length is decreased.

The characteristics in Figs. 5.10 and 5.11 cover the range from 100 to 1000 cycles, the lower value being the flare cutoff frequency. The finite
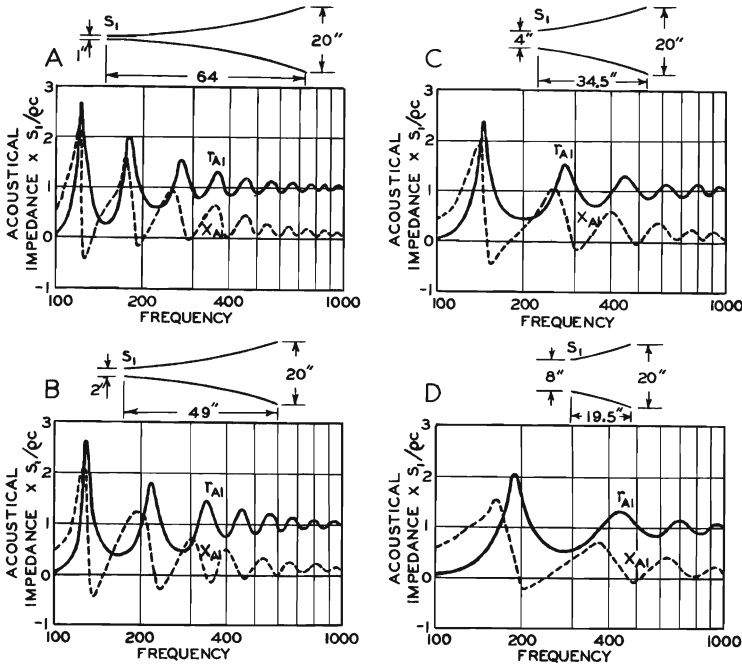
FIG. 5.11. The throat acoustical resistance and acoustical reactance frequency characteristics of a group of exponential horns, with a flare cutoff of 100 cycles and a mouth diameter of 20 inches, as a function of the throat mouth diameter. S_1 = the throat diameter, in square centimeters. r_{A1} = acoustical resistance. x_{A1} = acoustical reactance. Note: The characteristics shown are the throat acoustical resistance or acoustical reactance multiplied by S_1 and divided by ρc.

horn, of course, transmits below this frequency because the acoustical resistance is not zero. However, save for the case where the throat is comparable to the mouth, as for example, Fig. 5.11D, the value of the acoustical resistance, at and below the flare cutoff frequency, is quite small.

5.29. Exponential Connectors.—A transformer is used in electrical circuits to transfer between two acoustical impedances of different values

without appreciable reflection loss. In acoustical systems a horn may be used to transfer from one acoustical impedance to another. As a matter of fact a horn may be looked upon as an acoustical transformer, transforming large pressures and small volume currents to small pressures and large volume currents. It is the purpose of this section to show how an exponential horn or connector may be used to transfer from one acoustical impedance to another.

Fig. 5.10 shows an exponential horn coupled to an infinite tube. The acoustical impedance of an infinite tube is

$$\frac{\rho c}{S_2} \qquad 5.87$$

where ρ = density, in grams per cubic centimeter,

c = velocity of sound, in centimeters per second, and

S_2 = cross-sectional area of the infinite tube, in square centimeters.
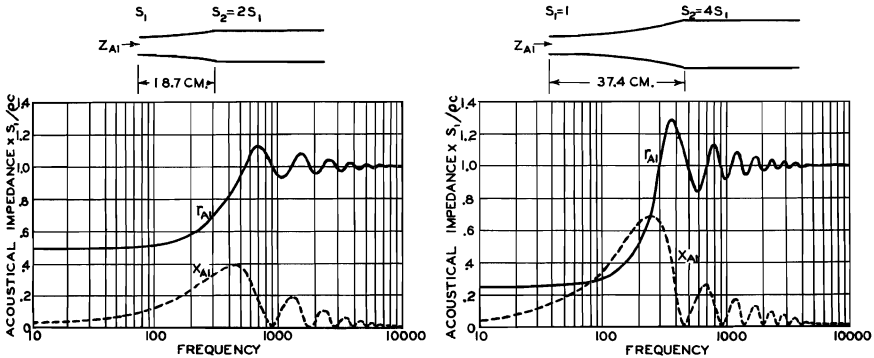
FIG. 5.12. The throat acoustical resistance and acoustical reactance frequency characteristics of two exponential connectors with a flare cutoff of 100 cycles. The mouth of the horn is connected to an infinite pipe. r_{A1} = acoustical resistance. x_{A1} = acoustical reactance. Note: The characteristics shown are the acoustical resistance or acoustical reactance multiplied by S_1 and divided by ρc.

Equation 5.87 is the mouth acoustical impedance of the exponential horn. Equation 5.80 then becomes

$$z_{A1} = \frac{\rho c}{S_1}\left[\frac{\cos{(bl+\theta)} + j\sin{(bl)}}{\cos{(bl-\theta)} + j\sin{(bl)}}\right] \qquad 5.88$$

For $b = 0$, equation 5.88 is indeterminate. To evaluate take the derivative of the numerator and denominator with respect to b and set $b = 0$. Then the expression for the throat acoustical impedance becomes

$$z_{A1} = \frac{\rho c}{S_1}\left[\frac{1 + j\dfrac{lm}{2} - \dfrac{lm}{2}}{1 + \dfrac{lm}{2} + j\dfrac{lm}{2}}\right] \qquad 5.89$$

Below the frequency corresponding to $b = 0$, b is imaginary. This portion of the range may be evaluated by employing equations 5.82, 5.83, 5.85, and 5.89.

The acoustical impedance characteristics of two exponential connectors with a flare cutoff of 100 cycles (that is $b = 0$ at 100 cycles) is shown in Fig. 5.12. Below 100 cycles the throat acoustical impedance is the same as that of the infinite pipe. However, at the high frequencies the throat acoustical impedance is the same as the surge acoustical impedance of a pipe of the diameter of the throat. In order to effect a constant transfer of acoustical impedance with respect to frequency over a certain frequency range the cutoff of the connector must be placed below the low-frequency limit of the frequency range.
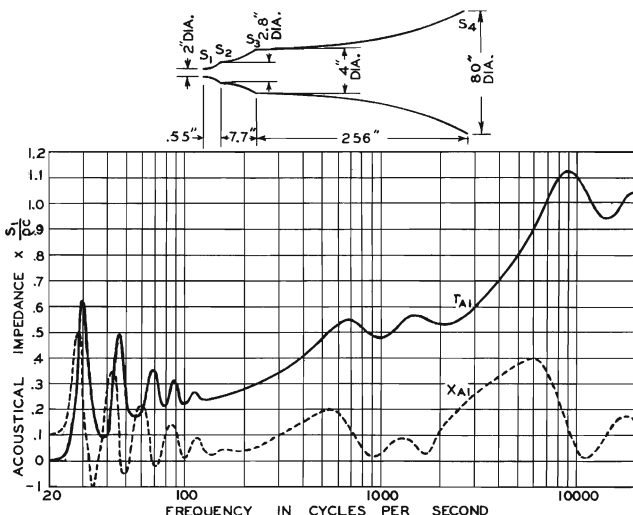
FIG. 5.13. The throat acoustical resistance and acoustical reactance frequency characteristics of a multiple flare exponential horn of three sections. The cutoffs due to flare of the three horns are 25, 100, and 1400 cycles. r_{A1} = acoustical resistance. x_{A1} = acoustical reactance. Note: The characteristics shown are the throat acoustical resistance or acoustical reactance multiplied by S_1 and divided by pc. S_1 = area at the throat of the small horn in centimeters.

5.30. A Horn Consisting of Manifold Exponential Sections.[37]—The efficiency of a horn loudspeaker is governed, among many other factors, by the throat acoustical resistance. To obtain the maximum efficiency at any frequency the effective acoustical reactance of the entire vibrating system should be equal to the effective acoustical resistance. This, in general, means that to obtain maximum efficiency the throat acoustical resistance

[37] Olson, H. F., *Jour. Soc. Mot. Pic. Eng.*, Vol. 30, No. 5, p. 511, 1938.

of the horn should be proportional to the frequency, since the acoustical reactance is primarily an inertive reactance and, therefore, proportional to the frequency. Practically any throat acoustical impedance frequency characteristic may be obtained by employing a horn consisting of manifold exponential sections.

A horn consisting of three rates of flare is shown in Fig. 5.13. The acoustical impedance characteristic at the throat of the small horn is obtained in stages. First, the throat acoustical impedance characteristic for the large horn is obtained by using equation 5.80. The throat acoustical impedance obtained for the large horn now becomes the mouth acoustical impedance of the intermediate horn. The acoustical impedance of the throat of the intermediate horn is obtained by employing equation 5.80. For the frequency corresponding to $b = 0$ of the intermediate horn the acoustical impedance at the throat of the intermediate horn becomes indeterminate. The expression can be evaluated as shown in Sec. 5.27 on the finite exponential horn. Next, the throat acoustical impedance at the throat of the small horn is obtained by again employing equation 5.80. The mouth acoustical impedance of the small horn is the throat acoustical impedance just obtained for the intermediate horn. The acoustical impedance characteristic of Fig. 5.13 shows three distinct steps depicting the surge acoustical impedance of each section.

5.31. Closed Pipe with a Flange.—The acoustical impedance of a pipe closed at one end and equipped with a flange at the open end may be considered to be the sum of the acoustical impedance of the pipe and the end correction. It will be assumed that the open end of the pipe is equipped with a massless piston.

The input acoustical impedance to the pipe at the massless piston may be obtained from equation 5.74 by setting $z_{A2} = \infty$. The input acoustical impedance of the pipe closed at the far end is

$$z_{A0} = -\frac{j\rho c}{\pi R^2} \cot kl \qquad 5.90$$

where l = length of the pipe, in centimeters,
R = radius of the pipe, in centimeters,
ρ = density, in grams per cubic centimeter,
c = velocity of sound, in centimeters per second,
$k = 2/\pi\lambda$, and
λ = wavelength, in centimeters.

The above equation is the acoustical impedance of a closed pipe when there is no end correction, as for example, when the pipe is used in a closed system.

When the open end is free and terminated in a large baffle the total acoustical impedance is the sum of equations 5.12 and 5.90.

$$z_{AT} = \frac{\rho c}{\pi R^2}\left[1 - \frac{J_1(kR)}{kR}\right] + j\frac{\omega\rho}{2\pi R^4 k^3}K_1(2kR) - j\frac{\rho c}{\pi R^2}\cot kl \qquad 5.91$$

The ratio of the pressure at the closed end of the tube to the free space pressure is useful in predicting the performance of pipes and cavities. The ratio of the pressure at the closed end to that in free space is

$$\frac{p_0}{p} = \sqrt{\left[\cos kl - \frac{\pi R^2}{\rho c} x_A \sin kl\right]^2 + \frac{(\pi R^2)^2}{(\rho c)^2} r_A{}^2 \sin^2 kl} \qquad 5.92$$

where p = pressure at the closed end, in dynes per square centimeter,

p_0 = pressure in free space, in dynes per square centimeter,

r_A = acoustical resistance, in acoustical ohms, equation 5.12, and

x_A = acoustical reactance, in acoustical ohms, equation 5.12.
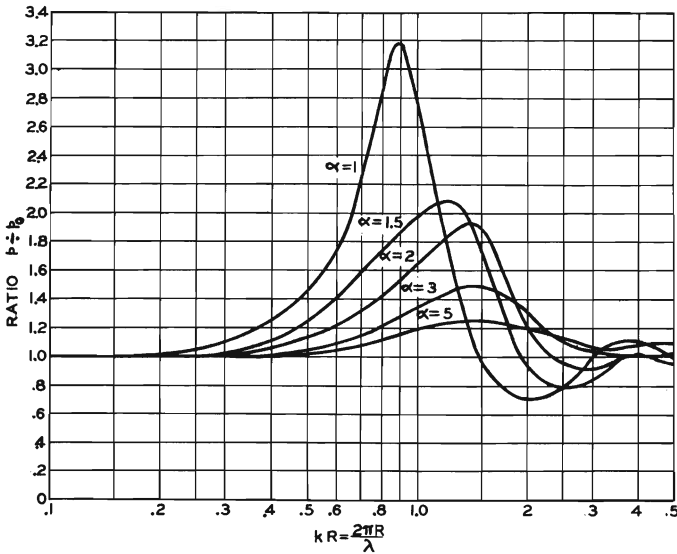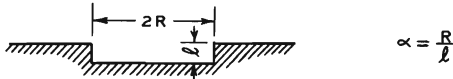
FIG. 5.14. Ratio of the pressure at the bottom of a cylindrical cavity to the free space pressure in the incident sound wave as a function of kR.

The characteristics of Fig. 5.14 depict the ratio of the pressure at the closed end of a cylindrical cavity to the pressure in free space as a function of the dimensions of the cavity and the wavelength of the sound.

5.32. Sound Transmission in Tubes.[38,39,40,41]—The effect of viscosity upon the characteristics of small holes and slits was considered in Secs. 5.3

[38] Crandall, " Vibrating Systems and Sound," D. Van Nostrand Company, Princeton, N.J., 1926.

[39] Lamb, " Theory of Sound," E. Arnold, London, 1931.

[40] Rayleigh, " Theory of Sound," Macmillan and Company, London, 1926.

[41] Mason, W. P., *Phys. Rev.*, Vol. 31, No. 2, p. 283, 1928.

and 5.4. The transmission loss in tubes of circular section is of interest in problems in acoustics involving the use of tubes. The equation[41] expressing the sound transmission in a tube is

$$A = A_0 \epsilon^{-\alpha x} \qquad 5.93$$

where A = amplitude (pressure or volume current) at a distance x centimeters from the amplitude A_0,

$\alpha = \dfrac{\gamma'}{Rc} \sqrt{\dfrac{\omega \mu}{2\rho}}$,

R = radius of the tube, in centimeters,

c = velocity of sound, in centimeters per second,

$\omega = 2\pi f$,

f = frequency, in cycles per second,

μ = viscosity coefficient, 1.86×10^{-4} for air,

ρ = density, in grams per cubic centimeters,

$\gamma' = 1 + 1.58 \, (\gamma^{1/2} - \gamma^{-1/2})$, and

γ = ratio of specific heats, 1.4 for air.

The attenuation characteristics of tubes of various diameters as a function of the frequency are shown in Fig. 5.15.
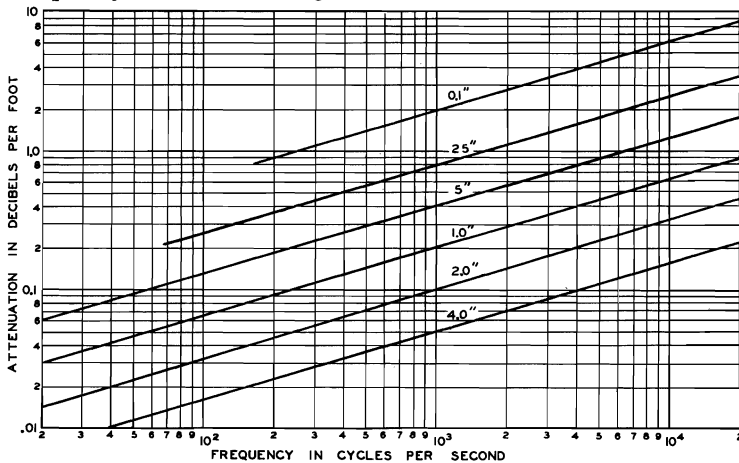
FIG. 5.15. The attenuation of a sound wave, in decibels per foot, as a function of the frequency, in cycles per second, in pipes of various diameters and filled with dry air at 20° Centigrade.

5.33. Transmission from One Pipe to Another Pipe of Different Cross-sectional Area.[42]

—Consider two pipes of cross sections S_1 and S_2 joined as shown in Fig. 5.16. Assume that sound travels from pipe S_1 to pipe S_2.

[42] Stewart and Lindsay, " Acoustics," D. Van Nostrand Company, Princeton, N.J., 1930.

The boundary conditions are

1. Continuity of pressure,
2. Continuity of volume current.

The condition for pressure may be written

$$p_1 + p_1' = p_2 \qquad 5.94$$

where p_1 = incident pressure in pipe S_1, in dynes per square centimeter,
　　p_1' = reflected pressure in pipe S_1, in dynes per square centimeter, and
　　p_2 = transmitted pressure in pipe S_2, in dynes per square centimeter.

The condition for volume current may be written

$$U_1 - U_1' = U_2 \qquad 5.95$$

where U_1 = incident volume current in S_1, in cubic centimeters per second,
　　U_1' = reflected volume current in S_1, in centimeters per second,
　　U_2 = transmitted volume current in S_2, in centimeters per second.

The acoustical resistance of the first pipe S_1 is
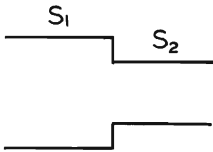
$$r_{A1} = \frac{\rho c}{S_1} = \frac{p_1}{U_1} \qquad 5.96$$

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

FIG. 5.16. Two connected pipes of cross-sectional areas S_1 and S_2.

c = velocity of sound in the medium, in centimeters per second, and

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

The acoustical resistance of the second pipe S_2 is

$$r_{A2} = \frac{\rho c}{S_2} = \frac{p_2'}{U_2} \qquad 5.97$$

where S_2 = cross-sectional area of the second pipe, in square centimeters.

Expressing equation 5.91 in terms of pressure

$$p_1 S_1 - p_1' S_1 = p_2 S_2$$

or

$$p_1 - p_1' = \frac{p_2 S_2}{S_1} \qquad 5.98$$

Eliminating p_1' from equations 5.94 and 5.98

$$p_2 = \frac{p_1(2S_1)}{S_1 + S_2} = \frac{2p_1}{1 + \dfrac{S_2}{S_1}} \qquad 5.99$$

Expressing in terms of volume current,

$$U_2 = \frac{2U_1}{1 + \dfrac{S_1}{S_2}} \qquad 5.100$$

Equations 5.99 and 5.100 show that the pressure and volume current of the transmitted wave in pipe S_2 is always in phase with the pressure and volume current of the incident wave in pipe S_1.

The reflected pressure in terms of the incident pressure is

$$p_1' = \left(\frac{S_1 - S_2}{S_1 + S_2}\right) p_1 \qquad 5.101$$

The reflected volume current in terms of the incident volume current is

$$U_1' = \frac{(S_1 - S_2)}{S_1 + S_2}\, U_1 \qquad 5.102$$

Equations 5.101 and 5.102 show that if $S_1 < S_2$ the reflected pressure or volume current are in phase with the incident pressure or volume current. If $S_1 > S_2$, the reflected pressure or volume current are opposite in phase with the incident pressure or volume current. If $S_1 = S_2$, there is no reflected wave.

The ratio of the transmitted power to the incident power is

$$P_{12} = \frac{4 r_{A1}\, r_{A2}}{(r_{A1} + r_{A2})^2} \qquad 5.103$$

5.34. Transmission Through Three Pipes.[43]—Consider three pipes of cross sections S_1, S_2, and S_3 as shown in Fig. 5.17. Assume that sound travels from pipe 1 to pipe 3. Let the boundary between S_1 and S_2 be denoted by A and between S_2 and S_3 by B.
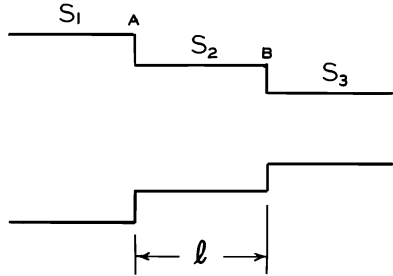
The boundary conditions are

   1. Continuity of pressure,
   2. Continuity of volume current.

At the boundary A the conditions for the pressure may be written

$$p_1 + p_1' = p_2 + p_2' \qquad 5.104$$

Fig. 5.17. Three pipes of cross-sectional areas S_1, S_2, and S_3. The pipe S_2 is of finite length l.

where $p_1 =$ incident pressure in S_1, in dynes per square centimeter,
   $p_1' =$ reflected pressure in S_1, in dynes per square centimeter,
   $p_2 =$ transmitted pressure in S_2, in dynes per square centimeter, and
   $p_2' =$ reflected pressure in S_2, in dynes per square centimeter.

At the boundary A the conditions for the volume current may be written

$$U_1 - U_1' = U_2 - U_2' \qquad 5.105$$

where $U_1 =$ incident volume current in S_1, in cubic centimeters per second,
   $U_1' =$ reflected volume current in S_1, in cubic centimeters per second,
   $U_2 =$ transmitted volume current in S_2, in cubic centimeters per second, and
   $U_2' =$ reflected volume current in S_2, in cubic centimeters per second.

[43] Stewart and Lindsay, " Acoustics," D. Van Nostrand Company, Princeton, N.J., 1930.

At the boundary B the conditions for the pressure may be written

$$p_2\epsilon^{-jkl} + p_2'\epsilon^{jkl} = p_3 \qquad\qquad 5.106$$

where p_2 = incident pressure in S_2, in dynes per square centimeter,

  p_2' = reflected pressure in S_2, in dynes per square centimeter,

  p_3 = transmitted pressure in S_3, in dynes per square centimeter,

  l = length of pipe S_2, in centimeters,

  $k = 2\pi/\lambda$,

  λ = wavelength, in centimeters.

At the boundary B the conditions for the volume current may be written

$$U_2\epsilon^{-jkl} - U_2'\epsilon^{jkl} = U_3 \qquad\qquad 5.107$$

where U_2 = transmitted volume current in S_2, in cubic centimeters per second,

  U_2' = reflected volume current in S_2, in cubic centimeters per second, and

  U_3 = transmitted volume current in S_3, in cubic centimeters per second.

From equations 5.104, 5.105, 5.106, and 5.107,

$$p_1 = \frac{p_3}{2}\left[\left(\frac{S_3}{S_1} + 1\right)\cos kl + j\left(\frac{S_2}{S_1} + \frac{S_3}{S_2}\right)\sin kl\right] \qquad 5.108$$

The ratio of the power transmitted in S_3 to the incident flow of power in S_1 is

$$P_{13} = \frac{4\dfrac{S_3}{S_1}}{\left(\dfrac{S_3}{S_1} + 1\right)^2 \cos^2 kl + \left(\dfrac{S_2}{S_1} + \dfrac{S_3}{S_2}\right)^2 \sin^2 kl} \qquad 5.109$$

If kl is small, the transmission is independent of the cross section of the channel S_2. If $\sin kl = \pm 1$, the power transmission is

$$P_{13} = \frac{4\dfrac{S_3}{S_1}}{\left(\dfrac{S_2}{S_1} + \dfrac{S_3}{S_2}\right)^2} = \frac{4S_3 S_2{}^2 S_1}{(S_2{}^2 + S_1 S_3)^2} \qquad 5.110$$

Equation 5.105 shows that $P = 1$, if $S_2{}^2 = S_1 S_3$. That is, if $\sin kl = \pm 1$ and providing the area of S_2 is a geometric mean of S_1 and S_3, the transmission is unity. .

5.35. Transmission from One Medium to Another Medium.[44]—The problem of transmission from one medium to another medium as shown in Fig. 5.18 is the same as the problem transmission from one pipe to another

[44] Stewart and Lindsay, " Acoustics," D. Van Nostrand Company, Princeton, N.J., 1930.

pipe of different cross-sectional area.   The boundary between the two media is assumed to be plane and parallel to the wavefront which is also assumed to be plane.

The ratio of the power transmitted in the medium 2 to incident flow of power in the medium 1 of Fig. 5.18 is

$$P_{12} = \frac{4 r_{A1} r_{A2}}{(r_{A1} + r_{A2})^2} \qquad\qquad 5.111$$

where r_{A1} = acoustical resistance of medium 1, and

r_{A2} = acoustical resistance of medium 2.
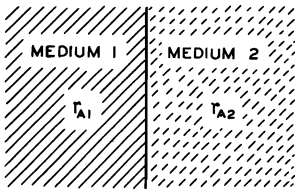
FIG. 5.18. Two media of acoustical resistances r_{A1} and r_{A2}.
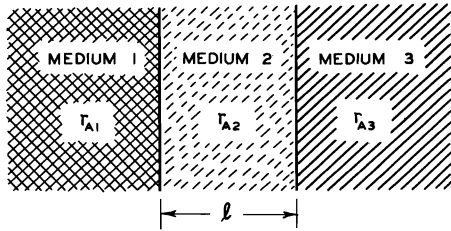
FIG. 5.19. Three media of acoustical resistances r_{A1}, r_{A2}, and r_{A3}. The length of the medium 2 is l.

5.36. Transmission Through Three Media.[45]—The problem of transmission through the three media of Fig. 5.19 is the same as that through the three pipes.   The ratio of the power transmitted in the medium 3 to the incident flow of power in the medium 1 is

$$P_{13} = \frac{4 \dfrac{r_{A1}}{r_{A3}}}{\left(\dfrac{r_{A1}}{r_{A3}} + 1\right)^2 \cos^2 kl + \left(\dfrac{r_{A1}}{r_{A2}} + \dfrac{r_{A2}}{r_{A3}}\right)^2 \sin^2 kl} \qquad 5.112$$

where r_{A1} = acoustical resistance of the medium 1,

r_{A2} = acoustical resistance of the medium 2, between 1 and 3,

r_{A3} = acoustical resistance of the medium 3,

l = length of the medium 2, in centimeters,

$k = 2\pi/\lambda$,

λ = wavelength in the medium 2, in centimeters.

5.37. Tubes Lined with Absorbing Material.—In ventilator and exhaust systems it is desirable to provide a high degree of attenuation for audio-frequency waves while offering low resistance to continuous flow of air.   For that purpose, the most satisfactory systems are ducts lined with

[45] Stewart and Lindsay, " Acoustics," D. Van Nostrand Company, Princeton, N.J., 1930.

absorbing material. Longitudinal isolation of the walls of the duct should be provided to prevent longitudinal transmission of sound by the walls of the duct. This can be accomplished by the use of rubber connectors at regular intervals. The walls of the duct should be rigid so that air-borne sounds are not transmitted through the walls. Very high attenuation can be obtained in ducts of this type.

The attenuation, in decibels per foot, in a square or rectangular conduit lined with absorbing material may be obtained from the following empirical formula,[46]

$$\frac{db}{ft} = 12.6\alpha^{1.4}\frac{P}{A} \qquad\qquad 5.113$$

where P = perimeter, in inches,

A = cross-sectional area in square inches, and

α = absorption coefficient of the material used for lining the duct.

Equation 5.113 holds for square ducts and rectangular ducts in which the ratio between the two sides is not greater than two.

The general subject[47,48] of tubes lined with absorbing material, with both rigid and vibratile walls, has been considered theoretically and experimentally.

5.38. Response of a Vibrating System of One Degree of Freedom.— Consider the electrical circuit, consisting of inductance, electrical resistance, and electrical capacitance and a voltage connected in series, as shown in Fig. 5.20. The resonant frequency, in cycles per second, is given by

$$f_r = \frac{1}{2\pi\sqrt{LC_E}} \qquad\qquad 5.114$$

where L = inductance, in henries, and

C_E = electrical capacitance, in farads.

The current in the circuit is given by

$$i = \frac{e}{r_E + j\left(\omega L + \frac{1}{j\omega C_E}\right)} \qquad\qquad 5.115$$

where r_E = electrical resistance, in ohms,

e = driving voltage, in volts, and

i = current, in amperes.

The quantity Q_r is given by

$$Q_r = \frac{\omega_r L}{r_E} \qquad\qquad 5.116$$

where $\omega_r = 2\pi f_r$.

[46] Sabine, H. J., *Jour. Acous. Soc. Amer.*, Vol. 12, No. 1, p. 53, 1940.
[47] Sivian, L. J., *Jour. Acous. Soc. Amer.*, Vol. 9, No. 2, p. 135, 1937.
[48] Molloy, C. T., *Jour. Acous. Soc. Amer.*, Vol. 16, No. 1, p. 31, 1944.

L C_E r_E

e   i

ELECTRICAL   CIRCUIT

VALUES
OF
Q_r

RESPONSE

$f \div f_r$
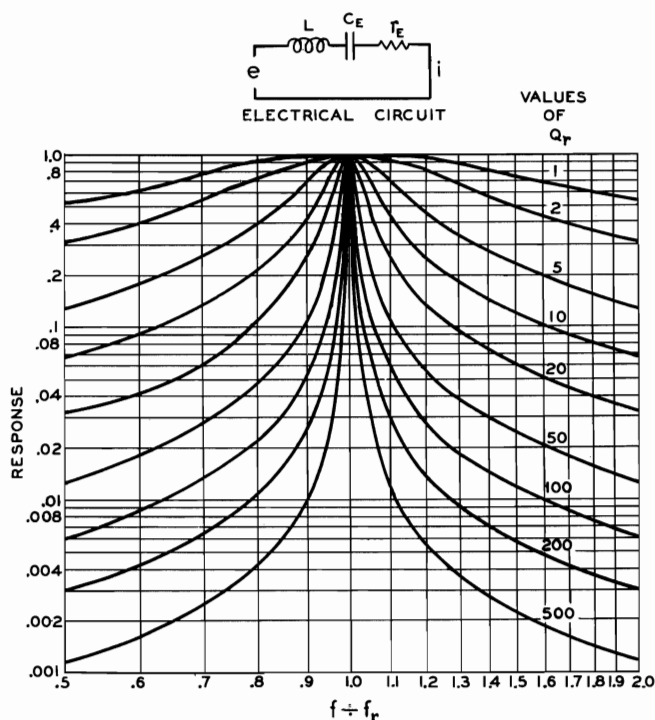
FIG. 5.20. The current response characteristics of a simple series circuit as a function of the ratio $f \div f_r$, where f_r = the resonant frequency, and f = the frequency under consideration. The numbers of the characteristics refer to the value of Q_r, $Q_r = 2\pi f_r L / r_E$. The above characteristics are applicable to acoustical and mechanical systems by the substitution of the elements and quantities which are analogous to the electrical system.

The current response characteristics as function of the ratio $f \div f_r$ for various values of Q_r are shown in Fig. 5.20.

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 (see Chapter IV).

6

DIRECT RADIATOR LOUDSPEAKERS

6.1. Introduction.[1]—A loudspeaker is an electroacoustic transducer designed to radiate acoustical energy into a room or open air. There are two general types of loudspeakers in use today, namely: the direct radiator and the horn type loudspeaker. The diaphragm of the direct radiator loudspeaker is coupled directly to the air. The diaphragm of the horn loudspeaker is coupled to the air by means of a horn. The direct radiator loudspeaker will be considered in this chapter and the horn loudspeaker will be considered in the following chapter.

The almost universal use of the direct radiator loudspeaker is due to the simplicity of construction, small space requirements, and the relatively uniform response characteristic. Uniform response over a moderate frequency band may be obtained with any simple, direct radiator dynamic loudspeaker. However, reproduction over a wide frequency range is restricted by practical limitations. The two extreme ends of the audio-frequency band are the most difficult to reproduce with efficiency comparable to that of the mid-audio-frequency range. Inefficiency at the low frequencies is primarily due to the small radiation mechanical resistance. There are a number of means available for increasing the radiation mechanical resistance at the low frequencies. A large radiation mechanical resistance may be obtained by using a large cone. A phase inverter consisting of a completely enclosed cabinet with ports provides a means for extending the low-frequency range. A horn may be used for presenting a large radiation mechanical resistance to a diaphragm at the low frequencies. The efficiency of a direct radiator loudspeaker at the high frequencies is limited by the mechanical mass reactance of the vibrating system. There are a number of arrangements suitable for reducing the mass of the vibrating system at the high frequencies. Two or more separate loudspeaker mechanisms may be used, each designed to reproduce a certain portion of the range. Multiple cones driven by a single voice coil may be arranged so that the mass of the system decreases at the high frequencies. The voice coil may be sectionalized to decrease the mass and inductance at the high frequencies and thereby increase the high-frequency range. Multiple coils and multiple cones

[1] Mawardi, Osman K., *Jour. Acous. Soc. Amer.*, Vol. 26, No. 1, p. 1, 1954.

combined into a single mechanism may be designed to yield uniform response to the upper limit of audibility.

It is the purpose of this chapter to outline the factors which influence the performance of the conventional, direct radiator loudspeaker, to illustrate systems for controlling the response with respect to frequency and to describe several means for decreasing the effective mass of the vibrating systems at the high frequencies and for improving the efficiency at the low frequencies.

6.2. Single-Coil, Single-Cone Loudspeaker.—The simple dynamic loudspeaker consists of a paper cone driven by a voice coil located in a magnetic field. A cross-sectional view, the voice coil circuit and the mechanical circuit of a dynamic loudspeaker are shown in Fig. 6.1. The
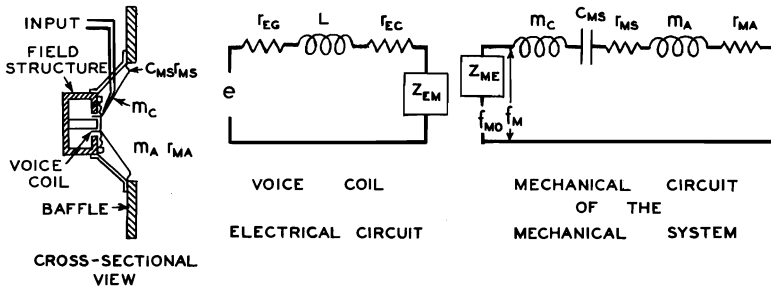
FIG. 6.1. Cross-sectional view of a single-coil, single-cone, direct radiator, dynamic, loudspeaker mechanism mounted in a baffle. In the voice coil circuit, $e =$ the internal voltage of the generator. $r_{EG} =$ the internal electrical resistance of the generator. r_{EC} and $L =$ the electrical resistance and inductance of the voice coil. $z_{EM} =$ the motional electrical impedance. In the mechanical circuit, $m_C =$ the mass of the cone and voice coil. $C_{MS} =$ the compliance of the suspension system. $r_{MS} =$ the mechanical resistance of the suspension system. $m_A =$ the mass of the air load. $r_{MA} =$ the mechanical resistance of the air load. $f_M =$ the mechanomotive force in the voice coil. $z_{ME} =$ the mechanical impedance due to the electrical circuit. $f_{M0} =$ the mechanomotive force of the mechanical generator.

total mechanical impedance, in mechanical ohms, of the vibrating system at the voice coil is

$$z_{MT} = r_{MS} + r_{MA} + j\omega m_C + j\omega m_A - \frac{j}{\omega C_{MS}} \qquad 6.1$$

where $r_{MS} =$ mechanical resistance of the suspension system, in mechanical ohms,

$r_{MA} =$ mechanical resistance of the air load, in mechanical ohms,

$m_C =$ mass of the cone and the voice coil, in grams,

$m_A =$ mass of the air load, in grams, and

$C_{MS} =$ compliance of the suspension system, in centimeters per dyne.

Equation 6.1 may be written as follows

$$z_{MT} = r_{MS} + r_{MA} + jx_{MC} + jx_{MA} - jx_{MS} \qquad 6.2$$

where r_{MS} = mechanical resistance of the suspension system, in mechanical ohms,

r_{MA} = mechanical resistance of the air load, in mechanical ohms,

$x_{MC} = \omega m_C$ = mechanical reactance of the voice coil and cone, in mechanical ohms,

$x_{MA} = \omega m_A$ = mechanical reactance of the air load, in mechanical ohms, and

$x_{MS} = \dfrac{1}{\omega C_{MS}}$ = mechanical reactance of the suspension system, in mechanical ohms.

The mechanical resistance and mechanical reactance of the air load may be obtained from Sec. 5.10 and Fig. 5.2.

The motional electrical impedance,[2] in abohms, of the mechanical system is

$$z_{EM} = \frac{(Bl)^2}{z_{MT}} \qquad 6.3$$

where B = flux density in air gap, in gausses,

l = length of the conductor in the voice coil, in centimeters, and

z_{MT} = total mechanical impedance of the mechanical system, in mechanical ohms.

The efficiency of the loudspeaker is the ratio of the sound power output to the electrical power input. The efficiency, in per cent, may be obtained from the voice coil circuit of Fig. 6.1 and expressed as follows,

$$\mu = \frac{r_{ER}}{r_{EC} + r_{EM}} \times 100 \qquad 6.4$$

where r_{ER} = component of the motional electrical resistance due to the radiation of sound, in abohms,

r_{EM} = total motional electrical resistance, in abohms, and

r_{EC} = damped electrical resistance of the voice coil, in abohms.

The components r_{ER} and r_{EM} may be obtained from equations 6.2 and 6.3.

From equations 6.2, 6.3, and 6.4, the efficiency, in per cent, of the loudspeaker is

$$\mu = \frac{(Bl)^2 r_{MA}}{(Bl)^2(r_{MS} + r_{MA}) + r_{EC}[(r_{MC} + r_{MA})^2 + (x_{MA} + x_{MC} - x_{MS})^2]} \times 100 \quad 6.5$$

[2] Olson, " Dynamical Analogies," D. Van Nostrand Company, Princeton, N.J., 1943.

To simplify the discussion assume that the mechanical reactance and mechanical resistance of the suspension system are zero. The mechanical impedance characteristics of the mechanical system are shown in Fig. 6.2. Since r_{MA} is small compared to x_{MA} and x_{MC}, equation 6.5 becomes

$$\mu = \frac{(Bl)^2 r_{MA}}{r_{EC}(x_{MA} + x_{MC})^2} \times 100 \qquad\qquad 6.6$$

SYSTEM	A	B	C
DIAMETER, INCHES	16	4	1
MASS OF CONE, GRAMS	40	1	.015
MASS OF VOICE COIL, GRAMS	4	.35	.015
COMPLIANCE, SUSPENSION	3.2×10^{-7}	8.0×10^{-7}	5.3×10^{-7}
MECH. RES., SUSPENSION	2400	200	110
VOICE COIL MATERIAL	CU.	CU.	AL.
AIR GAP FLUX GAUSSES	10000	10000	10000
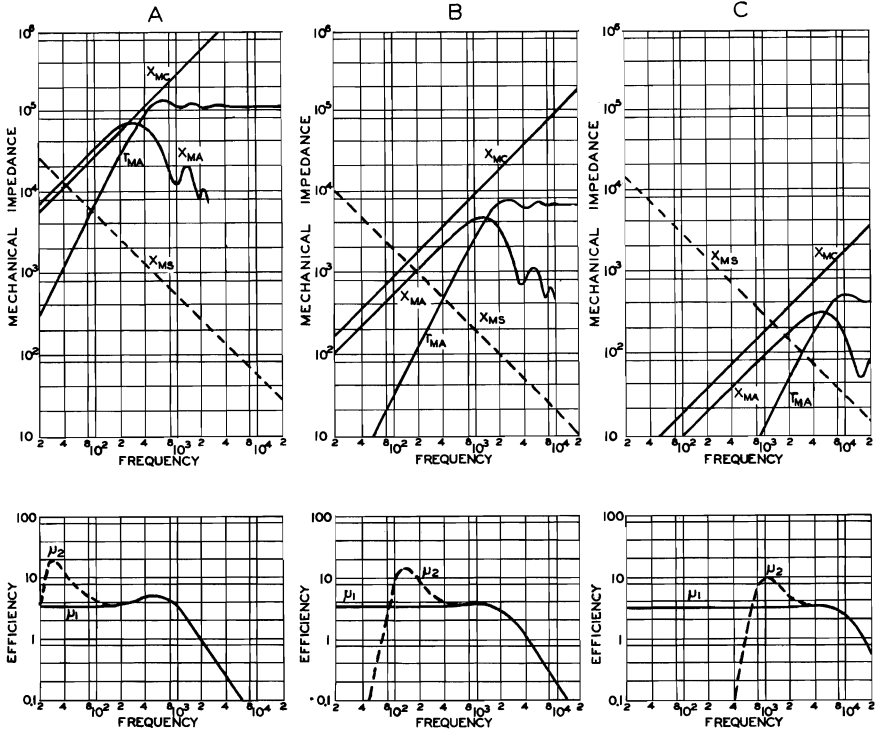
FIG. 6.2. The mechanical impedance frequency characteristics of three direct radiator loudspeakers having 1-inch, 4-inch, and 16-inch diameter cones. x_{MC} = the mechanical reactance due to the cone and coil. x_{MS} = the mechanical reactance due to the suspension system. x_{MA} = the mechanical reactance due to the air load. r_{MA} = the mechanical resistance due to the air load. The efficiency characteristics shown are for the constants as shown in the table and the graphs of the mechanical impedances. In the efficiency characteristics, μ_1 = the efficiency for x_{MS} equal to zero. μ_2 = the efficiency for x_{MS} as indicated by the graph.

In terms of the resistivity and density of the voice coil, equation 6.6 becomes

$$\mu = \frac{B^2 r_{MA} m_1}{\rho K_r (x_{MA} + x_{MC})^2 10^3} \times 100 \qquad 6.7$$

where m_1 = mass of the voice coil, in grams,

ρ = density of the voice coil conductor, in grams per cubic centimeter, and

K_r = resistivity of the voice coil conductor, in microhms per centimeter cube.

The density, resistivity, and density-resistivity product of various elements are shown in Table 6.1.

TABLE 6.1. DENSITY ρ, in GRAMS PER CUBIC CENTIMETER; RESISTIVITY K_r, IN MICROHMS PER CENTIMETER CUBE AND DENSITY-RESISTIVITY PRODUCT ρK_r OF VARIOUS ELEMENTS; TEMPERATURE, 20° C.

Element	ρ	K_r	ρK_r
Sodium	.97	4.6	4.5
Lithium	.53	9.4	5.0
Potassium	.87	7.1	6.2
Calcium	1.55	4.6	7.1
Aluminum	2.70	2.82	7.6
Magnesium	1.74	4.6	8.0
Titanium	4.5	3.2	14.4
Copper	8.89	1.72	15.2
Silver	10.5	1.63	17.1
Chromium	6.93	2.6	18.0
Beryllium	1.8	10.1	18.2
Barium	3.5	9.8	34.0
Manganese	7.2	5.0	36.0
Caesium	1.9	21.2	40.2
Zinc	7.14	5.9	42.0
Gold	19.3	2.44	47.0
Molybdenum	10.2	5.7	58.0
Cadmium	8.6	7.4	64.0
Nickel	8.8	7.8	69.0
Iron	7.9	9.8	78.0
Cobalt	8.7	9.7	84.0
Tin	7.3	11.5	84.0
Tungsten	19.0	5.5	105.0
Iridium	22.4	6.5	146.0
Platinum	21.3	9.8	208.0
Lead	11.0	22.0	242.0
Antimony	6.6	41.7	275.0
Bismuth	9.7	119.0	1116.0
Mercury	13.5	95.7	1290.0

The relation between the efficiency and the ratio of the mass of the voice coil to the mass of the cone and the air load may be obtained from equation 6.7 and is depicted in Fig. 6.3. The maximum efficiency occurs when the mass of the voice coil is equal to the mass of the cone and air load.
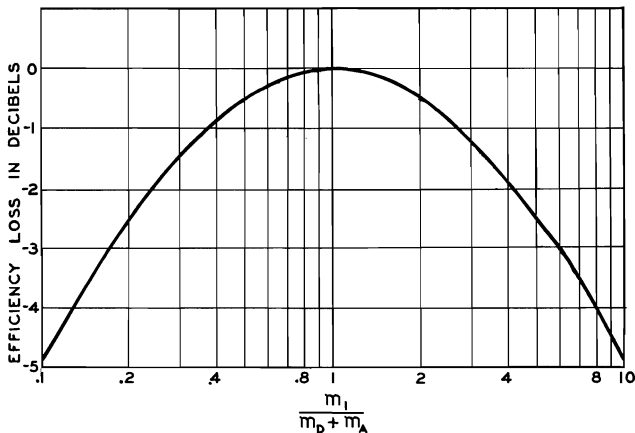
FIG. 6.3. The efficiency loss in a direct radiator loudspeaker as a function of the ratio $\dfrac{m_1}{m_D + m_A}$, where $m_1 =$ the mass of the voice coil, $m_D =$ the mass of the diaphragm, $m_A =$ the mass of the air load. The maximum efficiency is arbitrarily depicted as 0 db.

In general, in commercial loudspeakers[3,4] it is not practical to make the cone mass equal to the voice-coil mass. As a matter of fact, the cone mass is usually several times the voice-coil mass. A consideration of equation 6.7 shows that the efficiency can be increased by the use of a light-weight
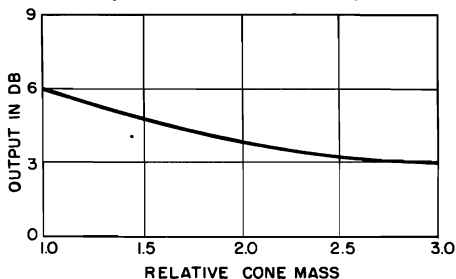
FIG. 6.4. Output of a typical direct radiator loudspeaker as a function of the mass of the cone.

cone. The relative output of a typical direct radiator loudspeaker as a function of the weight of the cone is shown in Fig. 6.4. In this example, the weight of the permanent magnet was kept constant. However, the mass of the voice coil and the air gap were selected to obtain maximum output.

[3] Olson, H. F., *Audio Engineering*, Vol. 34, No. 10, p. 5, 1950.
[4] Olson, Preston, and May, *Jour. Aud. Eng. Soc.*, Vol. 2, No. 4, p. 219, 1954.

There is a limit to the extent to which the reduction in mass of the cone can be carried because, as the cone mass is reduced, the strength of the cone is reduced and as a consequence the nonlinear distortion is increased due to overload of the material of the cone. High sensitivity and low distortion are not compatible. In order to obtain low nonlinear distortion, a relatively heavy cone must be used. The subject of nonlinear distortion and cone weight will be discussed in Sec. 6.26.

The mechanical impedance and corresponding efficiency characteristics assuming the mechanical reactance due to the suspension to be zero are shown in Fig. 6.2. The air load mechanical resistance and mechanical reactance are assumed to be the same as those on two sides of a vibrating piston with the diameter equal to the cone diameter (see Sec. 5.8). The weights of the cones and voice coils are typical of loudspeakers in actual use today. It will be seen that the efficiencies of all three systems are practically the same. Of course, the power-handling capacity of the smaller cones is very small at the lower frequencies.

In the preceding considerations the mechanical reactance due to the suspension system was assumed to be zero. The efficiency in which all the elements of the vibrating system are included may be obtained from equation 6.5. The mechanical, resistance r_{MC}, due to the suspension system is also a factor in the efficiency in the region of resonance. Typical values of r_{MC} for 16-, 4-, and 1-inch cones are shown in Fig. 6.2. The efficiency characteristics under these conditions are shown in Fig. 6.2. It will be noted that the efficiency is high at the resonant frequency. However, when coupled to a vacuum tube driving system the motional electrical impedance is also increased which reduces the power input to the voice coil. For this reason the response is not accentuated to the degree depicted by the peak in the efficiency characteristic. It will be seen that the efficiency decreases very rapidly below the resonant frequency. Therefore, in a direct radiator loudspeaker the limit at the low-frequency end of the frequency range is determined by the resonant frequency of the system.

The motional electrical impedance of a dynamic loudspeaker is given by equation 6.3. The normal electrical impedance, in abohms, of voice coil is given by

$$z_{EN} = z_{EM} + z_{ED} \qquad\qquad 6.8$$

where z_{EM} = motional electrical impedance, in abohms, and

$\quad z_{ED}$ = electrical impedance of the voice coil in the absence of motion, that is blocked, in abohms.

The components of the motional electrical impedance are shown in Fig. 6.5. At the resonant frequency the motional electrical impedance is large because the mechanical impedance is small. The current in the voice coil circuit may be determined from the voice coil electrical circuit, the driving voltage and the electrical resistance of the generator.

The mechamotive force or driving force,[5] in dynes, applied to the mechanical system is

$$f_M = Bli \qquad 6.9$$

where B = flux density in the air gap, in gausses,

$\quad l$ = length of the conductor, in centimeters, and

$\quad i$ = current in the voice coil circuit, in abamperes.

This is the driving force, f_M, applied to the mechanical system as shown in Fig. 6.1.
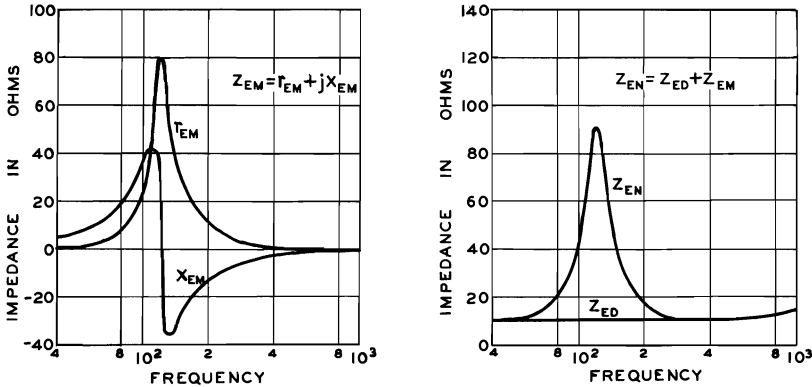
FIG. 6.5. The electrical impedance characteristics of the voice coil in a direct radiator loudspeaker. z_{EN} = the normal electrical impedance. z_{ED} = the damped electrical impedance. z_{EM} = the motional electrical impedance. r_{EM} = the resistive component of the motional electrical impedance. x_{EM} = the reactive component of the motional electrical impedance.

The mechanical impedance,[6] in mechanical ohms, due to the electrical circuit is

$$z_{ME} = \frac{(Bl)^2}{z_{ET}} \qquad 6.10$$

where $z_{ET} = r_{EC} + j\omega L + r_{EG}$,

$\quad r_{EC}$ = damped electrical resistance of the voice coil, in abohms,

$\quad L$ = damped inductance of the voice coil in abhenries, and

$\quad r_{EG}$ = electrical resistance of the generator, in abohms.

This mechanical impedance appears in the mechanical system as shown in Fig. 6.1. In calculating the steady state performance the driving force, f_M, applied to the mechanical system is used and the mechanical impedance due to the electrical system need not be considered. However, in computing the transient response of the system, the damping constant, etc., the

[5] Olson, " Dynamical Analogies," D. Van Nostrand Company, Princeton, N.J., 1943.

[6] *Ibid.*

mechanical impedance due to the electrical circuit must be included. The driving force of the generator in the mechanical system which will produce a force, f_M, across the mechanical system is

$$f_{M0} = f_M + \frac{f_M z_{ME}}{z_{MT}}$$ 　　　　6.11

The increase of electrical impedance of the voice coil, with frequency, in combination with the existing vacuum tube driving system, is another factor which reduces the response of a dynamic loudspeaker at the higher frequencies. The electrical impedance characteristics of the vacuum tube power amplifiers are generally designed so that the voltage across the loud-speaker, for constant voltage applied to the input of the power stage, is independent of the frequency. Therefore, the current in the voice coil decreases with frequency as the electrical impedance increases with frequency. The electrical impedance frequency characteristics of several voice coils are shown in Fig. 6.6. In the case of a large, heavy voice coil
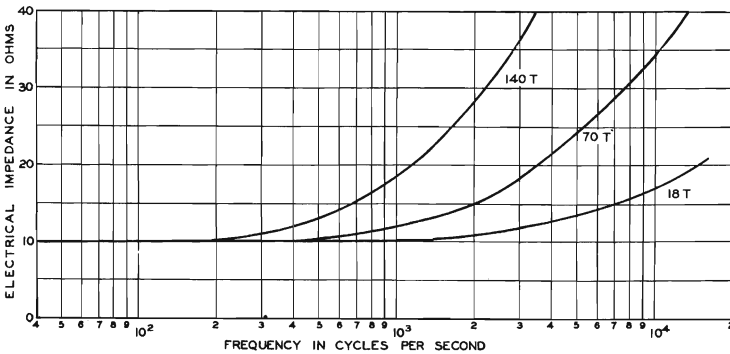
Fig. 6.6. The electrical impedance characteristics of $1\frac{1}{2}$-inch diameter voice coils of 140, 70, and 18 turns and all having 10 ohms d-c resistance.

the rapid increase of the electrical impedance at the higher frequencies causes a corresponding reduction in the driving force. To maintain the driving force at the higher frequencies requires a relatively low ratio of the inductive electrical reactance to the electrical resistance which for a constant value of the electrical resistance is equivalent to a reduction in the mass of the voice coil.

The response of a loudspeaker is a measure of the sound pressure produced at a designated position in the medium with the electrical input, frequency, and acoustic conditions specified. In general, the response is obtained on the axis of the cone. If the loudspeaker were nondirectional, the efficiency characteristic would also be the response frequency characteristic. The system is not nondirectional but is similar to that of a vibrating piston, in

that the directional becomes sharper with increase in frequency. However, the piston directional pattern cannot be used because there is considerable deviation from piston action in a cone loudspeaker. Measured directional characteristics of direct radiator loudspeakers, having the constants given
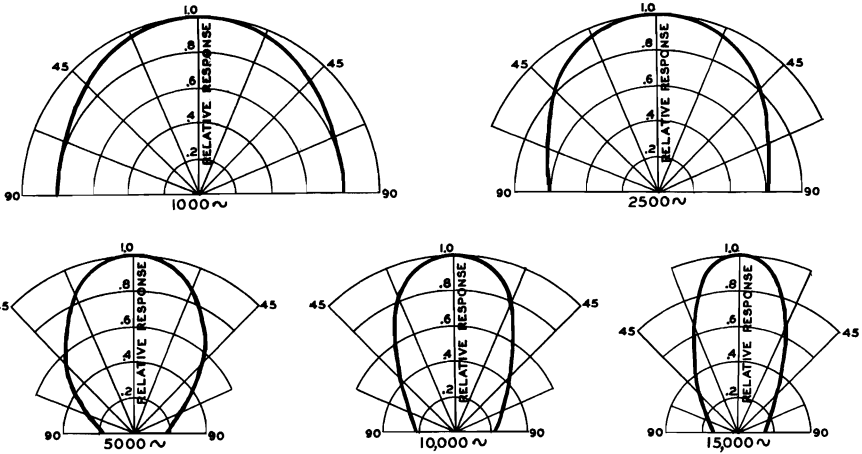
FIG. 6.7. Directional characteristics of a dynamic, direct radiator loudspeaker, with a 110° cone 4 inches in diameter, mounted in a large baffle.
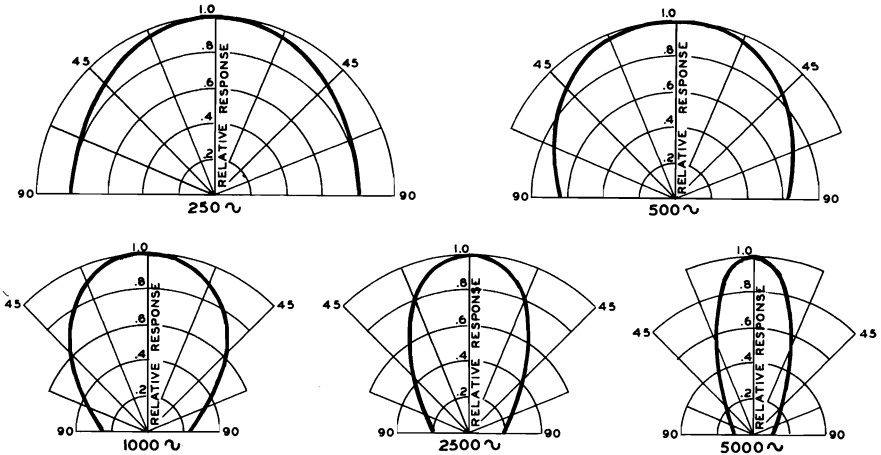
FIG. 6.8. Directional characteristics of a dynamic, direct radiator loudspeaker with a 110° cone 16 inches in diameter, mounted in a large baffle.

in Fig. 6.2, are shown in Figs. 6.7 and 6.8. Employing the mechanical circuit and the electrical circuit of Fig. 6.1 and the data of Fig. 6.2, the total output of the loudspeaker may be determined as outlined in this section. It is quite obvious that the response on the axis will be accentuated at the high frequencies due to the sharpening of the directional pattern.

The power output of a loudspeaker may be obtained from the directional pattern and the response frequency characteristic by considering the sound flow through a spherical surface in which the loudspeaker is located at the center (see Sec. 10.3D1). The surface is divided into incremental areas and the power transmitted through each area is determined from the sound pressure. The total power is equal to the summation of the incremental areas and may be expressed as

$$P = \frac{10^{-7}}{\rho c} \int \int p^2 dS \qquad\qquad 6.12$$

where P = total power, in watts,

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

   c = velocity of sound in the medium, in centimeters per second,

   p = root mean square sound pressure over the element of area dS, in dynes per square centimeter, and

   dS = element of area on the spherical surface, in square centimeters.

In the case under consideration the power output, P, as a function of the frequency may be determined from equation 6.5 and the electrical input. The directional patterns for the cones having diameters of 4 and 16 inches are shown in Figs. 6.7 and 6.8. From these data, the pressure on the axis may be determined from equation 6.12. The computed response frequency characteristics of the loudspeakers of Fig. 6.2 are shown in Fig. 6.9. These characteristics are quite similar to the actual response frequency characteristics.

Another factor of interest in a direct radiator is the power handling capacity. The sound power output, in watts, is given by

$$P = (r_{MA} \dot{x}^2) 10^{-7} \qquad\qquad 6.13$$

where r_{MA} = mechanical resistance, in mechanical ohms, obtained from Sec. 5.8, and

   $\dot{x}$ = root-mean-square velocity of the piston, in centimeters per second.

Equation 6.13 may be used to compute the power output of a direct radiator loudspeaker in the region were all parts of the cone move in phase. In general, the output is limited by the permissible amplitude. The greatest amplitude occurs at the low frequencies where the action is essentially that of a piston. Therefore, piston action may be assumed.

The peak amplitude characteristics of a 16-inch, a 4-inch, and a 1-inch piston mounted in an infinite baffle for 1 watt of sound output are shown in Fig. 6.10. These characteristics show that for practical amplitudes a relatively large piston is required to deliver adequate power at the low frequencies.

The directional pattern of a vibrating paper cone depends on three principal factors: the cone diameter, the cone angle, and the frequency.

Other factors, such as the paper pulp, the processing, the corrugations, the voice coil diameter, and the suspension also influence the directional pattern, but in a lesser degree. The directional patterns for various frequencies of 110° cones having diameters of 4 and 16 inches are shown in
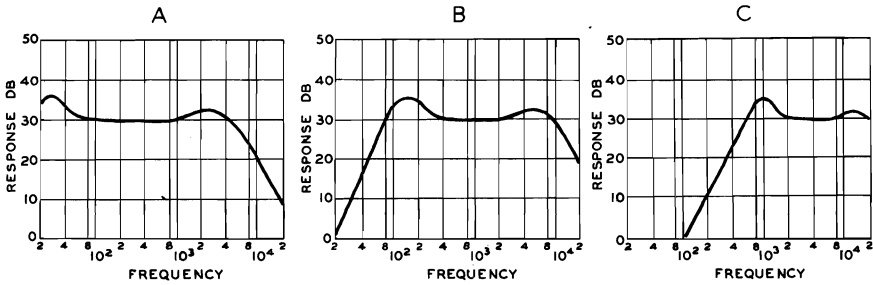
FIG. 6.9. Pressure response frequency characteristics of the loudspeakers of Fig. 6.2 having cone diameters of 1 inch, 4 inches, and 16 inches.
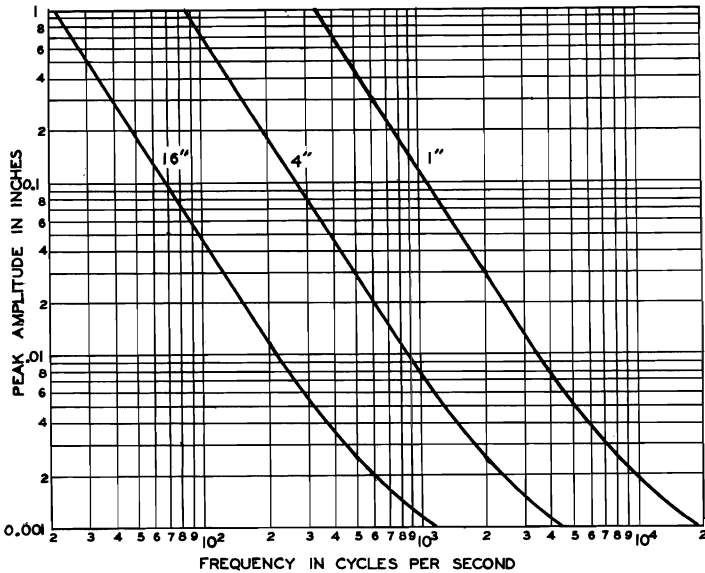
FIG. 6.10. The amplitude frequency characteristics of vibrating pistons of various diameters, mounted in an infinite wall, for 1-watt output on one side.

Figs. 6.7 and 6.8. It will be see that the directional pattern becomes sharper with increase in frequency. However, the pattern is broader than that of a vibrating piston of the same diameter due to the relatively low velocity of propagation of sound in the paper cone. The directional patterns

of 130° and 100° cones 4 inches in diameter are shown in Fig. 6.11. It
will be seen that the directional pattern at the high frequencies becomes
broader as the cone angle is increased. This is to be expected because the
velocity of propagation of sound in the paper cone is about two times
the velocity of sound in air. Under these conditions the delay between the
sound emitted from the outside and the center of the cone will increase as
the angle of the cone is increased. As a result the directional pattern will
be broadest for the cone with the widest angle. The preceding observations
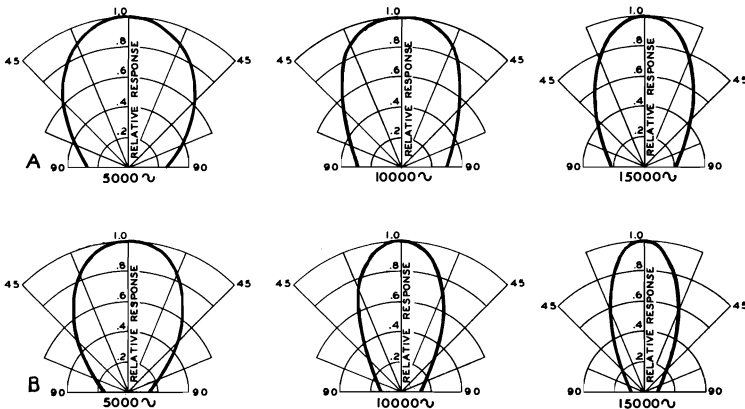
FIG. 6.11. Directional characteristics of dynamic direct radiator loudspeakers with
cones 4 inches in diameter for two different cone angles. Row A. 130° cone. Row B.
100° cone.

with regard to cone type vibrators may be substantiated by theoretical
considerations as outlined in Sec. 2.21.

The characteristics of Figs. 6.2 and 6.9 show that the low-frequency
efficiency may be maintained to the higher frequency ranges by employing
a small and relatively light weight cone and voice coil. On the other hand,
to obtain adequate power handling capacity at the low frequencies with
tolerable excursions of the vibrating system requires a cone of relatively
large area. To insure operation below the elastic limits of the materials,
a cone of large area must be of sturdy construction. Equation 6.7 and Fig.
6.3 show that a large heavy cone also requires a relatively large voice coil
in order to maintain a tolerable efficiency. The efficiency of this system
is low in the high-frequency range. Furthermore, the directional pattern
of a large cone becomes quite narrow in the high-frequency range. Where
the frequency range is confined within the limits of from 80 cycles to
4000 cycles, satisfactory efficiency, response, and directional characteristics
can be obtained from a single-cone, single-coil loudspeaker. The above
discussions show that to obtain adequate power handling capacity and
uniform response over a wide frequency range (greater than 80 to 4000

cycles) requires a relatively large diameter, heavy diaphragm, and large coil at the lower frequencies, and a relatively light diaphragm and coil to obtain good efficiencies at the higher frequencies. There are a number of direct radiator loudspeaker systems which may be built to satisfy these conditions. It is the purpose of the sections which follow to consider a number of these systems.

6.3. Multiple Single-Cone, Single-Coil Loudspeaker. — Several arrangements for obtaining uniform response, broad directional pattern, adequate power handling capacity, and tolerable efficiency are shown in Fig. 6.12.

The systems of Fig. 6.12A, C, and D consist of a large diameter heavy cone driven by large voice coil for the low-frequency range and a small
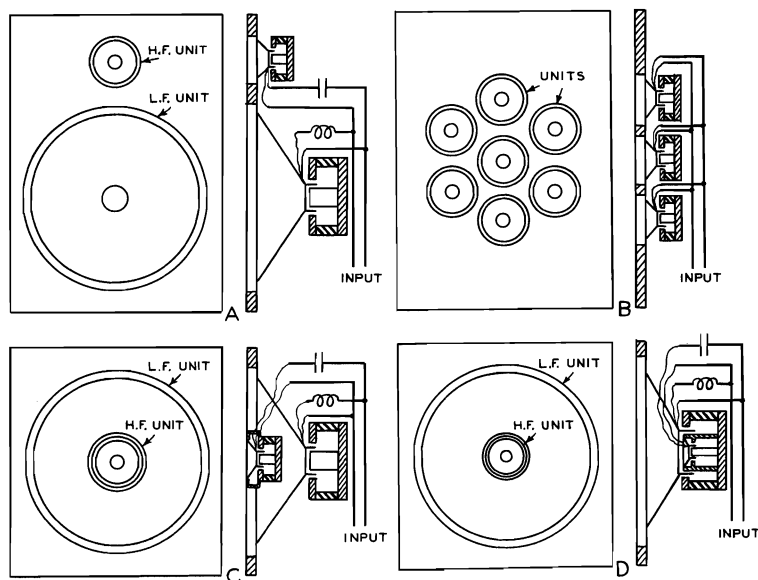
FIG. 6.12. Multiple single-cone, single-coil, direct radiator, dynamic, loud-speaker systems. A, C, and D. Large low-frequency unit, small high-frequency unit, and filter system. B. Seven small units connected in parallel.

diameter light cone and small voice coil for the high-frequency range and a filter system for allocating the power in the high- and low-frequency ranges to the respective low- and high-frequency units. The filter system consists of an inductance in series with the low-frequency unit and a condenser in series with the high-frequency unit. Due to the large inductance of the large voice coil, as shown in Fig. 6.6, it has been found that for most applications the inductance in series with the low-frequency unit may be omitted. On the other hand, if a more elaborate filter system is required, the circuit of Fig. 7.16 may be used.

In Fig. 6.12A the low- and high-frequency units are separated by a relatively large distance. In the overlap frequency region this distance may be more than 1 wavelength. The directional patterns of two sources shown in Fig. 2.3 are applicable to this system. These characteristics show that two separated sources exhibit directional patterns with one or more lobes with very low response between the lobes. The result is frequency discrimination, for points removed from the axis, in the overlap region. This condition is reduced in Fig. 6.12C but is not eliminated. However, a disadvantage of the system of Fig. 6.12C is that sound diffracts around the high-frequency unit and is reflected from the large cone causing a ragged response due to interference between the direct and reflected sound.

The objectional features of Fig. 6.12A and C referred to above have been eliminated in Fig. 6.12D. In this system[7] the large cone is geometrically a continuation of the small cone. Therefore, in the overlap
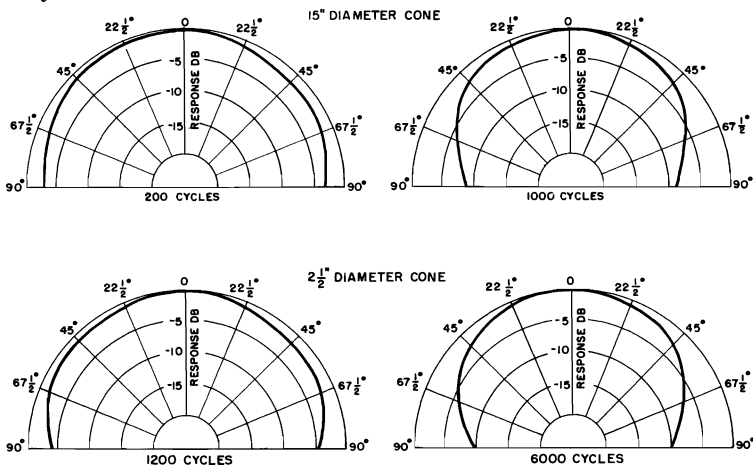
FIG. 6.13. Directional characteristics of direct radiator loudspeakers with cone diameters of 15 inches and 2.5 inches.

region the two cones vibrate together as a single cone. In this way phase and diffraction effects are eliminated.

In a two-unit loudspeaker, employing a large cone for the reproduction of the low-frequency range and a small cone for the reproduction of the high-frequency range, a uniform directivity pattern can be obtained over the entire audio-frequency range. This has been described in connection with Figs. 6.6 and 6.7. This is illustrated further in Fig. 6.13 in which the directivity patterns of 15-inch and 2½-inch cone loudspeakers are compared for a six to one ratio of frequency, that is, for a constant ratio of diameter to wavelength. Fig. 6.13 shows that the directivity pattern of a 15-inch loudspeaker at 200 to 1000 cycles corresponds to that of a 2½-inch

[7] Olson and Preston, *RCA Review*, Vol. 7, No. 2, p. 155, 1946.

loudspeaker at 1200 and 6000 cycles. These relationships were used in designing the two units of the system shown in Fig. 6.12D.

In the loudspeaker[8,9] shown in Fig. 6.12D, small cones may be attached to the large cone to reduce the velocity of wave propagation in the large cone. Fig. 6.14. This broadens the directivity pattern of the low-frequency cone. In the high-frequency range, the conical domes attached to the surface of the low-frequency cone improve the performance in three ways: by decreasing the angle into which the high-frequency cone feeds, thereby increasing the output of the high-frequency cone; by diffusely reflecting some of the sound emitted by the high-frequency cone, thereby eliminating discrete reflections; and by diffracting some of the sound emitted by the high-frequency cone, thereby broadening the directivity pattern.

The angles into which the high-frequency cone feeds, without and with the conical domes applied to the low-frequency cone of Fig. 6.14, are designated as ϕ_1 and ϕ_2 in Fig. 6.15A and Fig. 6.15B. Since ϕ_2 is smaller than ϕ_1, the acoustic radiation load upon the cone is greater with the conical domes than without them. When the acoustic radiation load upon a direct radiator loudspeaker is increased, the sound power output is increased. Thus, it will be seen that the conical domes increase the high-frequency sound radiated by the high-frequency cone. In other words, the high-frequency efficiency is improved.

Some of the sound emitted by the

FIG. 6.14. A perspective view of a duo-cone loudspeaker with domes attached to the low-frequency cone. (After Olson, Preston, and May.)

high-frequency cone is diffusely reflected by the conical domes, as shown in Fig. 6.16. Without the domes, there would be many similar reflections which would lead to reinforcements and cancellations with the direct radiation. The result would be corresponding peaks and dips in the response of the high-frequency cone. With the domes, the symmetry of the low-frequency cone is upset and there are many reflections in different directions and of different path lengths. The reflections, therefore, cancel out and the net result is a smooth response-frequency characteristic.

Some of the sound emitted by the high-frequency cone is diffracted by the conical domes as shown in Fig. 6.17. By diffraction is meant the bending

[8] Olson, H. F., *Radio and Television News*, Vol. 51, No. 2, p. 69, 1954.

[9] Olson, Preston, and May, *Jour. Aud. Eng. Soc.*, Vol. 2, No. 4, p. 219, 1954.

of the sound around an obstacle.   The pencils of sound designated 1 and 2 in Fig. 6.17 are diffracted.   The pencils of sound designated 3 to 7, inclusive, are radiated directly from the high-frequency cone.   It will be seen that the effect of the diffracted sound is to increase the curvature of the wavefront in the direction of 1, 2, and 3.   As a result, the directivity pattern is broadened.
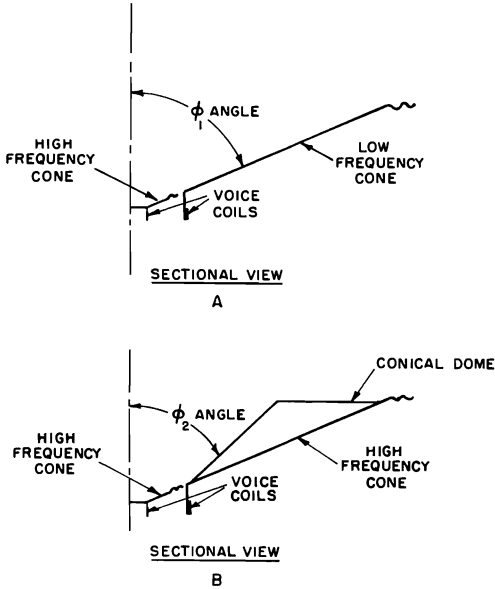
FIG. 6.15.   A.   Duo-cone loudspeaker with a plain low-frequency cone.   B.   Duo-cone loudspeaker with domes attached to the low-frequency cone.

Referring to Figs. 6.2 and 6.9 it will be seen that uniform response may be obtained over a wide frequency range by means of a light cone driven by a light coil and resonant at the lower limit of the frequency range. Of course, the power handling capacity of a single unit of this type is inadequate and a multiple set of units must be employed.   The number of units required may be determined from the required power output and the allowable excursion together with equation 6.13 and Fig. 6.10.   An arrangement of seven small loudspeaker units mounted in a flat baffle with the voice coils connected in parallel is shown in Fig. 6.12B.   The voice coils of the loudspeakers may, of course, be connected in parallel, series, or series-parallel. In order to obtain better high-frequency spatial distribution the units may be inclined at various angles, for example, the units may be mounted so that the resulting vibrating surface approximates a spherical surface (see Sec. 2.20).
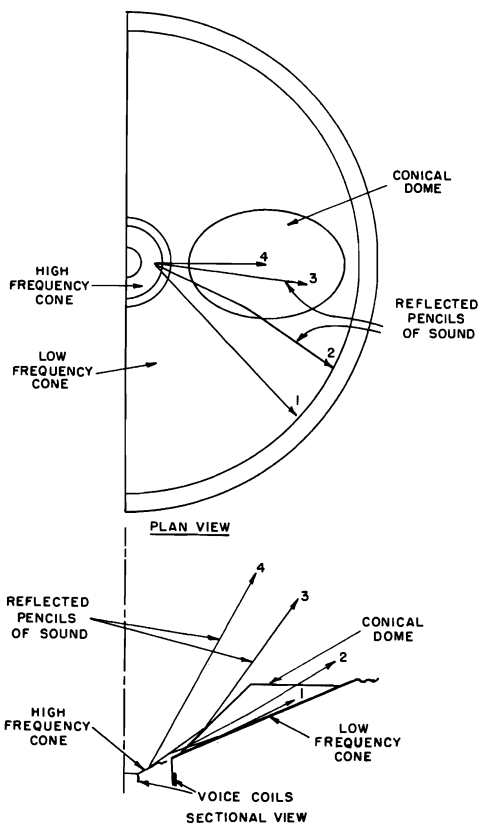
FIG. 6.16.   Diffuse reflections of the sound emitted by
the high-frequency cone by the domes attached to the
low-frequency cone.

The frequency range of a direct radiator loudspeaker may be increased
by sectionalizing the coil or cone or both and thereby reducing the mechanical
impedance and electrical impedance or both at the higher frequencies.
These systems will be considered in the sections which follow.

6.4. Single-Coil, Double-Cone Loudspeaker.[10]—A typical single-coil,
double-cone loudspeaker, Fig. 6.18B, consists of a single coil coupled to two
cones.   In this system an increase in frequency range is obtained by reducing
the mechanical impedance of the diaphragm by coupling a smaller cone to

[10] Olson, H. F., *Jour. Acous. Soc. Amer.*, Vol. 10, No. 4, p. 305, 1939.
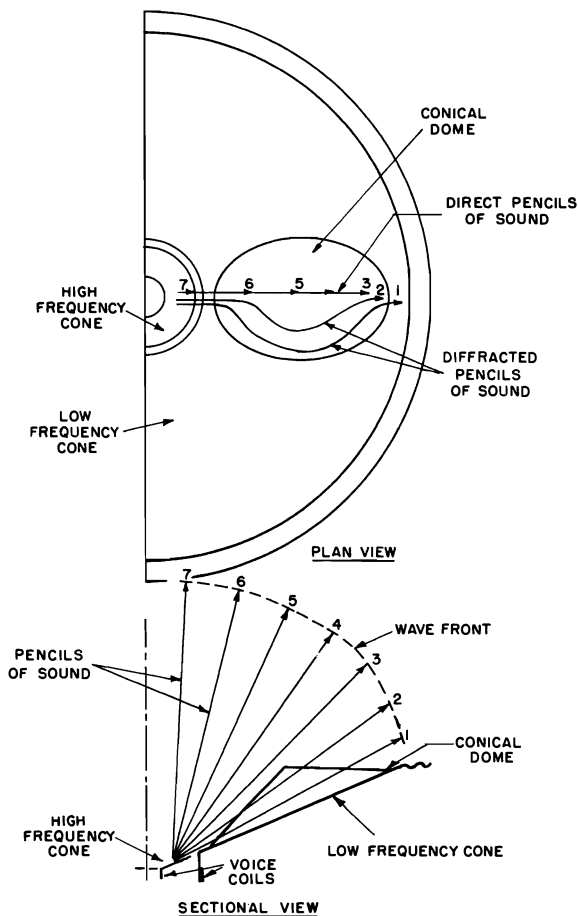
FIG. 6.17. Diffraction of the sound emitted by the high-frequency cone by the domes attached to the low-frequency cone.

the voice coil at the high frequencies. The two cones are separated by a compliance. At low frequencies the mechanical reactance of the compliance, C_M, is large compared to the mechanical impedance, z_{M1}, of the large cone and consequently the entire system moves as a whole. At high frequencies the mechanical reactance of the compliance, c_M, is small compared to the mechanical impedance, z_{M1}, of the large cone, and the small cone, z_{M2} moves while the large cone, z_{M1}, remains stationary. By means of this reduction in cone mechanical impedance the range may be extended almost a full octave, depending upon the mass and electrical impedance characteristics of the voice coil. The response characteristics of a single-coil, single-cone

loudspeaker is shown in Fig. 6.18A. The voice coil and large cone of Fig. 6.18B is the same as that of Fig. 6.18A. The high-frequency range has been extended about one-half octave without any sacrifice of power handling capacity by the addition of the small cone.

6.5. Double-Coil, Single-Cone Loudspeaker.[11]—The double-coil, single-cone loudspeaker, Fig. 6.18C, consists of a voice coil, divided into two parts separated by a compliance, coupled to a single corrugated cone. The inductance and electrical resistance of the larger portion of the voice coil, L_1, r_{E1}, is shunted by an electrical capacitance, C_E. At low frequencies the electrical reactance of the electrical capacitance is large compared to the electrical impedance of the larger portion of the voice coil, L_1, r_{E1}, and the mechanical reactance of the compliance, C_M, separating the two portions of the voice coil is large compared to the mechanical mass reactance of m_1, and the mechanical impedance z_{M1}. Therefore, in the low-frequency range the action is the same as that of a single-coil loudspeaker. At high frequencies the reactance of the electrical capacitance, C_E, is small compared to the electrical impedance of L_1, r_{E1} or L_2, r_{E2}; and the mechanical reactance of the compliance, C_M, is small compared to the mechanical reactance of large coil, m_1. The cone is driven by the lighter portion, m_2, of the voice coil and the heavy coil, m_1, remains stationary. In the mid-range there is a phase difference between the currents in the two portions of the voice coil. A corresponding phase shift occurs in the mechanical system. As a consequence, a smooth overlap is obtained in going from two-coil operation at the low frequencies to a single-coil operation at the high frequencies. Above the frequency of ultimate resistance the radiation resistance is a constant. In order to obtain uniform output in this range the mechanical impedance of the system must be independent of the frequency. This may be accomplished by embossing suitable corrugations in the cone which reduce the effective mass reactance. The double-coil system reduces the effective mass reactance of the voice coil as compared to a single coil, as well as the electrical impedance at the higher frequencies. A typical response characteristic of this loudspeaker is shown in Fig. 6.18C.

6.6. Double-Coil, Double-Cone Loudspeaker.[12]—The double-coil, double-cone loudspeaker, Fig. 6.18D, consists of a light coil coupled to a small cone, connected by a compliance to a heavy coil and large cone. In this system an increase in range is obtained by reducing the impedance of both the coil and the diaphragm at the higher frequencies. At low frequencies the electrical reactance of the capacitance, C_E, is large compared to the electrical impedance of the large portion of the voice coil, L_1, r_{E1}, and the same current flows in both coils. The mechanical reactance of the compliance, C_M, separating the two portions of the coil is large compared to the mechanical impedance of m_1, plus z_{M1}. Therefore, at low frequencies the system behaves as a single-coil, single-cone loudspeaker. Both parts

[11] Olson, H. F., *Proc. Inst. Rad. Eng.*, Vol. 22, No. 1, p. 33, 1034.
[12] Olson, H. F., *Jour. Acous. Soc. Amer.*, Vol. 10, No. 4, p. 305, 1939.
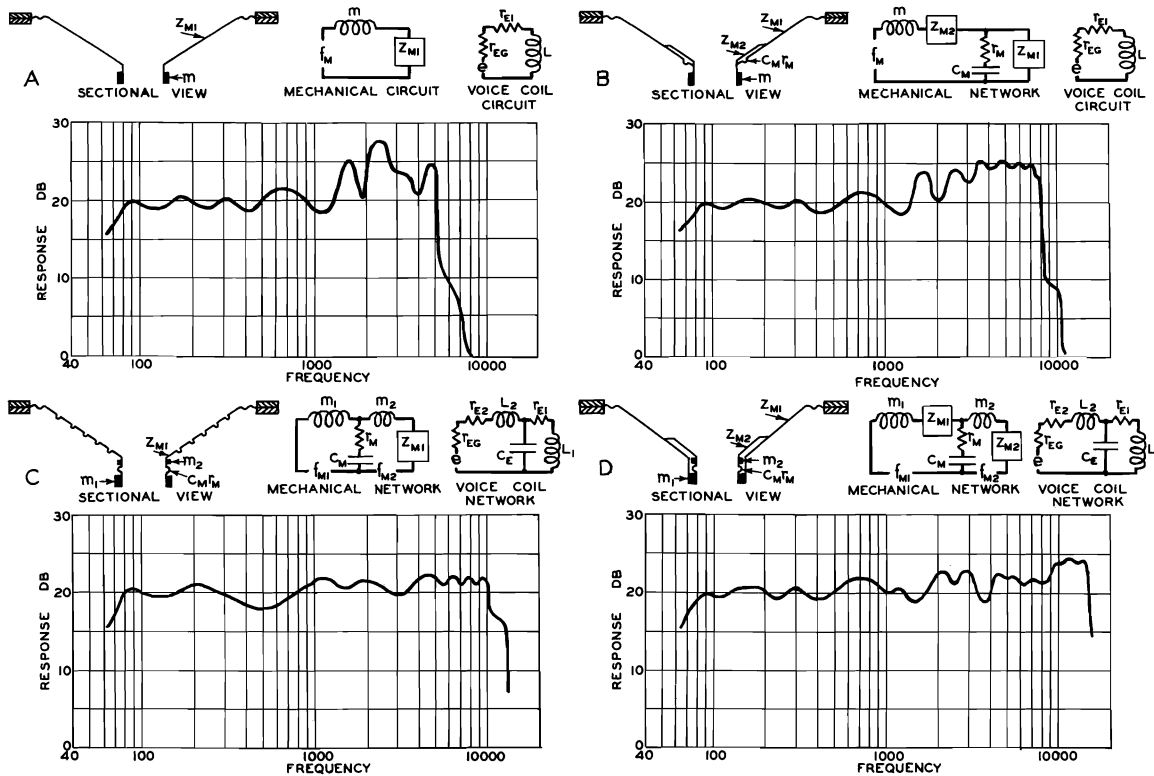
FIG. 6.18.   A. Cross-sectional view of a single-cone, single-coil loudspeaker with the voice coil electrical circuit and the mechanical circuit of the mechanical system.   In the voice coil electrical circuit, e = the internal voltage of the generator. r_{EG} = the internal electrical resistance of the generator. r_{E1} and L = the electrical resistance and the inductance of the voice coil.   In the mechanical

circuit, m = the mass of the voice coil. z_{M1} = the mechanical impedance of the cone at the voice coil. f_M = the force generated in the voice coil. The graph shows the pressure response frequency. B. Cross-sectional view of a double cone, single-coil loudspeaker with the voice coil electrical circuit and mechanical network of the mechanical system. In the voice coil electrical circuit, e = the internal voltage of the generator. r_{EG} = the internal electrical resistance of the generator. r_{E1} and L = the electrical resistance and inductance of the voice coil. In the mechanical network, m = the mass of the voice coil. z_{M1} and z_{M2} = the mechanical impedance of the large and small cones. C_M and r_M = the compliance and mechanical resistance of the corrugation in the large cone. f_M = the force-generated in the voice coil. The graph shows the pressure response frequency characteristic. C. Cross-sectional view of a single-cone, double-coil loudspeaker with the voice coil network and the mechanical network of the mechanical system. In the voice coil electrical network, e = the internal voltage of the generator. r_{EG} = the internal electrical resistance of the generator. r_{E1} and L_1 = the electrical resistance and the inductance of the large coil. r_{E2} and L_2 = the electrical resistance and inductance of the small coil. C_E = the electrical capacitance shunting the large coil. In the mechanical network, m_1 = the mass of the large coil. m_2 = the mass of the small coil. z_{M1} = the mechanical impedance of the cone at the voice coil. C_M and r_M = the compliance and mechanical resistance of the corrugation separating the large coil and small coil. f_{M1} = the force generated in the large coil. f_{M2} = the force generated in the small coil. The graph shows the pressure response frequency characteristic. D. Cross-sectional view of a double-cone, double-coil loudspeaker with the voice coil electrical network and mechanical network of the mechanical system. In the voice coil electrical network, e = the internal voltage of the generator. r_{EG} = the internal electrical resistance of the generator. r_{E1} and L_1 = the electrical resistance and the inductance of the large coil. r_{E2} and L_2 = the electrical resistance and the inductance of the small coil. C_E = the capacitance shunting the large coil. In the mechanical network, m_1 = the mass of the large coil. m_2 = the mass of the small coil. z_{M1} = the mechanical impedance of the large cone. z_{M2} = the mechanical impedance of the small cone. C_M and r_M = the compliance and mechanical resistance of the corrugation separating the large cone and coil and the small cone and coil. f_{M1} = the force generated in the large coil. f_{M2} = force generated in the small coil. The graph shows the pressure response frequency characteristic.

of the voice coil are used at low frequencies; no air-gap flux is wasted as would be the case if two separate units were used. At high frequencies the electrical capacitance, C_E, is small compared to the electrical impedance of either the heavy coil, L_2, r_{E2}, or the light coil, L_1, r_{E1}, and practically all the current flows in the light coil. The mechanical impedance of the compliance, C_M, is small compared to the mechanical reactance of m_1 and the mechanical impedance of z_{M1}. Therefore, at high frequencies the small cone, z_{M2}, is driven by the light coil, m_2, and the heavy coil, m_1, remain stationary. This system is equivalent to two separate loudspeakers. The advantage, of course, resides in the fact that only a single field structure of the size required to accommodate a coil the size of the two coils is needed. Such a field structure coil system and large cone would be required to give equivalent performance at the low frequencies. The response frequency characteristic of a double-cone, double-coil loudspeaker is shown in Fig. 6.18D. This system provides a means of obtaining good response to 14,000 cycles. At the same time, the diameter of the large cone can be chosen so that moderate power
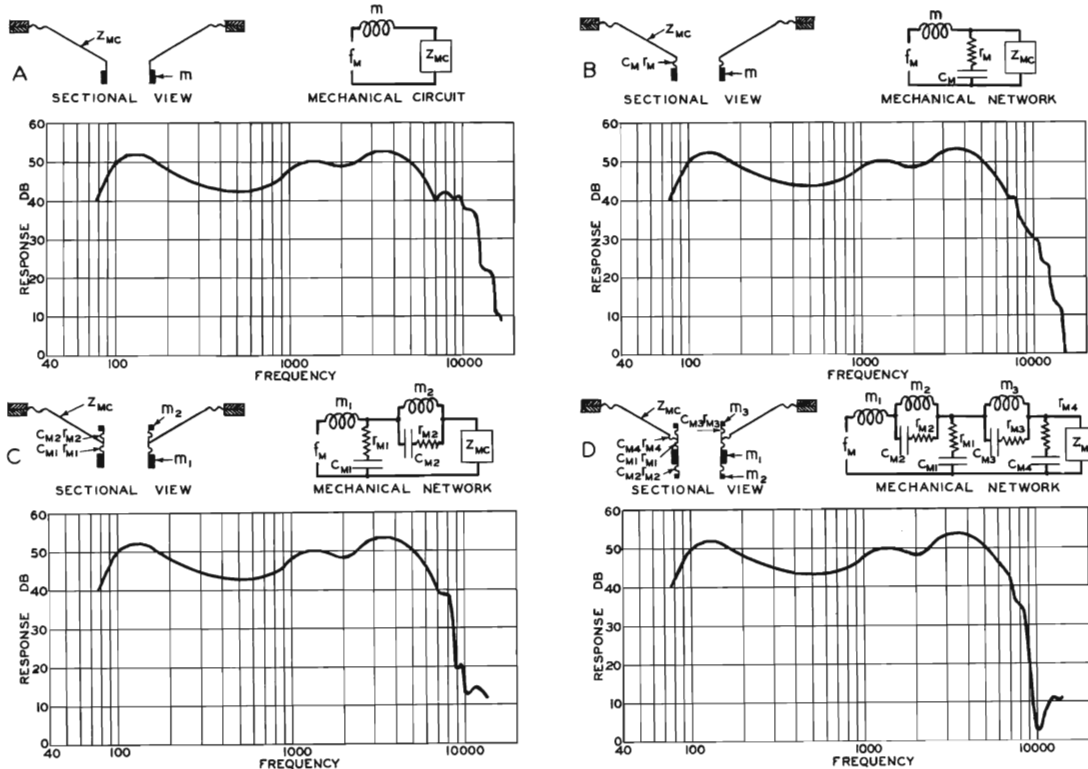
FIG. 6.19. A. Cross-sectional view of a conventional single-coil direct radiator, dynamic loudspeaker and the mechanical circuit of the mechanical system. In the mechanical circuit, m = the mass of the voice coil. z_{MC} = the mechanical impedance of the cone and suspension system. The graph shows the pressure response frequency characteristic. B. Cross-sectional view of the

loudspeaker in A with a corrugation rolled into the voice coil form separating the coil and cone. In the mechanical network, m = the mass of the coil. C_M and r_M = the compliance and mechanical resistance of the corrugation separating the coil and the cone. z_{MC} = the mechanical impedance of the cone and suspension system. The graph shows the pressure response frequency characteristic. C. Cross-sectional view of the loudspeaker in B with a mass connected by a corrugation to the front of the cone. C_{M1} and r_{M1} = the compliance and mechanical resistance of the corrugation separating the coil and the cone. C_{M2} and r_{M2} = the compliance of the corrugation connecting the mass m_2 to the cone. z_{MC} = the mechanical impedance of the cone. The graph shows the pressure response frequency characteristic. D. Cross-sectional view of the loudspeaker in C with a mass connected by a corrugation to the bottom end of the voice coil. In the mechanical network, m_1 = the mass of the voice coil. C_{M1} and r_{M1} = the compliance and mechanical resistance of the corrugation separating the coil and the cone. C_{M2} and r_{M2} = the compliance of the corrugation connecting the mass m_2 to the cone. C_{M3} and r_{M3} = the compliance and mechanical resistance of the corrugation connecting the mass m_3 to the cone. C_{M4} and r_{M4} = the compliance and mechanical resistance of the corrugation in the cone. z_{MC} = the mechanical impedance of the cone. The graph shows the pressure response frequency characteristic.

handling capacity as a direct radiator loudspeaker may be obtained at the low frequencies.

6.7. Mechanical Networks for Controlling the High-Frequency Response of a Loudspeaker.—In general, in radio and other forms of sound reproduction it is desirable to attenuate the response above a certain high-frequency limit. In some cases, it may be desirable to attenuate a certain band as, for example, 10,000 cycles in radio reproduction to eliminate the adjacent channel beat note. Electrical networks and filters are usually quite costly compared to mechanical filters for certain applications in sound reproduction. It is the purpose of this section to describe the construction and performance of several mechanical networks and filters for suppressing certain frequency bands or for attenuating the high-frequency response of a loudspeaker.

A relatively light weight, 8-inch loudspeaker was chosen for these tests. This type of loudspeaker is used in small radio receivers. Due to the small mass of the cone and coil the response is well maintained at the high frequencies. The principles involved are applicable to all loudspeakers. The loudspeaker was mounted in a 3-foot irregular baffle. The response was obtained employing a velocity microphone located on the axis of the speaker at a distance of two feet.

A. *Conventional Single-Coil Loudspeaker.*—The response frequency characteristic of the conventional loudspeaker, referred to above, is shown in Fig. 6.19A. The mechanical circuit of the mechanical system is also shown in Fig. 6.19A. The constants have been indicated as the mass of the voice coil, m_1, and the compliance of the centering suspensions, the cone mechanical impedance including the cone outside suspension and the radiation mechanical resistance, etc., lumped as z_{MC}. The response is well maintained to 12,000 cycles. For this reason, this loudspeaker is well adapted to illustrate the performance of mechanical networks for controlling the response at the higher frequencies.

B. *Loudspeaker with a Compliance Shunting the Cone Mechanical Imped-ance.*—One of the simplest means for attenuating the high-frequency response of a loudspeaker is a compliance inserted between the voice coil and the cone. This compliance, C_M, may take the simple form of a bead or corruga-tion pressed into the voice coil form. The response characteristic of a conventional loudspeaker with a compliance between the voice coil and cone is shown in Fig. 6.19B. In the mechanical network the compliance, C_M, shuts the cone mechanical impedance, z_{MC}. Comparing with Fig. 6.19A it will be seen that there is some attenuation at the higher frequencies. However, the attenuation is not large. This is due to the fact that the mechanical impedance, z_{MC}, does not increase appreciably with frequency. At the higher frequencies a light cone, in particular, does not vibrate as a piston. In a large diameter light cone the action changes gradually from piston action to wave propagation at the higher frequencies. As a conse-quence, the mechanical impedance does not increase directly with the frequency. In some loudspeakers the mechanical impedance, z_{MC}, actually decreases with frequency at the higher frequencies.

C. *Loudspeaker with a Compliance Shunting; a Compliance and Mass in Parallel, Connected in Series with the Cone Mechanical Impedance.*—In a radio receiver it is desirable to attenuate the response at 10,000 cycles so that the 10,000-cycle adjacent channel beat note will not be reproduced. A parallel circuit inserted in series with a line causes high attenuation at the resonance frequency. By inserting a parallel circuit in series with the voice coil and cone the response will be reduced at the resonant frequency. The amount of attenuation will depend upon the magnitude of the mechani-cal resistance in the compliance. An example of this system is shown in Fig. 6.19C. The mass and compliance are designated as m_2 and C_{M2}. Comparing with Fig. 6.19A the attenuation at 10,000 cycles is about 25 db. This system is also easy to fabricate. Two suitable corrugations are pressed into a single voice coil form.

D. *Loudspeaker with a "T" Type Filter Connecting the Voice Coil Mass and the Cone Mechanical Impedance.*—This system, Fig. 6.19D, consists of two parallel resonant mechanical circuits, or a parallel resonant mechani-cal circuit, m_2 and C_{M2}, connected to the bottom of the voice coil of the system of Fig. 6.19C. The mechanical network is also shown in Fig. 6.19D. The system then is a "T" type low-pass mechanical filter connecting the coil and cone. Very high attenuation is obtained at the resonant frequency of the arms. The response frequency characteristic of this system is shown in Fig. 6.19D. Comparing with Fig. 1.19A the attenuation at 10,000 cycles is 35 db. The attenuation is also quite high above 10,000 cycles. As in the other systems it is made by simply pressing three corrugations into a single voice coil form.

Several mechanical networks for controlling and suppressing the response of a loudspeaker at the high frequencies have been described. Some of these systems are in use in practically all loudspeakers. The cost of the system is very small compared to an electrical network for accomplishing

the same result because the mechanical networks are made by simply placing corrugations in the voice coil form. These examples also illustrate the value of analogies of electrical circuits in designing and in predicting the action of mechanical systems.

6.8. Loudspeaker Baffles.—A baffle is a partition which may be used with an acoustical radiator to increase the effective length of the acoustical transmission path between the front and back of the radiator. The term baffle is commonly applied to a plane surface. When a direct radiator loudspeaker is mounted in a baffle, there exists at 180° phase difference between the front and back of the cone. When the baffle is small compared to the wavelength the system is an acoustic doublet (see Secs. 2.3 and 5.14). In this frequency range the power output for constant velocity is proportional to the fourth power of the frequency (see Sec. 5.14). When the baffle is large compared to the wavelength, the two sides of the cone act independently and the sound power output is proportional to the square of the frequency (see Secs. 2.2 and 5.8). In the case of a mass controlled system the velocity is inversely proportional to the frequency. A mass controlled system is a system in which a positive mechanical reactance is the controlling mechanical impedance. Therefore, in the case of the large baffle the sound power output will be independent of frequency (see Sec. 6.2). However, when the dimensions of the baffle are small compared to the wavelength, the power output in the case of a mass controlled system is proportional to the square of the frequency. In this frequency range the low-frequency response falls off rapidly. The transition between doublet operation and independent operation is quite marked. This transition point occurs when the dimensions of the baffle are slightly less than one-half wavelength.

It is the purpose of this section to consider the action of various types of baffles and loud-speaker systems.

A. *Irregular Baffle.*—In the case of a cone in a square baffle the path from the front to the back is practically the same for all possible paths. Therefore, some peculiarities in the response would be expected when the acoustical path from the front to back is equal to a wavelength. At this frequency the sound that is diffracted around the baffle and transmitted forward will interfere destructively with the radiation from the front. The pressure response characteristics of Fig. 6.20A show "dips" in the response when the acoustical path from front to back is a wavelength. Using an irregular baffle, Fig. 6.20B, it is possible to reduce this interference and obtain a uniform response characteristic. In this baffle the various paths from front to back differ and the destructive interference is spread over a wide frequency range. The pressure response frequency characteristics of an irregular baffle, Fig. 6.20B, show that the dip in the response frequency characteristic of the square baffle is eliminated by the use of an irregular baffle.

B. *Large Baffle, Different Resonant Frequencies.*—The radiation mechanical resistance of a vibrating piston in an infinite baffle is proportional to the square of the frequency in the range below the frequency where the

radiation resistance attains its ultimate value. Referring to equation 6.5 it will be seen that the power output of a direct radiator loudspeaker will be independent of the frequency in the frequency range above the resonant frequency up to the frequency of ultimate mechanical resistance, and will
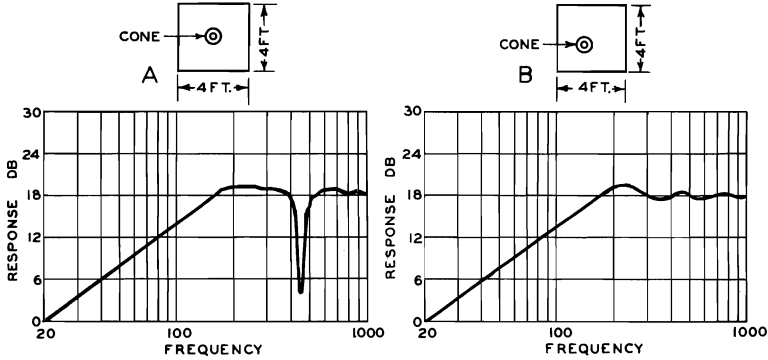
FIG. 6.20. Pressure response frequency characteristics of mass-controlled, direct radiator, dynamic loudspeaker mechanisms, with 10-inch diameter cones, mounted in square baffles. In A, the loudspeaker mechanism is mounted in the center of the baffle. In B, the loudspeaker mechanism is mounted unsymmetrically to eliminate interference.

be proportional to the fourth power of the frequency below the resonant frequency. The measured pressure response frequency characteristic of a direct radiator loudspeaker having a fundamental resonance of 50, 100, and 200 cycles is shown in Fig. 6.21. It will be seen that the pressure
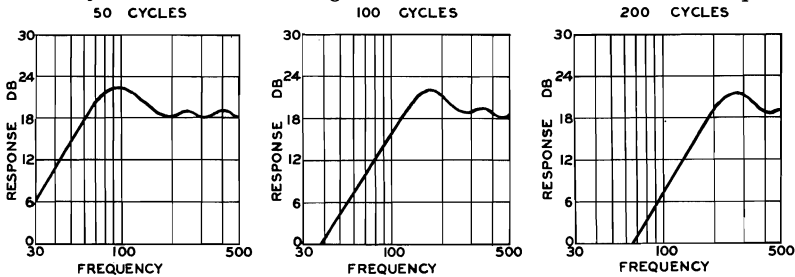
FIG. 6.21. Pressure response frequency characteristics of direct radiator, dynamic loudspeaker mechanisms, with 10-inch diameter cones, with resonant frequencies of 50, 100, and 200 cycles, mounted in very large baffles.

response is independent of the frequency in the frequency range above the resonant frequency. Below the resonant frequency the pressure response falls off 12 db per octave. These results agree with that predicted by theory.

C. *Low Resonant Frequency, Different Baffle Sizes.*—The radiation mechanical resistance of a vibrating piston in a finite baffle is proportional to the fourth power of the frequency when the dimensions of the baffle are small

compared to the wavelength, doublet operation, and proportional to the square of the frequency when the dimensions are comparable to or greater than the wavelength in the range below the ultimate mechanical resistance (see Secs. 2.2, 5.8, and 5.14). If the considerations are confined to the frequency range above the resonant frequency of the mechanism, the velocity

FIG. 6.22. Pressure response frequency characteristics of mass-controlled, direct radiator, dynamic loudspeaker mechanisms, with 10-inch diameter cones, mounted in square baffles of 8, 4, and 2 feet on a side.

of the cone will be inversely proportional to the frequency. Under these conditions, the pressure response will be proportional to the frequency in the range where the system behaves as a doublet and independent of the frequency where the system behaves as a simple radiator. The experimental results of Fig. 6.22 substantiates these predictions for 2-, 4-, and 8-foot baffles. Above the range where the system changes from doublet to singlet

FIG. 6.23. Pressure response frequency characteristics of direct radiator, dynamic loudspeaker mechanisms, with 10-inch diameter cones, operating under the following conditions: A. Square baffle 8 feet on a side and a loudspeaker resonant frequency of 50 cycles. B. Square baffle 4 feet on a side and a loudspeaker resonant frequency of 100 cycles. C. Square baffle 2 feet on a side and a loudspeaker with a resonant frequency of 200 cycles.

operation the pressure response is independent of the frequency. Below this transition point the pressure response falls off 6 db per octave.

D. *Different Resonant Frequencies and Different Baffle Sizes.*—If the resonant frequency of the loudspeaker is placed near the doublet-singlet transition frequency the pressure response will be independent of the frequency above this frequency and will be proportional to the cube of the frequency below this frequency. The experimental results of Fig. 6.23 combines the loudspeaker mechanism of Fig. 6.21 with the baffles of Fig. 6.22. The resonant frequency is placed slightly lower than the doublet-singlet transition frequency so that the output is quite uniform above the resonant frequency. Below the resonant frequency and the doublet-singlet transition frequency the pressure response falls off 18 db per octave. Again the experimental results are in agreement with theory.

6.9. Cabinet Loudspeakers.[13,14,15,16]—The most common housing for a direct radiator loudspeaker is the conventional open-back cabinet which
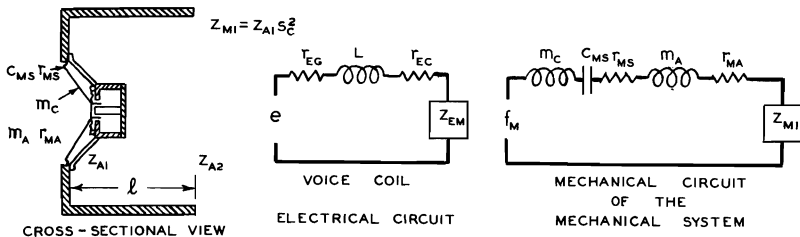
FIG. 6.24. Cross-sectional view of a single-coil, single-cone, direct radiator, dynamic loudspeaker mechanism mounted in open-back cabinet. In the voice coil circuit, e = the internal voltage of the generator. r_{EG} = the internal electrical resistance of the generator. r_{EC} and L = the electrical resistance and inductance of the voice coil. z_{EM} = the motional electrical impedance. In the mechanical circuit, m_C = the mass of the cone and voice coil. C_{MS} = the compliance of the suspension system. r_{MS} = the mechanical resistance of the suspension system. m_A = the mass of the air load. r_{MA} = the mechanical resistance of the air load. f_M = the mechanomotive force in the voice coil. z_{M1} = the mechanical impedance due to the cabinet load on the cone. z_{A1} = the acoustical impedance at the closed end of the cabinet. z_{A2} = the acoustical impedance at the open end of the cabinet. S_c = the area of the cone.

also houses the radio chassis or phonograph mechanism. These range in size from the largest console type to the smallest midget. From the standpoint of sound reproduction the principle is the same in all, namely, to provide a baffle for the loudspeaker. In the case of the midget cabinets the sound path from the front to the back is very small and the low-frequency sounds are not reproduced. In the case of the large console cabinets the

[13] Olson and Preston, *Radio and Tel. News.*, Vol. 51, No. 2, p. 69, 1954.
[14] Olson, H. F., *Audio Eng.*, Vol. 35, No. 11, p. 34, 1951.
[15] Olson, H. F., *Radio and Tel. News*, Vol. 45, No. 5, p. 53, 1951.
[16] Meeker, Slaymaker, and Merrill, *Jour. Acous. Soc. Amer.*, Vol. 22, No. 2, p. 206, 1950.

acoustic path length is sufficiently large to insure good reproduction of low frequencies.   One of the most troublesome acoustical factors in conventional cabinets is the resonance in the enclosure back of the cone.   This resonance is termed cabinet resonance.   The system may be considered from the standpoint of lumped or distributed constants.   In the case of most systems, the latter viewpoint seems to yield better agreement with experiment.   The cabinet enclosing the back of the cone may be considered to be a pipe with distributed constants.

A cross-sectional view of a direct radiator loudspeaker mounted in an open-back cabinet and the mechanical circuit of the mechanical system is shown in Fig. 6.24.   The input acoustical impedance of a finite cylindrical pipe has been considered in Sec. 5.25.   In this chapter it has been more convenient to use mechanical impedance instead of acoustical impedance. The mechanical impedance due to the cabinet in terms of the acoustical impedance is

$$z_{M1} = z_{A1} S_c^2 \qquad\qquad 6.14$$

where z_{M1} = mechanical input impedance of the cabinet, in mechanical ohms.

z_{A1} = acoustical impedance of the cabinet, in acoustical ohms, and

S_c = area of the cone in square centimeters.

The power output of the system may be determined from the mechanical and electrical circuits of Fig. 6.24 and the constants of the system.

It is the purpose of the sections which follow to consider the performance of various types of cabinets and loudspeaker systems.

A. *Low Resonant Frequency, Different Cabinet Sizes.*—The pressure response frequency characteristics of a direct radiator loudspeaker mechanism, having a resonant frequency of 20 cycles mounted in cabinets of various sizes, is shown in Fig. 6.25.   The resonant frequencies at 80, 150,
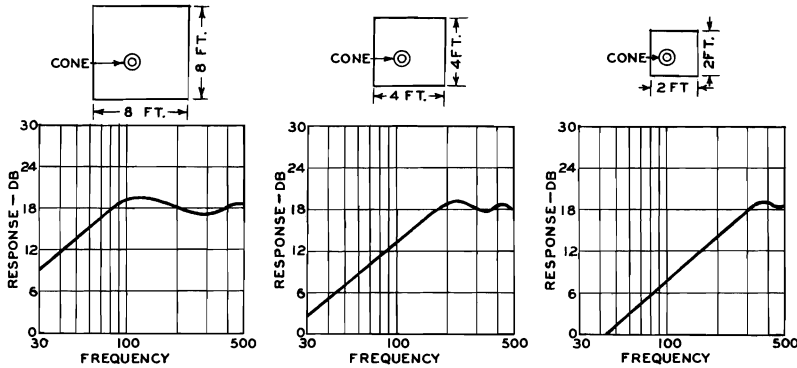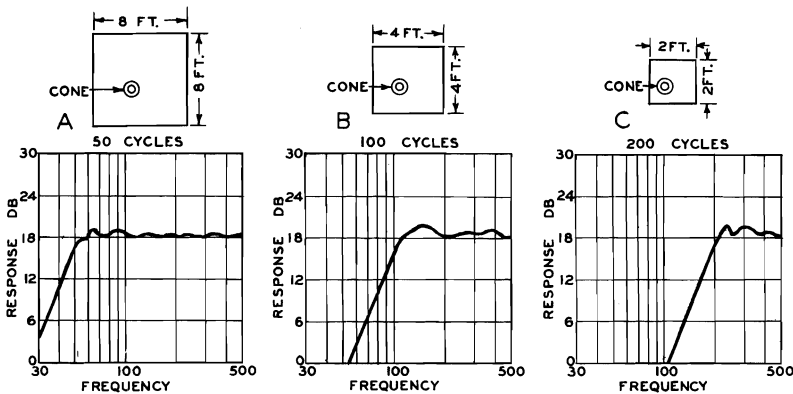

FIG. 6.25.   Pressure response frequency characteristics of mass-controlled, direct radiator, dynamic loudspeaker mechanisms with 10-inch cones mounted in square open-back cabinets.   A. Cabinet, 4 feet × 4 feet × 12 inches in depth.   B. Cabinet, 2 feet × 2 feet × 8 inches in depth.   C. Cabinet, 1 foot × 1 foot × 6 inches in depth.

and 250 cycles for the 4-, 2-, and 1-foot cabinets, respectively, is quite evident. In this region the output is somewhat exaggerated in spite of the fact that the cabinets are relatively shallow. Below the resonant frequency the system behaves as a doublet. Therefore, with a mass-controlled mechanism the response falls off 6 db per octave.

Fig. 6.26. Pressure response frequency characteristics of direct radiator, dynamic loudspeaker mechanisms, with 10-inch diameter cones, operating in open-back cabinets under the following conditions: A. Cabinet, 4 feet × 4 feet × 12 inches in depth and a loudspeaker resonant frequency of 50 cycles. B. Cabinet, 2 feet × 2 feet × 8 inches in depth and a loudspeaker resonant frequency of 100 cycles. C. Cabinet, 1 foot × 1 foot × 6 inches in depth and a loudspeaker resonant frequency of 200 cycles.

B. *Different Resonant Frequencies and Different Cabinet Sizes.*—In most of the cabinets and mechanisms in use today the resonant frequencies of the two systems are quite close together. This situation comes about in a perfectly natural way due to manufacture procedures and design limitations involved in low-cost, direct radiator mechanisms. The pressure response
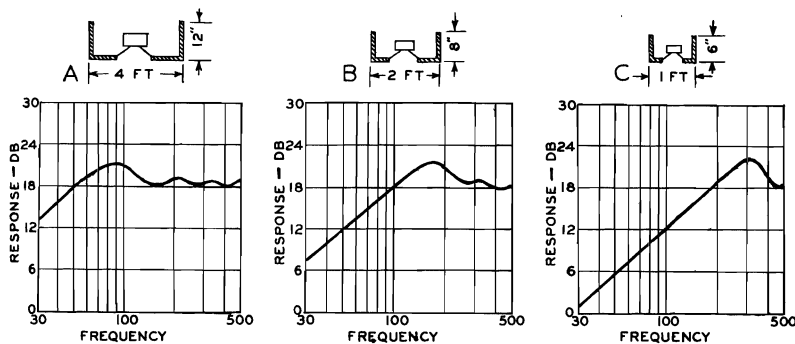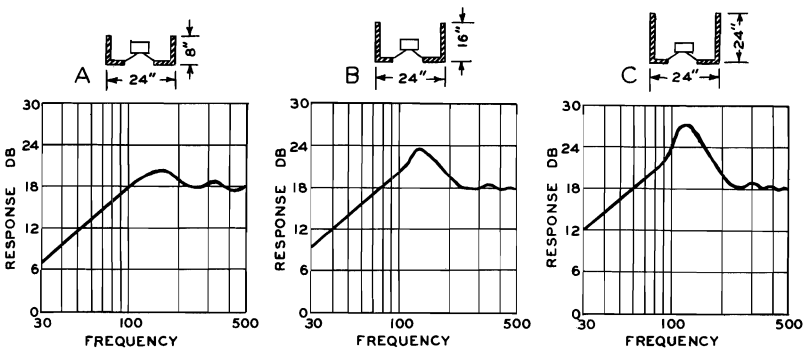

Fig. 6.27. Pressure response frequency characteristics of mass-controlled, direct radiator, dynamic loudspeaker mechanisms with 10-inch diameter cones operating in open-back cabinets 2 feet × 2 feet and the following depths: A. 8 inches. B. 16 inches. C. 24 inches.

frequency characteristics of combinations of various cabinets and mechanisms having different resonant frequencies are shown in Fig. 6.26. These characteristics show a marked increase in output of about 6 db at the region of cabinet and mechanism resonance. Below this frequency range the pressure response falls off 18 db per octave.

C. *Effect of the Depth of the Cabinet.*—A consideration of the open-back cabinet system of Fig. 6.24 shows that the depth of the cabinet will influence the response, particularly at the resonant frequency. The pressure response frequency characteristics of a mass-controlled loudspeaker mechanism mounted in 2-foot cabinets with depths of 8, 16, and 24 inches are shown in Fig. 6.27. It will be seen that the accentuated response in the region of cabinet resonance becomes more pronounced as the depth of the cabinet is increased.

6.10. Back-Enclosed Cabinet Loudspeaker.—A loudspeaker mechanism with the back of the cone completely enclosed by the cabinet is shown in Fig. 6.28. At the low frequencies the system is a simple source (see
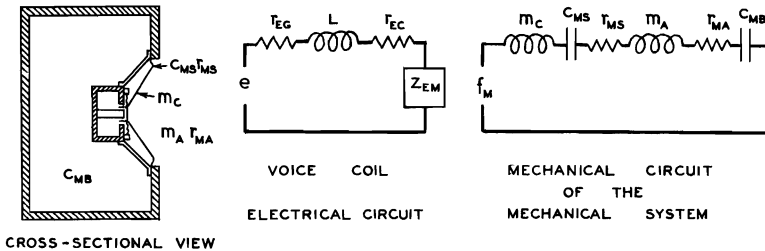
FIG. 6.28. Cross-sectional view of a single-coil, single cone, direct radiator, dynamic loudspeaker mechanism mounted in closed-back cabinet. In the voice coil circuit, e = the internal voltage of the generator. r_{EG} = the internal electrical resistance of the generator. r_{EC} and L = the electrical resistance and inductance of the voice coil. z_{EM} = the motional electrical impedance. In the mechanical circuit, m_C = the mass of the cone and voice coil. C_{MS} = the compliance of the suspension system. r_{MS} = the mechanical resistance of the suspension system. m_A = the mass of the air load. r_{MA} = the mechanical resistance of the air load. C_{MB} = the compliance of the cabinet. f_M = the mechanomotive force in the voice coil.

Sec. 2.2). Under these conditions the radiation mechanical resistance is proportional to the square of the frequency up to the frequency of ultimate mechanical resistance. The mechanical circuit of Fig. 6.28 shows that, under these conditions, the output will be independent of the frequency above the resonant frequency of the system.

A consideration of the mechanical circuit shows that the fundamental resonance is influenced by the compliance of the cone suspension, and the compliance of the enclosure. The compliance of the enclosure in terms of the acoustical capacitance is given by

$$C_{MB} = \frac{C_A}{S_c^2} \qquad\qquad 6.15$$

where C_{MB} = compliance of the cabinet, in centimeters per dyne,

C_A = acoustical capacitance of the cabinet, in (centimeters)^5 per dyne, and
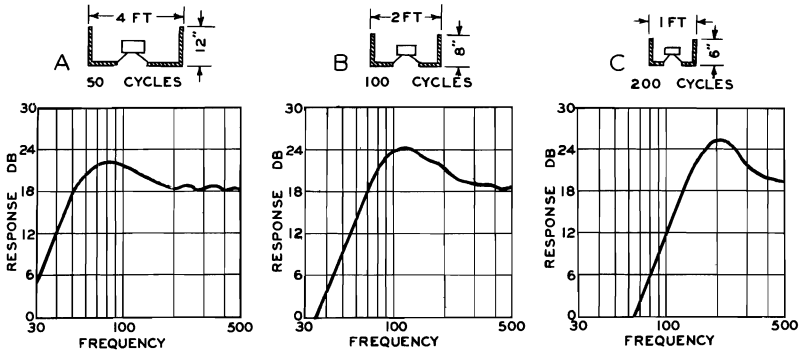
S_c = area of the cone, in square centimeters.
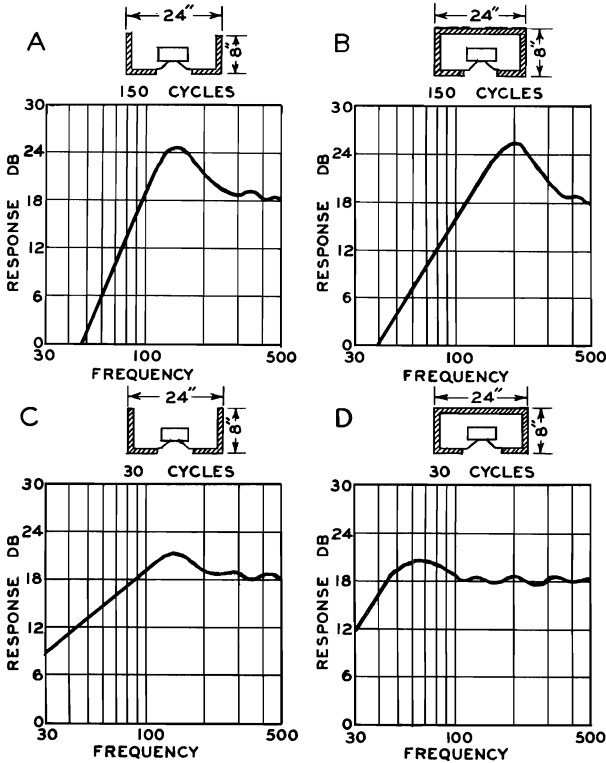
FIG. 6.29. Pressure response frequency characteristics of direct radiator, dynamic loudspeaker mechanisms with 10-inch diameter cones operating in open- and closed-back cabinets. A. Open-back cabinet, 2 feet × 2 feet × 8 inches in depth and a loudspeaker resonant frequency of 150 cycles. B. Closed-back cabinet, 2 feet × 2 feet × 8 inches in depth and a loudspeaker resonant frequency of 150 cycles. C. Open-back cabinet, 2 feet × 2 feet × 8 inches in depth and a loudspeaker resonant frequency of 30 cycles. D. Closed-back cabinet, 2 feet × 2 feet × 8 inches in depth and a loudspeaker resonant frequency of 30 cycles.

From the expression for the acoustical capacitance of an enclosure, equation 5.7 and equation 6.15, the compliance of the cabinet is given by

$$C_{MB} = \frac{V}{\rho c^2 S_c^2} \qquad 6.16$$

where V = volume, in cubic centimeters,

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

    c = velocity of sound, in centimeters per second.

The pressure response frequency characteristic of a loudspeaker mechanism having a resonant frequency of 150 cycles mounted in a 24-inch open-back cabinet is shown in Fig. 6.29A. The response falls off 18 db per octave below 150 cycles.

The pressure response frequency characteristic of a loudspeaker mechanism, having a resonant frequency of 150 cycles, mounted in a completely enclosed 24-inch cabinet is shown in Fig. 6.29B. The fundamental resonant frequency of the system is 200 cycles. The increase in the resonant frequency is due to the addition of the compliance of the enclosure. The response falls off 12 db per octave below the resonance frequency.

The pressure response frequency characteristic of a loudspeaker mechanism having a resonance frequency of 30 cycles mounted in a 24-inch open-back cabinet is shown in Fig. 6.29C. In this case the response falls off 6 db per octave below the doublet-singlet transition frequency.

The pressure response frequency characteristic of a loudspeaker mechanism having a resonant frequency of 30 cycles mounted in a completely enclosed cabinet is shown in Fig. 6.29D. The compliance of the cabinet raises the fundamental resonant frequency of the entire system to 70 cycles. The response is maintained down to 40 cycles. The response of this system is superior to that of the 8-foot baffle with a low-frequency resonant mechanism as shown in Fig. 6.16 or to the 4-foot open cabinet with a low resonance mechanism, Fig. 6.19.

6.11. Compound Direct Radiator Loudspeaker.[17]—A consideration of the back-enclosed cabinet loudspeaker has been given in Sec. 6.10. The analysis shows that cabinet volume influences the response in the low-frequency range when the compliance of the cabinet is comparable to the compliance of the suspension system. In order to obtain adequate response in the low-frequency range, some means must be provided for increasing the effective volume of the cabinet. The effective volume of a cabinet may be increased by decreasing the stiffness presented to the radiating loudspeaker mechanism by the cabinet in the low-frequency range. It is possible to reduce the effective stiffness of the cabinet by the use of an auxiliary loudspeaker mechanism which drives the radiating loudspeaker mechanism. This system has been termed a compound direct radiator loudspeaker.

A front and sectional view of a compound direct radiator loudspeaker is shown in Fig. 6.30. A schematic sectional view and the electrical and the mechanical network of the system is shown in Fig. 6.31. It will be seen that the two voice coils are connected in series. Thus, in the low-frequency range the same current flows in both voice coils. Referring to equation 6.16, it will be seen that the compliance of the cabinet is inversely proportional to the square of the cone area. Thus, it will be seen that the compliance

[17] Olson, Preston, and May, Unpublished Report.
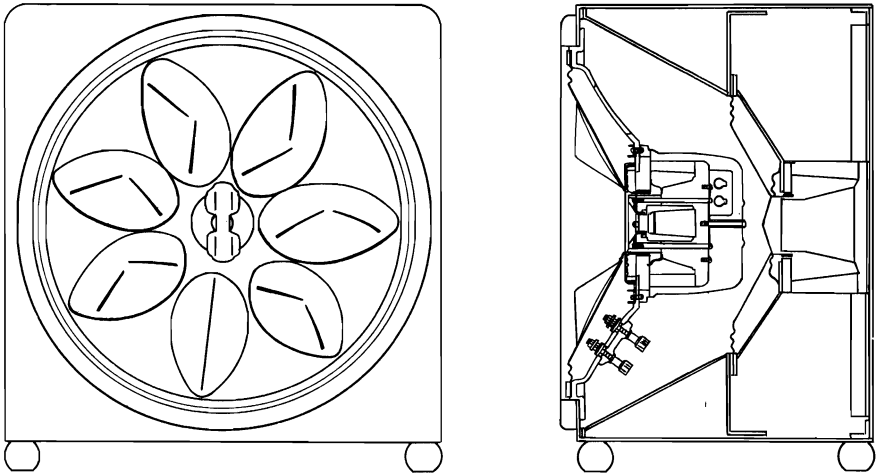
FIG. 6.30. Front and sectional views of a compound direct radiator loudspeaker. (After Olson, Preston, and May.)

CROSS-SECTIONAL VIEW
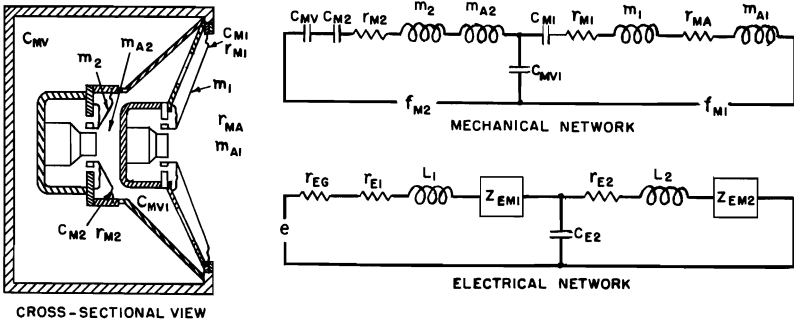
FIG. 6.31. Cross-sectional view and the electrical and acoustical networks of a compound direct radiator loudspeaker. In the electrical network, e = internal voltage of the generator. r_{EG} = internal electrical resistance of the generator. r_{E1} and L_1 = electrical resistance and inductance of the voice coil of the radiating loudspeaker. z_{EM1} = electrical motional impedance of the radiating loudspeaker. r_{E2} and L_2 = electrical resistance and inductance of the secondary loudspeaker. z_{EM2} = electrical motional impedance of secondary loudspeaker. C_{E2} = electrical capacitance. In the acoustical network, r_{MA} and m_{A1} = mechanical resistance and mass of the air load upon the cone of the radiating loudspeaker. m_1 = mass of the cone and coil of the radiating loudspeaker. r_{M1} and C_{M1} = mechanical resistance and acoustical resistance and compliance of the suspensions of the radiating loudspeaker. f_{M1} = mechanomotive force in the voice coil of the radiating loudspeaker. C_{MV} = compliance of the cabinet volume. r_{M2} and C_{M2} = mechanical resistance and compliance of the suspensions of the secondary loudspeaker. m_2 and m_{A2} = masses of the cone and air load of the secondary loudspeaker. C_{MV1} = compliance of the volume between the two loudspeakers. f_{M2} = mechanomotive force in the voice coil of the secondary loudspeaker.

of the cabinet can be increased by making the diameter of the driving loud-speaker small. By this expedient the resonant frequency of the driving loudspeaker and cabinet can be made lower than the radiating loudspeaker and the cabinet. By this two-stage system the response in the low-frequency range can be extended. A 15-inch loudspeaker is the radiating loudspeaker in the system of Fig. 6.30. The response-frequency characteristic of the duo-cone loudspeaker mechanism in a back-enclosed cabinet of two cubic feet is shown in Fig. 6.32. The response frequency characteristic of the compound direct radiator loudspeaker housed in the same
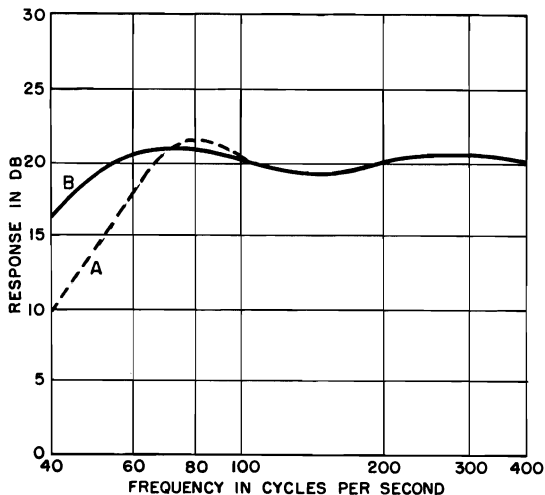
FIG. 6.32. A. Response frequency characteristic of a 15-inch duo-cone loudspeaker mechanism in a cabinet of 2 cubic feet. B. Response frequency characteristic of a compound direct radiator loudspeaker housed in the same cabinet and employing the same radiating loudspeaker mechanism.

cabinet is also shown in Fig. 6.32. It will be seen that a substantial gain in low-frequency response is obtained by the use of the compound direct radiator loudspeaker system.

6.12. Acoustical Phase Inverter Loudspeaker.—The acoustical phase inverter loudspeaker[18] system consists of a direct radiator loudspeaker mechanism mounted in a completely closed cabinet save for a port coupling the cabinet volume to the air, Fig. 6.33. The phase of the velocities on the two sides of the cone differs by 180°. Referring to the mechanical network of Fig. 6.33, it will be seen that the velocities in the branches 1 and 2 may differ by as much as 180° for positive mechanical reactances and no mechanical resistances in branches 1 and 2 and a pure compliance in

[18] Dickey, Caulton, and Perry, *Radio Engineering*, Vol. 8, No. 2, p. 104, 1936.

branch 3.   The phase angle will be reduced as mechanical resistance is introduced.   However, the mechanical resistance in direct radiator loudspeaker systems is small compared to the mechanical reactance and the constants may be chosen so that the phase angle between the velocity of the cone and the port is very small.   This system increases the radiation mechanical resistance of a direct radiator loudspeaker at the low frequencies.   The pressure response frequency characteristic of a loudspeaker mechanism mounted in an open-back cabinet is shown in Fig. 6.34A.   The pressure
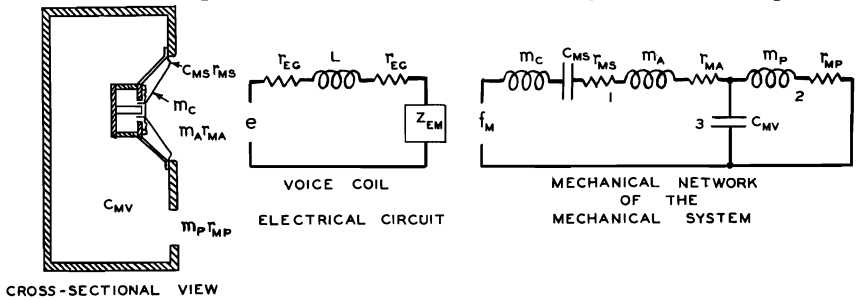
FIG. 6.33.   Cross-sectional view of a single-coil, single-cone, direct radiator, dynamic loudspeaker mechanism mounted in closed-back cabinet with a port.   In the voice coil circuit, e = the internal voltage of the generator. r_{EG} = the internal electrical resistance of the generator. r_{EC} and L = the electrical resistance and inductance of the voice coil. z_{EM} = the motional electrical impedance.   In the mechanical circuit, m_C = the mass of the cone and voice coil. C_{MS} = the compliance of the suspension system. r_{MS} = the mechanical resistance of the suspension system. m_A = the mass of the air load. r_{MA} = the mechanical resistance of the air load. C_{MV} = the compliance of the cabinet. m_P = the mass of the air in the port. r_{MP} = mechanical resistance of the air load on the port. f_M = the mechanomotive force in the voice coil.
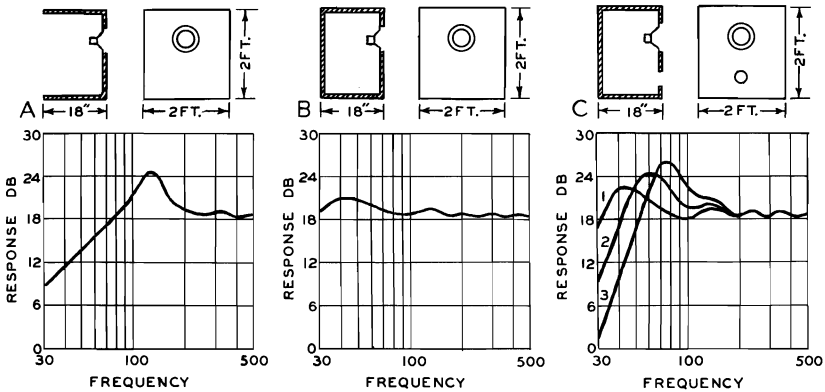
FIG. 6.34.   Pressure response frequency characteristics of a direct radiator, dynamic loudspeaker mechanism with a 10-inch diameter cone and a resonant frequency of 30 cycles operating under the following conditions: A. Open cabinet, 2 feet × 2 feet × 18 inches in depth.   B. Closed cabinet, 2 feet × 2 feet × 18 inches in depth. C. Phase inverter cabinet, 2 feet × 2 feet × 18 inches in depth and various port openings, 1. Small port.   2. Medium port.   3. Large port.

response frequency characteristic for the same mechanism mounted in a closed cabinet of the same dimensions is shown in Fig. 6.34B.   The pressure response frequency characteristic of the same mechanism and cabinet used as an acoustical phase inverter for various port openings is shown in Fig. 6.34C.   The low-frequency range is extended, the output is increased and cabinet resonance is eliminated by the phase inverter system.

6.13. Drone Cone Phase Inverter.[19]—The acoustical phase inverter loudspeaker system has been described in the preceding section.   A study of this system has shown that the particle velocity over the area of the port

CROSS-SECTIONAL VIEW
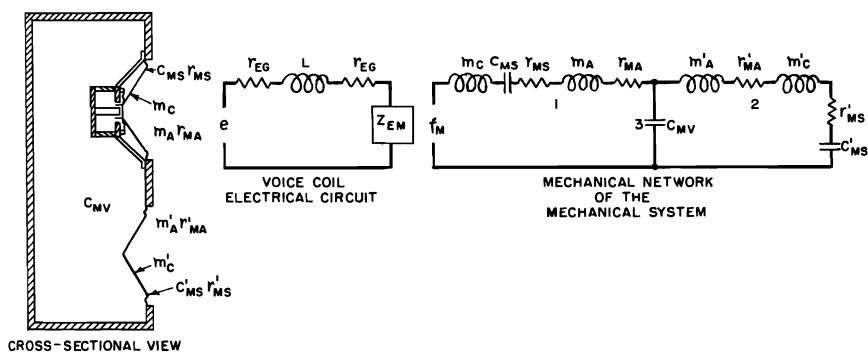
Fig. 6.35.   Cross-sectional view, electrical circuit and mechanical network of a phase inverter cabinet equipped with a drone cone.   In the electrical circuit, e = internal voltage of the electrical generator. r_{EG} = internal electrical resistance of the generator. L and r_{EC} = inductance and electrical resistance of the voice coil. z_{EM} = motional electrical impedance.   In the mechanical circuit, m_C and m_A = mass of the cone and voice coil and air load. C_{MS} and r_{MS} = compliance and mechanical resistance of the suspension system. r_{MA} = mechanical resistance of the air load. C_{MV} = compliance of the cabinet. m'_A = mass of the air load upon the drone cone. r'_{MA} = mechanical resistance load of the air load upon the drone cone. r'_{MS} = mechanical resistance of the suspension of the drone cone. m'_C = mass of drone cone. C'_{MS} = compliance of the suspension of the drone cone.   (After Olson, Preston, and May.)

is not uniform, either with respect to phase or amplitude.   The result is a loss in energy due to phase shift and friction incurred by viscosity.   Another problem in the ported cabinet is the difficulty of providing a port of large cross-sectional area so that the particle velocity in the port will be relatively low.   The appropriate inertance in the port can be obtained with a large cross-sectional area if the length of the port is increased by the required amount.   When this is done, the port becomes very long; and, as a result, the losses due to viscosity are very large.   Thus, it will be seen that simple port in the phase inverter or bass reflex cabinet is not a satisfactory system from the standpoint of maximum performance.   These objectionable features can be overcome by the use of an undriven cone, termed a drone cone, instead of the port, as shown in Fig. 6.35.   In this system the port

[19] Olson, Preston, and May, *Jour. Aud. Eng. Soc.*, Vol. 2, No. 4, p. 219, 1954.

area of the drone cone can be made the same as the active cone.   The phase
and amplitude of the particle velocity are the same over the entire area of
the drone cone.   Furthermore, the particle velocity is relatively low because
the area of the drone cone is large compared to a port.   As a result, the losses
are low in the drone cone phase inverter.

A typical response-frequency characteristic of the drone cone phase
inverter is shown in Fig. 6.36.   Two response-frequency characteristics of
the same loudspeaker in the same cabinet but with two different ports are
also shown in Fig. 6.36.   In one case the frequency range of the port is the
same as the drone cone, but the output which is obtained with the port is
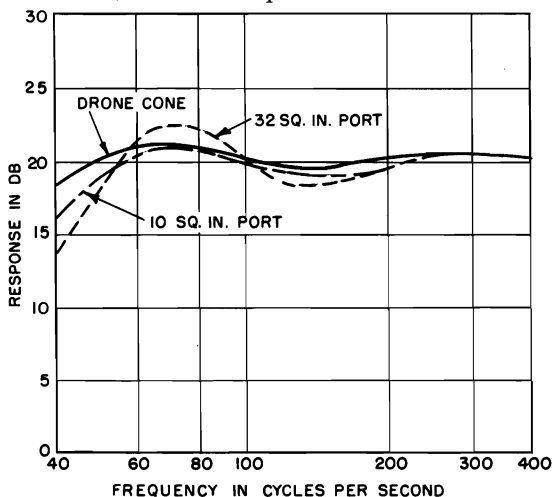
FIG. 6.36.   Response frequency characteristics of a
15-inch duocone loudspeaker housed in a cabinet of
seven cubic feet comparing a drone cone-type phase
inverter with a port-type phase inverter.

lower.   In the other case the output of the port is the same as the drone
cone, but the frequency range which is obtained is less.   To summarize,
these characteristics show that a wider frequency range with greater output
can be obtained with the drone cone type of phase inverter, as compared
to the port type, the reason being that the losses in the drone cone are less
than in the port.

6.14. Acoustical Labyrinth Loudspeaker.[20]—The acoustical labyrinth
loudspeaker consists of an absorbent walled conduit with one end tightly
coupled to the back of the cone of a direct loudspeaker mechanism and the
other end opening in front or at the bottom of the cabinet within which it
is folded (Fig. 6.37).   The labyrinth is a piston driven tube with absorbing
walls.   At the first half wavelength resonance, the velocity at the open
end is in phase with that of the front of the cone.   The radiation, then,

[20] Olney, Benj., *Jour. Acous. Soc. Amer.*, Vol. 8, No. 2, p. 104, 1936.

from both sources is additive and the response is increased.   An increase in response can be obtained over about an octave.   The rising absorption of the tube lining with increase in frequency damps out the higher resonances.   The transmission through the tube is very low above 150 cycles. An anti-resonance occurs when the tube is one-quarter wavelength long. The deleterious effect of the fundamental resonance of the cone with its suspension system upon the response may be eliminated by choosing the constants so that fundamental resonance of the loudspeaker coincides with the quarter wavelength anti-resonance of the tube.   The pressure response characteristic of a direct radiator loudspeaker with and without a labyrinth is shown in Fig. 6.37.   These characteristics show that the accentuated response due to cabinet resonance has been eliminated and that the low-frequency range has been extended.
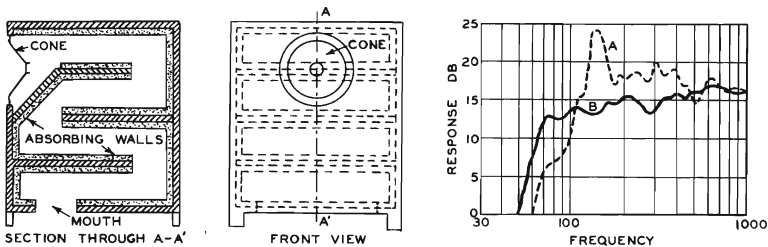
FIG. 6.37.   Acoustical labyrinth loudspeaker.   The pressure response frequency characteristic of an acoustical labyrinth loudspeaker is labeled B on the graph.   The pressure response frequency characteristic of the corresponding open-back cabinet loudspeaker is labeled A on the graph.   (After Olney.)

6.15. Combination Horn and Direct Radiator Loudspeaker.[21]—One form of the combination horn and direct radiator loudspeaker consists of a horn coupled to the back side of a direct radiator loudspeaker mechanism and an acoustical capacitance for changing the output from the horn to the open side of the cone for reproduction of the mid- and high-frequency ranges (Fig. 6.38).

At low frequencies the mechanical reactance of the compliance, C_{M1}, is large compared to the mechanical impedance, z_{M1}, at the throat of the horn.   Therefore, the cone is coupled directly to the horn in this frequency range.   In the system shown in Fig. 6.38 the mechanical reactance of the compliance, C_{M1}, becomes equal to the throat mechanical impedance, z_{M1}, at 150 cycles.   Therefore, above 150 cycles, the response from the horn is attenuated and the major portion of the output issues from the front of the cone and the system behaves as a direct radiator loudspeaker.   The use of a horn as a coupling means makes it possible to obtain large power

[21] Olson and Hackley, *Proc. Inst. Rad. Eng.*, Vol. 24, No. 12, p. 1557, 1936.

outputs from a small diameter cone. In addition, the combination of a horn and a direct radiator loudspeaker mechanism yields high efficiency and smooth response at the low frequencies. A cone with a single coil may be used for reproduction to 7000 cycles. For reproduction to 12,000 cycles a double voice coil is used. The pressure response frequency characteristic of the combination horn and direct radiator loudspeaker with double voice coil driving system is shown in Fig. 6.38.
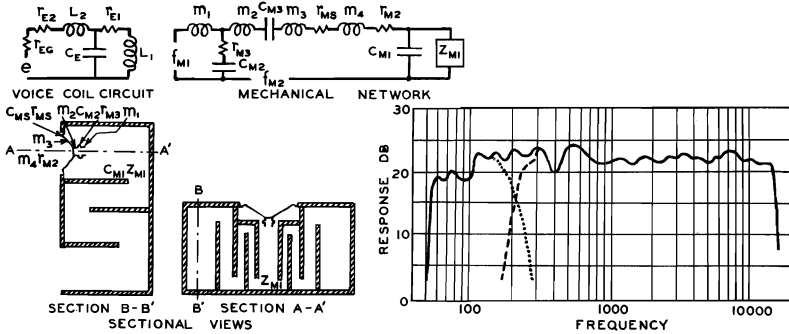
FIG. 6.38. Sectional views of the combination horn and double-voice coil, direct radiator loudspeaker. In the voice coil circuit, e = the internal voltage of the generator. r_{EG} = the internal electrical resistance of the generator. r_{E1} and L_1 = the electrical resistance and inductance of the large coil. r_{E2} and L_2 = the electrical resistance and inductance of the small coil. C_E = electrical capacitance. In the mechanical network, m_1 = the mass of the large coil. m_2 = the mass of the small coil. C_{M2} and r_{M3} = the compliance and mechanical resistance of the corrugation separating the large and small coils. m_3 = the mass of the cone. r_{MS} and C_{MS} = the mechanical resistance and compliance of the suspension system. m_4 and r_{M2} = the mass and mechanical resistance of the air load on the front of the cone. C_{M1} = the compliance of the chamber behind the cone. z_{M1} = the mechanical impedance at the throat of the horn. f_{M1} and f_{M2} = the mechanomotive forces generated in the large and small voice coil sections. The graph shows the pressure response frequency characteristic of the combination horn and direct radiator loudspeaker. The overlap between the horn and direct radiator action is shown by the dotted and dashed characteristics. (After Olson and Hackley.)

A combination horn and direct radiator loudspeaker for operation in the corner of a room is shown in Fig. 6.39. A sectional view depicting the horn of the combination horn and direct radiator loudspeaker is shown in Fig. 6.40. The theory and action is the same as the system shown in Fig. 6.38. There is some advantage in the operation of a relatively small horn loudspeaker in the corner of the room in that the radiation resistance presented to the mouth is increased. See Secs. 2.2D and 5.24. There is one unique feature of the horn system[22] shown in Figs. 6.39 and 6.40, namely, that it provides true corner operation in that the horn mouth feeds into the boundaries of the loudspeaker and the two walls and the floor as well. If

[22] Olson, H. F., Unpublished Report.

the floor is not employed as the case of most corner systems, full advantage of corner operation cannot be realized.

In a modification of the systems of Figs. 6.38, 6.39, and 6.40 direct radiator operation is not used, that is, one side of the cone is coupled to the horn and the other side is coupled to an enclosed cavity.   See Sec. 7.4B.

A phase inverter loudspeaker may also be operated in the corner of the room to obtain increased output in the very low audio-frequency range.

Another form of the combination horn and direct radiator loudspeaker[23] consists of a direct radiator loudspeaker with a large cone for the reproduction of the low-frequency range and a small horn loudspeaker for the
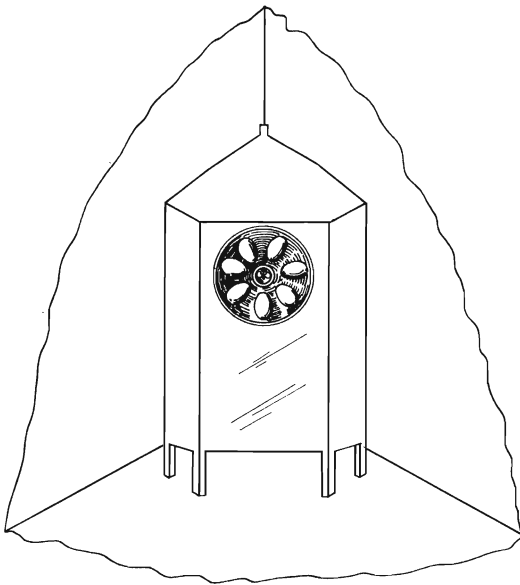
FIG. 6.39.   A perspective view of a corner-type combination horn and direct radiator loudspeaker.

reproduction of the high-frequency range.   Two different designs of this type of loudspeaker are shown in Fig. 6.41.   In Fig. 6.41A the center pole for the low-frequency loudspeaker constitutes the small portion of the horn and a flared type cone in the direct radiator loudspeaker provides a continuation of the horn.   In Fig. 6.41B the center pole also constitutes the small portion of the horn.   A small cellular horn, coupled to the small portion in the pole, completes the horn.   An electrical dividing network is used to allocate the input to the low- and high-frequency units in the appropriate frequency ranges.

[23] Lansing, J. B., *Jour. Soc. Mot. Pic. Eng.*, Vol. 46, No. 3, p. 212, 1946.
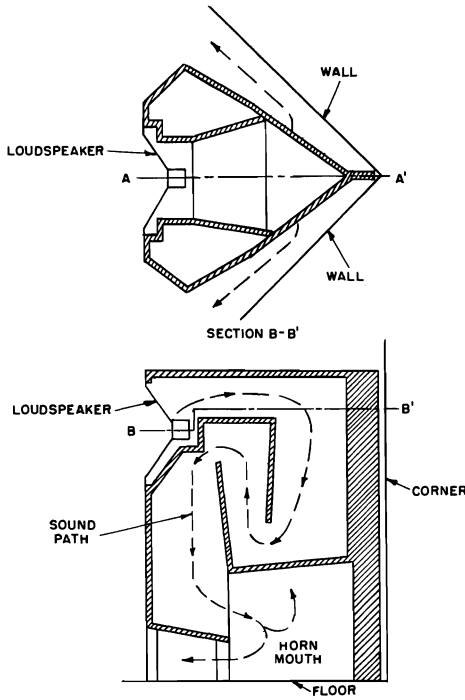
FIG. 6.40. Cross-sectional views of a corner-type combination horn and direct radiator loudspeaker.
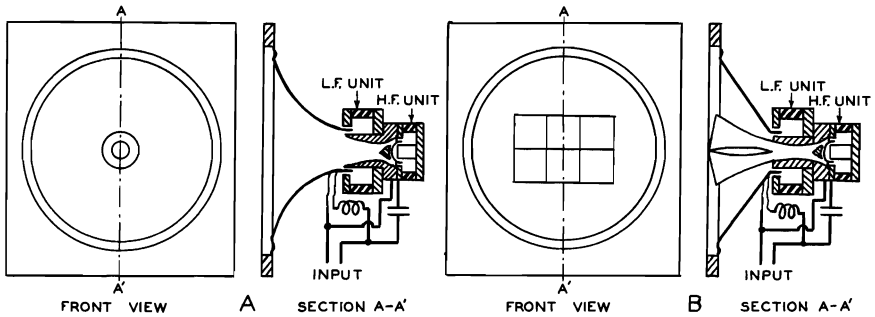
FIG. 6.41. Combination horn and direct radiator loudspeakers. A. A direct radiator loudspeaker is used for the reproduction of the low-frequency range; and a horn loudspeaker, in which the pole and low-frequency cone form the horn, is used for the reproduction of the high-frequency range. B. A direct radiator loudspeaker is used for the reproduction of the low-frequency range and a cellular horn loudspeaker is used for the reproduction of the high-frequency range.

A modification of the system of Fig. 6.41A, shown in Fig. 6.42, includes a diverging acoustic lens[24,25] at the mouth of the high-frequency horn unit. Referring to Sec. 2.19, it will be seen that the high-frequency radiation from a simple horn is emitted in a relatively narrow beam. It has been shown in Sec. 1.11 that a diverging acoustic lens will spread out the impinging wavefront. An acoustic lens placed in the mouth of the high-frequency horn will increase the spread of the high-frequency radiation and thus provide improved directivity.

6.16. Loudspeaker Mechanisms for Small Space Requirements.[26]— There are many applications in the field of sound reproduction where space is limited and the volume occupied by the loudspeaker must be kept to a minimum. This is particularly the case in pocket type personal radio receivers where the loudspeaker is the largest single component. For these applications, an inverted magnetic-field structure has developed as shown in Fig. 6.43. The field structure consists of three parts, namely, the top and bottom plates and the magnet. The cone is located between the top and bottom plates. The top and bottom plates are perforated for efficient transmission of sound. The performance of the loudspeaker is the same
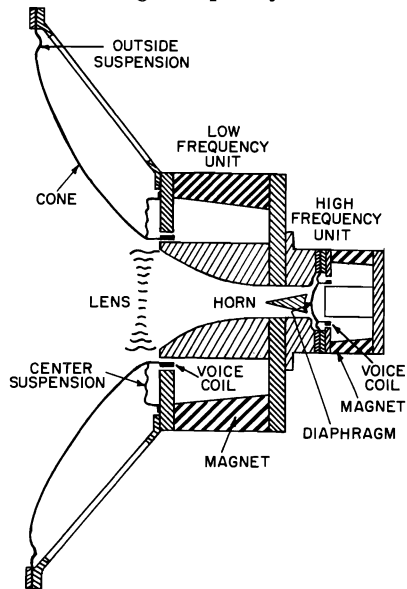
Fig. 6.42. Combination horn and direct radiator loudspeaker equipped with a lens in the mouth of the high-frequency horn.
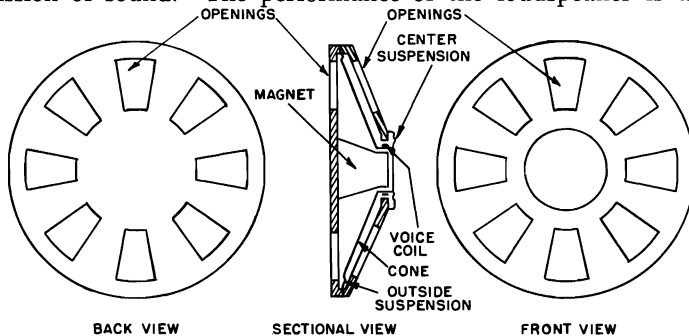
FIG. 6.43. Loudspeaker mechanism with an inverted field structure. (After Bleazey and Preston.)

[24] Koch and Harvey, *Jour. Acous. Soc. Amer.*, Vol. 21, No. 5, p. 471, 1949.
[25] Frayne and Locanthi, *Jour. Soc. Mot. Pic. Tel. Eng.*, Vol. 63, No. 3, p. 82, 1954.
[26] Bleazey and Preston, *RCA Review*, Vol. 17, No. 2, p. 211, 1956.

as that of the conventional design with a cone of the same diameter and weight and a magnet of the same weight. An examination of Fig. 6.43 reveals that the space occupied by the inverted-magnet field structure is considerably less than in the case of the conventional design of Fig. 6.1.

6.17. Feedback Applied to a Loudspeaker.—Feedback in a transmission system or a section thereof is the returning of a fraction of the output to the input. Negative feedback is feedback which results in decreasing the amplification. Among the sources of nonlinear distortion and nonuniform response in a reproducing system may be the power amplifier and loudspeaker. It is possible to reduce distortion and improve the response as a function of the frequency of an amplifier by making the amplification deliberately higher than necessary and then feeding the output back in such a way as to throw away excess gain. In the same way this system may be made to include the loudspeaker. It is not an easy proposition
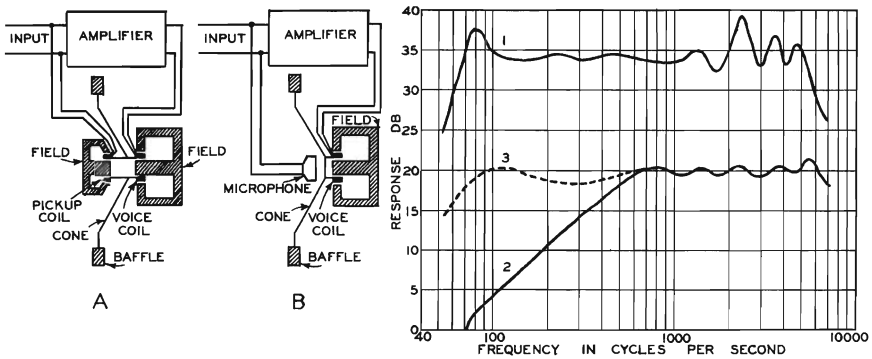
FIG. 6.44. Loudspeaker and amplifier feedback systems. A. The output of the pickup coil is fed into the input side of the amplifier. B. The output of the microphone is fed into the input side of the amplifier. The graph shows the pressure response frequency characteristic of System A: 1. Without feedback. 2. With feedback. 3. With feedback and compensation.

to employ feedback in this way because of the very special control required of phase shifts in the amplifier and loudspeaker system. Unless certain phase relations[27,28] are maintained, oscillation will occur. Fig. 6.44 shows feedback systems applied to an amplifier and loudspeaker. In Fig. 6.44A a pickup coil is attached to the cone. The output from the pickup coil is fed into the input of the amplifier out of phase with the signal input. The response characteristic of the amplifier loudspeaker without feedback is shown in Fig. 6.44. The same loudspeaker and amplifier with 15 db negative feedback from the pickup coil are also shown in Fig. 6.44. It will be seen that the response at the high frequencies is improved. This system tends to drive the cone at constant velocity for constant signal voltage

[27] Nyquist, H., *Bell Syst. Tech. Jour.*, Vol. 11, No. 1, p. 126, 1932.
[28] Black, H. C., *Bell Syst. Tech. Jour.*, Vol. 13, No. 1, p. 1, 1934.

input. Therefore, the response will fall off below the point of ultimate resistance, because the radiation resistance falls off 6 db per octave in this range (see Fig. 6.2). The response may be made uniform with respect to frequency by compensation of the input to the system.

A feedback system employing an amplifier, loudspeaker and microphone is shown in Fig. 6.44B. If a pressure operated microphone having uniform sensitivity with respect to frequency is used the response characteristic of the loudspeaker will become more uniform as the amount of feedback is increased.

6.18. Cabinet Configuration.[29],[30],[31]—The outside configuration of the cabinet influences the response due to diffraction effects introduced by the edges of the cabinet. The effects of diffraction for various shapes are given in Sec. 1.11. The response frequency characteristics of Fig. 1.7 shows that cabinet diffraction effects may introduce variations of several decibels in the response of a loudspeaker mechanism with an otherwise smooth response frequency characteristic. As a result of a study of cabinet shapes, a cabinet has been developed in which the deleterious effects of diffraction have been reduced to practical limits. The cabinet which has been evolved is shown in Fig. 6.45. It will be seen that the sharp front edges of a standard rectangular parallelopiped type cabinet, which should set up diffracted waves, have been eliminated. As a result, the variations in the response due to diffraction effects have been reduced to a negligible amount.
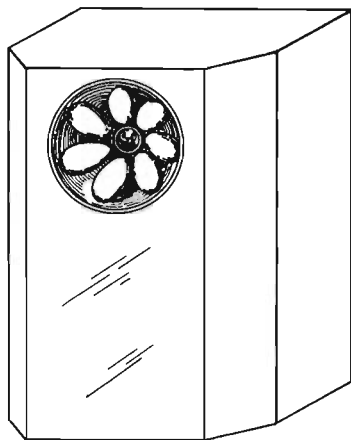
FIG. 6.45. A cabinet designed to eliminate the deleterious effects of diffraction in the response of the loudspeaker.

6.19. Loudspeaker Mounting Arrangement in the Cabinet Wall.[32],[33]—The mounting arrangement of the loudspeaker mechanism in the front wall of the cabinet influences the response due to the resonances of the cavity in front of the mechanism. In addition, variations in the response are produced by reflections and diffractions from the circular boundary of this cavity. The standard mounting arrangement for loudspeaker mechanisms which has been used for years is shown in Fig. 6.46A. Referring to Fig. 6.46A, it will be seen that the cabinet wall forms a cavity in front of the loudspeaker. The resonances and anti-resonances of this cavity, as well

[29] Olson, H. F., *Radio and Television News*, Vol. 45, No. 5, p. 53, 1951.
[30] Olson, H. F., *Audio Engineering*, Vol. 35, No. 11, p. 34, 1951.
[31] Olson, Preston, and May, *Jour. Aud. Eng. Soc.*, Vol. 2, No. 4, p. 219, 1954.
[32] Olson, H. F., *Radio and Television News*, Vol. 45, No. 5, p. 53, 1951.
[33] Olson, Preston, and May, *Jour. Aud. Eng. Soc.*, Vol. 2, No. 4, p. 219, 1954.

A   B
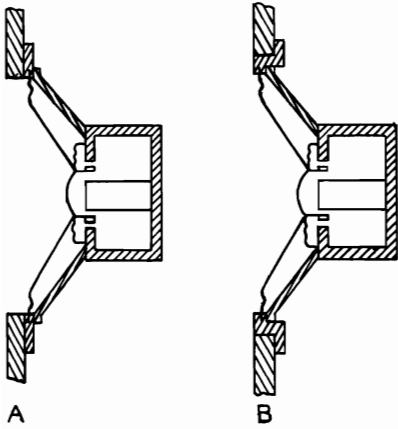
FIG. 6.46. A. Direct radiator loudspeaker mechanism mounted on the back of the cabinet wall. B. Direct radiator loudspeaker mechanism mounted flush with the front of the cabinet wall.

as reflections and diffractions of this wall edge, introduce variations in the response frequency characteristic as shown by curve A in Fig. 6.47. These variations in response can be reduced by the improved loudspeaker mechanism mounting arrangement, as shown in Fig. 6.46B. It will be seen that the cavity in front of the loudspeaker mechanism has been materially reduced. The reflecting edge of the cut-out in the cabinet wall has been completely eliminated. The sharpness of the edge has also been reduced which mitigates the diffraction effects due to this edge. The response frequency characteristic of a loudspeaker mechanism mounted as shown in Fig. 6.41B is shown by curve B in Fig. 6.47. Comparing the response frequency characteristics of curves A and B of Fig. 6.47, it will be seen that a considerable improvement in response can be obtained with the mounting arrangement shown in Fig. 6.46B.
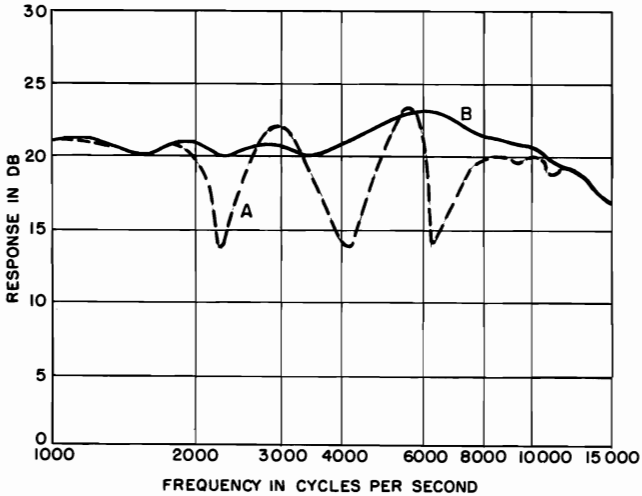
FIG. 6.47. A. Response frequency characteristic of a direct radiator loudspeaker mechanism mounted as shown in Fig. 6.46A. B. Response frequency characteristic of a direct radiator loudspeaker mechanism mounted as shown in Fig. 6.46B.

6.20. Loudspeaker Locations in Television Receivers.—The proper placement of the loudspeaker mechanism in television receivers is difficult to achieve because of the large area of the surface of the kinescope on the front of the receiver.   This is a particularly difficult problem in table model television receivers where there is no space on the front of the receiver for the loudspeaker mechanism.   As a consequence the loudspeaker mechanism must be placed in one or more of the four locations LS1, LS2, LS3, and LS4, shown in Fig. 6.48.   In all of these locations the listener-viewer is located at approximately 90° from the axis of the loudspeaker.   The
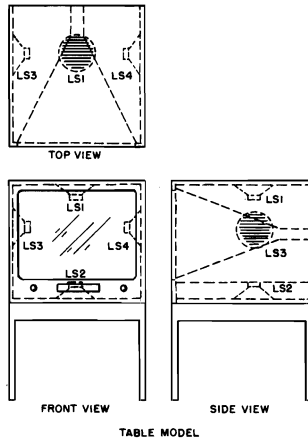
FIG. 6.48.   Loudspeaker mechanism locations in a table model television receiver.

loudspeaker mechanisms employed are usually small ones with 4- or 5-inch diameter cones.   Referring to the directional patterns of a four-inch cone of Fig. 6.11, it will be seen that it is practically nondirectional in the low-frequency range.   However, in the high-frequency range it becomes quite directional.   Under these conditions with the mounting arrangements of Fig. 6.48 there will be considerable frequency discrimination in the high-frequency range.   Some compensation in high-frequency areas is provided by accentuating the high-frequency response.

The placement of the loudspeaker mechanism in consoles is a somewhat simpler problem, as shown in Fig. 6.49, in that the loudspeaker mechanism may be located on the front of the cabinet.   The most common arrangement is a single loudspeaker LS1.   Other arrangements are as follows: Two loudspeaker mechanisms LS2 and LS3 are placed in the corner and angled to increase the coverage.   Three loudspeaker mechanisms LS1, LS2, and LS3 placed in the front and corners of the cabinet.   Three loudspeakers with the loudspeaker mechanism LS1 in the front of the cabinet and loudspeaker mechanisms LS4 and LS5 in the sides of the cabinet.   If separate

loudspeaker mechanisms are used to cover the low- and high-frequency ranges, the low-frequency loudspeaker mechanisms may be placed in one or more of the locations LS1, LS2, LS3, LS4, and LS5 and the high-frequency
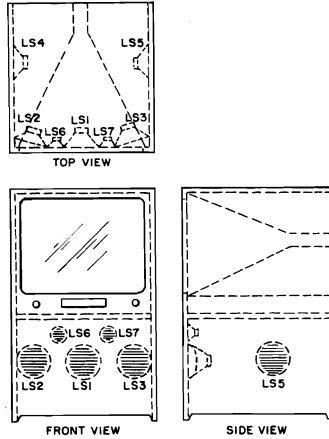
FIG. 6.49. Loudspeaker mechanism locations in a console model television receiver.
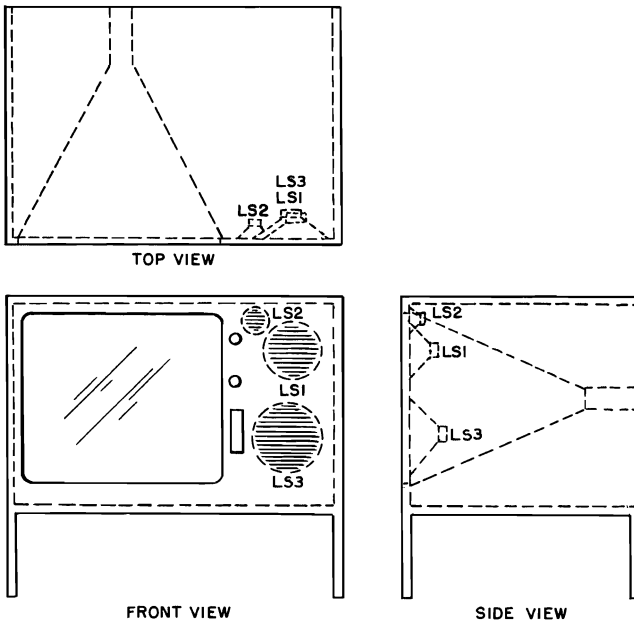
FIG. 6.50. Loudspeaker mechanism locations in a console model television receiver.

loudspeaker mechanisms may be placed in either or both locations LS6 and LS7.

Telecasts are usually viewed within an angle of $\pm 45°$ with respect to the axis of the kinescope. For that reason there appears to be no object in the use of loudspeaker locations LS2, LS3, LS4, and LS5 because a single loudspeaker mechanism will give adequate coverage over a total angle of 90°. Furthermore, multiple loudspeaker mechanisms decrease the intelligibility of speech reproduction. There is one objectionable feature in the cabinet of Fig. 6.49, namely, that the television chassis and kinescope are so large that the loudspeaker mechanism must be located very close to the floor. It has been established that the most natural sound reproduction is obtained when the loudspeaker mechanism is located at ear level of the listener.

A cabinet design in which the loudspeaker may be located at a greater distance from the floor is shown in Fig. 6.50. In the most common arrangement, a simple loudspeaker LS1 is used to cover the entire frequency range and placed near the top of the cabinet. In another arrangement an additional loudspeaker mechanism LS2 may be added. In a further modification one loudspeaker mechanism may be used to cover the high-frequency range and another loudspeaker mechanism may be used to cover the low-frequency range. Another loudspeaker location is LS3. This loudspeaker may be used in various combinations with LS1 and LS2. For example, in one arrangement, each of the three loudspeakers covers a section of the frequency range. In another combination LS2 may be omitted and LS1 and LS3 may be operated in parallel.

In another arrangement of Fig. 6.50, loudspeakers may be symmetrically located on both sides of the kinescope.

6.21. Loudspeaker Locations in Phonographs.—There are two general types of phonograph cabinets, namely, table and console models. There are many possible locations for loudspeaker mechanisms in phonograph cabinets as depicted in Figs. 6.51, 6.52, and 6.53.

The seven most common loudspeaker mechanism locations in table model phonographs are shown in Fig. 6.51. Obviously, not all of these loudspeaker locations are employed in a single instrument. For example, the most common arrangement is that of a single loudspeaker mechanism in location LS1 of Fig. 6.51. In a multiple arrangement, two loudspeaker mechanisms LS2 and LS3 of Fig. 6.51 are placed in the corners of the cabinet and angled to increase the coverage. In another multiple arrangement, two loudspeaker mechanisms LS4 and LS5 are placed in the two sides of the cabinet. In a multiple arrangement of three loudspeaker mechanisms, LS1, LS4, and LS5 are placed in the front and the two sides of the cabinet, respectively, to obtain wide angle coverage. In another multiple arrangement the three loudspeakers LS1, LS2, and LS3 are placed in the front of the cabinet. If separate loudspeaker mechanisms are used to cover the low- and high-frequency ranges, the low-frequency loudspeaker mechanisms may be placed in one or more of the locations LS1, LS2, LS3, LS4,

and LS5 and the high frequency loudspeaker mechanisms may be placed in either one or both of the locations LS6 and LS7.

The two most common console type phonograph cabinets and the loudspeaker locations in these cabinets are shown in Figs. 6.52 and 6.53. Seven loudspeaker locations are shown in Fig. 6.52. The locations LS1, LS2, LS3, LS4, and LS5 are employed for mechanisms with full frequency range. A single loudspeaker at location LS1 is the most common arrangement. Other arrangements are as follows: Two loudspeaker mechanisms LS2 and LS3 are placed in the corner and angled to increase the coverage. The two loudspeaker mechanisms LS4 and LS5 are placed in the two sides of the
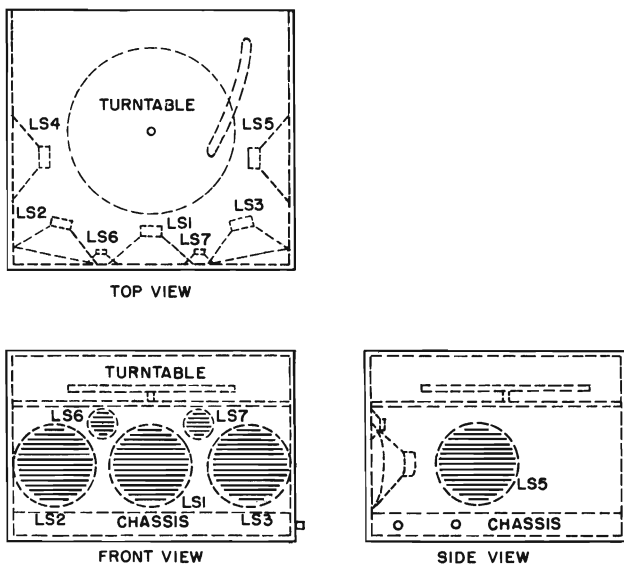
FIG. 6.51. Loudspeaker mechanism locations in a table model phonograph.

cabinet. The latter arrangement is seldom used. Three loudspeaker mechanisms LS1, LS4, and LS5 are placed in the front and the two sides of the cabinet, respectively, to obtain wide angle coverage. Three loudspeaker mechanisms LS1, LS2, and LS3 are placed in the front of the cabinet. If separate loudspeaker mechanisms are used to cover the low- and high-frequency ranges, the low-frequency loudspeakers may be placed in one or more of the locations LS1, LS2, LS3, LS4, and LS5 and the high-frequency loudspeakers placed in locations LS6 and LS7. In the simplest arrangement a single high-frequency loudspeaker LS6 and a single low-frequency loudspeaker LS1 are used. Other arrangements include various arrangements of LS1 to LS7. In all arrangements the loudspeakers should be located at as large a distance from the floor as possible. The advantage of the console phonograph shown in Fig. 6.53 is that the loudspeaker
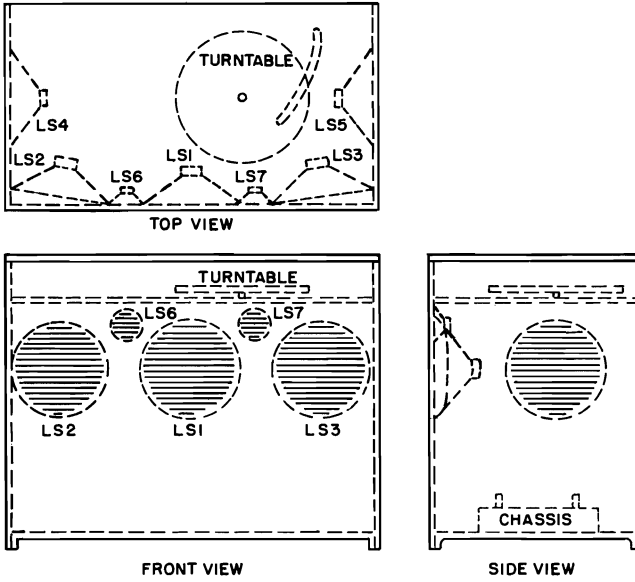
FIG. 6.52. Loudspeaker mechanism locations in a console model phonograph.
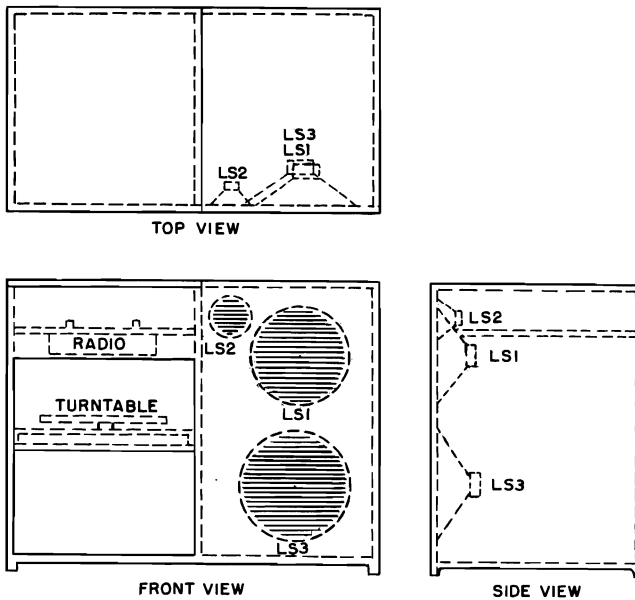
FIG. 6.53. Loudspeaker mechanism locations in a console model phonograph.

mechanism can be placed at a greater distance from the floor. It has been established that the most natural sound reproduction is obtained when the loudspeaker is located at a distance from the floor corresponding to the ear level of the listener. In the cabinet of Fig. 6.53 the most common arrangement is that of a single loudspeaker mechanism covering the entire frequency range and placed near the top of the cabinet. In another arrangement a high-frequency loudspeaker mechanism LS1 is used to cover the low-frequency range. Another loudspeaker location is LS3. This loudspeaker may be used in various combinations with LS1 and LS2. For example, in one arrangement, each of the three loudspeakers covers a section of the frequency range. In another combination LS2 may be omitted and LS1 and LS3 may be operated in parallel.

6.22. Loudspeaker Locations in Radio Receivers.—Radio receivers may be classed as follows: console, table, portable, personal, and automobile. Each of these types requires a different arrangement and location for the loudspeaker mechanism.

In general, the loudspeaker mechanism locations in the console type radio receiver are the same as the console type phonograph. See Sec. 6.21 and Figs. 6.52 and 6.53.

The six most common locations for loudspeaker mechanisms in table model radio receivers are shown in Fig. 6.54. The most common arrangement of the loudspeaker mechanism in a table model radio receiver is the use of a single loudspeaker in any of the locations LS1, LS2, LS3, LS4, LS5, or LS6. In a multiple arrangement, two loudspeaker mechanisms LS2 and LS3 are placed and angled to increase the coverage. In another multiple arrangement two loudspeaker mechanisms LS4 and LS5 are placed in the two sides of the cabinet. In a multiple arrangement of three loudspeaker mechanisms, the loudspeaker mechanisms LS1, LS4, and LS5 are placed in the front and the two sides of the cabinet, respectively, to obtain wide angle coverage. In another multiple arrangement of three loudspeakers, the loudspeaker mechanisms LS1, LS2, and LS3 are used.

Since the cabinet size of a table model receiver is relatively small, the response falls off rapidly in the low-frequency region. See Sec. 6.9. Suitable compensation may be employed in the audio amplifier to maintain the low-frequency response. Employing this expedient, the limitation is the maximum allowable excursion of the loudspeaker cone. The amplitude of the cone for a certain sound power output is inversely proportional to the area of the cone. Thus, it will be seen that the low-frequency response that can be obtained is a function of the total radiating surface.

Personal radio receivers employ very small enclosures in the form of a rectangular parallelopiped. The cubical content of the smallest receivers is of the order of 25 cubic inches. The loudspeaker mechanism is usually placed in the front panel of the radio receiver. The order of power available for feeding the loudspeaker is of the order of 5 to 50 milliwatts. Since the power is limited to a very low value and the loudspeaker mechanism and cabinet are both small, it is impossible to obtain adequate low-frequency

response for natural sound reproduction.   In the personal radio the principal use is for the reproduction of speech.   Therefore, the acoustical problem is to obtain good intelligibility on speech together with adequate sound level output.   This can be accomplished in the personal radio receiver by proper and suitable design of the loudspeaker and case.   See Sec. 6.9.

The acoustic problem in the larger portable radio receiver is similar to the personal radio receiver.   Of course, the larger the case and loudspeaker the greater the low-frequency response.

Automobile radio receivers are housed in an enclosure of about 1000 cubic inches or less.   In general, a loudspeaker mechanism is also placed in the same housing.   In some cases the radio chassis and loudspeaker are
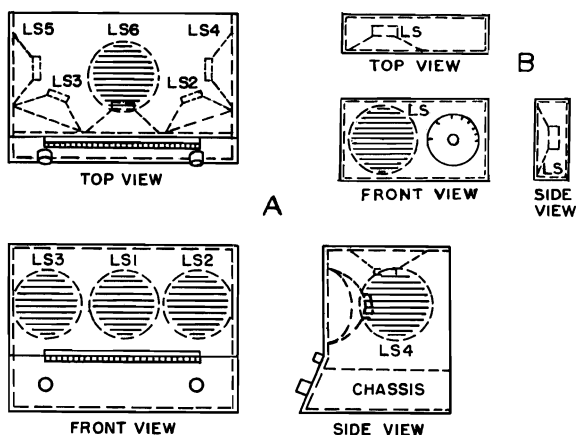
FIG. 6.54.   A. Loudspeaker mechanism locations in a table model radio receiver.   B. Loudspeaker mechanism location in a personal radio receiver.

mounted in separate enclosures.   An additional loudspeaker is sometimes placed in the deck behind the rear seat.   See Sec. 11.25.   Since the loudspeaker in the rear deck operates in the trunk compartment, there is no enclosure problem.   However, the space allocated in the loudspeaker mechanism is limited when it is mounted with the chassis or separately for reproduction in the forward portion of the motor car.   Employing a low resonant eight-inch loudspeaker or the equivalent in an elliptical loudspeaker, very good reproduction of low frequencies can be obtained in an enclosure of 1000 cubic inches.

6.23. Loudspeaker Locations in Combination Instruments.—The location of the loudspeaker mechanism in combination radiophonograph cabinets is similar to that of the phonograph of Sec. 6.21.   In both the table model and console phonographs it is customary to add a radio tuner so that both radio and phonograph reproduction can be obtained.

6.24. Concentrated Source Loudspeaker.[34]—There are applications where it is necessary to produce a high sound signal level over a limited zone without producing a high sound signal level surrounding the zone. These conditions may be obtained by means of the system shown in Fig. 6.55. In the particular example of Fig. 6.55, 67 small direct radiator loudspeaker mechanisms are mounted on a section of a spherical surface. The output of the loudspeakers are all in phase at the center of the spherical surface or the focus of the system. By means of this system a gain of more than 20
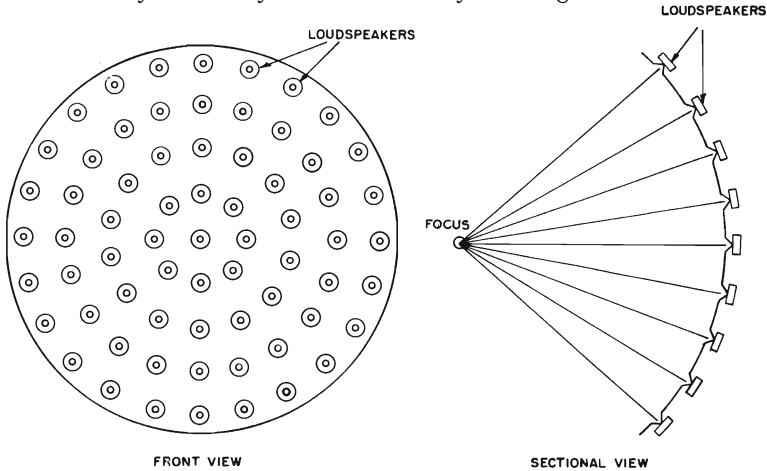
FIG. 6.55. Front and sectional views of a concentrated source loudspeaker.

db in signal level at focus compared to other locations removed from the focus may be obtained.

6.25. Transient Response.[35]—The subject of transient response embraces a wide variety of physical phenomena. Electrical transients concern electrical circuits and the components of electrical systems. Acoustical transients concern acoustical and mechanical systems. In view of the fact that the sound reproducing and collecting systems are mechanical, the general tendency is to assume that these systems exhibit very poor transient response characteristics. In properly designed acoustical elements the performance is very often far superior to the other components used in sound reproducing systems.

The behavior of a loudspeaker may be analyzed by solving the differential equations of the dynamical system. In other words, find the velocities of the elements of the system which, when substituted in the differential equations, will satisfy the initial and final conditions. The solution of a differential equation may be divided into the steady state term and the

[34] Olson, H. F., Unpublished Report.
[35] Olson, " Dynamical Analogies," D. Van Nostrand Company, Princeton, N.J., 1943.

transient term. The operational calculus is of great value in obtaining the transient response of a mechanical or acoustical system to a suddenly impressed force or pressure.

The general analysis used by Heaviside is applicable to any type of vibrating system whether electrical, mechanical, or acoustical. It is the purpose of this section to show the response of the conventional direct radiator loudspeaker to a suddenly applied unit force.

The mechanical circuit of the dynamic loudspeaker at the low frequencies is shown in Fig. 6.1. The differential equation for the system of 6.1 is

$$m\ddot{x} + r_{MT}\dot{x} + \frac{x}{C_M} = f_{MO} \qquad\qquad 6.17$$

where x = displacement,

f_{MO} = mechanical driving force, in dynes,

m = total mass, in grams,

C_M = compliance of the suspension system, in centimeters per dyne, and

r_{MT} = total mechanical resistance, in mechanical ohms.

The total mechanical resistance is

$$r_{MT} = r_{MS} + r_{MR} + r_{ME} \qquad\qquad 6.18$$

where r_{MS} = mechanical resistance due to losses in the suspension system, etc., in mechanical ohms,

r_{MR} = mechanical radiation resistance, in mechanical ohms, and

r_{ME} = mechanical resistance due to the electrical system, in mechanical ohms.

The mechanical resistance, r_{MS}, is the sum of all the losses in the suspension, the viscosity of the grill and cloth coverings and the viscosity loss due to the air forced through the slit formed by the air gap and voice coil.

From equation 6.10 the mechanical impedance, z_{ME}, due to the electrical circuit is

$$z_{ME} = r_{ME} = \frac{(Bl)^2}{r_{ET}} \qquad\qquad 6.19$$

where B = flux density, in gausses,

l = length of the voice coil conductor, in centimeters,

$r_{ET} = r_{EC} + r_{EG}$,

r_{EC} = damped electrical resistance of the voice coil, in abohms, and

r_{EG} = internal electrical resistance of the generator (the vacuum tube), in abohms.

The inductance of the voice coil is negligible. The mechanical radiation resistance, r_{MR}, is given by the equation 5.10. It may be obtained directly from the graph of Fig. 5.2.

The mass, m, is the sum of the cone mass and the mass of the air load upon the cone. The mechanical mass reactance of the air load upon a cone may be obtained from equation 5.10. It may be obtained directly from the graph of Fig. 5.2.

Heaviside's unextended problem[36],[37],[38] is as follows: Given a linear network of n meshes in a state of equilibrium, find its response when a unit force is applied to any mesh. The unit function is defined to be a force which is zero for $t < 0$ and unity for $t \geq 0$.

The indicial mechanical admittance of the mechanical circuit of Fig. 6.1 is

$$A(t) = \frac{1}{r_{MT} + mp + \dfrac{1}{C_M p}} \qquad 6.20$$

where p is employed as a symbol for the differentiation with respect to the independent variable, time.

Let

$$\alpha = \frac{r_{MT}}{2m}$$

$$\omega = \sqrt{\frac{1}{mC_M} - \alpha^2}$$

The indicial mechanical admittance may be written

$$A(t) = \frac{1}{m\omega} \frac{p\omega}{(p+\alpha)^2 + \omega^2} \qquad 6.21$$

From tables of operational formulas, the solution is

$$A(t) = \frac{1}{m\omega} \epsilon^{-\alpha t} \sin \omega t \qquad 6.22$$

Fig. 6.56 shows the effect of the electrical impedance of the vacuum tube upon the transient response of a loudspeaker. In this case the loudspeaker is connected to the following generators: a very high electrical impedance corresponding to pentode or Class "B" operation; a generator of one-half the electrical resistance of the loudspeaker corresponding to class "A" operation; and to a generator of very low electrical impedance corresponding to inverse feedback operation. The electrical impedance characteristic of the loudspeaker is shown by the uppermost left-hand graph of Fig. 6.57. This example shows that the damping exerted by the electrical system is of consequence. However, there is very little difference between Class "A" and feedback operation. When a loudspeaker operates from a high

[36] Carson, "Electric Circuit Theory and Operational Calculus," McGraw-Hill Book Company, New York, N.Y., 1926.

[37] Bush, "Operational Circuit Analysis," John Wiley and Sons, New York, N.Y., 1937.

[38] Berg, "Heaviside's Operational Calculus," McGraw-Hill Book Company, New York, N.Y., 1936.

electrical impedance vacuum tube amplifier, the internal mechanical resistance of the loudspeaker is the major factor influencing the transient response.
Fig. 6.57 shows response of a 12-inch (10-inch diameter cone) loudspeaker
to a unit force for various values of mechanical resistance.   In order to
correlate the response with actual systems, the electrical impedance frequency
characteristic for each system is also shown.   These characteristics are for
a loudspeaker coupled to a generator with very high internal electrical
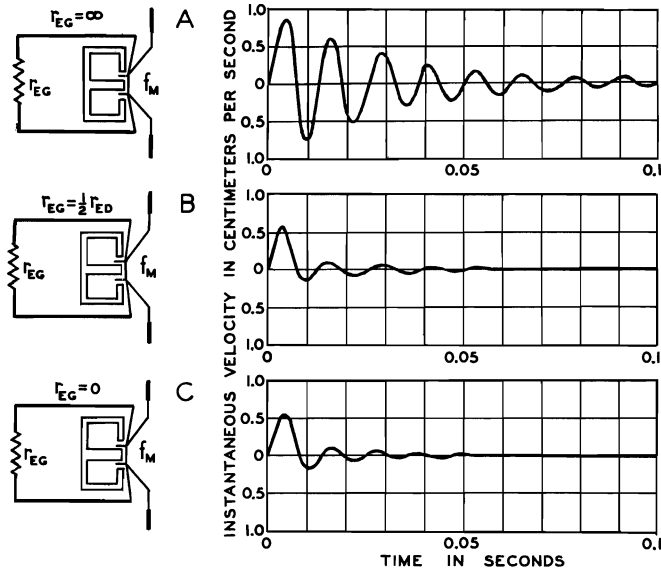
FIG. 6.56.   The transient response of a direct radiator, dynamic
loudspeaker, with a 12-inch diameter cone, to a unit force for various
types of electrical generators.   A. Generator of very high electrical
resistance.   B. Generator having an electrical resistance of one half
of the loudspeaker electrical impedance.   C. Generator of zero electrical impedance.

impedance.   For this type of operation it is customary to provide a large
mechanical resistance, r_{MS}, the second and third conditions of Fig. 6.57.

Figs. 6.56 and 6.57 show that the "hangover" in properly designed and
operated loudspeakers is very small.   Of course, the systems are improved
as the fundamental resonant frequency is lowered.   In some of the small
receivers employing relatively high electrical impedance power amplifiers
driving loudspeakers having the fundamental resonance above 100 cycles,
the response to transients is usually very poor because the internal mechanical
resistance is not sufficiently large.   Of course, the steady state response
with respect to frequency is usually not very smooth and the nonlinear
distortion is quite large in these receivers.   As a consequence, the poor
transient response is masked by these distortions.

Another means for depicting the transient response of a loudspeaker is the application of a tone burst signal.[39,40,41]   A tone burst signal consists of a sine wave with a rectangular envelope containing ten or more cycles.   The deviation in the sound output from the rapid growth and decay characteristics and steady state characteristic of an applied tone burst signal depicts the transient response of a loudspeaker.   Referring to the response frequency characteristic of a loudspeaker with an undamped suspension
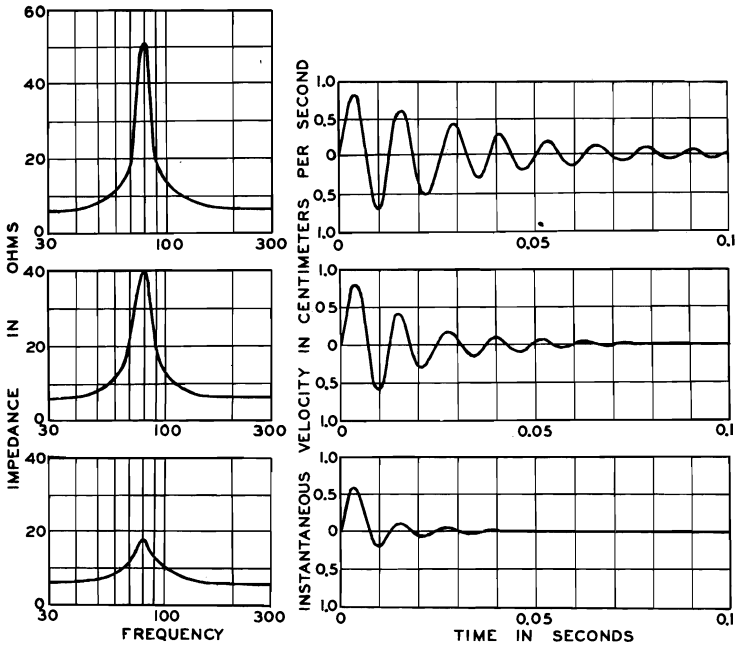
FIG. 6.57.   The transient response of a direct radiator, dynamic loudspeaker, with a 12-inch diameter cone, for different values of the suspension mechanical resistance.   The electrical impedance frequency characteristic indicates the degree of internal damping.

system shown in Fig. 6.74A, it will be seen that there is a peak in the response at 800 cycles and a dip at 1100 cycles.   The sound outputs from this loudspeaker with applied tone burst signals of 800 cycles and 1100 cycles are shown in Fig. 6.58.   In the case of the peak in the response frequency characteristic there is a slow growth and a slow decay in the response of the loudspeaker to an applied tone burst signal.   In the case of a dip in the response frequency characteristic there is a rapid growth followed by a decrease in output then followed again by an increase in output in the response to an applied tone burst signal.   It has been suggested that a

[39] Olson, H. F., *Audio Engineering*, Vol. 34, No. 10, p. 5, 1940.
[40] Olson, Preston, and May, *Jour. Aud. Eng. Soc.*, Vol. 2, No. 4, p. 219, 1954.
[41] Corrington, M. S., *Jour. Aud. Eng. Soc.*, Vol. 3, No. 1, p. 35, 1955.

measure of the transient distortion be designated as the remaining response after the applied tone burst signal has been cut off, as shown in Fig. 6.58. Apparatus for measuring the transient response by means of a tone burst signal is described in Sec. 10.3G.
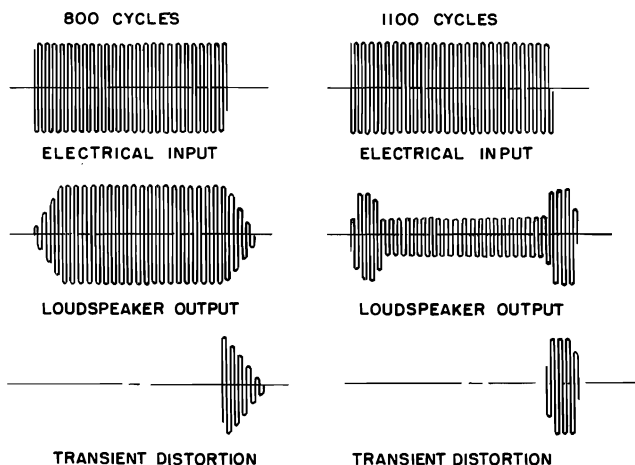
FIG. 6.58. The transient response of a loudspeaker having the response frequency characteristic of Fig. 6.74A at 800 cycles and 1100 cycles. The waves show the tone burst input to the loudspeaker, the sound output from the loudspeaker, and the output from the loudspeaker after the input to the loudspeaker has been stopped.

6.26. Distortion.—The general trend in all types of radio receivers and phonographs is more output without a corresponding increase in the size of the loudspeaker. As a result, the maximum amplitude of the loudspeaker is also increased. Many apparently peculiar activities are manifested by the loudspeaker when the amplitude or excursion of the cone is large. Under these conditions, the operation exceeds the linear portion of the stress-strain characteristic of the cone material.

Most of the unusual phenomena are due to the nonlinear characteristics of the cone suspension system. One of the effects of a nonlinear cone suspension system is a jump phenomena in the response characteristic. Another effect is the production of harmonics and subharmonics due to the nonlinear cone suspension system. Frequency modulation of a high-frequency signal by a large low-frequency amplitude of the cone is another form of distortion. The nonlinear characteristics of the air also introduce distortion. It is the purpose of this section to consider the various types of distortion produced in a direct radiator type loudspeaker system.

A. *Distortion Due to Nonlinear Cone System.*[42,43]—Nonlinear distortion is

[42] Olson, H. F., *Audio Engineering*, Vol. 34, No. 10, p. 5, 1950.
[43] Olson, Preston, and May, *Jour. Aud. Eng. Soc.*, Vol. 2, No. 4, p. 219, 1954.

generated in the cone when the operation exceeds the linear portion of the stress-strain characteristic of the cone.   The strength of the cone depends

FIG. 6.59.   The sound output of a typical direct radiator loudspeaker, for a certain fixed value of nonlinear distortion, as a function of the mass of the cone.

upon the thickness of the paper. The weight in turn is proportional to the thickness of the cone.   The sound output of a typical direct radiator loudspeaker for a certain value of nonlinear distortion as a function of the weight of the cone is shown in Fig. 6.59.   The relative output of a direct radiator as a function of the weight of the cone is shown in Fig. 6.4.   A review of Figs. 6.4 and 6.59 shows that high sensitivity by the use of a light cone is not compatible with low nonlinear distortion.   In order to obtain low nonlinear distortion, a relatively heavy cone must be used.

B. *Nonlinear Suspension System.*[49]—The force displacement characteristic of a typical, direct radiator loudspeaker cone suspension system
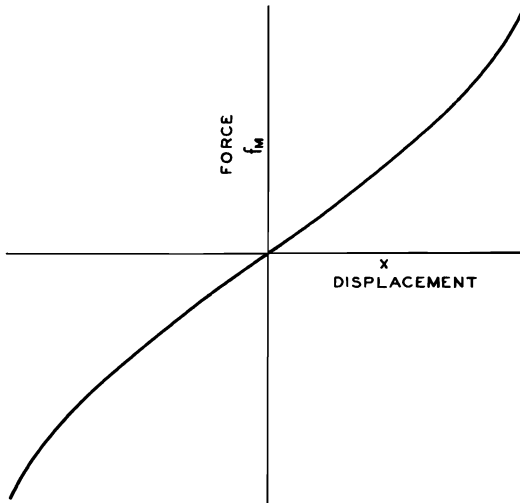
FIG. 6.60.   Force displacement characteristic of the suspension system of a direct radiator loudspeaker.

is shown in Fig. 6.60.   It will be seen that for small amplitudes the suspension system is linear.   However, for large amplitudes the suspension system is nonlinear.

[44] Olson, H. F., *Jour. Acous. Soc. Amer.*, Vol. 16, No. 1, p. 1, 1944.

The force deflection characteristic of the loudspeaker cone suspension system of Fig. 6.60 may be approximately represented by the expression

$$f_M = f(x) = \alpha x + \beta x^3, \qquad 6.23$$

where α = constant > 0, β = constant > 0, and f_M = applied force which produces the displacement x.

The compliance of the suspension system of Fig. 6.60 may be obtained from equation 6.23 as follows:

$$C_M = \frac{x}{f_M} = \frac{1}{\alpha + \beta x^2} \qquad 6.24$$

The differential equation of the vibrating system in Fig. 6.1 is

$$m\ddot{x} + r_M\dot{x} + \frac{x}{C_M} = F \cos \omega t, \qquad 6.25$$

where x = displacement,

   $\dot{x}$ = velocity,

   $\ddot{x}$ = acceleration,

   m = mass of the cone, coil, and air load,

   r_M = mechanical resistance due to dissipation in the air load and suspension system,

   C_M = compliance of the suspension system,

   $F = Bli$,

   B = magnetic flux density in the air gap,

   l = length of the voice coil conductor,

   i = amplitude of the current in the voice coil,

   $\omega = 2\pi f$,

   f = frequency, and

   t = time.

Substituting the expression for C_M of equation 6.24 in equation 6.25, the differential equation becomes

$$m\ddot{x} + r_M\dot{x} + \alpha x + \beta x^3 = F \cos \omega t \qquad 6.26$$

Since the mechanical resistance, r_M, is quite small compared to the mechanical reactance, save over a very narrow frequency range near the resonant frequency, equation 6.26 can be written as follows:

$$m\ddot{x} + \alpha\dot{x} + \beta x^3 = F \cos \omega t. \qquad 6.27$$

A number of investigators have obtained an approximate solution of this differential equation.

If β is considered to be small, the relation

$$\omega^2 = \frac{\alpha}{m} + \frac{\frac{3}{4}\beta A^2}{m} - \frac{F}{Am} \qquad 6.28$$

between the arbitrary amplitude A and ω may be obtained,

An approximate solution of the differential equation, for unit mass, is

$$x = A \cos \omega t + \frac{1}{32} \frac{\beta A^3}{\alpha + \frac{3}{4}\beta A^2 - (F/A)} \cos 3\omega t. \qquad 6.29$$

The sections which follow will show that these equations predict the performance of a loudspeaker with a nonlinear cone suspension system.

C. *Distortion Characteristics of Nonlinear Suspension System.*—The well-known experimental result of a nonlinear cone suspension system is the production of odd order harmonics when a sinusoidal input is applied to the loudspeaker. The wave shape under these conditions is shown in Fig. 6.61.
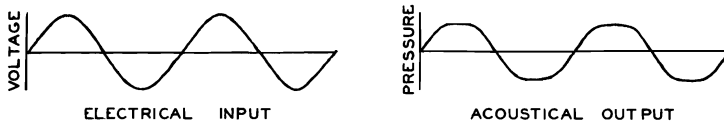
FIG. 6.61. The wave shapes of the electrical input and the sound pressure output of a loudspeaker with a nonlinear suspension system.

The third harmonic is the preponderant distortion component. Equation 6.29 shows that a third harmonic term is introduced due to the suspension system. In the case of a direct radiator loudspeaker, the amplitude is inversely proportional to the square of the frequency for constant sound power output in the frequency region below the frequency of ultimate
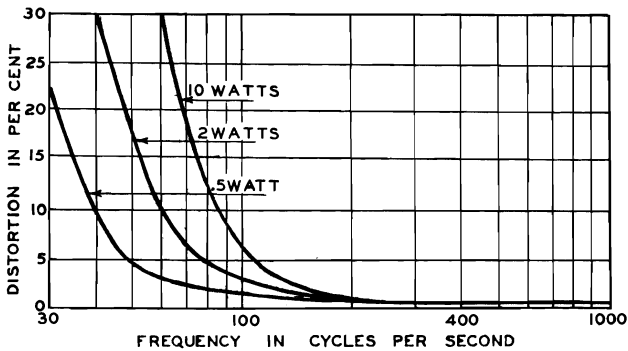
FIG. 6.62. Distortion frequency characteristics of a direct radiator, dynamic loudspeaker mechanism with a 10-inch diameter cone and a nonlinear suspension system for electrical inputs of 2, 5, and 10 watts.

resistance. Consequently, the greatest distortion will occur at the low-frequency end of the frequency range as shown by typical, experimental, nonlinear distortion frequency characteristics of Fig. 6.62. The manifestation and effect of this type of distortion upon the reproduction of sound are well known. Distortion occurs in all amplifiers as well as loudspeakers.

As a matter of fact, it is more troublesome in amplifiers because the distortion occurs over the entire audio frequency range, whereas the distortion is confined to the low-frequency range in loudspeakers.

In the above considerations, the distortion produced by the nonlinear element comprises harmonics of the fundamental. Distortion components with frequencies of $\frac{1}{2}$, $\frac{1}{3}$, $\frac{1}{4}$ ... $1/n$ of the frequency of the applied force also occur in nonlinear systems. Those familiar with the performance of loudspeakers have noticed the production of subharmonics. In general, these are very pronounced in the mid-frequency range. In the mid-frequency range the subharmonics are due to the nonlinear properties of the cone. Particular solutions of equation 6.26 have been obtained which show that subharmonics are possible in a loudspeaker due to a nonlinear cone suspension system. As pointed out above, the amplitude of the cone of a direct radiator loudspeaker is inversely proportional to the square of the frequency for constant sound output. The large amplitudes are confined to the low-frequency range. Therefore, these subharmonics will be of a very low frequency and difficult to detect. Careful experimental investigations have shown the existence of subharmonics due to a nonlinear cone suspension system as predicted from theoretical considerations.
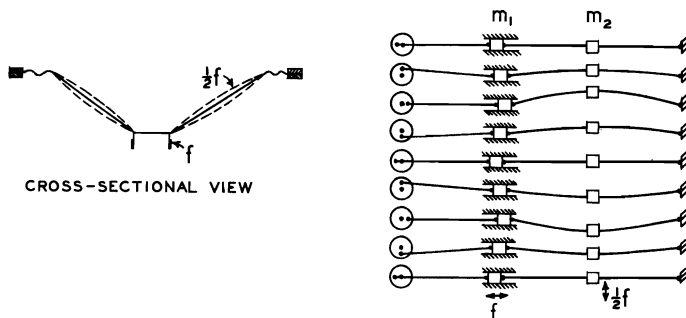
FIG. 6.63. A system consisting of a mass m_1 driven by a crank at a frequency f and a second mass m_2 supported by a spring coupled to m_1 vibrating with a frequency $\frac{1}{2}f$. The cross-sectional view of the cone shows a similar system and illustrates how subharmonics may be produced by a loudspeaker.

Fig. 6.63 illustrates the mechanism of one type of subharmonic. The driven mass m_1 at the end of the bar vibrates at a frequency f while the mass m_2 vibrates at a frequency $\frac{1}{2}f$. In the same way a cone, Fig. 6.63, will vibrate at a subharmonic frequency. The existence of subharmonics in direct radiator loudspeakers is well known. However, in horn loudspeakers the diaphragms are relatively small and quite rigid. Consequently the conditions for the production of subharmonics are not particularly favourable. Circular corrugations in the diaphragm or cone may be used to increase the stiffness and thereby reduce the tendency to break into subharmonic vibrations.

D. *Response Frequency Characteristics of a Direct Radiator Loudspeaker With a Nonlinear Suspension System.*—The velocity frequency characteristic of a loudspeaker with a nonlinear suspension system may be obtained from the equation 6.29. A theoretical response frequency characteristic is shown in Fig. 6.64.

Suppose that a constant current is applied to the voice coil of the loudspeaker and at a low-frequency point A of Fig. 6.64. Then as the frequency is increased, the velocity increases steadily to the point C. At this point the velocity drops suddenly, in a jump, to point E. From point E on, the velocity steadily decreases. Now start at F and decrease the frequency. The velocity steadily increases to the point D. At point D the velocity suddenly jumps to the point B. From point B on, the velocity steadily decreases.

Typical experimental velocity frequency characteristics are shown in Fig. 6.65. The velocity frequency characteristic for an increase in frequency is shown in Fig. 6.65. The velocity frequency characteristic for a decrease in frequency, is also shown in Fig. 6.65. These characteristics are quite similar to the theoretical characteristic of Fig. 6.64.

E. *Distortion Due to Inhomogeneity of the Air-Gap Flux.*—Inhomogeneity of the flux density through which the voice coil moves is another source of distortion. The result is that the driving force does not correspond to the voltage developed by the generator in the electrical driving system. Furthermore, the motional electrical impedance is a function of the amplitude. This type of distortion is similar to that due to a nonlinear suspension system. The wave-shape distortion is similar to that of Fig. 6.61.

FIG. 6.64. Theoretical response frequency characteristic of a direct radiator loudspeaker mechanism with a nonlinear suspension system. The unstable portion of the response frequency characteristic is indicated by a dashed line.

The force, in dynes, developed by the interaction of the current in the voice coil and the magnetic field is

$$f = Bli \qquad 6.30$$

where B = flux density, in gausses,

   l = length of the voice coil conductor, in centimeters, and

   i = current, in abamperes.

Equation 6.30 shows that the force is directly proportional to the current if Bl is a constant. If the Bl product varies with the position of the voice
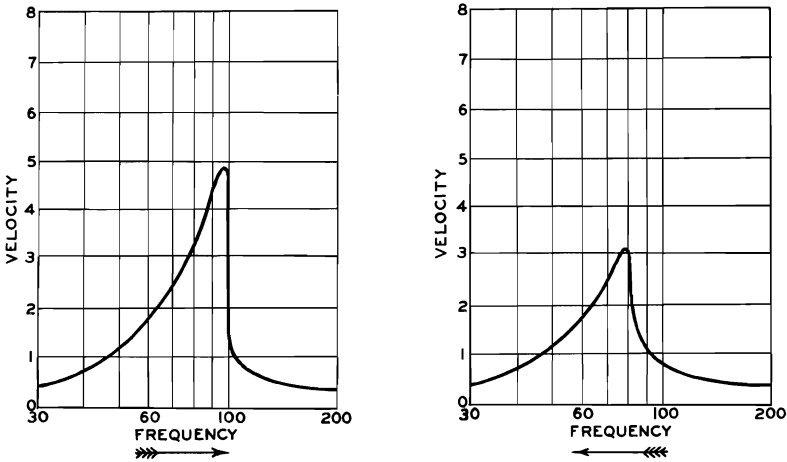
FIG. 6.65. Experimental response frequency characteristic of a direct radiator loudspeaker mechanism with a nonlinear suspension system for an increasing and a decreasing applied frequency.

coil, the force will not be proportional to the current and distortion will be produced.

A typical flux distribution in an air gap is shown in Fig. 6.66. A consideration of the flux distribution shows that the Bl product will be practically a constant if the voice coil is made longer than the gap, as shown in Fig. 6.66B, because, as the coil moves into the weaker tufting field on one end, it moves into a stronger field on the other end. From the standpoint of efficiency at the higher frequencies, this method is not particularly desirable
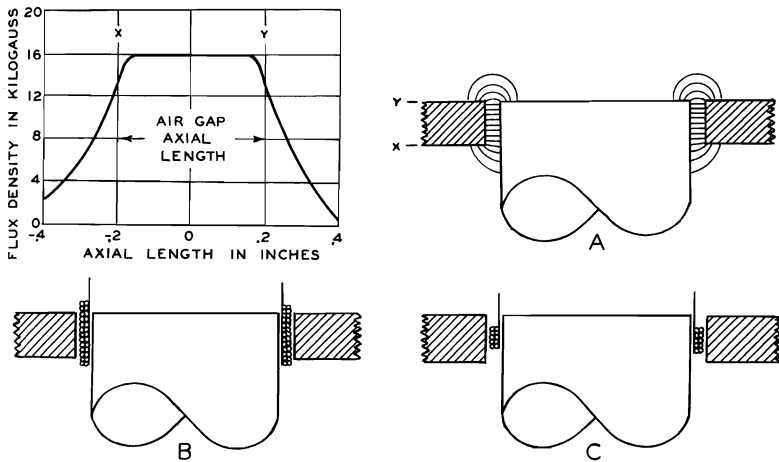
FIG. 6.66. Graph of the flux distribution in an air gap. A. Typical field map of the flux lines in an air gap. B. A voice coil longer than the air gap. C. A voice coil shorter than the air gap.

because part of the voice coil is in a weak field. This type of distortion can also be eliminated by making the air gap of sufficient axial length so that the voice coil remains at all times in a uniform field as shown in Fig. 6.66C. The latter method is used for high-frequency loudspeakers of high efficiency.

F. *Frequency-Modulation Distortion.*[45]—The amplitude of the cone in a direct radiator loudspeaker for constant sound output, in the frequency range below the ultimate radiation resistance, is inversely proportional to the square of the frequency. If the cone is radiating both at high- and low-frequency, the source of high-frequency energy may be considered to be moving back and forth at the low frequency. The high-frequency energy will be modulated. The resulting frequency-modulated wave may be represented by a carrier and a double infinity of sidebands.

The square root of the ratio of the power in the sidebands to the total power in the sound wave, in per cent, is

$$D = 2900 \frac{f_2 \sqrt{p_1}}{f_1^2 d^2} \qquad\qquad 6.31$$

where f_2 = modulated frequency, in cycles,

   f_1 = modulating frequency, in cycles,

   p_1 = acoustical output at f_1, in watts, and

   d = cone diameter, in inches.
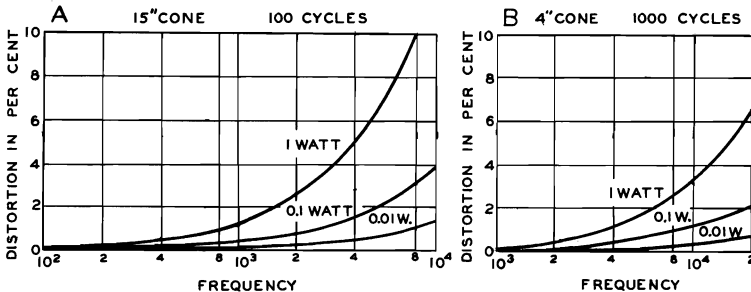
FIG. 6.67. A. Frequency modulation distortion characteristics of a 15-inch diameter cone with outputs of 1, 0.1, and 0.01 acoustical watts at 100 cycles for a second applied frequency over the range 100 to 10,000 cycles. B. Frequency modulation distortion characteristics of a 4-inch diameter cone with outputs of 1, 0.1, and 0.01 acoustical watts at 1000 cycles for a second applied frequency over the range of 1000 to 10,000 cycles.

Frequency modulation distortion characteristics for a cone 15 inches in diameter and a cone 4 inches in diameter for acoustical outputs of 1, 0.1, and 0.01 watts are shown in Fig. 6.67.

G. *Air Nonlinear Distortion.*—In general, the distortion generated in the air between the cone of a direct radiator loudspeaker and the listener is considered to be negligible. It is very much less than this type of distortion in a horn type loudspeaker. However, if small distortions are of

[45] Beers and Belar, *Proc. IRE*, Vol. 31, No. 4, p. 132, 1943.

consequence, then some consideration must be given to the distortion generated in the air between the cone and the air of the listener.

The ratio of the second harmonic pressure to the fundamental pressure, at a distance r centimeters from a spherical radiator of radius r_1 centimeters, is

$$\frac{p_{2r}}{p_{1r}} = \frac{(\gamma + 1)\omega p_{1r} r}{2\sqrt{2}\gamma p_0 c} \log_e \frac{r}{r_1} \qquad 6.32$$

where γ = ratio of specific beats (1.4 for air),

p_{1r} = fundamental sound pressure at a distance r, in dynes per square centimeter.

p_{2r} = second harmonic sound pressure at a distance r, in dynes per square centimeter.

p_0 = atmospheric pressure, in dynes per square centimeter,

$\omega = 2\pi f$,

f = frequency, in cycles per second,

c = velocity of sound, in centimeters per second.

Equation 6.32 applies to any diverging wave system in which the sound pressure varies inversely as the distance. It may be mentioned in passing that the pressure in a sound wave in free space varies inversely as the distance.

The second harmonic pressure,[46] in dynes per square centimeter, generated in a distance x, in centimeters, in a plane wave is

$$p_2 = \frac{(\gamma + 1)\omega}{2\sqrt{2}\gamma p_0 c} p_1^2 x \qquad 6.33$$

where p_1 = fundamental pressure, in dynes per square centimeter, and the other quantities are the same as equation 6.32.

Equation 6.33 applies to a plane wave, as, for example, a sound wave in a pipe.

In the case of a direct radiator loudspeaker the wave is diverging. At a distance equal to the radius of the cone the system can, from the standpoint of distortion, be replaced by a spherical radiator equal to the radius of the cone. The distortion generated in the volume between the cone and the spherical surface may be determined by approximations by employing equation 6.33. The complete expression for the second harmonic distortion, in per cent, generated between a cone of radius r_1, in centimeters, and an observation point at a distance r, in centimeters, from the front of the baffle is

$$D = \frac{p_{2r}}{p_{1r}} 100 = \frac{(\gamma + 1)\omega p_1}{2\sqrt{2}\gamma p_0 c} \left[.85r + r \log_e \frac{r}{r_1} \right] 100 \qquad 6.34$$

where p_{1r} = pressure at the observation point, in dynes per square centimeter.

[46] Thuras, Jenkins, and O'Neil, *Jour. Acous. Soc., Amer.* Vol. 6, No. 3, p. 173, 1935.

The distortion frequency characteristics, for a distance of 3 meters (about 10 feet) and various pressures at the observation point, for direct radiator loudspeakers with cones having diameters of 2 inches and 8 inches, are shown in Fig. 6.68.
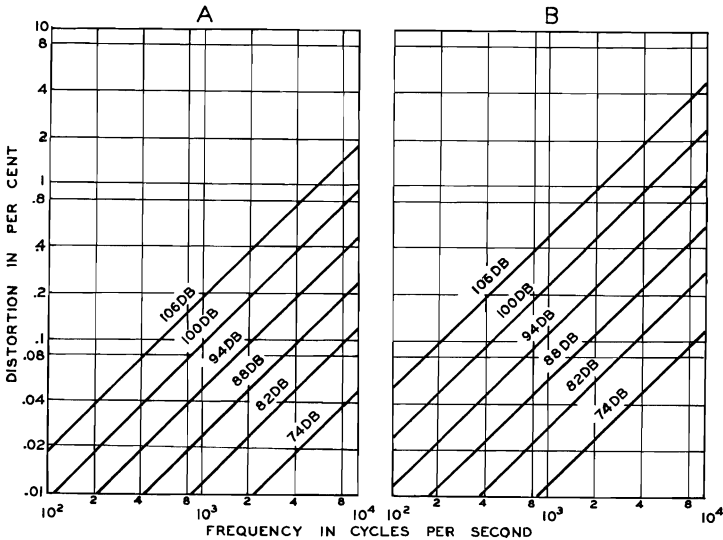
FIG. 6.68.   Distortion frequency characteristics depicting the distortion generated in the air between the cone and the observer in a direct radiator loudspeaker.   A. Loudspeaker with a cone 8 inches in diameter.   B. Loudspeaker with a cone 2 inches in diameter.   Labels on characteristics indicate sound levels at observation point.   0 db = .0002 dyne per square centimeter.   Distance 3 meters.

6.27. Diaphragms, Suspensions, and Voice Coils.—The diaphragm or cone of practically all direct radiator loudspeakers is made of paper.   Typical cones shown in Fig. 6.69, are made by a felting process employing a master screen having the shape of the diaphragm.   The mixture of pulp and water is drawn through the screen leaving a thin deposit of compressed pulp. When this deposit is dried it can be removed from the screen and the result is the finished diaphragm.   The outside suspension system can also be felted as part of the cone.

There are two types of felted diaphragms in general use—namely, the circular and the elliptical cone shown in Fig. 6.69.   In certain cabinets it is possible to obtain a larger diaphragm area by employing the elliptical cone.   The directional pattern of the elliptical cone is sharper in the plane containing the major axis of the ellipse and axis of the cone and is broader in the plane containing the minor axis of the ellipse and axis of the cone than the circular cone with the same area.

There are three types of cross sections in general use in felted cones— namely: the conical shape, Fig. 6.70A; the flared shape, Fig. 6.70B; the

corrugated conical shape, Fig. 6.70C. The shapes of Fig. 6.70 may be employed in either circular or elliptical cones.

The flared shape is somewhat more rigid than the conical shape. For this reason, the directional pattern in the high-frequency range is very much sharper. The use of corrugations increases the radial rigidity and slows

FRONT VIEW   FRONT VIEW

SECTIONAL VIEW   SECTION A - A'   SECTION B - B'
       A   B
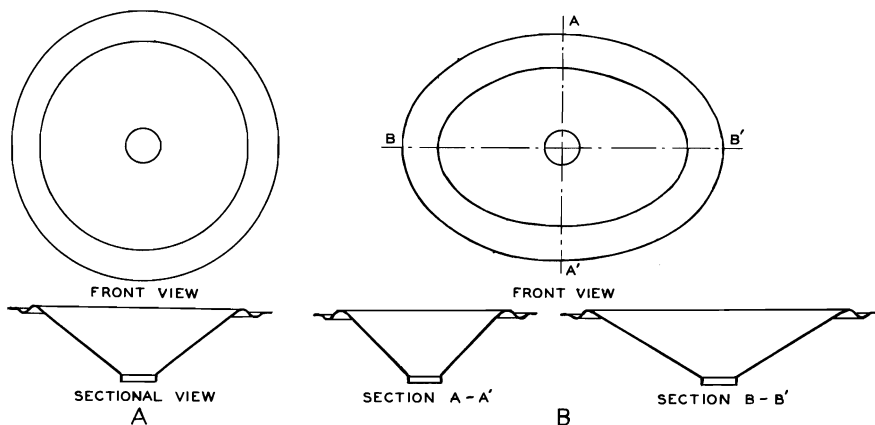
FIG. 6.69.   Felted cones for direct radiator, dynamic loudspeakers.   A. Circular cone. B. Elliptical cone.

CORRUGATIONS

       A   B   C
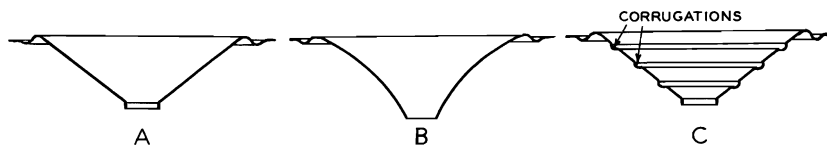
FIG. 6.70.   Sectional views of felted cones for direct radiator, dynamic loud-speakers.   A. Conical shape.   B. Flared Shape.   C. Conical shape with corrugations.

propagation of the wave in the cone and thereby broadens the directional pattern.

The three types of suspension systems, shown in Fig. 6.71, are in general use. The leather or kidskin suspension system shown in Fig. 6.71A has gradually gone out of use. It has been displaced by the one-piece felted cone and suspension system shown in Fig. 6.71B. The latter system is much simpler and less costly in manufacture. One of the principal disadvantages of the felted suspension system is the nonlinear characteristics which introduce distortion (see Sec. 6.16). The stiffness of the suspension system may be decreased and the distortion reduced by means of a folded or double suspension system as shown in Fig. 7.61C. The reduction in
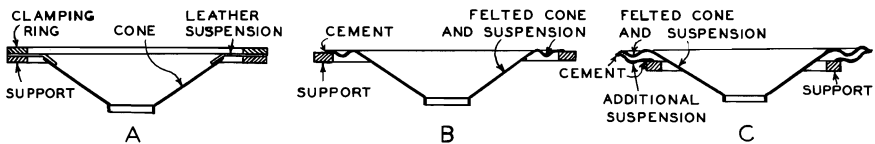
FIG. 6.71. Sectional views of cone suspension systems for direct radiator loudspeakers. A. Leather suspension. B. Felted corrugated suspension. C. Folded or double-felted corrugated suspension system.
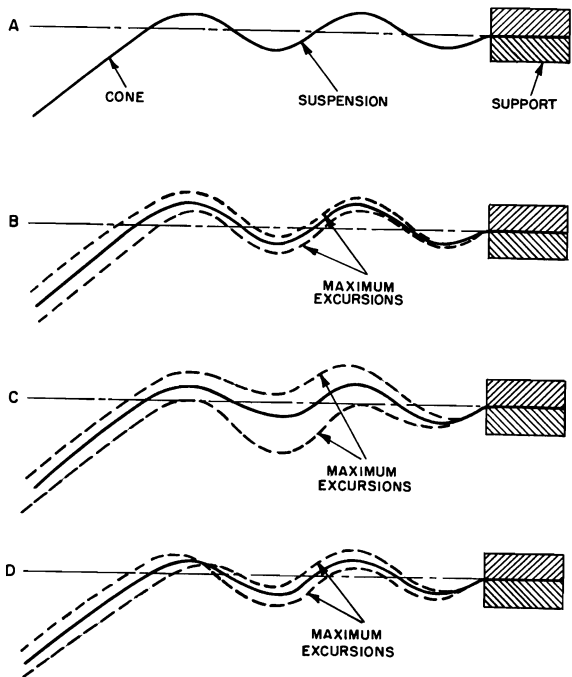
FIG. 6.72. A. Outer suspension system for a cone of a direct radiator loudspeaker. B. Normal vibration of the suspension. C. Resonant vibration of the suspension in phase with the cone. D. Resonant vibration of the suspension out of phase with the cone.

stiffness makes it possible to obtain a low fundamental resonant frequency in small light cones and thereby extend the low-frequency range (see Sec. 6.2).

An enlarged view of the corrugated outside suspension system of the type of Fig. 6.71B is shown in Fig. 6.72A.

The normal mode of vibration of the cone and suspension is shown in Fig. 6.72B. The maximum excursions of each part of the suspension and

cone are shown by the dotted lines. It will be seen that the amplitude falls off gradually in the suspension from the edge of the cone to the fixed outside edge. Unfortunately, a limp suspension employed to obtain a low resonant frequency does not behave in this manner throughout the frequency range, but breaks into resonance in the mid-frequency range. The amplitude of the suspension may be greater than that of the cone, as shown in Fig. 6.72C. The vibration of the suspension is in phase with the cone, as shown in Fig. 6.72C, and out-of-phase, as shown in Fig. 6.72D.

In the past, the procedure has been to coat the suspension with some highly viscous material, thereby providing damping which reduces the amplitude at resonance. In this way, the response is smooth and free from the peak and dip. The objection to the use of viscous materials is that these materials tend to dry out, with the result that the damping efficiency is reduced. The vibration[47,48] can be controlled and the effects of any resonance reduced to a negligible amount by means of a special rubber damping ring, as shown in Fig. 6.73. The curve of Fig. 6.74A taken without
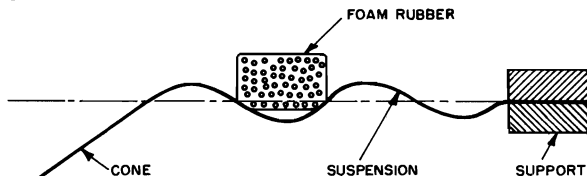
FIG. 6.73. The output suspension system of the cone of a direct radiator loudspeaker equipped with a foam rubber damper. (After Olson, Preston, and May.)

the damping ring can be compared with that of Fig. 6.74B taken with the damping ring. It will be seen that the response frequency characteristic without the damping ring exhibits a peak and a dip at 800 and 1100 cycles, respectively.

In the direct radiator loudspeaker the sound vibrations start at the voice coil, flow out in the cone, and then into the suspension system. In the low-frequency range the phase shift, in degrees, along the cone is relatively small and the cone behaves essentially as a piston. However, in the high-frequency range the phase difference between the voice and suspension may be several radians. In this frequency range it is important that the wave that travels into the suspension system is absorbed and not reflected back. The latter condition would lead to standing waves which would produce a ragged response frequency characteristic. The sponge-rubber damping ring serves as a suitable acoustical termination, thereby absorbing the vibrations which flow into the suspension system. The response frequency characteristics without and with the damping ring are shown in Figs. 6.75A and 6.75B, respectively. It will be seen that response is smoother with the damping ring.

[47] Olson, H. F., *Radio and Television News*, Vol. 51, No. 2, p. 69, 1954.
[48] Olson, Preston, and May, *Jour. Aud. Eng. Soc.*, Vol. 2, No. 4, p. 219, 1954.
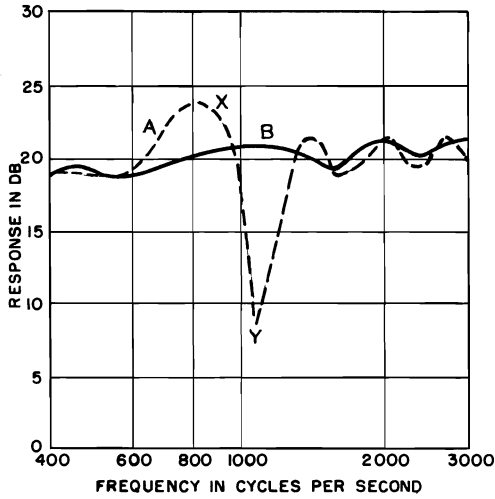
FIG. 6.74. Mid-frequency response frequency characteristics of direct radiator loudspeakers. A. A conventional suspension system. B. A suspension equipped with a foam rubber damper.
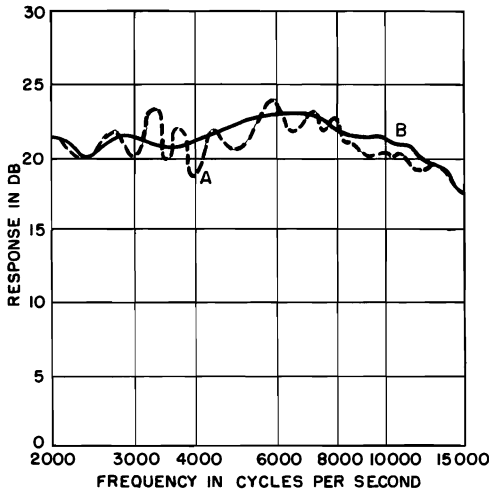
FIG. 6.75. High-frequency response frequency characteristics of direct radiator loudspeakers. A. A conventional suspension system. B. A suspension equipped with a foam rubber damper.

Centering suspensions for keeping the voice coil aligned in the air gap are shown in Fig. 6.76. An inside slotted centering suspension, usually made of fiber, is shown in Fig. 6.76A. An inside felted paper centering

suspension with corrugations is shown in Fig. 6.76B.   This type of suspen-
sion is usually employed where the amplitude is small.   An outside felted
paper suspension with corrugations is shown in Fig. 6.76C.   The outside
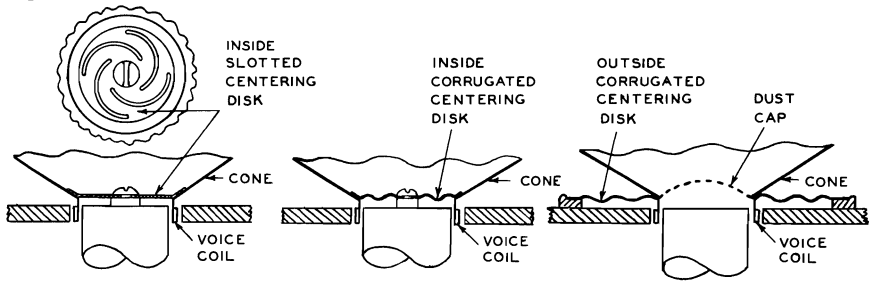
FIG. 6.76.   Sectional views of voice coil centering systems for direct radiator dynamic
loudspeakers. A. Inside slotted fiber disk.   B. Inside felted corrugated centering disk.
C. Outside felted corrugated centering disk.

centering suspension can be made large in diameter and thereby obtain a
very low value of stiffness.
    Voice coil construction used in direct radiator loudspeakers are shown
in Fig. 6.77.   A voice coil wound on a cylindrical paper form with round wire
is shown in Fig. 6.77A.   Cement is used to bind the voice coil to the form.
The cement also serves to bind the adjacent turns of wire together.   Three
types of self-supporting voice coils are shown in Fig. 6.77A, C, and D.
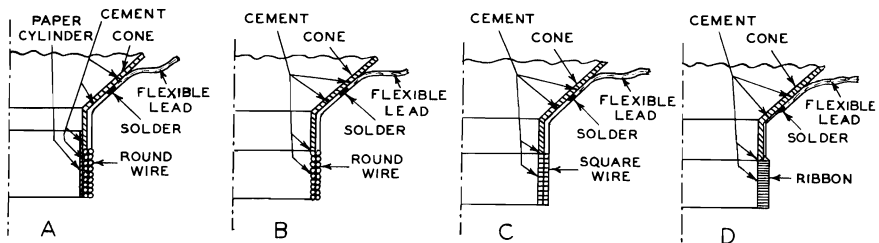
FIG. 6.77.   Sectional views of voice coil constructions.   A. Round enameled wire
wound on a paper form.   B, C, and D.   Self-supporting voice coils held together with
thermosetting cement.   B. Round wire.   C. Square wire.   D. Edgewise wound ribbon.

Thermosetting cement is used to bind the entire assembly.   The use of a
self-supporting coil eliminates the cylindrical paper form and thereby re-
duces the space required in the air gap.   The use of square wire or ribbon
effects a further reduction in the space required in the air gap.
    6.28. High-Frequency Sound Distributor.—The diameter of the
vibrating surface of multiple cones decreases with increase in frequency and
as a result the directional pattern is essentially independent of the frequency.
When a single uncorrugated cone is used to cover the high-frequency range
the directional pattern becomes quite narrow at the higher frequencies.

By means of a distributor consisting of vanes it is possible to spread the high-frequency radiation and thereby maintain uniform directional characteristics with respect to the frequency. The high-frequency contours of equal phase for a cone with and without a distributor are shown in Figs. 6.78A and 6.78B. The radius of curvature of the wavefront with the distributor being considerably less than that of the plane cone shows that the
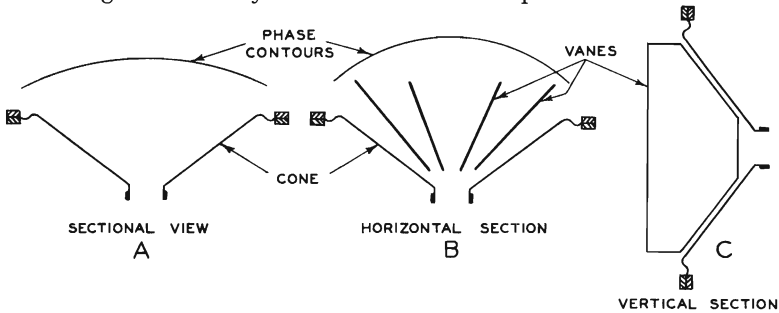
FIG. 6.78. High-frequency sound distributors for direct radiator loudspeakers. A. The contour of equal phase for a plain cone. B. The horizontal cross-sectional view of a cone with a vane distributor and the contour of equal phase. C. The vertical cross-sectional view of a cone with a vane distributor.

distributor broadens the radiation pattern. The vertical section, Fig. 6.78C, shows that the distributor will not broaden the pattern in this direction. In general, in radio or phonograph reproduction, the required vertical plane of spread is quite small. If a broader pattern is required in this plane crossed vanes may be used.

6.29. Field Structures.—Six typical electromagnetic and permanent magnet field structures used in direct radiator dynamic loudspeakers are shown in Fig. 6.79. For many years the loudspeakers employing the field structures shown in A and B of Fig. 6.79 were universally used in a-c powered radio receivers and phonographs. The field coil was used as a choke in the filter system of the high voltage power supply. The advent of the new permanent magnet alloys consisting of combinations of aluminum, nickel, cobalt, and iron and termed "Alnico" made it possible to design efficient magnetic structures of reasonable size. The air-gap flux densities obtained with field coil excitation could be duplicated with permanent magnets. Another important factor is the lower cost of the permanent magnet loudspeaker as compared to the field coil excitation even when the power supply is considered as part of the problem. As a consequence, the magnetic structures shown in C, D, E, and F of Fig. 6.79 are universally used in dynamic direct radiator loudspeakers today.

The design of field structures involves both empirical and theoretical considerations because the leakage flux cannot be predicted without some experimental data. It is beyond the scope of this book to give a comprehensive discussion on the design of field structures. However, since field
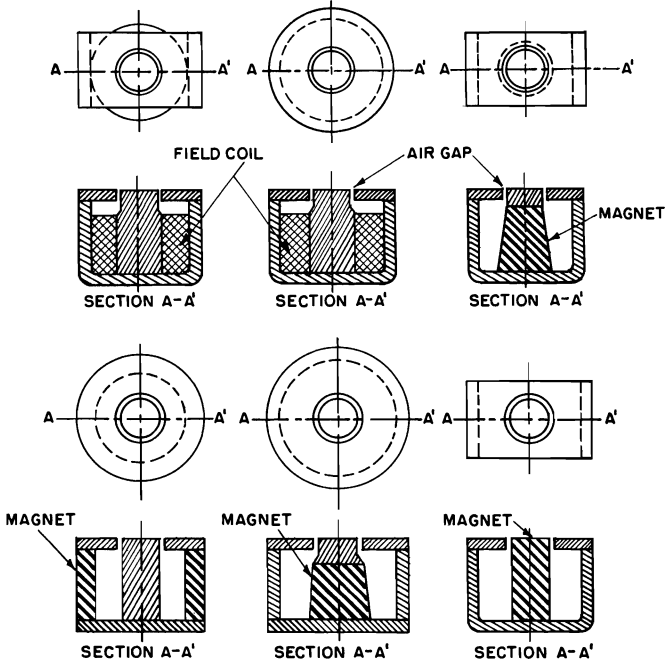
FIG. 6.79. Field structures. A and B, electromagnetic types. C, D, E, and F, permanent magnet types.

structures are used in loudspeakers, microphones, and other transducers it seems worthwhile to outline the fundamentals of magnetic circuits.

A few of the terms used in magnet systems will be defined.

Magnetic Flux—Magnetic flux is the physical manifestation of a condition existing in a medium or material subjected to a magnetizing influence. The quantity is characterized by the fact that an electromotive force is induced in a conductor surrounding the flux during any time that the flux changes in magnitude. In the cgs system the unit of magnetic flux is the maxwell.

Magnetomotive Force—Magnetomotive force in a magnetic circuit is the work required to carry a unit magnetic pole around the circuit against the magnetic field. In the cgs system, the unit of magnetomotive force is the gilbert.

Reluctance—Reluctance is the property of the magnetic circuit to resist magnetization. Thus the amount of magnetic flux resulting from a given magnetomotive force acting on a magnetic circuit is determined by the magnetic reluctance of the circuit.

Maxwell—The maxwell is the cgs unit of magnetic flux. It is the flux produced by a magnetomotive force of 1 gilbert in a magnetic circuit of unit reluctance.

Line—Line is a term commonly used interchangeably for a maxwell.

Gilbert—The gilbert is the cgs unit of magnetomotive force. It is the magnetomotive force required to produce 1 maxwell of magnetic flux in a magnetic circuit of unit reluctance.

Oersted—The oersted is the unit of field strength in the cgs system. It is the magnetomotive force equivalent to 1 gilbert per centimeter of length.

Gauss—The gauss is the unit of flux density. One gauss equals 1 maxwell per square centimeter.

Flux—Flux is the term applied to the physical manifestation of the presence of magnetic induction.

Flux Density—Flux density is the number of lines or maxwells per unit area in a section normal to the direction of the flux. In the cgs system the unit is the gauss.

Ampere-Turn—Ampere-turn is the unit of magnetomotive force. It is a product of the number of turns on a coil and the amperes passing through the turns.

Magnetizing Force—Magnetizing force is the magnetomotive force per unit length at any given point in a magnetic circuit. In the cgs system the unit of magnetizing force is the oersted.

Leakage—Leakage is that portion of the magnetic field that is not useful.

Leakage Coefficient—Leakage coefficient is the ratio of the total flux produced to the useful flux.

Induction, Intrinsic—Also known as ferric induction. Intrinsic induction is that portion of the induction in excess of the induction in a vacuum for the same magnetizing force.

Induction, Magnetic—Magnetic induction is the magnetic flux per unit area of a section normal to the direction of flux, resulting when a substance is subjected to a magnetic field. This is also known as magnetic flux density. In the cgs system the unit of magnetic flux density is the gauss.

Coercive Force—Coercive force is the magnetomotive force which must be applied to a magnetic material in a direction opposite to the residual induction to reduce the latter to zero.

Demagnetization—Demagnetization is the reduction of magnetization. It may be either partial or complete.

Demagnetization Curve—The demagnetization curve is that portion of the normal hysteresis loop in the second quadrant showing the induction in a magnetic material as related to the magnetizing force applied in a direction opposite to the residual induction.

Permeability—Permeability is the ratio of the magnetic induction in a given medium to the induction which would be produced in a vacuum by the action of the same magnetizing force.

The subject of analogies and the importance in the solution of problems in vibrating systems have been discussed in Chapter IV. Analogies between electrical and magnetic quantities may be used to solve problems in magnetic systems by reducing the system to an electrical network. The performance of the network may be determined by electrical circuit theory.

The fundamental equation of magnetic circuits is given by

$$\phi = \frac{M}{R} \qquad 6.35$$

where ϕ = total lines of flux, in maxwells,

M = magnetomotive force, in gilberts, and

R = reluctance, no unit.

Equation 6.35 is analogous to Ohm's Law in electrical circuits expressed as follows:

$$i = \frac{e}{r_E} \qquad 6.36$$

where i = current, in abamperes,

e = electromotive force, in abvolts, and

r_E = electrical resistance, in abohms.

The quantities, units, and symbols in electrical and magnetic systems are shown in Table 6.2.

TABLE 6.2

Electrical			Magnetic		
Quantity	Unit	Symbol	Quantity	Unit	Symbol
Electromotive Force	Volts $\times 10^8$	e	Magnetomotive Force	Gilberts	M
Electrical Resistance	Ohms $\times 10^9$	r_E	Reluctance	No Unit	R
Current Resistance	Amperes $\times 10^{-1}$	i	Flux	Maxwells	ϕ

The analogies between electrical and magnetic systems will be used to solve problems in magnetic systems in this book. The first illustration will be the solution of two simple magnetic systems, shown in Fig. 6.80.

The magnetomotive force, in gilberts, in an electromagnetic system, as shown in Fig. 6.80A is given by

$$M_T = 4\pi ni \qquad 6.37$$

where n = number turns, and

i = current, in abamperes.

The magnetizing force, in oersteds, developed by the permanent magnet in the permanent magnet system of Fig. 6.80B may be obtained from the demagnetization curves of Fig. 6.81. In order to use the minimum amount of material it is necessary to operate at the maximum value of $B \times H$. For Alnico V this is 470 oersteds per centimeter of length.

The total lines in the system at the coil or magnet, in maxwells, is

$$\phi_T = \phi_1 + \phi_2 \qquad\qquad 6.38$$

where $\phi_1 =$ lines in the air gap, in maxwells, and

$\phi_2 =$ lines in the leakage field, in maxwells.

The total magnetomotive force, in gilberts, developed by the energized coil or permanent magnet is

$$M_T = M_1 + M_2 \qquad\qquad 6.39$$
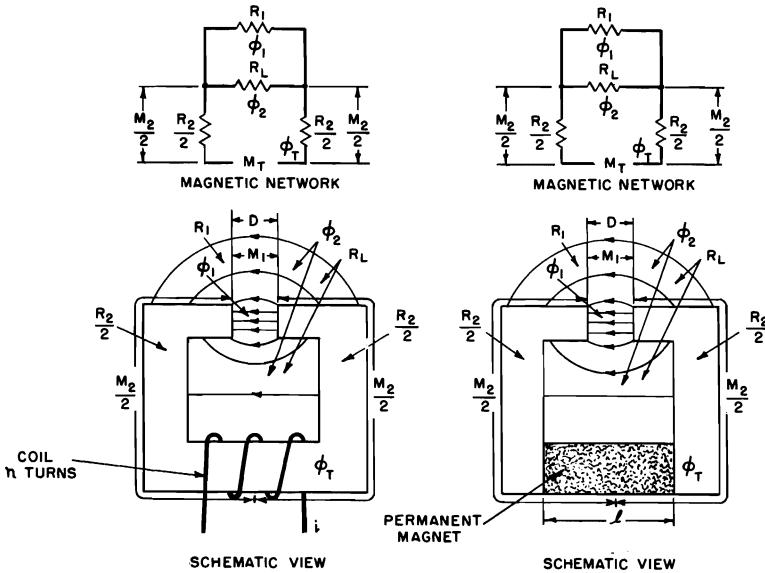
FIG. 6.80. Schematic views and magnetic networks of electromagnetic and permanent field structures. $M_1 =$ magnetomotive force drop in the air gap. $M_2 =$ magnetomotive force drop in the iron path. $M_T =$ magnetomotive force developed by the current i in the coil or by the permanent magnet. $\phi_1 =$ total flux lines in the air gap. $\phi_2 =$ total leakage flux lines. $\phi_T =$ total lines delivered by the coil or permanent magnet. $R_1 =$ reluctance of the air gap. $R_L =$ reluctance of the leakage paths. $R_2 =$ reluctance of the iron path.

The number of lines in the air gap, in maxwells, is

$$\phi_1 = \frac{M_1}{R_1} \qquad\qquad 6.40$$

where $R_1 =$ reluctance of the air gap, and

$M_1 =$ magnetomotive force across the air gap in gilberts.

The reluctance of the air gap is

$$R_1 = \frac{l}{A_1} \qquad 6.41$$

where l = length of the air gap in the direction of the flux in centimeters, and

A_1 = cross-sectional area of the air gap, in square centimeters.
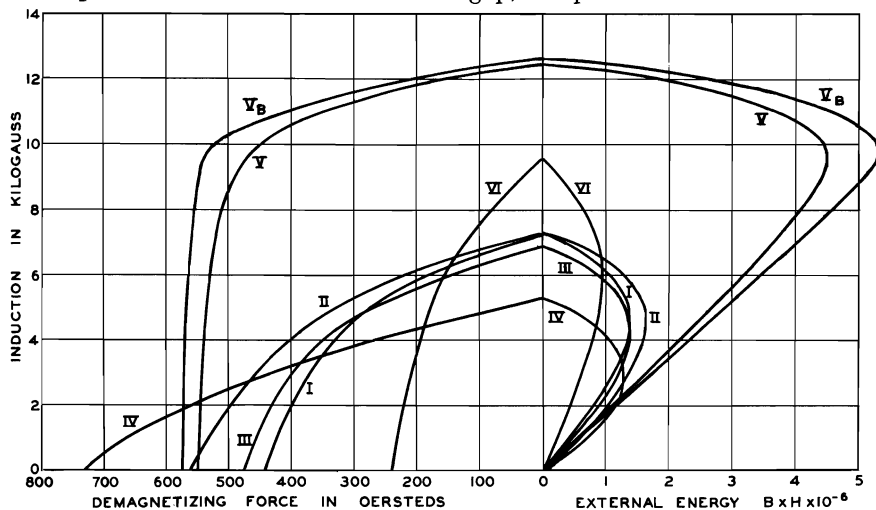
FIG. 6.81. Demagnetization and energy characteristics of permanent magnet materials. I. Alnico I; II. Alnico II; III. Alnico III; IV. Alnico IV; V. Alnico V; V_B. Alnico V_B; VI. 36 per cent cobalt.

The reluctance in the iron structure

$$R_2 = \frac{l}{\mu A_2} \qquad 6.42$$

where l = length of the path in the iron, in centimeters,

A_2 = cross-sectional area, in square centimeters, and

μ = permeability.

The curves[49] of Fig. 6.82 show the relation between the intrinsic flux density and the magnetizing force for different magnetic materials. The permeability may be obtained from the curves of Fig. 6.82 and the following relation

$$\mu = \frac{B}{H} \qquad 6.43$$

where B = flux density, in gausses, and

H = magnetizing force in oersteds.

[49] Kentner, A. E., *Gen. Elec. Rev. XLV*, No. 11, p. 633, 1942.

The magnetomotive force drop in the iron is given by

$$M_2 = R_2 \phi_2 \qquad\qquad 6.44$$

where R_2 = reluctance in the iron structure, and

ϕ_2 = lines in the iron, in maxwells.

The magnetomotive force drop in the iron should be made small compared to the magnetomotive force drop in the air gap.
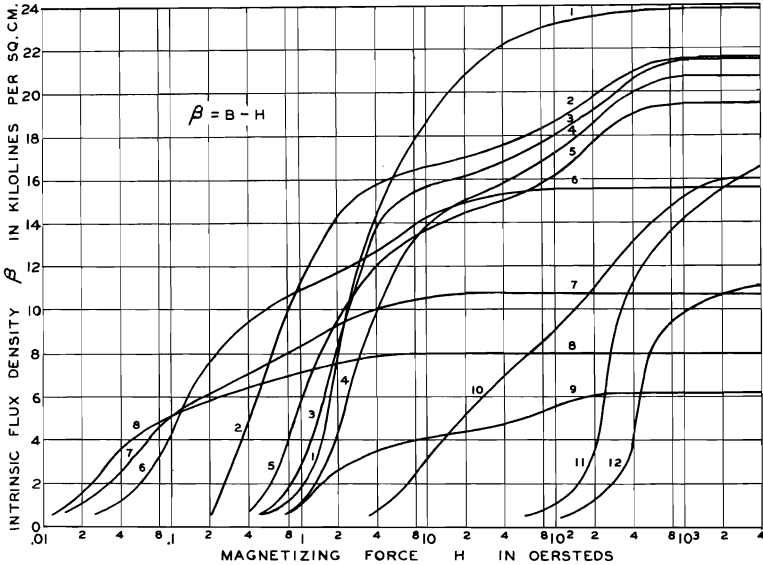
FIG. 6.82.   D-c magnetization characteristics for various magnetic materials.
1. Permandur, 50 per cent cobalt and 50 per cent iron.   2. Electrolytic iron.
3. Armco iron.   4. Annealed cold drawn steel.   5. Medium hard silicon steel.
6. Nickel iron alloy, 47 per cent nickel.   7. Permalloy, 79 per cent nickel.
8. Allegheny, mumetal.   9. Pure nickel.   10. Cast iron.   11. Cobalt permanent magnet steel.   12. Alnico permanent magnet alloy.   (After Kentner.)

The air-gap flux density, in gausses, is given by

$$B = \frac{\phi_1}{A_1} = \frac{M_1}{l} \qquad\qquad 6.45$$

By means of the above equations and the leakage flux it is possible to design the field structure. The starting point is usually the desired air-gap flux density. The air-gap flux, ϕ_1, and the magnetomotive force, M_1, required to produce this flux may be obtained from equation 6.45 and the flux density, B. If the leakage lines, ϕ_2, are known then the total lines, ϕ_T, are given by equation 6.38. The magnetomotive drop, M_2, in the iron can be obtained from the number of lines, ϕ_2, in the iron and the reluctance, R_2, as shown in equation 6.44. Then the total magnetomotive, M_T, can

be determined from equation 6.39. The number of ampere turns required to produce this magnetomotive is obtained from equation 6.36. In the case of a permanent magnet, the length of magnet which delivers the required magnetomotive force is

$$l = \frac{M_T}{H} \qquad 6.46$$

where l = length, in centimeters,

M_T = total magnetomotive force required, in gilberts, and

H = demagnetizing force, in oersteds (see Fig. 6.81).

For Alnico V, the demagnetizing force is 470 oersteds per centimeter of length, when the flux density magnet is 9500 gausses. The required cross-sectional area of the permanent magnet is the total flux ϕ_T divided by 9500. For relatively long magnets or large air gaps, the leakage flux of the magnet must also be considered in obtaining the cross-sectional area. This consideration results in a larger cross section at the center of the magnet.

Measurements of the air-gap flux density, the leakage flux and the flux density in various parts of the magnetic circuit can be made by means of a calibrated flux meter and a loop or coil. The air-gap flux density can be obtained by means of a calibrated search coil and fluxmeter. The flux in any part of the magnetic circuit can be obtained by placing a loop of one or more turns around the section to be tested. The loop is connected to the fluxmeter. This coil is then pulled out to a point where there is no flux. The flux can be obtained from the deflection and calibration of the fluxmeter and the number of turns in the coil. From these measurements, data can be obtained which will give the total flux, the leakage flux, the air-gap flux, and the flux density in the iron and permanent magnet structure. This data together with the equations and data in the preceding considerations will indicate the direction of improvement from the standpoint of air-gap flux density, leakage flux, and optimum cross section of the iron and permanent magnet.

6.30. Electrostatic Loudspeaker.[50,51,52]—All of the considerations in this chapter have been concerned with the dynamic direct radiator loudspeaker. During the early stages of sound reproduction, magnetic and electrostatic loudspeaker mechanisms were also employed. However, during the past 25 years the dynamic direct radiator loudspeaker has been universally employed for the reproduction of sound. Recently, some attention has again been given to electrostatic loudspeakers. The advent of thin plastic sheets with excellent electrical properties has revived interest in electrostatic loudspeakers. In view of these developments it seems logical to analyze the action and describe some of the new developments in

[50] Olson, " Dynamical Analogies," D. Van Nostrand Company, Princeton, N.J., 1943.

[51] Hanna, C. R., *Jour. Acous. Soc. Amer.*, Vol. 2, No. 2, p. 143, 1930.

[52] Hunt, " Electroacoustics," John Wiley and Sons, New York, N.Y., 1954.

electrostatic loudspeakers. Consider the system of Fig. 6.83, consisting of a vibrating surface moving normal to its plane and separated from a fixed conductor. The force, in dynes, between the plates is

$$f_M = \frac{e^2 A}{8\pi a^2} \qquad 6.47$$

where $e =$ electromotive force between plates, in statvolts,

$\quad a =$ normal distance between the plates, in centimeters, and

$\quad A =$ area of the plate, in square centimeters.
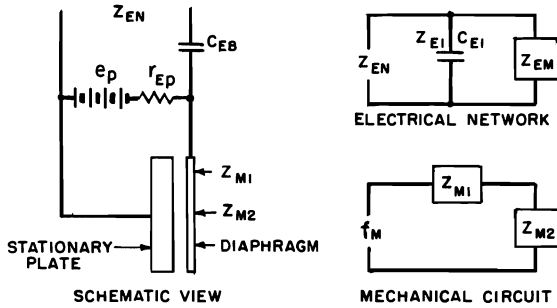
Fig. 6.83. Schematic view, electrical network, and mechanical circuit of an electrostatic loudspeaker. In the electrical network, $z_{EN} =$ the normal electrical capacitance of the loudspeaker. $z_{EM} =$ the motional electrical impedance of the loudspeaker. $z_{E1} =$ the damped electrical impedance of the loudspeaker. $z_{E1} = 1/j\omega C_{E1}$. $C_{E1} =$ the damped electrical capacitance of the loudspeaker. In the mechanical circuit, $f_M =$ driving force. $z_{M1} =$ the mechanical impedance of the movable plate. $z_{M2} =$ the mechanical impedance of the load. (After Olson, "Dynamical Analogies", D. Van Nostrand Company, Princeton, N.J., 1942.)

Assume that the polarizing voltage is e_0 and that the alternating voltage is $e = e_{max}\sin \omega t$. The force, in dynes, between the plates is

$$f_M = \frac{(e_0^2 + e_{max} \sin \omega t)^2 A}{8\pi a^2} \qquad 6.48$$

$$f_M = \frac{e_0^2 + 2e_0 e_{max} \sin \omega t + 1/2\, e^2_{max} - 1/2\, e^2_{max} \cos 2\omega t}{8\pi a^2} A \qquad 6.49$$

The first and third terms in the numerator of equation 6.49 represent steady forces. The fourth term is an alternating force of twice the frequency of the impressed voltage. The second term is an alternating force of the frequency of the impressed voltage. If the polarizing electromotive e_0 is large compared to the alternating electromotive force $e_{max}\sin \omega t$, the fourth term will be negligible. The useful force, in dynes, then is the second term

which causes the moving surface to vibrate with a velocity which corresponds to the impressed electromotive force.

$$f_M = \frac{e_0 e_{max} \sin \omega t}{4\pi a^2} A = \frac{e_0 e}{4\pi a^2} \qquad 6.50$$

The electrical motion impedance of this system will now be considered. The charge, in statcoulombs, on the condenser is

$$q = C_E e_0 \qquad 6.51$$

where e_0 = potential difference between the plates, in statvolts, and
$\quad C_E$ = capacity per unit area, in statfarads.

The current, in statamperes, generated due to motion is

$$i = \frac{dq}{dt} \qquad 6.52$$

From equations 6.51 and 6.52 the generated current is

$$i = e_0 \frac{dC_E}{dx} \frac{dx}{dt} \qquad 6.53$$

The electrical capacitance of the condenser, in stratfarads, is

$$C_{E1} = \frac{A}{4\pi a} \qquad 6.54$$

Let the movable plate be deflected a distance Δx away from the fixed plate. The electrical capacitance is

$$C_{E1} - \Delta C_{E1} = \frac{A}{4\pi(a + \Delta x)} \qquad 6.55$$

Now let the movable plate be deflected a distance Δx toward the fixed plate. The electrical capacitance is

$$C_{E1} + \Delta C_{E1} = \frac{A}{4\pi(a - \Delta x)} \qquad 6.56$$

The difference between the two conditions is

$$\Delta C_{E1} = \frac{A\Delta x}{4\pi a^2 - (\Delta x)^2} = \frac{A\Delta x}{4\pi a^2} \qquad 6.57$$

The change in electrical capacitance with respect to x is

$$\frac{dC_{E1}}{dt} = \frac{A}{4\pi a^2} \qquad 6.58$$

Substituting equation 6.58 in 6.53, the generated current, in statamperes, is

$$i = \frac{e_0 A}{4\pi a^2} \dot{x} \qquad 6.59$$

From the mechanical circuit of Fig. 6.83, the mechanical rectilineal impedance of the vibrating system is

$$z_M = z_{M1} + z_{M2} \qquad\qquad 6.60$$

where z_M = total mechanical rectilineal impedance of the vibrating system, in mechanical ohms,

z_{M1} = mechanical rectilineal impedance of the vibrating plate, in mechanical ohms, and

z_{M2} = mechanical rectilineal impedance of the load, in mechanical ohms.

The mechanical rectilineal impedance at the plate is

$$z_M = \frac{f_M}{\dot{x}} \qquad\qquad 6.61$$

From equations 6.50 and 6.59

$$\frac{e}{i} = \frac{16\pi^2 a^4}{e_0{}^2 A^2} \frac{f_M}{\dot{x}} \qquad\qquad 6.62$$

From equations 6.60, 6.61, and 6.62

$$z_{EM} = \frac{16\pi^2 a^4}{e_0{}^2 A^2} z_M \qquad\qquad 6.63$$

where z_{EM} = motional electrical impedance, in statohms, and

z_M = total mechanical rectilineal impedance presented to the vibrating surface including the vibrating surface.

The normal electrical impedance z_{EN}, in statohms, of the condenser is

$$z_{EN} = \frac{z_{E1} z_{EM}}{z_{E1} + z_{EM}} \qquad\qquad 6.64$$

where z_{E1} = damped electrical impedance of the condenser, in statohms, and

z_{EM} = motional electrical impedance of the condenser, in statohms.

The motional electrical impedance as given by equation 6.64 may be represented as in parallel with the blocked or damped electrical impedance of the condenser as depicted by the electrical network in Fig. 6.83.

The preceding considerations show that the efficiency is proportional to the square of the area of the plates, proportional to the square of the polarizing voltage, inversely proportional to the fourth power of the spacing between the plates, and inversely proportional to the mechanical impedance of the vibrating system. Thus, it will be seen that in order to obtain high efficiency the spacing must be small because the relationship between spacing and efficiency is a fourth-power function. This, of course, limits the amplitude. For example, the maximum amplitude of a 15-inch dynamic loudspeaker is about $\pm\frac{3}{8}$ inch. To provide for this amplitude in a bilateral electrostatic loudspeaker would require a spacing between the plates of at least one inch, which is, of course, impractical. About 25 years ago a large

bilateral condenser loudspeaker was built in which the diaphragm area was 100 square feet. The spacing between the fixed plates was $\frac{1}{4}$ inch. The vibrating membrane was .010 rubber with gold leaf as the conductor. The polarizing voltage was 4000 volts. In the low-frequency range below 100 cycles the performance of the electrostatic loudspeaker was not comparable to a 12-inch dynamic loudspeaker from the standpoints of efficiency and power handling capacity. Employing a thinner membrane would not have
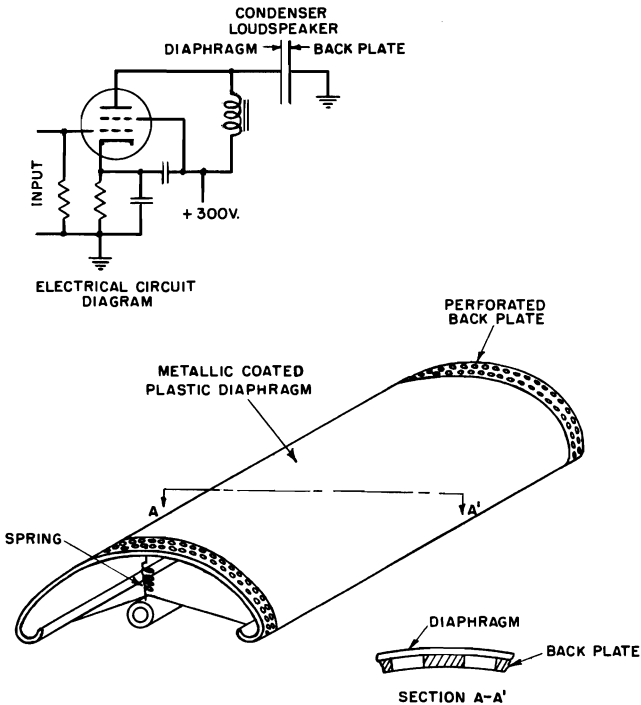
FIG. 6.84. Perspective and sectional views and electrical circuit diagram of a high-frequency unilateral electrostatic loudspeaker.

increased the efficiency because the mass of the air load was the major factor in the mechanical impedance below 100 cycles. Using more than 4000 volts polarizing voltage leads to corona and other leakage effects. These tests substantiate the theoretical consideration, namely, that the operation of a practical electrostatic loudspeaker must be confined to the upper portion of the audio frequency range, say above 1000 cycles.

A commercial electrostatic loudspeaker[53] for the frequency range above 7500 cycles is shown in Fig. 6.84. The diaphragm is a .001-inch plastic with a thin metallic coating. The diaphragm rests directly upon the perforated metal back plate. The electrostatic loudspeaker is coupled

[53] Bobb, Goldman, and Roop, *Jour. Acous. Soc. Amer.*, Vol. 27, No. 6, p. 1128, 1955.

directly to the vacuum tube; the plate supplies the polarizing voltage. Since the effective spacing is about .001 inch, the sensitivity is comparable to conventional dynamic loudspeakers. This small spacing limits the maximum amplitude and confines the operation to the frequency range above 7500 cycles.

Front and sectional views and the electrical circuit diagram of a bilateral or push-pull electrostatic loudspeaker[54] is shown in Fig. 6.85. In commercial versions of this loudspeaker the area of the diaphragm is one square foot. The spacing between the movable and fixed plates is about $\frac{1}{16}$ inch.
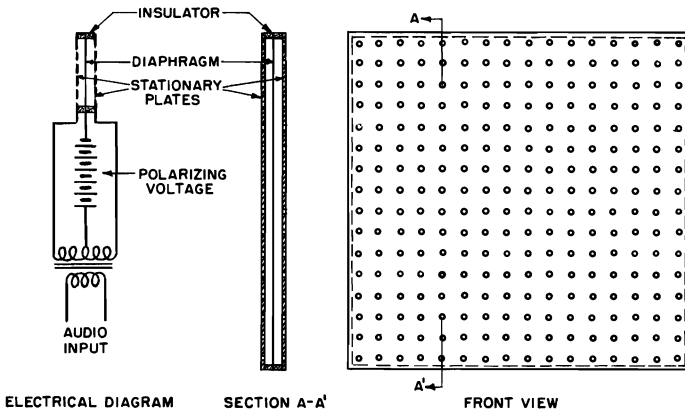
FIG. 6.85. Front and sectional views and electrical diagram of a bilateral electrostatic loudspeaker.

A polarizing voltage of about 3000 volts is used. Under these conditions the sensitivity must be confined to the frequency range above 1000 cycles because the power handling capacity is inadequate below 1000 cycles due to the limited amplitude. Since the directivity pattern of a diaphragm of these dimensions is quite sharp in the high-frequency region, two or more units are used and directed so as to obtain the required coverage.

6.31. Sound Power Emitted by a Loudspeaker.—The sound power, P_{AN}, in ergs per second, emitted by a point source or by a nondirectional loudspeaker is given by

$$P_{AN} = \frac{4\pi r^2 p^2}{\rho c} \qquad 6.65$$

where p = sound pressure, in dynes per square centimeter at the distance r, in centimeters.

If the sound source or loudspeaker is directional, that is, if the sound emitted varies with the direction with respect to some axis of the system, then the total sound power from the geometry of Fig. 6.86, is given by

$$P_{AD} = \frac{r^2}{\rho c}\int_0^{2\pi}\int_0^{\pi} p^2(\theta,\,\psi,\,r)\,\sin\theta\,d\theta\,d\psi \qquad 6.66$$

[54] Janszen, A. A., *Jour. Aud. Eng. Soc.*, Vol. 3, No. 2, p. 87, 1955.

where $p(\theta, \psi, r)$ = sound pressure, in dynes per square centimeter, at a distance r, in centimeters, and the angle θ and angle ψ, and

θ and ψ = angular polar coordinates of the system. The loudspeaker axis coincides with the x axis.
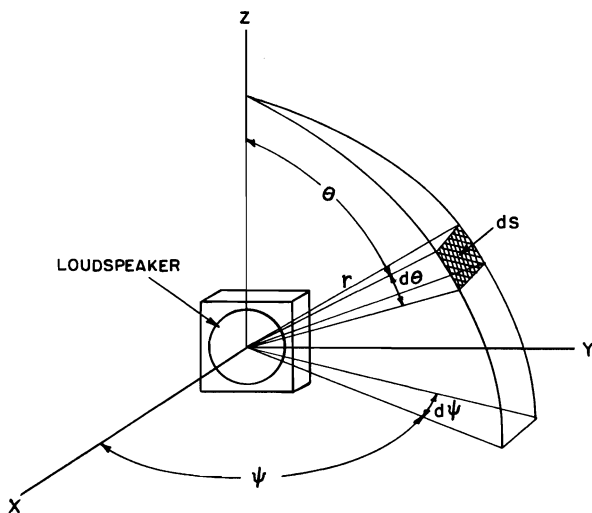
FIG. 6.86. The geometry for obtaining the total sound output of a loudspeaker.

6.32. Loudspeaker Directivity Index.—The directivity factor of a loudspeaker is the ratio of the sound power which would be radiated if the free space axial sound pressure were constant over 4π solid angles to the actual sound power radiated. The directivity factor, Q, is given by

$$Q = \frac{P_{AN}}{P_{AD}} \qquad 6.67$$

where P_{AN} and P_{AD} are given by equations 6.65 and 6.66.

The directivity index can be computed from the directional characteristics of the loudspeaker.

The directivity index of a loudspeaker can be obtained from the directivity factor as follows:

$$DI_{DB} = \log_{10} Q \qquad 6.68$$

7

HORN LOUDSPEAKERS

7.1. Introduction.—Large-scale reproduction of sound, involving several acoustical watts, is quite commonplace. Since high power audio-frequency amplifiers are costly, it is logical to reduce the amplifier output to a minimum by the use of high-efficiency loudspeakers. At the present time, horn loudspeakers seem to be the only satisfactory high-efficiency system for large-scale sound reproduction. A horn loudspeaker[1,2,3,4,5] 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 generator. This feature is extremely valuable for obtaining maximum over-all efficiency in the design of the acoustical system. Employing a suitable combination of horns, directional characteristics which are independent of the frequency, as well as practically any type of directional pattern, may be obtained. The combination of high efficiency and the possibility of any directional pattern makes the horn loudspeaker particularly suitable for large-scale reproduction. For applications requiring high quality reproduction of intense sound, some consideration should be given to the introduction of frequencies not present in the output due to nonlinearity of the operating characteristics of the elements which constitute the vibrating system of the loudspeaker. It is the purpose of this chapter to consider the principal factors which influence and govern the efficiency, distortion, and power handling characteristics of a horn loudspeaker and to describe several horn loudspeaker systems.

7.2. Efficiency.[6,7,8]—The efficiency of a loudspeaker is the ratio of the useful acoustical power output to the electrical power input. For all large-scale reproduction of sound, efficiency is an important consideration. Specifically, the efficiency depends primarily upon the flux density, the

[1] Hanna and Slepian, *Jour. A.I.E.E.*, Vol. 43, No. 3, p. 251, 1924.
[2] Wente and Thuras, *Bell Syst. Tech. Jour.*, Vol. 7, No. 1, p. 1940, 1928.
[3] Olson, H. F., *Jour. Acous. Soc. Amer.*, Vol. 2, No. 4, p. 242, 1931.
[4] Wente and Thuras, *Jour. A.I.E.E.*, Vol. 53, No. 1, p. 17, 1934.
[5] Olson, H. F., *RCA Review*, Vol. 2, No. 2, p. 265, 1937.
[6] Wente and Thuras, *Jour. A.I.E.E.*, Vol. 43, No. 3, p. 251, 1924.
[7] Olson, H. F., *RCA Review*, Vol. 2, No. 2, p. 265, 1937.
[8] Massa, F., *Electronics*, Vol. 10, No. 4, p. 30, 1937.

mass and the density-resistivity product of the voice coil, the mass of the diaphragm, the ratio of the diaphragm to the throat area, the dimensions of the air chamber, the area of the diaphragm, and the voice coil temperature. Some of the factors are interrelated and others are independent; as a consequence, it is impossible to depict in one set of characteristics the effect of the various parameters. Therefore, the design of a horn loudspeaker is usually a long and tedious task. The labor is further increased when economic considerations are involved. It is believed that a general consideration of the problem, together with a series of characteristics, is valuable for initiating the design of a loudspeaker and for facilitating the determination of the ultimate constants. The throat acoustical impedance and directional characteristics of a large number of representative horns were given in Secs. 5.28 and 2.19. From these characteristics it is possible to interpolate the characteristic of practically any horn and thus eliminate considerable initial work in the design of a horn loudspeaker. It is the purpose of this section to consider the effect of the various parameters, referred to above, upon the efficiency of a horn loudspeaker and to include characteristics depicting the influence of these parameters upon the performance.

A. *The Relation Between the Voice Coil Mass, the Load Mechanical Resistance, and the Initial Efficiency.*—Initial efficiency is the ratio of sound power output to electrical power input in the system in which the mechanical reactance is negligible and in which all the mechanical resistance may be attributed to radiation. In most loudspeakers the mechanical reactance of the vibrating system is negligible in the upper low-frequency range. Near the cutoff of the horn the mechanical reactive component at the throat of the horn is relatively large. Furthermore, the mechanical reactance due to the stiffness of the diaphragm may be comparable to the other mechanical impedances in the system. Therefore, the starting point in most horn loudspeaker designs is a determination of the initial efficiency. This is logical because the mechanical reactances referred to above are usually chosen so their effect upon the efficiency characteristic in the upper low-frequency range is very small. It is the purpose of this section to discuss briefly the factors which influence the initial efficiency and to include a family of curves showing the effect of the flux density, the voice coil mass, the throat area, and the diaphragm diameter upon the initial efficiency. The motional electrical impedance,[9] z_{EM}, in ohms, is given by

$$z_{EM} = \frac{(Bl)^2}{z_M} \times 10^{-9} \qquad 7.1$$

where B = flux density, in gausses,

$\quad l$ = length of wire in the voice coil, in centimeters, and

$\quad z_M$ = mechanical impedance of the vibrating system, in mechanical ohms, at the point f_M in Fig. 7.1.

[9] Olson, "Dynamical Analogies," D. Van Nostrand Company, Princeton, N.J., 1943.

The efficiency, μ, in per cent, is

$$\mu = \frac{r_{EM}}{r_{ED} + r_{EM}} \times 100 \qquad\qquad 7.2$$

where r_{EM} = electrical resistance component of the motional electrical resistance, in ohms, and

r_{ED} = damped electrical resistance of the voice coil, in ohms.
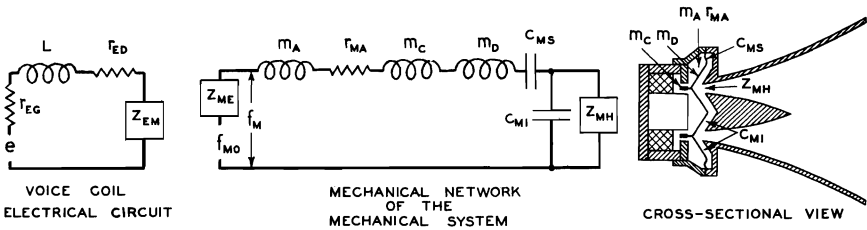
FIG. 7.1   Cross-sectional view of a horn loudspeaker, the electrical circuit and mechanical network of the vibrating system.   In the voice coil circuit: e = the internal voltage of the generator. r_{EG} = the internal electrical resistance of the generator.   L = the inductance of the voice coil. r_{ED} = the damped electrical resistance of the voice coil. z_{EM} = the motional electrical impedance.   In the mechanical network: m_A and r_{MA} = the mass and mechanical radiation resistance due to the air load on the back of the diaphragm. m_C and m_D = the masses of the voice coil and diaphragm. C_{MS} and C_{M1} = the compliances of the suspension and air chamber. z_{MH} = the mechanical impedance at the throat of the horn. z_{ME} = the mechanical impedance due to the electrical circuit. f_M = the force generated in the voice coil. f_{MO} = the force of the mechanical generator.

In the mechanical network, Fig. 7.1, the mechanical impedance, z_M, in mechanical ohms, at f_M is given by

$$z_M = j\omega m_A + r_{MA} + j\omega m_C + j\omega m_D + \frac{1}{j\omega C_{MS}} + \frac{z_{MH}}{j\omega C_{M1} z_{MH} + 1} \qquad 7.3$$

where m_A = mass of the air load on the back of the diaphragm, in grams,

$\quad m_C$ = mass of the voice coil, in grams,

$\quad m_D$ = mass of the diaphragm, in grams,

$\quad r_{MA}$ = mechanical resistance load on the back of the diaphragm, in mechanical ohms,

$\quad C_{MS}$ = compliance of the suspension, in centimeters per dyne,

$\quad C_{M1}$ = compliance of the air chamber, in centimeters per dyne,

$\quad z_{MH} = r_{MH} + jx_{MH}$ = mechanical impedance of the throat of the horn, in mechanical ohms,

$\quad r_{MH}$ = mechanical resistance of the throat of the horn, in mechanical ohms, and

$\quad x_{MH}$ = mechanical reactance of the throat of the horn, in mechanical ohms.

For initial efficiency considerations, the mechanical reactance of the mechanical system is assumed to be negligible compared to the radiation mechanical resistance, that is, m_A, m_C, m_D, C_{M1}, $1/C_{MS}$, and x_{MH} are zero. r_{MA} is also negligible. Then

$$z_M = r_{MH} = 42 \frac{A_D{}^2}{A_T} \qquad\qquad 7.4$$

where A_D = area of the diaphragm, in square centimeters, and
$\quad\ A_T$ = area of the throat, in square centimeters.

Substituting equations 7.1 and 7.4 in equation 7.2

$$\mu = \frac{B^2}{\left(\dfrac{42 A_D{}^2 r_{ED}}{l^2 A_T}\right) 10^9 + B^2} \times 100 \qquad\qquad 7.5$$

The electrical resistance,[10] r_{ED}, in ohms, is given by

$$r_{ED} = \frac{K_r l}{S} \times 10^{-6} \qquad\qquad 7.6$$

where K_r = resistivity of the voice coil material, in microhms, per centimeter cube (see Table 6.1),
$\quad l$ = length of the conductor, in centimeters, and
$\quad S$ = area of the conductor, in square centimeters.

Then equation 7.5 becomes

$$\mu = \frac{B^2}{\left(\dfrac{42 A_D{}^2 K_r}{l S A_T}\right) 10^3 + B^2} \times 100 \qquad\qquad 7.7$$

The mass of the coil, m_C, is

$$m_C = l S \rho \ \text{grams} \qquad\qquad 7.8$$

where ρ = density, in grams per cubic centimeter (see Table 6.1). The efficiency may be written, employing equation 7.8, as

$$\mu = \frac{B^2}{\left(\dfrac{42 A_D{}^2 K_r \rho}{m_C A_T}\right) 10^3 + B^2} \times 100 \qquad\qquad 7.9$$

For a particular material, $K_r \rho$ is a constant. Equation 7.9 gives the efficiency in terms of B^2, m_C, and $A_D{}^2/A_T$. The efficiency as a function of

---

[10] The voice coil electrical circuit is shown in Fig. 7.1. r_{ED} is the total damped electrical resistance of the voice coil and includes skin effect and hysteresis losses in the iron. L is the inductance of the voice coil. As shown in Fig. 6.6, the electrical impedance of the voice coil increases at the high frequencies due to the electrical reactance of L and an increase in electrical resistance due to skin effect and hysteresis losses in the iron circuit. In order to simplify these considerations the damped electrical resistance will be assumed to be the same as the ohmic (d-c) electrical resistance.

$A_D{}^2/A_T$ for aluminum voice coils of 0.1, 0.25, 0.5, 1, 2, 4, and 8 grams and flux densities of 22,000 and 14,000 gausses is shown in Fig. 7.2. The characteristics of Fig. 7.2 also apply to a copper voice coil if the abscissa are multiplied by 0.5. Equation 7.9 and Fig. 7.2 show the factors which influence the initial efficiency of a horn loudspeaker.
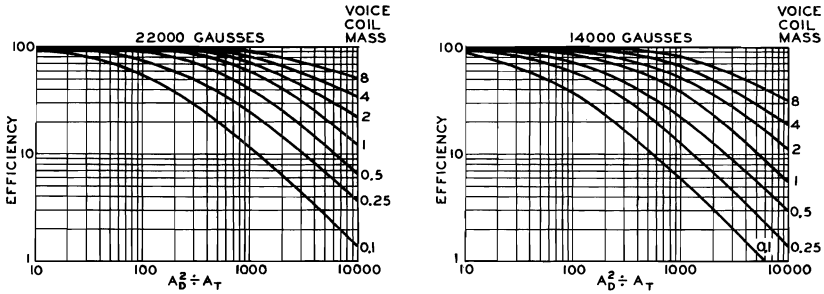
Fig. 7.2. The initial efficiency, in per cent, of a horn loudspeaker as a function of $A_D{}^2/A_T$ for aluminum voice coils having masses of 0.1, 0.25, 0.5, 1.2, 4, and 8 grams and flux densities of 22,000 and 14,000 gausses. A_D and A_T are the areas of the diaphragm and throat, respectively, in square centimeters. The above graphs may be applied to a copper voice coil by multiplying the ratio $A_D{}^2/A_T$ by one-half.

B. *The Effect of the Mass of the Vibrating System upon the Efficiency.*—In the preceding section the mechanical reactance of the vibrating system was assumed to be negligible compared to the mechanical resistance. The mechanical mass reactance of the diaphragm and voice coil influences the efficiency when this mechanical reactance becomes comparable to the mechanical resistance. It is the purpose of this section to consider the effect of the mechanical reactance of the vibrating system upon the efficiency.

The real part of the motional electrical impedance, equation 7.1, is

$$r_{EM} = \left(\frac{(Bl)^2 r_M}{r_M{}^2 + x_M{}^2} \right) 10^{-9} \text{ ohms} \qquad 7.10$$

where r_M = mechanical resistance of the vibrating system, in mechanical ohms, and

x_M = mechanical reactance of the vibrating system, in mechanical ohms.

In this discussion, let

$$x_M = \omega \, (m_D + m_C) \qquad 7.11$$

At the high frequencies the mechanical reactance due to C_{MS} and x_{MA} is negligible compared to the mechanical reactance due to the mass of the diaphragm. In order to divorce the effect of the air chamber from the effect of the mass of the diaphragm, the compliance, C_{M1}, will be assumed to be

zero. For the same reason r_{MA} will be assumed to be zero. These effects will be considered in following sections. The mechanical resistance, r_M, then becomes the horn throat resistance r_{MH}. The throat mechanical resistance is given by

$$r_{MH} = 42 \frac{A_D{}^2}{A_T} \qquad 7.12$$

where $A_T =$ area of the throat, in square centimeters. Substituting equation 7.10 in 7.2, the efficiency, in per cent, becomes

$$\mu = \frac{(Bl)^2 r_{MH}}{r_{ED}(r_{MH}{}^2 + x_M{}^2)10^9 + (Bl)^2 r_{MH}} \times 100 \qquad 7.13$$

This expression shows that the efficiency is a function of the flux density, the coil mass and material, the diaphragm mass, the throat resistance, and
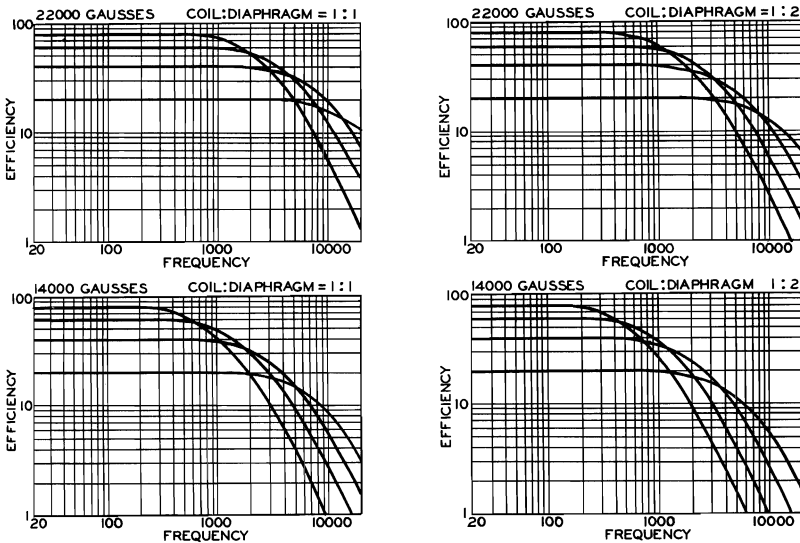
FIG. 7.3. The efficiency, in per cent, as a function of the frequency of horn loud-speaker systems having ratios of voice coil mass to diaphragm mass of 1:2 and 1:1, flux densities of 22,000 and 14,000 gausses, and initial efficiencies of 20, 40, 60, and 80 per cent for an aluminum coil. The above graphs may be applied to a copper voice coil by multiplying the frequency by one-half.

the frequency. The efficiency characteristics for ratios of voice coil mass to diaphragm mass of 1 : 1 and 1 : 2, and flux densities of 22,000 and 14,000 gausses for an aluminum voice coil are shown in Fig. 7.3. The characteristics of Fig. 7.3 are applicable to a copper voice coil by multiplying the abscissa by 0.5. In order to connect with the characteristics of initial efficiency of Fig. 7.2, these curves are depicted in terms of the initial efficiency (20, 40, 60, and 80 per cent). These data show that it is a

comparatively simple matter to obtain high efficiencies at the lower frequencies. However, at the higher frequencies the efficiency is limited by the mass of the diaphragm and voice coil.

C. *The Effect of the Air Chamber upon the Efficiency.*[11,12,13,14,15,16]—The results of the preceding sections were obtained by assuming the compliance of the air chamber to be zero. In general, it is impractical to design a high efficiency loudspeaker to cover a wide frequency range without an air chamber, because the diaphragm area is usually larger than the throat area. In order to eliminate interference, the dimensions of the elements of the air chamber are usually made small compared to the wavelength. When these conditions obtain, the volume of the air chamber appears as a compliance. At the higher frequencies, the mechanical impedance at the throat of the horn is resistive, the mechanical reactance of the suspension is very small, and the mechanical impedance of the diaphragm system is a mechanical mass reactance. The mechanical network reduces to a mechanical resistance and compliance in parallel connected in series with a mass. It is the purpose of this section to show the effect of the air chamber upon the efficiency from the standpoint of this mechanical network. The mechanical impedance of a mechanical resistance and compliance in parallel, which is the equivalent of the throat mechanical resistance and compliance of the air chamber, is given by

$$z_M = \frac{r_{MH}}{1 + j\omega r_{MH}C_{M1}} \qquad 7.14$$

where r_{MH} = mechanical resistance at the horn throat, in mechanical ohms, and

C_{M1} = compliance of the air chamber, in centimeters per dyne.

The throat mechanical resistance, r_{MH}, is given by equation 7.12. The mechanical compliance, Sec. 5.7, of the air chamber is given by

$$C_{M1} = \frac{C_A}{A_D{}^2} = \frac{V}{\rho c^2 A_D{}^2} \qquad 7.15$$

where C_A = acoustical capacitance of the air chamber, in (centimeters)^5 per dyne.

V = volume of the air chamber, in cubic centimeters,

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

c = velocity of sound, in centimeters per second.

[11] Hanna and Slepian, *Jour. A.I.E.E.*, Vol. 43, No. 3, p. 251, 1924.
[12] Wente and Thuras, *Bell Syst. Tech. Jour.*, Vol. 7, No. 1, p. 140, 1928.
[13] Olson, H. F., *Jour. Acous. Soc. Amer.*, Vol. 2, No. 4, p. 242, 1931.
[14] Wente and Thuras, *Jour. A.I.E.E.*, Vol. 53, No. 1, p. 17. 1934.
[15] Olson, H. F., *RCA Review*, Vol. 2, No. 2, p. 265, 1937.
[16] Smith, Bob H., *Jour. Acous. Soc. Amer.*, Vol. 25, No. 2, p. 305, 1953.

Referring to the system shown in Fig. 7.1, it is obvious that the effect of the air chamber will be to reduce the mechanical reactance of the system at the high frequencies and thereby increase the efficiency over a wide range. Figure 7.4 shows the efficiency characteristics of a system consisting of a voice coil and diaphragm having a mechanical reactance of 1 ohm at 1000 cycles coupled to the throat of a horn having a mechanical resistance of 1 ohm and an air chamber having a mechanical reactance of 1 ohm at 1000 cycles, 2000 cycles and an infinite mechanical reactance for an initial efficiency of 20 per cent, 40 per cent, 60 per cent, and 80 per cent. These characteristics are applicable to other frequencies by multiplying the
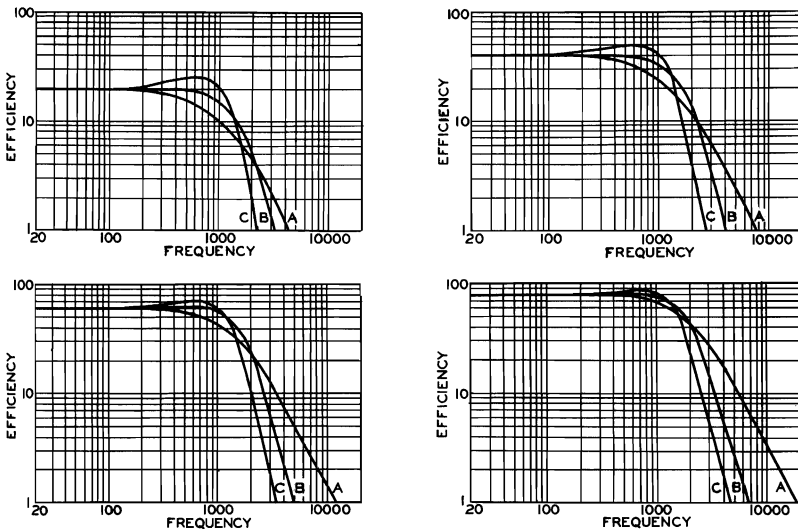
FIG. 7.4. The efficiency, in per cent, as a function of the frequency of a voice coil and diaphragm having a mechanical reactance of 1 mechanical ohm at 1000 cycles coupled to a throat of a horn having a mechanical resistance of 1 mechanical ohm and an air chamber having the following mechanical reactances: A. An infinite mechanical reactance. B. A negative mechanical reactance of 1 mechanical ohm at 2000 cycles. C. A negative mechanical reactance of 1 mechanical ohm at 1000 cycles for initial efficiencies of 20, 40, 60, and 80 per cent. These characteristics are applicable to other frequencies by multiplying the frequency by any number and multiplying the mass and the compliance by the reciprocal of the number.

abscissa by any number and, of course, multiplying the mass and the compliance by the reciprocal of the number. These characteristics are also applicable for other values of mass and mechanical resistance by simply multiplying these two factors by the same number and the compliance by the reciprocal of that number. The characteristics shown in Fig. 7.4 have included mass-compliance products which cover the useful range of values.

—larger products result in a peaked characteristic, smaller values do not show much deviation from zero value of compliance.

D. *The Effect of the Generator Electrical Impedance and the Mechanical Impedance at the Throat of the Horn upon the Efficiency.*—Due to the impracticability of a horn mouth diameter comparable to the wavelength for low-frequency loudspeakers, it is interesting to note that a relatively smooth output response frequency characteristic can be obtained from a horn having a mechanical impedance characteristic varying over wide limits. Near the cutoff of both finite and infinite exponential horns, the radiation mechanical resistance at the throat is small and the positive mechanical reactance large. The compliance of the suspension system should be chosen so that its negative mechanical reactance balances the positive mechanical reactance due to the throat. For example, consider a moving coil mechanism coupled to the throat of a horn and fed by a vacuum tube amplifier; the sound power output is the real part of

$$\text{Power} = \left(\frac{e}{|z_{ET}|}\right)^2 z_{EM} \qquad\qquad 7.16$$

where the motional electrical impedance, z_{EM}, in ohms, from equation 7.1, is

$$z_{EM} = \frac{(Bl)^2}{A_D{}^2(r_{AH} + jx_{AH}) + jx_M} \, 10^{-9}$$

where B = air gap flux, in gausses,

l = length of wire in the voice coil, in centimeters,

A_D = area of the diaphragm, in square centimeters,

r_{AH} = acoustical resistance at the throat, in acoustical ohms,

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

x_M = mechanical reactance of the diaphragm, suspension and coil system, in mechanical ohms.

From the voice coil electrical circuit, Fig. 7.1, the total electrical impedance, z_{ET}, in ohms, at e is

$$z_{ET} = r_{ED} + r_{EG} + j\omega L + z_{EM} \qquad\qquad 7.17$$

where r_{ED} = voice coil resistance, in ohms,

r_{EG} = amplifier output resistance, in ohms,

L = inductance of the voice coil, in henries, and

e = amplifier open circuit voltage, in volts.

Equation 7.16 shows that the throat acoustical resistance may vary over wide limits without introducing large variations in the power output. As a specific example, Fig. 7.5 shows the power output as a function of the frequency for a horn, having all dimensions two and one-half times

that of Fig. 5.11B and driven by a vacuum tube having the constants indicated by the caption of Fig. 7.5. Although the variation in acoustical resistance is 6 to 1, the variation power output is only 2 db.

E. *The Effect of the Voice Coil Temperature upon the Efficiency.*[17] —The effect of the temperature of the voice coil upon the efficiency is usually ignored in considerations of the performance of a loudspeaker. In high power loudspeakers, where the temperature of the voice coil becomes quite high, considerable loss in efficiency may result as will be shown in the discussion which follows.

FIG. 7.5. Acoustical power output frequency characteristic of the horn (Fig. 5.11B with all dimensions multiplied by 2½) coupled to a 10½-inch diameter, 10-gram diaphragm driven by a 5-gram aluminum voice coil in a field of 20,000 gausses. Damped electrical resistance of voice coil 20 ohms. Electrical impedance of vacuum tube through a transformer, 35 ohms.

The efficiency, in per cent, of a loudspeaker, when the temperature correction is added, may be expressed.

$$\mu = \frac{r_{EM}}{r_{ED0}(1 + \alpha t) + r_{EM}} \times 100 \qquad 7.18$$

where r_{ED0} = damped electrical resistance of the voice coil at 0° Centigrade,

α = temperature coefficient of resistance, 0.00423 for aluminum and 0.00427 for copper,

t = temperature of the voice coil, in degrees Centigrade, and

r_{EM} = motional electrical resistance of the voice coil.

The efficiency as a function of the temperature for various values of initial efficiency at 0° Centigrade is shown in Fig. 7.6. These characteristics show that the relative loss in efficiency with increase in temperature is considerably greater for a loudspeaker with low efficiency.

F. *The Effect of the Sound Radiation from the Unloaded Side of the Diaphragm upon the Efficiency.*—In the consideration of the efficiency, usually very little cognizance is taken of the sound radiation from the back of the diaphragm of a horn loudspeaker. In view of the large amount of sound that is radiated from the back of the diaphragm, some consideration should be given to the effect of this radiation upon the efficiency. Since this radiation cannot be used, it must be considered as a loss the same as the electrical resistance loss in the voice coil. The loss due to the reactive component of the mechanical impedance is usually small compared to the mechanical reactance of the remainder of the system.

[17] Olson, H. F., *RCA Review*, Vol. 1, No. 4, p. 68, 1937.

FIG. 7.6. The efficiency, in per cent, as a function of the temperature of a voice coil for various values of initial efficiency at 0° Centigrade.
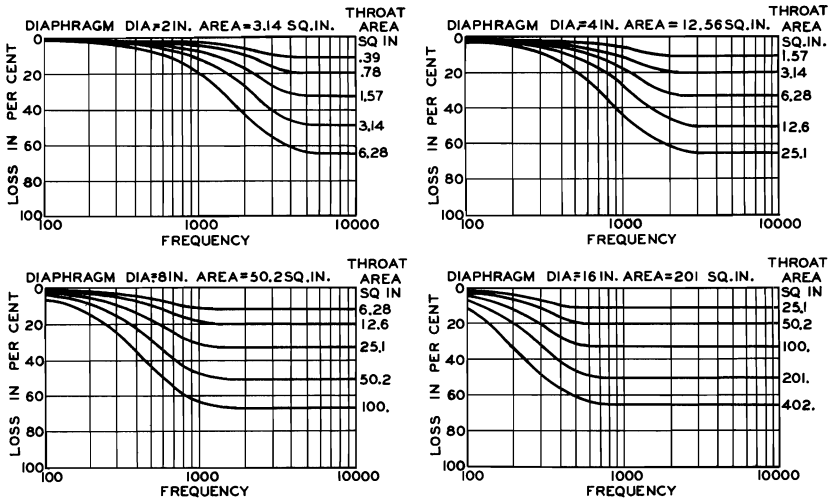
FIG. 7.7. Characteristics depicting the loss in per cent of the total sound radiation, due to the radiation of sound from the back of the diaphragm of a horn loudspeaker for diameters of 2, 4, 8, and 16 inches and throat areas of 2, 1, $\frac{1}{2}$, $\frac{1}{4}$, and $\frac{1}{8}$ times the diaphragm area.

The radiation from the back of the diaphragm may be assumed to be the same as that from a piston in an infinite baffle (see Sec. 5.8 and Fig. 5.2). The percentage of the total radiation which is lost due to the radiation from the back is given by.

$$\text{Efficiency loss} = \frac{r_{MA}}{r_{MA} + r_{MH}} \times 100 \qquad 7.19$$

where r_{MH} = radiation mechanical resistance at the throat of the horn, in mechanical ohms, and

r_{MA} = radiation mechanical resistance of the back of the diaphragm from Sec. 5.8, in mechanical ohms.

The characteristics depicting the loss due to radiation from the back of the diaphragm as a function of the frequency for diaphragm diameters of 2, 4, 8, and 16 inches and various ratios of throat area to diaphragm area are shown in Fig. 7.7. These characteristics show that the loss is indeed quite high.

7.3. Distortion.—In general, the electrical power input to (or the acoustical power output of) a loudspeaker is limited by the generation of spurious harmonics or subharmonics. The limiting factor may be due to air overload, excessive amplitudes where Hooke's law no longer holds, nonlinear elements, variable voice coil air-gap flux product, or nonfundamental vibration modes of the diaphragm. It is the purpose of this section to consider the most common forms of distortion in horn loudspeakers.

A. *Distortion Due to Air Overload in the Horn*.[18,19,20]—A sound wave of large amplitude cannot be propagated in air without a change in the wave form and, as a result, the production of harmonics. If equal positive and negative changes in pressure are impressed upon a mass of air the resultant changes in volume will not be the same. The volume change for an increase in pressure will be less than the volume change for an equal decrease in pressure. From a physical viewpoint the distortion may be said to be due to the nonlinearity of the air.

In the derivation of the fundamental wave equation the second order terms were omitted. If these terms are included the magnitude of the harmonic frequencies may be determined from the differential equation. The subject has been investigated both theoretically and experimentally by a number of investigators. In the case of an exponential horn for constant sound power output, the distortion is proportional to the frequency. Further, the nearer the observation frequency is to the cutoff frequency the smaller the distortion.

The distortion due to nonlinearity of the air is at the present time one of the most important as well as the most troublesome factors in the design of high-efficiency loudspeakers for large outputs. In order to obtain high efficiency, particularly at the higher frequencies, it is necessary to couple the relatively heavy diaphragm to a throat small in area compared to the diaphragm. For a certain allowable distortion the power output is directly proportional to the area of the throat. Obviously, to deliver large sound outputs with small distortion requires a very large throat which may be suitably coupled to a correspondingly large diaphragm or a large number of lightly driven small throat units.

[18] Rocard, *Comtes Rendus*, Vol. 196, p. 161, 1933.
[19] Thuras, Jenkins, and O'Neil, *Jour. Acous. Soc. Amer.*, Vol. 6, No. 3, p. 173, 1935.
[20] Goldstein and McLachlin, *Jour. Acous. Soc. Amer.*, Vol. 6, No. 4, p. 275, 1935.

The second harmonic distortion, at the mouth, in per cent of the fundamental, generated in an exponential horn is given by

$$D = \frac{p^2}{p_1} 100 = \frac{(\gamma + 1)p_{1t}\omega}{\sqrt{2}\gamma p_0 cm} (1 - \epsilon^{-mx/2})100 \qquad 7.20$$

where γ = ratio of specific beats, $\gamma = 1.4$ for air,

p_{1t} = sound pressure at the throat, in dynes per square centimeter,

$\omega = 2\pi f$,

f = frequency, in cycles per second,

m = flare constant of the exponential horn (see Sec. 5.22),

x = length of the horn, in centimeters,

p_0 = atmospheric pressure, in dynes per square centimeter, and

c = velocity of sound, in centimeters per second.

The power[21] which can be transmitted per square centimeter of throat area of an infinite exponential horn as a function of the ratio of the frequency under consideration to the cutoff frequency with the production
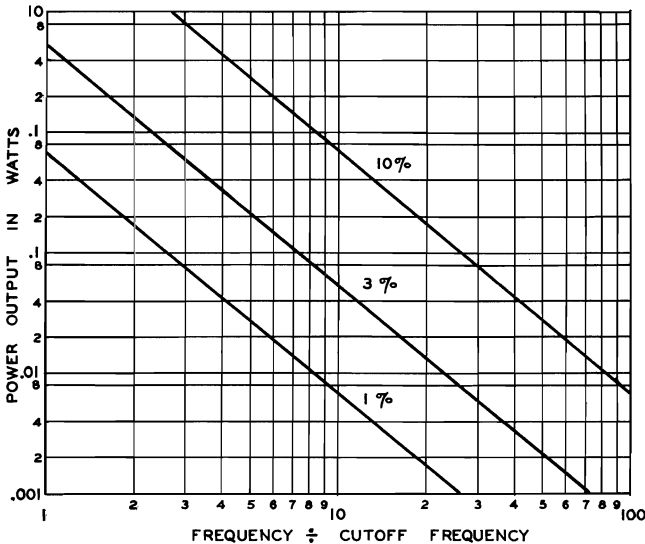
FIG. 7.8. The power output of infinite exponential horns, per square centimeter of throat area, for 1, 3, and 10 per cent distortion, as a function of the ratio of the frequency under consideration to the cutoff frequency.

of 1, 3, and 10 per cent distortion is shown in Fig. 7.8. For the sake of generality the curves shown in Fig. 7.8 refer to an infinite horn. However, the increase in power which may be transmitted by a practical finite

[21] Olson, H. F., *RCA Review*, Vol. 2, No. 2, p. 265, 1937.

horn is only a few per cent greater than that shown in Fig. 7.8, because very little distortion is generated in the large cross-sectional area near the mouth of the horn.

It may be mentioned in passing that the multiple flare horn (see Sec. 5.25) provides a means of decreasing the distortion because the rate of flare is very rapid near the diaphragm and, therefore, the pressures are rapidly reduced with respect to the distance from the diaphragm.

B. *Distortion Due to Variation in Volume of the Air Chamber.*[22]—In general, acoustical, mechanical, and electrical networks are assumed to be invariable; that is, the constants and connections of the network do not vary or change with time. A network which includes a circuit element that varies continuously or discontinuously with time is called a variable network. In some cases the variable elements are assumed to be certain functions of the time; that is, the variations are controlled by outside forces which do not appear in the equations or statement of the problem. In another type of variable circuit element the variation is not an explicit time function, but a function of the current and (its derivatives) which is flowing through the circuit.

An example of the latter type of circuit element in an acoustical system is the air chamber capacitance in a horn loudspeaker. The excursions of the diaphragm change the acoustical capacitance. The acoustical capacitance of the air chamber of Fig. 7.9 is given by

$$C_{A1} = \frac{V}{\rho c^2} = \frac{A(d + x)}{\rho c^2} \qquad 7.21$$

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

c = velocity of sound, in centimeters per second,

V = volume of the air chamber, in cubic centimeters,

A = projected area of the air chamber upon the diaphragm, in square centimeters,

d = distance between the diaphragm and front boundary of the air chamber in the absence of motion, in centimeters, and

x = displacement of the diaphragm, in centimeters.

The acoustical network of the acoustical system shows the effect of the nonlinear element upon the sound power output. In the case of a single frequency the distortion which this element introduces is small, because, for constant sound power output, the amplitude of the diaphragm is inversely proportional to the frequency. At low frequencies where the amplitude of the diaphragm may be so large that the volume of the air chamber becomes alternately zero and two times the normal volume, the acoustical reactance of the acoustical capacitance is very large compared to the acoustical resistance of the horn (see Fig. 7.9). At the high frequencies

[22] Olson, H. F., *RCA Review*, Vol. 2, No. 2, p. 265, 1937.

where the acoustical reactance of the acoustical capacitance is comparable to the acoustical resistance, the amplitude of the diaphragm for the same output is so small that the variation in acoustical capacitance may be neglected (see Fig. 7.9). However, the conditions are different when both a high and a low frequency are impressed upon the same system. Under these conditions considerable change in the acoustical capacitance occurs due to the large amplitudes of the diaphragm for the impressed low frequency. The resultant change in acoustical capacitance introduces a variable element for the impressed high frequency which may have variations in acoustical impedance as large as the impedance of the other elements of the system. The result is shown in Fig. 7.9. When this condition
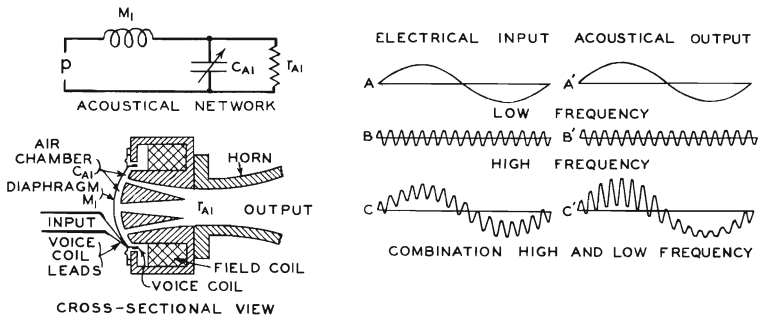
FIG. 7.9. A mechanism with an air chamber coupling the diaphragm to the horn. In the acoustical network: M_1 = the inertance of the diaphragm and voice coil. C_{A1} = the acoustical capacitance of the air chamber. r_{A1} = the acoustical resistance at the throat of the horn. p = the driving pressure. $p = Bli/A$. B = the flux density in the air gap. l = the length of the voice coil conductor. i = the current in the voice coil. A = the area of the diaphragm. The variation in volume of the air chamber introduces a nonlinear element in the form of the acoustical capacitance C_{A1}. The wave shapes of the electrical input and acoustical output for a low, high, and a combination of a high and a low frequency illustrates the effect of the nonlinear element upon the acoustical output.

obtains, particularly with close spacing between the diaphragm and the front boundary of the air chamber, the distortion may be tremendous. Physically the low frequency modulates the high frequency.

In the above discussion the air chamber is assumed to be a pure acoustical capacitance. This assumption is not correct at the higher frequencies where the dimensions of the air chamber are comparable to the wavelength. Regardless of the form of the acoustical impedance, it is, nevertheless, a function of the spacing between the diaphragm and the air chamber and is therefore a nonlinear element.

C. *Distortion Due to the Diaphragm Suspension System.*[23]—The outside suspension is another example of a variable circuit element in a mechanical

[23] Olson, H. F., *RCA Review*, Vol. 2, No. 2, p. 265, 1937.

system. In certain types or, as a matter of fact, for unlimited amplitudes in all types of suspension systems the stiffness is not a constant, but a function of the amplitude and, in general, increase for the larger amplitudes (see Sec. 6.26).

In the case of a horn loudspeaker the amplitude of the diaphragm for constant sound power output is inversely proportional to the frequency. Furthermore, the mechanical impedance of the suspension system becomes comparable to the other mechanical impedances in the system at the lower frequencies. Consequently, the greatest distortion due to the suspension system will occur at the low-frequency end of the working range.

The mechanical circuit of the mechanical system, Fig. 7.10, shows the effect of the nonlinear element. When the stiffness of the suspension system increases with amplitude, the third harmonic is the preponderant distortion (see Sec. 6.26C). The wave shape under these conditions is shown in Fig. 7.10. A distortion frequency characteristic of a diaphragm coupled to a large throat horn is shown in Fig. 7.10.
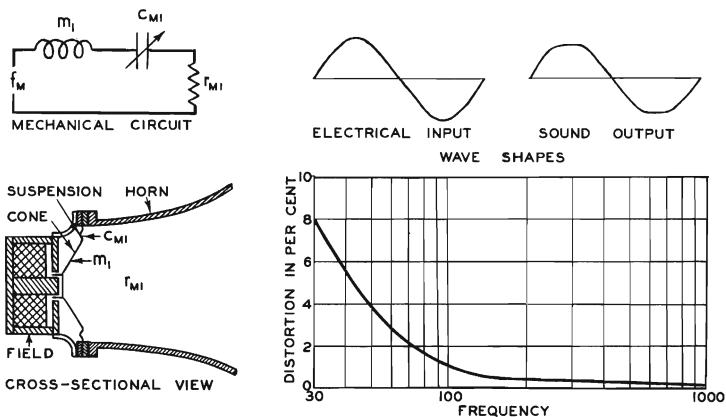
FIG. 7.10. Mechanism having a diaphragm with a nonlinear suspension system. In the mechanical circuit: m_1 = the mass of the diaphragm and voice coil. C_{M1} = the compliance of the diaphragm suspension system. r_{M1} = the mechanical resistance at the throat of the horn. f_M = the driving force. The mechanical circuit of the vibrating system and the wave shapes indicate the effect of the nonlinear element. The graph shows a typical distortion characteristic obtained on an 8-inch diameter diaphragm coupled to a large throat horn and delivering an acoustical power output of 3 watts.

D. *Distortion Due to a Nonuniform Magnetic Field in the Air Gap.*— Inhomogeneity of the flux density through which the voice coil moves is another source of distortion. The result is that the driving force does not correspond to the voltage developed by the generator in the electrical system.

The force, in dynes, developed by the interaction of the current in the voice coil and the magnetic field is

$$f = Bli \qquad\qquad 7.22$$

where B = flux density, in gausses,

l = length of the voice coil conductor, in centimeters, and

i = current, in abamperes.

Equation 7.22 shows that the force is directly proportional to the current if Bl is a constant. If the Bl product varies with the position of the voice coil the force will not be proportional to the current and distortion will result. A typical flux distribution in an air gap is shown in Fig. 6.66. A consideration of the flux distribution shows that the Bl product will be practically a constant if the voice coil is made longer than the air gap because, as the coil moves into the weaker tufting field on one side, it moves into a stronger field on the other side. From the standpoint of efficiency at the higher frequencies this method is not particularly desirable because part of the voice coil is in a weak field. This type of distortion can also be eliminated by making the air gap of sufficient axial length so that the voice coil remains at all times in a uniform field as shown in Fig. 6.66C. The latter method is usually used for high-frequency loudspeakers of high efficiency (also see Sec. 6.26E).

E. *Subharmonic Distortion.*—The distortions referred to above have been concerned with higher harmonics, that is, multiples of the fundamental. It has been shown in Sec. 6.26C that subharmonics are generated in vibrating systems with nonlinear elements. The existence of subharmonics in direct radiator loudspeakers is well known. However, in horn loudspeakers the diaphragms are relatively small and quite rigid. Consequently, the conditions for the production of subharmonics are not particularly favorable.

F. *Power Handling Capacity and the Voice Coil Temperature.*[24]—The maximum allowable distortion may determine the power rating for the loudspeaker. However, in certain loudspeakers the maximum allowable temperature of the voice coil determines the power rating. This is particularly true of high-frequency loudspeakers.

By making the efficiency a maximum, the dissipation in, and the resulting temperature of, the voice coil for a certain acoustical output will be a minimum. Practically all the heat energy developed in the voice coil is transmitted across the thin air film between the voice coil and the pole pieces and from the pole pieces to the field structure and thence into the surrounding air. In this heat circuit practically all the drop in temperature occurs in the thin air film. The temperature of the voice coil approaches the temperature of the pole pieces as the thickness of the air film is decreased. The temperature rises as a function of the power dissipated in the voice coil for various clearances between the voice coil and pole pieces.

[24] Olson, H. F., *RCA Review*, Vol. 2, No. 2, p. 265, 1937.

This is shown in Fig. 7.11. These results are obtained for no motion of the voice coil. When motion occurs, the thermal impedance of the air film is reduced and the temperature of the voice coil is diminished.
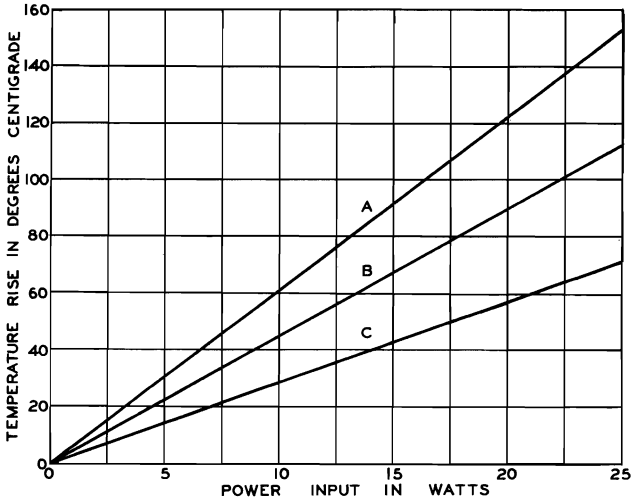
FIG. 7.11. The temperature rise as a function of the power delivered to a voice coil for air-gap clearances as follows: A. 0.021 inch, B. 0.015 inch, C. 0.009 inch. Coil $1\frac{1}{2}$ inches in diameter, and 0.25 inch in length.

G. *Power Handling Capacity and the Amplitude of the Diaphragm.*[25]— The maximum allowable amplitude of the diaphragm is another factor which may determine the maximum allowable acoustical power output. The acoustical power output, in watts, of a horn loudspeaker in which the diaphragm is terminated in an acoustical resistance is

$$P = \frac{\rho c (2\pi f)^2 d^2 A_D^2}{2A_H} 10^{-7} \qquad 7.23$$

where ρ = density, in grams per cubic centimeter,

   c = velocity of sound, in centimeters per second,

   f = frequency, in cycles per second,

   d = maximum amplitude from its mean position, in centimeters,

   A_D = area of the diaphragm, in square centimeters, and

   A_H = area of the throat of the horn, in square centimeters.

The amplitude of various diameter diaphragms coupled to a horn throat of 1 square inch for 1 acoustical watt output is shown in Fig. 7.12.

[25] Massa, F., *RCA Review*, Vol. 3, No. 2, p. 196, 1938.

FIG. 7.12. The peak amplitude frequency characteristics of vibrating pistons of various areas in square inches, coupled to the throat of a horn having an area of 1 square inch, for 1 watt output.

7.4. Horn Loudspeaker Systems.—A. *Single-Horn, Single-Channel System.*—The single-horn, single-channel system consists of a single horn driven by a single diaphragm. A diaphragm coupled to an exponential horn constitutes the simplest and most wide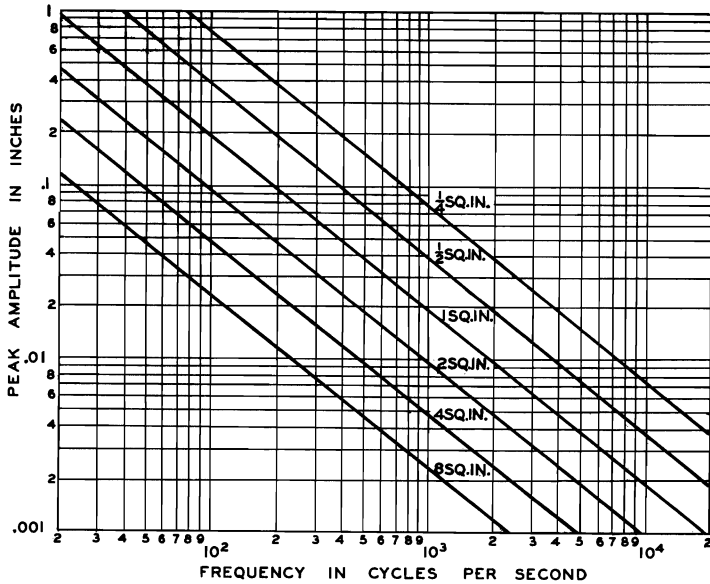ly used system. The efficiency frequency characteristic of a simple exponential horn coupled to a diaphragm and coil having a mass ratio of 2 operating in a field of 22,000 gausses is shown in Fig. 7.13. Two efficiency frequency characteristics are shown with initial efficiencies of 80 per cent and 50 per cent. Although it is possible to obtain reasonably high efficiency over a wide frequency range with a single horn coupled to a diaphragm, the efficiency can be increased by employing a multiple flare horn.

To obtain maximum efficiency in a horn loudspeaker at any frequency, the effective mechanical reactance of the entire system should be equal to the effective mechanical resistance. This, in general, means that to obtain maximum efficiency the throat mechanical resistance of the horn should be proportional to the frequency, since the mechanical reactance is primarily mass reactance and, therefore, proportional to the frequency. The surge mechanical resistance of the exponential horn is independent of the frequency. However, the acoustical resistance[26] of a multiple flare horn increases with frequency as shown in Sec. 5.26. Therefore, the efficiency is higher over a wide range than in the case of a horn with a single rate of flare. The effi-

[26] Olson, H. F., *Jour. Soc. Mot. Pict. Eng.*, Vol. 30, No. 5, p. 511, 1938.
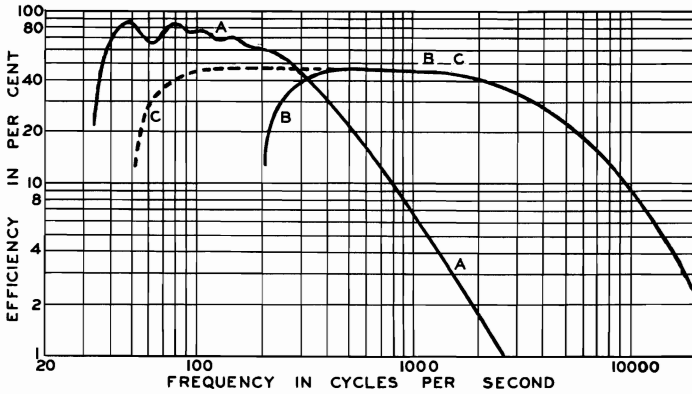
FIG. 7.13. A. Efficiency frequency characteristic of a horn loudspeaker employing the horn of Fig. 5.11D with the dimensions multiplied by three and driven by 4 cones, 12 inches in diameter, with 5-gram copper voice coils operating in a field of 14,000 gausses. B. Efficiency frequency characteristic of a horn loudspeaker employing the horn of Fig. 5.10D with the dimensions multiplied by one-half and driven by a diaphragm and an aluminum voice coil having a mass ratio of two to one operating in a field of 22,000 gausses. C. Same as B except that the horn dimensions of Fig. 5.10D are multiplied by two.

ciency frequency characteristic of the multiple flare horn described in Sec. 5.30 coupled to a diaphragm and coil having a mass ratio of 2 operating in a field of 22,000 gausses is shown in Fig. 7.14. This efficiency frequency characteristic is only a few per cent below the ultimate efficiency frequency characteristic obtained from the enevelope of the family of characteristics shown in Fig. 7.3.
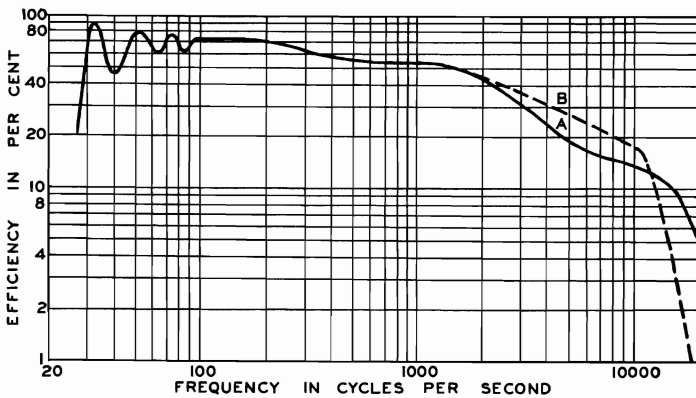
FIG. 7.14. Efficiency frequency characteristic of a diaphragm coupled to the horn of Fig. 5.13 and driven by an aluminum voice coil of one-half the diaphragm mass in a field of 22,000 gausses. A. Without air chamber. B. With air chamber.

The two preceding horn loudspeakers are suitable for high quality repro-
duction of speech and music.   For certain types of announce installations
it is desirable to project intelligible speech over very great distances (1
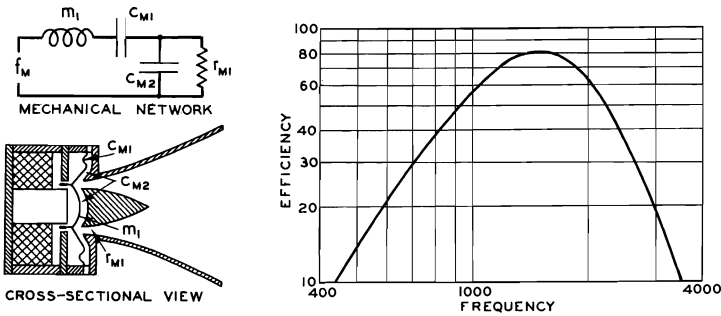
FIG. 7.15.   Cross-sectional view and mechanical circuit of a loudspeaker of
2 degrees of freedom.   In the mechanical network: m_1 = the mass of the
diaphragm and voice coil. C_{M1} = the compliance of the diaphragm
suspension system. C_{M2} = the compliance of the air chamber. r_{M1} =
the mechanical resistance at the throat of the horn. f_M = the driving
force.   The graph shows the efficiency frequency characteristic.
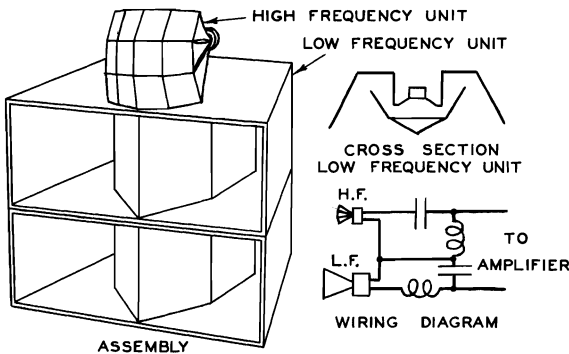
FIG. 7.16.   A two-channel, theater loudspeaker system
consisting of a folded low-frequency horn unit and a
multicellular horn high-frequency unit.   The wiring
diagram shows the electrical filter used to allocate the
power, as a function of the frequency, to the two units.

to 2 miles) under all manner of conditions.   This requires acoustical out-
puts of the order of from 500 to 1000 watts.   The characteristics of Fig.
7.12 show that it is not practical to build a horn loudspeaker of this capacity
for the reproduction of the lower frequencies.   A cross-sectional view of a
high power announce loudspeaker and the simplified mechanical network
is shown in Fig. 7.15.   The mechanical network shows a system of two
degrees of freedom.   The compliance of the suspension system and the

compliance of the air chamber are chosen so that very high efficiency is obtained over the range required for intelligible speech.   A typical efficiency frequency characteristic of this type of loudspeaker suitable for acoustical outputs of 500 to 1000 watts is shown in Fig. 7.15.   Due to the large audio-power amplifier requirements, high loudspeaker efficiency is an extremely important economic factor.

B. *Multiple-Horn, Multiple-Channel System.*—The two-channel or "two-way" system,[27],[28] is the most common example of a multichannel system.   This loudspeaker, Fig. 7.16, consists of a low-frequency folded

VOICE COIL CIRCUIT   MECHANICAL NETWORK

SECTION A–A'   FRONT VIEW
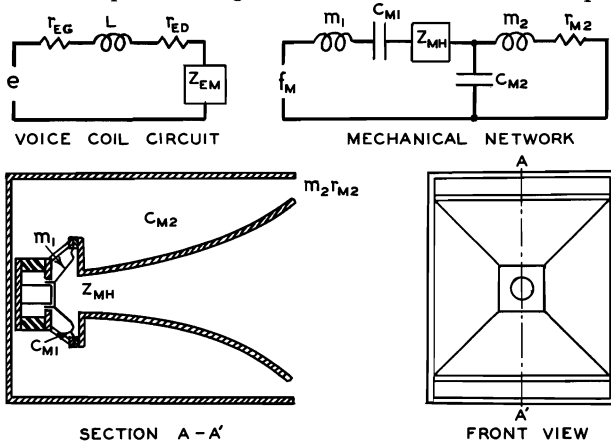
FIG. 7.17.   Combination horn and phase inverter low-frequency loudspeaker.   In the voice coil circuit, $e =$ the internal voltage of the electrical generator. $r_{EG} =$ the electrical resistance of the electrical generator. L and $r_{ED} =$ the damped inductance and electrical resistance of the voice coil. $z_{EM} =$ the electrical motional impedance.   In the mechanical network: $m_1 =$ the mass of the diaphragm. $C_{M1} =$ the compliance of the diaphragm suspension system. $z_{MH} =$ the mechanical impedance at the throat of the horn. $C_{M2} =$ the compliance of the air chamber. m_2 and $r_{M2} =$ the mass and mechanical resistance of the port opening. $f_M =$ the driving force.

horn unit for reproduction from 40 to 300 cycles and a multicellular horn unit for reproduction from 300 to 8000 cycles.

In order to minimize time delay and phase distortion due to a large path length difference between the low- and high-frequency horns, the effective length of the low- and high-frequency horns must be practically the same. The difference in path length in the system shown in Fig. 7.16 is made relatively small by employing a short folded horn coupled to a large diameter dynamic speaker mechanism.   A further reduction in path length between

[27] Wente and Thuras, *Jour. A.I.E.E.*, Vol. 53, No. 1, p. 17, 1934.

[28] Hilliard, J. K., *Tech. Bul. Acad. Res. Coun.*, March, 1936.

a short, straight axis high-frequency horn may be obtained by shifting the high-frequency unit backwards.

The high-frequency horn consists of a cluster of relatively small horns coupled to a common throat, Fig. 7.16. The directional characteristics of this type of loudspeaker were discussed in Sec. 2.20. Fig. 7.16 shows a 12-cell high-frequency unit. The throat is coupled to one or more mechanisms depending upon the power requirements.

An electric filter or dividing network is used to allocate the power to the

FRONT VIEW
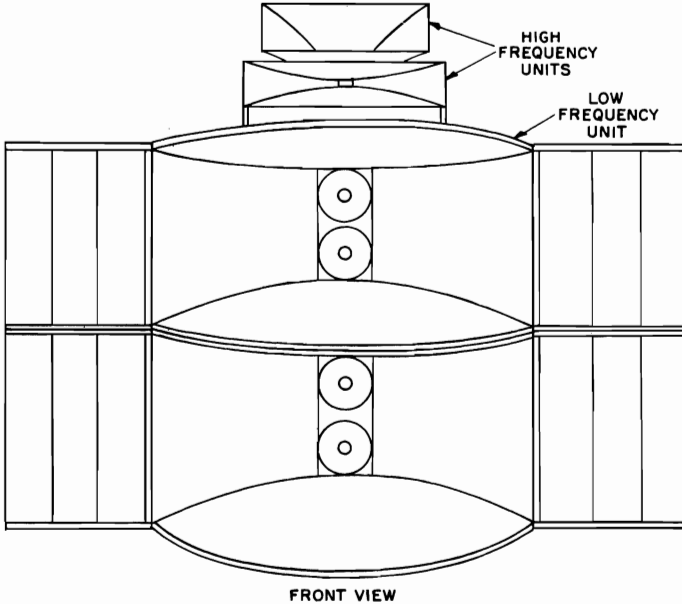
FIG. 7.18. A two-channel, theater loudspeaker system consisting of the combination of a straight axis horn and phase inverter, low-frequency unit and a two-layer, straight-side horn, high-frequency unit. (After Volkmann.)

high- and low-frequency units. The filter introduces phase shift as well as a loss in power of 2 db or more.

The efficiency frequency characteristics of the high- and low-frequency units of this loudspeaker without the filter are shown in Fig. 7.13, characteristics B. and A.

The low-frequency loudspeaker in the system depicted in Fig. 7.16 employs a short folded horn. Although the horn is short, there is still a path difference between the low- and high-frequency horns of about 1 wavelength at the overlap frequency of 300 cycles. The path difference can be obviated by the use of a high- and low-frequency horn of the same length. In order to conserve space the over-all depth must not be too

great. Under these conditions the flare cutoff at the low-frequency horn will be about 80 cycles. The radiation mechanical resistance can be increased and the output in the frequency range below the flare cutoff maintained by the use of a phase inverter system in combination with the horn as shown in Fig. 7.17. The action of the system may be determined from the mechanical network of Fig. 7.17. By a suitable choice of constants uniform response may be maintained in the low-frequency range down to 40 cycles.

A theater loudspeaker system[29] employing a low-frequency loudspeaker of the type shown in Fig. 7.17 is shown in Fig. 7.18. The low- and high-

FRONT VIEW
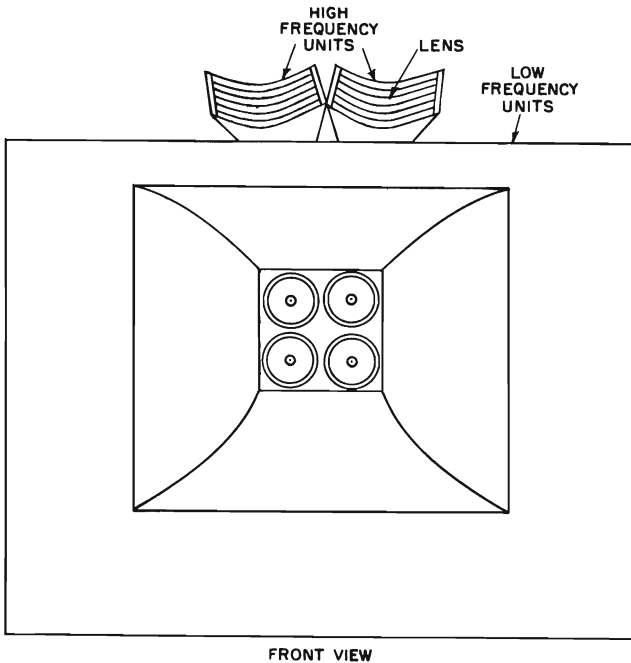
FIG. 7.19. A two-channel, theater loudspeaker system consisting of a straight-axis horn, low-frequency unit, and horn and lens high-frequency units. (After Frayne and Locanthi.)

frequency horns are of the type with straight sides and, therefore, exhibit a reasonably uniform directivity pattern in the horizontal plane. See Sec. 2.19C. This design makes it possible to obtain the directional characteristics of cellular horns without the complex construction. The low- and high-frequency horns are of the same length which obviates the transient distortion inherent in two channel systems in which there is a path length difference between the low and high frequency. This feature also simplifies the problem of obtaining uniform directivity in the overlap frequency region.

[29] Volkmann, J., Unpublished Report.

A theater loudspeaker system[30] employing a straight axis, low-frequency horn and high-frequency horns equipped with diverging acoustic lenses is shown in Fig. 7.19. The throat area of the low-frequency horn is the same as the total area of the four cone loudspeaker units. The high-frequency horns are equipped with slant plate-type diverging lenses which broadens the directivity pattern. In addition, by suitable design of the lens and the horn it is possible to obtain reasonably uniform coverage in the theater.

A folded horn loudspeaker[31] for operation in the corner of the room for the reproduction of the low-frequency range is shown in Fig. 7.20. The horn

FIG. 7.20. Perspective and sectional views of a folded horn for operation in the corner of the room. (After Klipsch.)

[30] Frayne and Locanthi, *Jour. Soc. Mot. Pic. Tel. Eng.*, Vol. 63, No. 3, p. 82, 1954.
[31] Klipsch, Paul W., *Trans. IRE*, Prof. Group on Audio, Vol. AU-1, No. 3, p. 16, 1953.

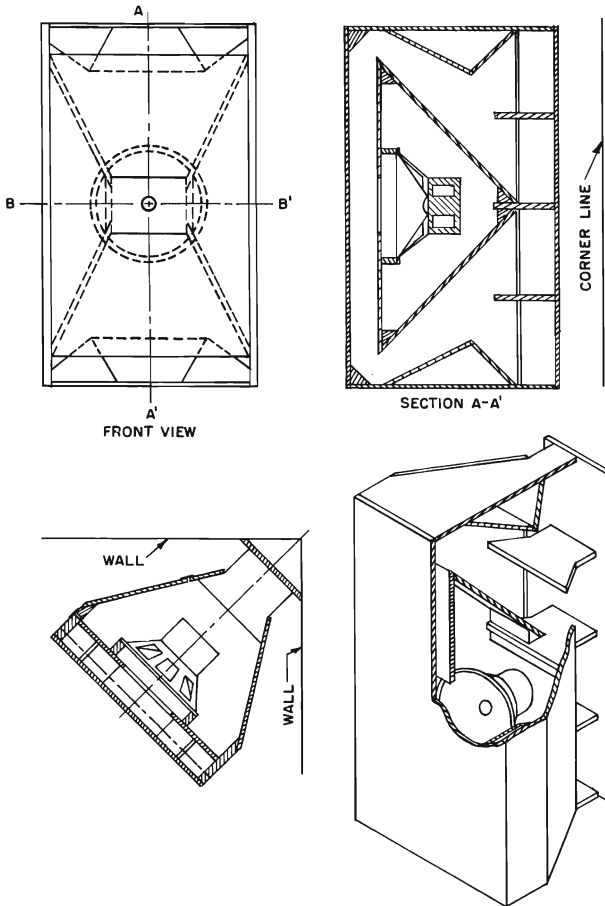is of the folded type. The acoustical radiation resistance presented to a loudspeaker operating in the corner of a room at the intersection of the floor and two walls is two times the acoustical radiation resistance presented to a loudspeaker operating along a wall of a room at the intersection of the wall and floor. See Sec. 2.2. This increased radiation can only be realized at the very low-frequency portion of the frequency range.

Some wide frequency range systems divide the frequency range into three sections, namely, low, mid, and high frequency. Direct radiator, dynamic and electrostatic, and horn loudspeaker units are used to cover the high-frequency range. A high-frequency horn loudspeaker for the frequency range above 10,000 is shown in Fig. 7.21. The diaphragms are of the order
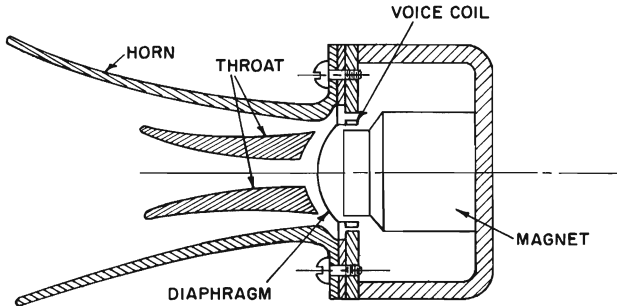
FIG. 7.21. Sectional view of a high-frequency horn loudspeaker.

of $\frac{3}{4}$ inch in diameter. The horns are about two inches in length. The mouth area of the horn is about three square inches.

C. *Compound Horn Loudspeaker*.[32]—The compound horn loudspeaker consists of a single diaphragm mechanism with one side of the diaphragm coupled to a straight axis horn and the other side coupled to a long folded horn, Fig. 7.22. The equivalent of the system is shown in Fig. 7.22. The functional acoustical network of the vibrating system is also shown in Fig. 7.22. At the low frequencies the acoustical reactance of the acoustical capacitance, C_{A2}, is large compared to the throat acoustical impedance, z_{A2}, of the low-frequency horn and sound radiation issues from the low-frequency horn. At the high frequencies the acoustical reactance of the acoustical capacitance, C_{A2}, is small compared to the acoustical impedances, z_{A1} and z_{A2}, and, therefore, shuts out the low-frequency horn and radiation issues from the high-frequency horn. In the mid-range, radiation issues from both horns. The response frequency characteristic, Fig. 7.22, shows the response range of the two horns. The throats of the two horns may be chosen so that the efficiency characteristic of this loudspeaker will be the same as that of the two-channel system discussed in the preceding section. However, the power handling capacity is somewhat smaller because the size of the diaphragm must be a compromise between high-and low-frequency requirements.

[32] Olson and Massa, *Jour. Acous. Soc. Amer.*, Vol. 8, No. 1, p. 48, 1936.

FIG. 7.22.   Cross-sectional view of a compound horn loudspeaker, the developed equivalent of the high- and low-frequency horns, and the acoustical network of the acoustical system.   In the acoustical network: M = the inertance of the diaphragm. C_{A1} = the acoustical capacitan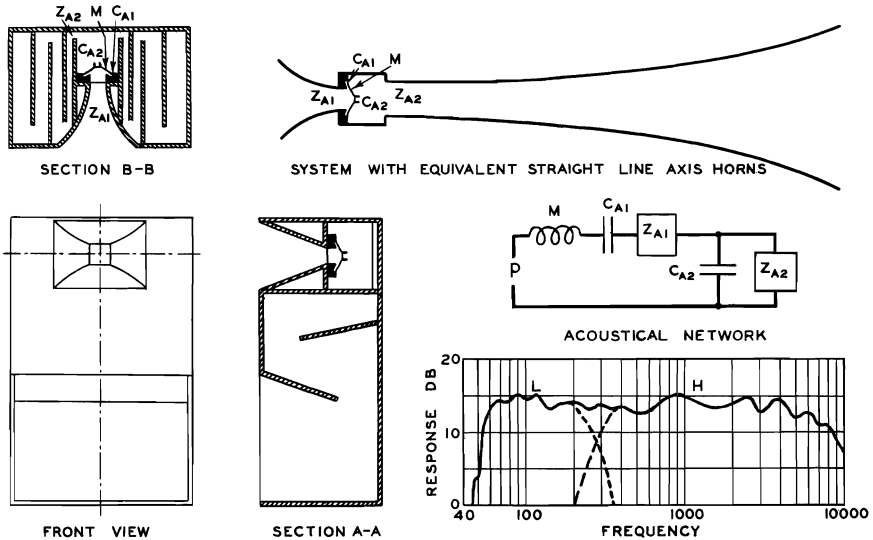ce of the diaphragm suspension system. z_{A1} = the acoustical impedance at the throat of the small horn. z_{A2} = the acoustical impedance at the throat of the large horn. C_{A2} = the acoustical capacitance of the chamber behind the diaphragm. P = the driving pressure. $p = Bli/A$. B = the flux density. l = the length of the conductor in the voice coil. i = the current in the voice coil. A = the area of the diaphragm.   The sections A-A and B-B refer to the horizontal and vertical cross sections of the front view.   The graph shows the frequency ranges of the high-frequency and low-frequency horns and the over-all pressure response frequency characteristic.

D. *Multiple-Horn, Single-Channel System.*—The multiple-horn, single-channel system consists of a large number of multiple flare horns, each driven by a diaphragm, Fig. 7.23.   A comparison of the efficiency characteristics of a multiflare horn loudspeaker, Fig. 7.23, with a multichannel system, Fig. 7.23, shows that the efficiencies are of the same order.   The multiple-horn, single-channel system eliminates many of the following disadvantages of the multichannel system: the phase difference due to the difference in path length between the two channels, the phase difference and power loss in the filters and dividing network, the nonuniform directional characteristics due to the small size of the high-frequency unit, and the distortion in the relatively small throat of the high-frequency horn. The space required for the single-channel system is greater than that for the multichannel system.   However, from a technical standpoint the single-channel system is far superior to the multiple-channel system.

A multiple-horn, single-channel system loudspeaker suitable for high-power announce systems is shown in Fig. 7.24.   This loudspeaker performs

the same function as the system shown in Fig. 7.15. The stresses in the diaphragm and the voice coil system are reduced by the use of a number of smaller units as contrasted to a single large unit. The possibility of failure of the system is reduced by the use of a multi-unit driving system. The use
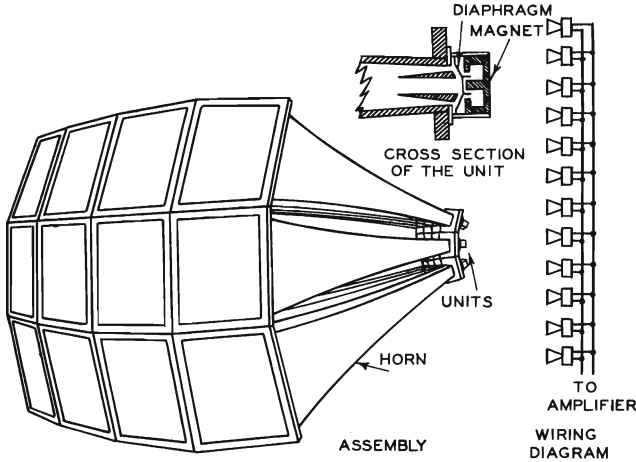
FIG. 7.23. A multiple-horn, single-channel, wide frequency range, loudspeaker system consisting of a cluster of multiflare horns, each coupled to a small diaphragm.
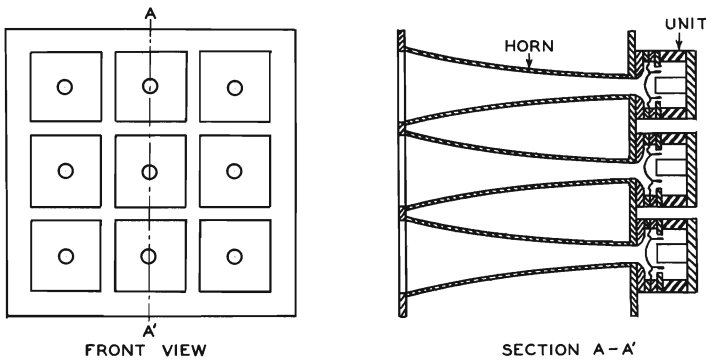
FIG. 7.24. A multiple-horn, single-channel system for high-power announce systems.

of a multiple-horn system makes it possible to obtain a greater variety of directional patterns than is possible in the single-horn system of Fig. 7.15.

E. *Horn Loudspeaker for Personal Radio Receivers.*[33]—The term personal radio receiver is used to designate a complete radio receiver with self-contained power supply, and of such physical dimensions that it can be easily

[33] Olson, Bleazey, Preston, and Hackley, *RCA Review*, Vol. 11, No. 1, p. 80, 1950.

carried by hand or in the pocket. The performance and compactness of personal radio receivers are limited by the efficiency with which electrical power is converted into sound power by the loudspeaker. Since the electrical power output is limited in the personal receiver, the efficiency of the loudspeaker is an important factor. The specifications indicate the use of a horn loudspeaker.

From the considerations in the preceding sections it appears that a combination horn and phase inverter loudspeaker would be the logical solution for a high-efficiency loudspeaker for personal receivers.

Perspective and sectional views and the mechanical network of a combination horn and direct radiator loudspeaker for personal radio receivers are shown in Fig. 7.25. The lid with two sides and the case form a horn

MECHANICAL NETWORK

PERSPECTIVE VIEW
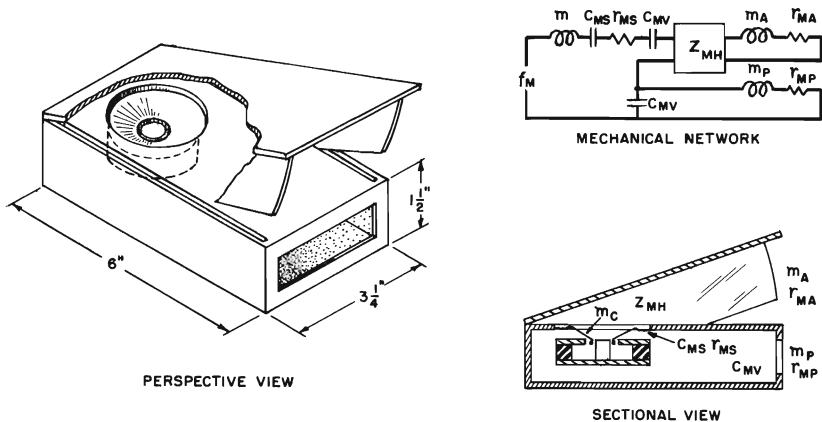
SECTIONAL VIEW

FIG. 7.25. Perspective and sectional views and mechanical network of a horn loudspeaker for a personal radio receiver. In the mechanical network: m_C = the mass of the cone and voice coil. r_{MS} and C_{MS} = the mechanical resistance and compliance of the suspension for the cone. C_{MV} = the compliance of the air in the case volume. z_{MH} = the quadripole representing the horn. m_A and r_{MA} = the mass and mechanical resistance of the air load on the mouth of the horn. m_p and r_{MP} = the mass and mechanical resistance of the air in the port and the air load upon the port. f_m = the driving force developed in the voice coil. (After Olson, Bleazey, and Preston.)

when the lid is open. With the horn collapsed, that is, the lid placed against the case, the dimensions are the same as those of a direct radiator loudspeaker system. The experimentally determined response frequency characteristic is shown in Fig. 7.26. The efficiency is of the order of 25 per cent. It is possible to obtain a sound level of 84 decibels at a distance of three feet with 10 milliwatts.

F. *Folded Horns.*—There are innumerable ways of folding or curling a horn. The different types of folded horns are shown in Figs. 7.16 and 7.22. The principal purpose of folding or curling a horn is to use the volume occupied by the horn more efficiently. Three more different types of folding

are shown in Fig. 7.27. A simple folded horn is shown in Fig. 7.27A. A folded horn with a ring-shaped mouth is shown in Fig. 7.27B. The directional characteristics of a ring-shaped mouth are sharper than those of the rectangular or circular shapes having equivalent areas (see Secs. 2.9 and 2.10). The horn shown in Fig. 7.27C is used for sending out radiation
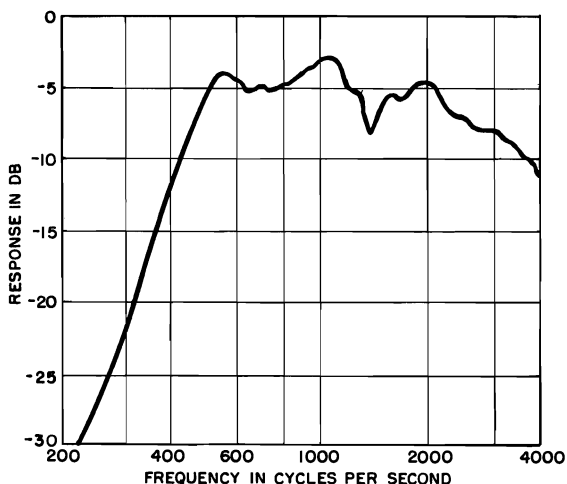
FIG. 7.26. Response frequency characteristic of the personal radio loudspeaker shown in Fig. 7.25.
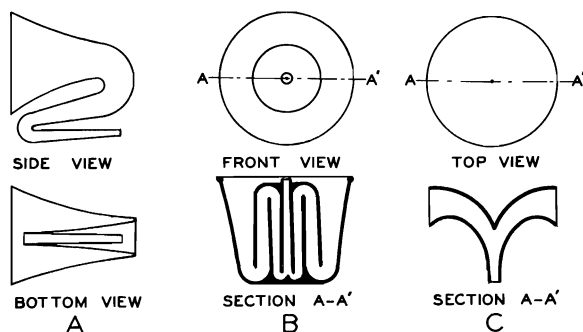
FIG. 7.27. Folded horns.

over 360° normal to the axis. It is customary to mount this loudspeaker on a pole.

The high-frequency response is usually attenuated in a folded horn due to destructive interference incurred by the different path lengths of the sounds traversing the bends. In order to eliminate destructive interference the same phase should exist over any plane normal to the axis. This condition is practically satisfied providing the radial dimensions at any bend

are a fraction of the wavelength. Wide range reproduction of sound requires a large-mouth horn for efficient reproduction of low-frequency sounds and small dimensions at the bends of a folded horn for efficient reproduction of high-frequency sounds. Obviously, it is practically impossible to incorporate both of these features into a single folded horn. It is true that folded horns have been used for years, but, in general, the response at either or both the low- or high-frequency ranges has been attenuated.

G. *Horn Loudspeaker Mechanisms.*—The diaphragm, voice coil, magnet structure, and air chamber of a horn loudspeaker mechanism may be built in a wide variety of ways. The variations in path length from any part of the diaphragm to the horn throat should be less than a quarter wavelength in order to eliminate destructive interference in the air chamber. Several
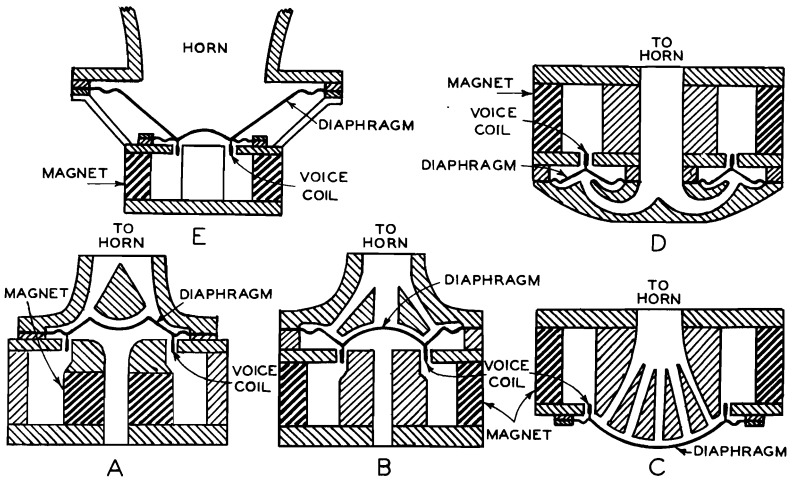
FIG. 7.28. Horn loudspeaker driving mechanisms. Mechanisms A, B, C, and D depict various types of air chambers and diaphragms for coupling to a small throat horn. Mechanism E depicts a large diaphragm coupled to a large throat horn.

different methods for reducing interference in the air chamber are shown in Fig. 7.28A, B, C, and D and Figs. 7.1, 7.9, 7.15, and 7.21. These expedients are necessary for efficient reproduction at the high-frequency portion of the audio range where the wavelength is relatively small. For the low-frequency portion of the audio-frequency range a large-throat horn may be coupled to a large diaphragm, as shown in Fig. 7.28E, without incurring any loss due to interference, notwithstanding the large size, because the dimensions are small compared to the wavelength.

H. *Diaphragms and Voice Coils.*—The diaphragms or cones of horn loudspeaker mechanisms are made of aluminum alloys, molded bakelite with various bases, molded styrol, fiber, paper, and felted paper. Typical diaphragm shapes are shown in Figs. 7.1, 7.9, 7.10, 7.15, 7.23, 7.24 and 7.28.

Round, square, and ribbon wire voice coil conductors are used as shown in Fig. 6.77.

I. *Field Structures.*—Permanent magnet and electromagnetic field structures used in horn loudspeaker mechanisms are shown in Figs. 7.1, 7.9, 7.10, 7.15, 7.23, 7.24, 7.28, and 6.79. In general, it is customary to use higher flux densities in the gap in horn loudspeakers than in direct radiator loudspeakers. Soft iron may be used for the pole tips for flux densities up to 20,000 gausses (See Fig. 6.82). For flux densities from 20,000 to 23,000 gausses, a special alloy, Permandur[34] (see Fig. 6.82), is employed for the pole tip material in order to obtain these high densities with tolerable efficiency.

J. *Horn Walls. Vibration and Absorption.*[35]—In the theoretical analysis carried out in this chapter it has been assumed that the horn walls are rigid and nonabsorbing. In the case of certain materials such as wood, paper, and fiber the absorption of sound by walls of the horn may introduce an attenuation of several decibels. The absorption may be reduced by the application of lacquers and varnishes. The attenuation in metallic horns due to dissipation is negligible. The vibration of the walls of the horn distorts the response frequency characteristic and introduces "hangover" and reverberation. The response to transients is usually poor when the walls of the horn vibrate. This vibration may be reduced by increasing the thickness of the walls and by suitable bracing. The vibrations and ring in metallic horns may be reduced by coating the outside of the horn with deadening material such as asphalt or pitch compounds.

7.5. Throttled Air Flow Loudspeaker.—A throttled air flow loudspeaker consists of a valve mechanism, actuated by the electrical signal, which modulates a steady air stream so that the undulations in the throttled air stream correspond to the variations in the electrical input, see Fig. 7.29. The throttled air stream is usually coupled to a horn to improve the efficiency of the system. In order to obtain a constant relationship between the electrical input and the acoustical output as a function of the frequency, the ratio of the volume current to the applied current must be independent of the frequency. This means that the ratio of the amplitude of the valve to the current must be independent of the frequency. This is, in general, difficult to accomplish in the high-frequency region in view of the fact that the valve mechanism must be stiffness controlled. This in turn means that the mass of the valve must be small and at the same time be sufficiently rugged to withstand the steady air pressure. The outstanding advantage of this system is the large acoustical output which can be obtained for a small electrical input. Efficiencies of more than 100 per cent can be realized if the ratio of acoustical output to the electrical input is considered. However, in addition to the electrical power which must be supplied to the throttling mechanism, there is the power that must be furnished in supplying the steady stream of air. If this mechanical power is added

[34] Elmen, G. W., *Bell Syst. Tech. Jour.*, Vol. 15, No. 1, p. 113, 1936.
[35] Phelps, W. D., *Jour. Acous. Soc. Amer.*, Vol. 12, No. 1, p. 68, 1940.

to electrical input, the efficiency will, of course, be considerably less than 100 per cent.

7.6. Ionophone Loudspeaker.[36]—The ionophone loudspeaker consists of an audio-modulated corona discharge coupled to a horn. Fig. 7.30. The corona is produced in a specially designed quartz envelope. The corona
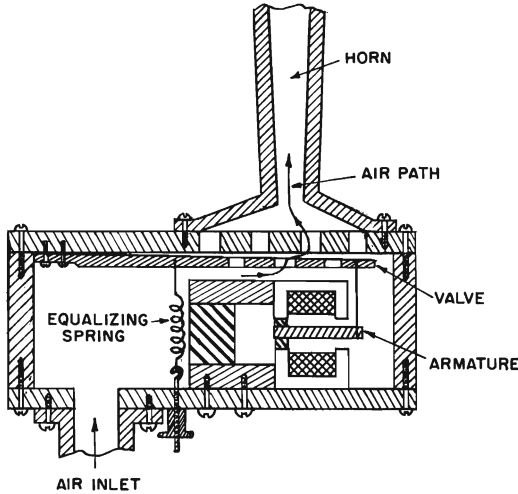
FIG. 7.29. Sectional view of a throttled air flow loudspeaker.
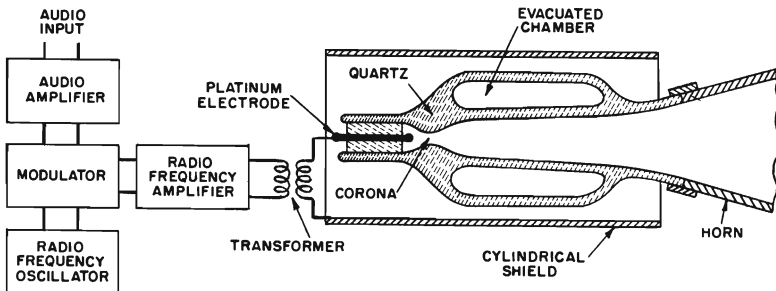
FIG. 7.30. Schematic and sectional view of an audio modulated corona discharge loudspeaker. (After Klein.)

is maintained by a radio-frequency power from a high power voltage amplifier. One terminal of the amplifier is coupled to the platinum electrode located in the quartz envelope and the other terminal is connected to the cylinder around the quartz envelope. An audio-modulated high-frequency signal is coupled to a radio-frequency oscillator and an audio signal. The

[36] Klein, S., *Acustika*, Vol. 4, No. 1, p. 77, 1954.

intensity of the corona varies in accordance with the amplitude of the audio signal with the result that the air expands and contracts in corresponding manner. A sound wave is thus produced in the throat of the horn. The sound generator is of the constant amplitude type, that is, for constant electrical input the amplitude is independent of the frequency. When this type of generator is coupled to a horn, the output will be proportional to the frequency. For practical systems, the amplitude is limited. This limitation restricts the use of this loudspeaker to the upper portion of the audio-frequency range.

8

MICROPHONES

8.1. Introduction.—A microphone is an electroacoustic transducer actuated by energy in an acoustical system and delivering energy to an electrical system, the wave form in the electrical system being substantially equivalent to that in the acoustical system. A pressure microphone is a microphone in which the electrical response is caused by 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 a 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.

Microphones may also be classified as directional or nondirectional. The particular configuration of the acoustical elements which constitute the vibrating system determines the directional properties of the microphone. It is the purpose of this chapter to consider the microphones in most common use today from the standpoint of the above classifications.

8.2. Pressure Microphones.—A. *Carbon Microphones.*—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 uniform response over a wide frequency range. For high-quality reproduction the distortion may be reduced by employing two buttons in a push-pull arrangement. It is the purpose of this section to consider single- and double-button carbon microphones.

1. *Single-Button Carbon Microphone.*—A typical carbon microphone is shown in Fig. 8.1. The carbon button consists of a cylindrical cavity filled with carbon granules. The carbon granules are usually made from anthracite coal. The carbon granules make contact with the diaphragm and the cylindrical cup. Suitable washers are used to prevent leakage of the carbon granules between the diaphragm and carbon cup without im-

peding the motion of the diaphragm. A displacement of the diaphragm produces a change in the pressure between the carbon granules which changes the electrical resistance from granule to granule. The net result is a change in the electrical resistance between the diaphragm and the carbon cup. For small displacements the change in resistance is proportional to
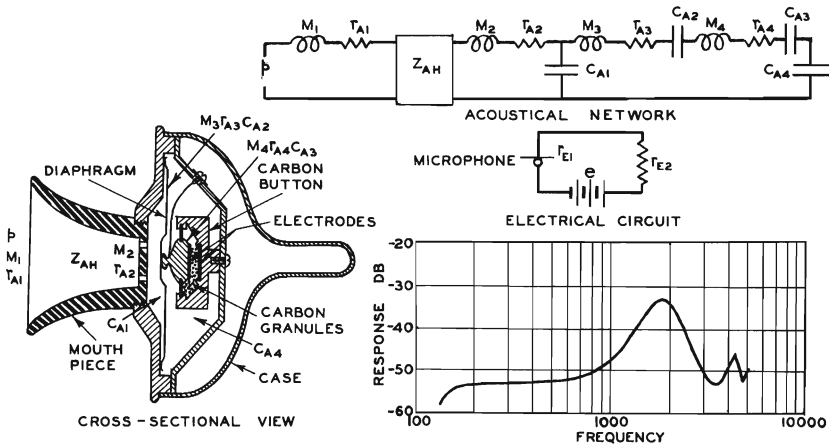
FIG. 8.1. Cross-sectional view, the electrical circuit and the acoustical network of a single-button carbon microphone. In the electrical circuit, r_{E1} = the electrical resistance of the carbon element, r_{E2} = the electrical resistance of the load, and e = the polarizing voltage of the battery. In the acoustical network M_1 and r_{A1} = the inertance and acoustical resistance at mouthpiece opening. z_{AH} = the acoustical quadripole representing the horn or mouthpiece. M_2 and r_{A2} = the inertance and acoustical resistance of the holes in the mouthpiece. C_{A1} = the acoustical capacitance of the air chamber in front of the diaphragm. M_3, r_{A3}, and C_{A2} = the inertance, acoustical resistance, and acoustical capacitance of the diaphragm. M_4, r_{A4}, and C_{A3} = the inertance, acoustical resistance, and acoustical capacitance of the carbon element. C_{A4} = the acoustical capacitance of the case. p = sound pressure. The graph shows the open circuit voltage response frequency characteristic for constant sound pressure in free space.

the displacement. Consider the electrical circuit of Fig. 8.1, for sinusoidal motion of the diaphragm, the current, in amperes, in the circuit is given by

$$i = \frac{e}{r_{E0} + hx \sin \omega t} \qquad 8.1$$

where e = voltage of the battery, in volts,

r_{E0} = total electrical resistance of the circuit when $x = 0$, in ohms,

x = amplitude of the diaphragm, in centimeters,

h = constant of the carbon element, in ohms per centimeter,

$\omega = 2\pi f$, and

f = frequency, in cycles per second.

Equation 8.1 may be expanded as follows,

$$i = \frac{e}{r_{E0}} \left(1 - \frac{hx}{r_{E0}} \sin \omega t + \frac{h^2 x^2}{r_{E0}^2} \sin^2 \omega t \ldots \right)$$

$$= \frac{e}{r_{E0}} \left(1 - \frac{hx}{r_{E0}} \sin \omega t + \frac{h^2 x^2}{2 r_{E0}^2} - \frac{h^2 x^2}{2 r_{E0}^2} \cos 2\omega t \ldots \right) \qquad 8.2$$

Equation 8.2 shows that there is a steady direct current, an alternating current of the frequency of the diaphragm vibration and harmonics of this vibration. For a limited frequency range of speech reproduction, the nonlinear distortion is not particularly objectionable.

The acoustical network of the acoustical system is shown in Fig. 8.1. The mouthpiece is a short exponential horn and is represented as an acoustical quadripole, z_{AH} (see Sec. 5.27). The performance of the system may be obtained from the acoustical circuit.

The diaphragm of the microphone is a circular plate supported at the edge (see Sec. 3.5). The effective mass and effective area of the diaphragm is one-third the total mass and total area of the diaphragm. Below the fundamental resonant frequency the acoustical capacitance of the diaphragm C_{A2}, is the controlling acoustical impedance. Under these conditions the displacement is proportional to the pressure. Since the change in electrical resistance of the carbon button and the resultant developed voltage is proportional to the amplitude, the output for constant sound pressure will be independent of the frequency below the fundamental resonant frequency of the system. These observations are supported by the response frequency characteristic of Fig. 8.1 which depicts uniform response in the low-frequency range below the fundamental resonant frequency of the system. In the region of resonance the output is accentuated. In the frequency range above the fundamental resonant frequency the response falls off rapidly in a series of peaks which are due to the higher modes of the diaphragm and the acoustical system.

An improved type of single-button carbon microphone[1] has been developed in which the response is quite uniform over a wide frequency range (Fig. 8.2). The conical diaphragm is made of a thin aluminum alloy. At low frequencies the diaphragm vibrates as a single unit. However, at the higher frequencies it is necessary to consider it to be made up of three separate masses. These masses consist of the central portion m_5, the ribbed intermediate portion m_2, and the outer portion m_4. The central portion includes the mass of the movable electrode and is coupled to the ribbed portion by the compliance, C_{M6}, which in turn is coupled to the outer portion by the compliance, C_{M2}. The paper books which support the edge of the diaphragm have a compliance, C_{M4}, and a mechanical resistance, r_{M4}. Their mass is included in the outer portion of the diaphragm, m_4. The internal mechanical resistance of the portions which form the coupling compliances, C_{M2} and C_{M6}, are represented by r_{M2} and r_{M6}, respectively.

[1] Jones, W. C., *Jour. A.I.E.E.*, Vol. 57, No. 10, p. 559, 1939.

A hole is provided in the diaphragm to permit rapid equalization of low-frequency pressures of high intensity and prevent damage to the diaphragm and other parts. The mass and the mechanical resistance of this hole, m_3 and r_{M3}, are so chosen that their effects on the response are confined to frequencies below 300 cycles. The controlling compliance, C_{M3}, is that of the cavity between the diaphragm and the die-cast frame. The carbon
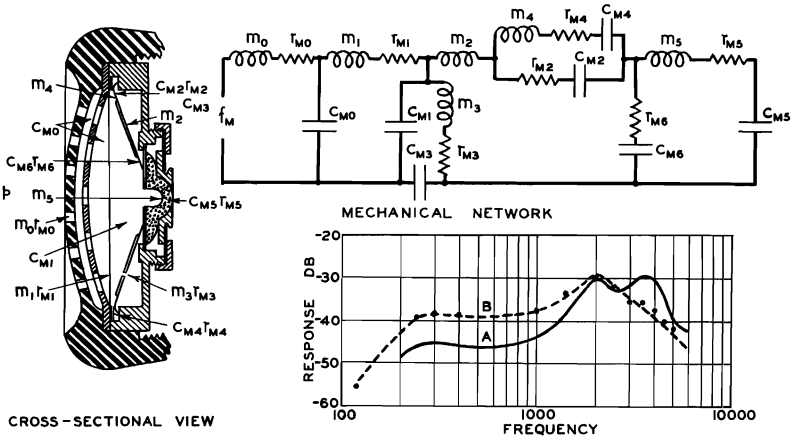

FIG. 8.2. Cross-sectional view and the mechanical network of an improved single-button carbon microphone. The electrical circuit is the same as that of Fig. 8.1. In the mechanical network, m_0 and r_{M0} = the mass and mechanical resistance of the holes in the outer grill. C_{M0} = compliance of the air chamber between the grill and the membrane. m_1 and r_{M1} = the mass and mechanical resistance of the waterproof membrane. C_{M1} = the compliance of the air chamber between the membrane and the diaphragm. m_2, r_{M2}, and C_{M2} = the mass, mechanical resistance, and compliance of the central and outer portion of the diaphragm. m_3 and r_{M3} = the mass and mechanical resistance of the hole in the diaphragm. C_{M3} = the compliance of the air chamber behind the diaphragm. C_{M4} and r_{M4} = the compliance and mechanical resistance of the paper book suspension. m_4 = the mass of the paper book suspension and the outer part of the diaphragm. m_5, r_{M5}, and C_{M5} = the mass, mechanical resistance, and compliance of the center portion of the diaphragm and the carbon granules. C_{M6} and r_{M6} = the compliance and mechanical resistance of the center of the diaphragm. f_M = the driving force. $f_M = pA$. A = the area of the diaphragm. p = the sound pressure. The open-circuit voltage response characteristics are shown in the graph. A. Response in free space. B. Response for constant sound pressure at the diaphragm. Dots computed from the mechanical network.

granules are represented by a compliance, C_{M5}, and a mechanical resistance, r_{M5}. The mass of the carbon granules is lumped with that of the central portion of the diaphragm. The holes in the inner grid are sufficiently large so that there is no reaction upon the response. The holes in the outer grill add the mass, m_0, and the mechanical resistance, r_{M0}. These holes are coupled to a moisture-proof membrane of mass, m_1, and mechanical

resistance, r_{M1}, by means of the compliance, C_{M0}, of the enclosed cavity. The cavity compliance, C_{M1}, couples the membrane to the diaphragm.

The response of this microphone computed from the mechanical network is shown in Fig. 8.2. The response for constant sound pressure on the diaphragm is also shown in Fig. 8.2. It will be seen that the agreement between the computed and measured characteristics is very good and substantiates this type of analysis. The response is very much smoother than in the case of the plate or disk type of diaphragm.
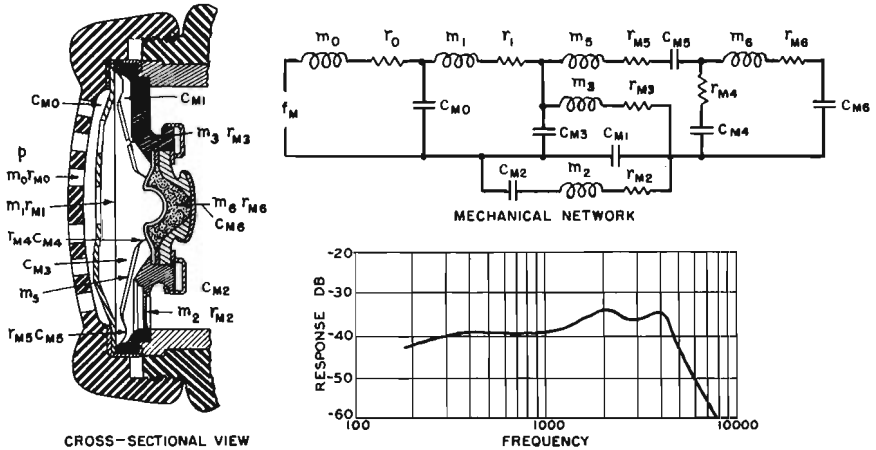

FIG. 8.3. Cross-sectional view and the mechanical network of a new type single-button carbon microphone. The electrical circuit is the same as that of Fig. 8.1. In the mechanical network, m_0 and r_{M0} = the mass and mechanical resistance of the holes in the outer grill. C_{M0} = the compliance of the air chamber between the inner and outer grids. m_1 and r_{M1} = the mass and mechanical resistance of the waterproof membrane. C_{M1} = the compliance of the air chamber behind the diaphragm. C_{M3} = the compliance of the air chamber behind the waterproof membrane and the diaphragm. m_2 and r_{M2} = the mass and mechanical resistance of the cloth. m_3 and r_{M3} = the mass and mechanical resistance of the hole in the diaphragm. m_5 = the mass of the diaphragm. r_{M4} and C_{M4} = the mechanical resistance and compliance of the diaphragm to carbon cup couples. C_{M2} = the compliance of the air chamber behind the microphone unit. r_{M5} and C_{M5} = the mechanical resistance and compliance of the diaphragm suspension system. m_6, r_{M6}, and C_{M6} = the mass, mechanical resistance, and compliance of the carbon cup and granules. f_M = the driving force. $f_M = pA$. p = the sound pressure. A = the area of the diaphragm. The graph shows the open circuit voltage response frequency characteristic for constant sound pressure on the diaphragm.

The free-space response shown in Fig. 8.2 indicates the diffraction effect of the microphone as an obstacle in increasing the pressure on the diaphragm (see Sec. 1.11 and Fig. 1.5).

In addition to the smoother response the sensitivity of the new unit is higher because of the reduction in mass of the vibrating system. Due to the shape of the carbon chamber the performance of the microphone is less affected by angular position.

A new carbon microphone,[2] shown in Fig. 8.3, has been developed which in appearance is similar to the microphone of Fig. 8.2. However, there are many important features which lead to improved performance. One of the most important elements that has been added to the microphone of Fig. 8.2 is the mechanical resistance, r_{M2}. The vibrating system is stiffness controlled below the resonant frequency. At the resonant frequency the mechanical resistance provides the controlling element. As a consequence, the response in the frequency region 2000 and 4000 cycles, depicted in Fig. 8.3, is smoother than that of the microphone of Fig. 8.2. The sensitivity of the microphone of Fig. 8.3 is somewhat greater than the microphone of Fig. 8.2. This has been accomplished by an improved design of the carbon cup.
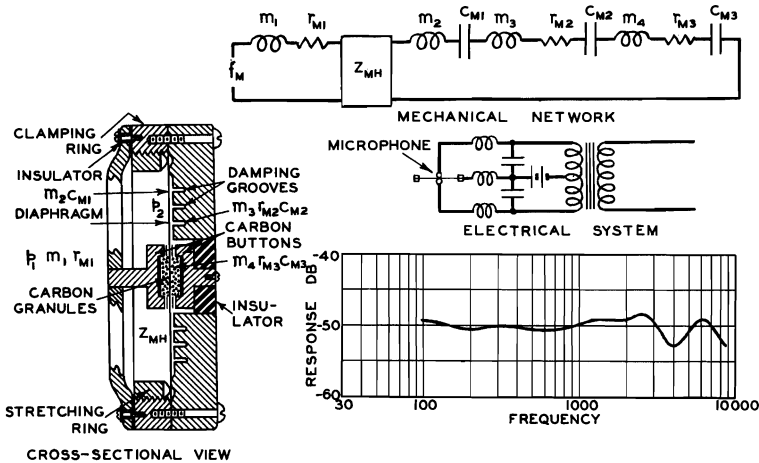

FIG. 8.4. Cross-sectional view, the electrical circuit, and the mechanical circuit of a double-button, stretched diaphragm, carbon microphone. In the mechanical circuit, m_1 and r_{M1} = the mass and mechanical resistance of the air load. z_{MH} = the mechanical quadripole representing the cylindrical cavity or pipe. m_2 and C_{M1} = the mass and compliance of the diaphragm. m_3, r_{M2}, and C_{M2} = the mass, mechanical resistance, and compliance of the carbon granules. m_4, r_{M3}, and C_{M3} = the mass, mechanical resistance, and compliance due to the damping plate. f_M = the driving force. $f_M = pA$. A = the area of the diaphragm. p = the sound pressure. The graph shows the open-circuit voltage response-frequency characteristic for constant sound pressure at the diaphragm.

2. *Double-Button Carbon Microphone.*—For applications requiring both high quality and large power output the single-button carbon microphone is not suitable due to the large nonlinear distortion. Uniform response and low distortion may be obtained in a carbon microphone[3] by means of a system consisting of a stretched diaphragm and two carbon buttons as shown in Fig. 8.4. The performance of the system may be obtained from a consideration of the mechanical network of the vibrating system. The

[2] Inglis and Tuffnel, *Bell Syst. Tech. Jour.*, Vol. 30, No. 2, p. 209, 1951.

[3] Jones, W. C., *Bell Syst. Tech. Jour.*, Vol. 10, No. 1, p. 46, 1931.

mechanical impedance of a stretched diaphragm, below its resonant frequency, is a stiffness mechanical reactance. Therefore, a constant sound pressure on the diaphragm will produce substantially constant displacement. Since the change in electrical resistance of the carbon buttons and the resultant developed voltage is proportional to the displacement, the voltage output will be independent of the frequency. To provide damping at the resonant frequency of the diaphragm the damping plate is placed very close to the back of the diaphragm. As the diaphragm moves, air is forced through this small space. The high viscosity loss in a small slit provides the damping (see Sec. 5.4). In order to reduce the stiffness, in the small space, suitable grooves are provided which reduce the length of the slit. The rear button is enclosed in the damping plate while the front button is supported by the bridge. The duraluminum diaphragm is gold-plated over the area occupied by the carbon buttons to insure contact between the carbon granules and the diaphragm. The resonant frequency of the stretched diaphragm is usually placed between 5000 and 8000 cycles (see Sec. 3.4). In the absence of the damping plate the amplitude for a constant force at the resonant frequency would be greater than that below the resonant frequency. By means of the damping plate the amplitude at the resonant frequency can be reduced to correspond to that of the remainder of the range. A response frequency characteristic of this microphone is shown in Fig. 8.4.

The electrical circuit diagram for this microphone is shown in Fig. 8.4. For a sinusoidal motion of the diaphragm the current, in amperes, in one of the buttons may be written as

$$i_1 = \frac{e}{r_{E0} + hx \sin \omega t} \qquad 8.3$$

when e = voltage of the battery, in volts,

r_{E0} = electrical resistance of the circuit, when $x = 0$, in ohms,

x = amplitude of the diaphragm, in centimeters,

h = constant of the carbon element, in ohms per centimeter,

$\omega = 2\pi f$, and

f = frequency, in cycles per second.

The current in the other button is

$$i_2 = \frac{e}{r_{E0} - hx \sin \omega t} \qquad 8.4$$

The difference between equations 8.3 and 8.4 after expanding is

$$i_2 - i_1 = \frac{2e}{r_{E0}} \left(\frac{hx}{r_{E0}} \sin \omega t + \frac{h^3 x^3}{r_{E0}^3} \sin^3 \omega t \ldots \right)$$

$$= \frac{2e}{r_{E0}} \left(\frac{hx \sin \omega t}{r_{E0}} + \frac{3}{4} \frac{h^3 x^3}{r_{E0}^3} \sin \omega t - \frac{h^3 x^3}{4 r_{E0}^3} \sin 3\omega t \ldots \right) \qquad 8.5$$

Comparing equation 8.5 with equation 8.2 shows that the large second harmonic term has been eliminated by the use of a push-pull two-button microphone.

One common cause of faulty operation of the carbon microphone is due to the cohering of the carbon granules caused by the breaking of the circuit when the current is flowing. The use of electric filters as shown in the circuit diagram will protect the microphone against cohering.

The frequency range and response of the double-button carbon microphone compares favorably with the condenser microphone. The carbon microphone is several times more sensitive than the condenser microphone. However, the limitation is carbon noise.
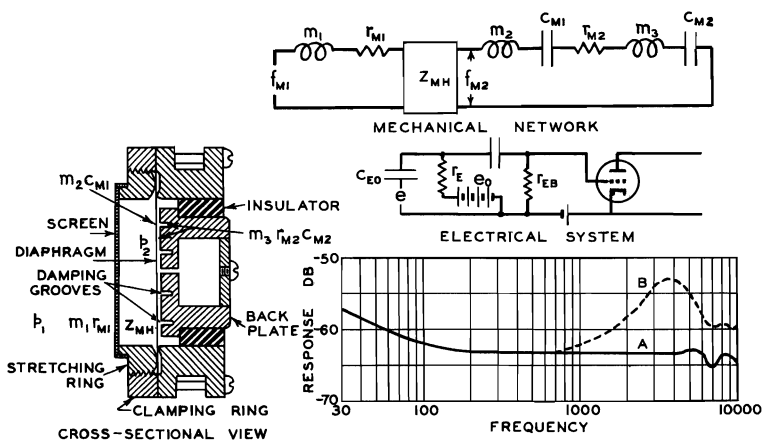

FIG. 8.5. Cross-sectional view, electrical system, and mechanical circuit of a condenser microphone. In the electrical system: e_0 = the polarizing voltage, r_E = the polarizing electrical resistance, r_{EB} = the bias electrical resistance. C_{E0} = the electrical capacitance of the microphone. In the mechanical circuit, m_1 and r_{M1} = the mass and mechanical resistance of the air load. z_{MH} = the mechanical quadripole representing the cylindrical cavity or pipe. m_2 and C_{M1} = the mass and compliance of the diaphragm. r_{M2}, m_3, and C_{M2} = the mass, mechanical resistance, and compliance of the air film. f_M = the driving force. $f_M = pA$. A = the area of the diaphragm. p = the sound pressure. The graph shows the open-circuit voltage response frequency characteristics. A. Response for constant sound pressure on the diaphragm. B. Response for constant sound pressure in free space.

B. *Condenser Microphone (Electrostatic Microphone)*.—A condenser microphone, also termed an electrostatic microphone, is a microphone which depends for its operation on variations in electrical capacitance. The typical condenser microphone[4] consists of a thin stretched plate separated from a parallel rigid plate (Fig. 8.5). The electrical system of this microphone is shown in Fig. 8.5.

[4] Wente, E. C., *Phys. Rev.*, Vol. 10, No. 1, p. 39, 1917.

The electrical capacitance, in statfarads, at any instant is given by

$$C_E = C_{E0} + C_{E1} \sin \omega t \qquad 8.6$$

where C_{E0} = electrical capacitance in the absence of an applied pressure, in statfarads,

C_{E1} = maximum change in the electrical capacitance due to the external applied sinusoidal pressure, in statfarads,

$\omega = 2\pi f$, and

f = frequency, in cycles per second.

From the electrical circuit

$$e_0 - r_E i = \frac{1}{C_E} \int i \, dt \qquad 8.7$$

where e_0 = polarizing voltage, in statvolts,

r_E = electrical resistance of the polarizing resistor, in statohms,

i = current, in statamperes, and

t = time, in seconds.

Equation 8.7 assumes that the bias resistor, r_{EB}, and the input electrical impedance of the vacuum tube is very large compared with r_E. Then e_0 may be considered to be in series with C_{E0} and r_E. Substituting the value of C_E from equation 8.7 in equation 8.6 and differentiating

$$(C_{E0} + C_{E1} \sin \omega t) r_E \frac{di}{dt} + (1 + r_E C_{E1} \omega \cos \omega t) i - e_0 C_{E1} \omega \cos \omega t = 0 \quad 8.8$$

The solution of equation 8.8 is

$$i = \frac{e_0 C_{E1}}{C_{E0}\sqrt{(1/C_{E0}\omega)^2 + r_E^2}} \sin (\omega t + \phi_1)$$

$$- \frac{e_0 C_{E1} r_E}{C_{E0}^2 \sqrt{[(1/C_{E0}\omega)^2 + 4r_E^2][(1/C_{E0}\omega)^2 + r_E^2]}} \sin (2\omega t + \phi_1 - \phi_2)$$

$$+ \text{ terms of higher order} \qquad 8.9$$

where $\phi_1 = \tan^{-1} 1/C_{E0}\omega r_E$ and $\phi_2 = \tan^{-1} 1/2C_{E0}\omega r_E$.

For small diaphragm amplitudes, the generated voltage, in statvolts, is

$$e' = r_E i = \frac{e_0 C_{E1} r_0}{C_{E0}\sqrt{\dfrac{1}{C_{E0}^2 \omega^2} + r_0^2}} \sin (\omega t + \phi_1) \qquad 8.10$$

Equation 8.10 shows that the condenser microphone[5] may be considered as a generator with an internal open circuit voltage of

$$e = e_0 \left(\frac{C_{E1}}{C_{E0}}\right) \sin (\omega t + \phi_1), \text{ in statvolts,} \qquad 8.11$$

and an internal electrical impedance of $1/C_{E0}\omega$, in statohms.

[5] Wente, E. C., *Phys. Rev.*, Vol. 19, No. 5, p. 498, 1922.

The mechanical network of the mechanical system of the condenser microphone is shown in Fig. 8.5. The performance of the vibrating system may be obtained from a consideration of the mechanical network. Equation 8.11 shows that the voltage is proportional to the amplitude. Therefore, to obtain a microphone in which the sensitivity is independent of the frequency, the amplitude, for a constant applied pressure, must be independent of the frequency. In the range below the resonant frequency the amplitude of a stretched membrane for a constant applied force is independent of the frequency (see Sec. 3.4). The addition of the back plate with very close spacing introduces mechanical resistance[6,7] due to the viscosity loss in the narrow slit (see Sec. 5.4). This mechanical resistance reduces the amplitude at the resonant frequency. The back plate also introduces stiffness due to the entrapped air. This stiffness can be reduced without reducing the mechanical resistance by cutting grooves in the back of the plate. If the damping is made sufficiently large the amplitude at the fundamental resonant frequency of the diaphragm can be made to correspond to that of the remainder of the range.

The amplitude of the diaphragm, in centimeters, is given by

$$x = \frac{f_{M2}}{\left[r_{M2} + j\omega(m_2 + m_3) + \dfrac{1}{j\omega}\left(\dfrac{1}{C_{M1}} + \dfrac{1}{C_{M2}}\right) \right] j\omega} \qquad 8.12$$

where f_{M2} = applied force, in dynes,

$f_{M2} = pA$,

p = sound pressure on the diaphragm, in dynes per square centimeter,

A = area of the diaphragm, in square centimeters,

r_{M2} = damping mechanical resistance of air film, in mechanical ohms,

m_2 = effective mass of the diaphragm, in grams,

C_{M1} = compliance due to stiffness of the diaphragm, in centimeters per dyne,

m_3 = mass of air film, in grams,

C_{M2} = compliance due to stiffness of the air film, in centimeters per dyne,

$\omega = 2\pi f$, and

f = frequency, in cycles per second.

Equation 8.12 shows that the sensitivity below the resonant frequency is inversely proportional to the stiffness and the mechanical resistance. For the same fundamental resonant frequency the stiffness can be reduced

[6] Crandall, I. B., *Phys. Rev.*, Vol. 11, No. 6, p. 449, 1918.
[7] Crandall, " Vibrating Systems and Sound," D. Van Nostrand Company, Princeton, N.J., 1926.

by decreasing the mass. This procedure also reduces the amount of mechanical resistance required to damp the fundamental resonance and thereby obtain uniform response. Aluminum alloys, due to the low density and high tensile strength, are the logical materials for use in diaphragms. The minimum diaphragm thickness suitable for the manufacture of condenser microphones is about .001 inch. The electrical capacitance of a microphone with a diaphragm diameter of $1\frac{5}{8}$ inches and a spacing of from .001 to .002 inch is from 400 to 200 mmfds. Due to the high electrical impedance of this capacitance it is necessary to locate the microphone near the vacuum tube amplifier. The electrical capacitance of a long connecting cable reduces the sensitivity without frequency discrimination because the internal electrical impedance of the microphone is also an electrical capacitance. The response frequency characteristics of a condenser microphone for
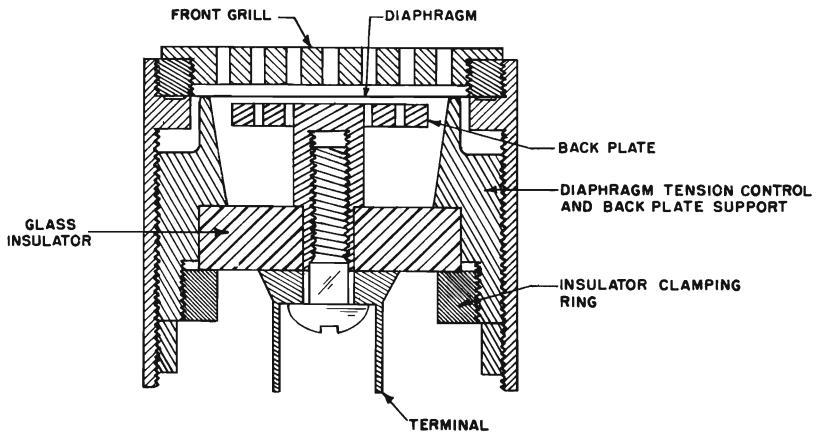

FIG. 8.6. Cross-sectional view of a miniature condenser microphone with a stretched membrane-type diaphragm.

constant sound pressure on the diaphragm and for constant free wave sound pressure are shown in Fig. 8.5.

The condenser microphone[8,9] shown in Fig. 8.5 employs a diaphragm with a diameter of $1\frac{5}{8}$ inches. The over-all diameter of the condenser microphone unit is about 3 inches. These microphones were developed about twenty-five years ago and were employed in the early days of sound reproduction. The condenser microphone was replaced by the electrodynamic (voice coil and ribbon) and piezoelectric microphone. During the past decade, smaller condenser microphones have been developed. A miniaturized version of the microphone shown in Fig. 8.5 is shown in Fig. 8.6. The over-all diameter of the microphone unit is a little less than 1 inch. The fundamental resonant frequency of the diaphragm is about 9000 cycles. The system is highly damped so that uniform response is maintained to over 15,000 cycles. The

[8] Harrison and Flanders, *Bell Syst. Tech. Jour.*, Vol. 11, No. 3, p. 451, 1932.
[9] Veneklasen, Paul S., *Jour. Acous. Soc. Amer.*, Vol. 20, No. 6, p. 807, 1948.

deviations in response are smooth and can be easily compensated by electrical means to obtain a response frequency characteristic which is independent of frequency. The amplifier which may be used with this microphone is shown in Fig. 8.7. The cathode follower type of operation provides a
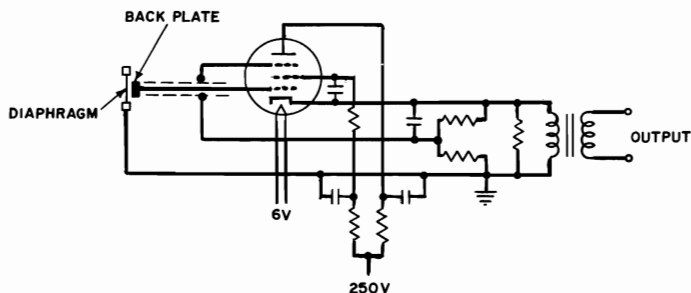

FIG. 8.7. Circuit diagram of a vacuum tube amplifier with a very large input electrical impedance.

system having a high input electrical impedance. This is necessary for the small condenser microphones in which the capacitance is only about 50 mmfds in order to maintain the response in the low-frequency region. See equation 8.11. The condenser microphone shown in Fig. 8.6 is used as a standard microphone in pressure calibration of laboratory standard microphones. See Sec. 10.2A1d.

Another miniature condenser microphone[10] is shown in Fig. 8.8. This microphone employs a plate instead of a stretched diaphragm. See Sec.
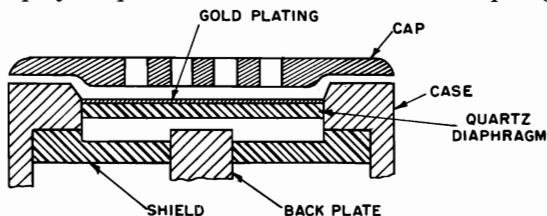
FIG. 8.8. Cross-sectional view of a miniature condenser microphone with plate-type diaphragm.

3.5. The over-all diameter of the microphone unit is about $\frac{3}{4}$ inch. The amplifier used with these microphones is of the type shown in Fig. 8.7.

C. *Piezoelectric* (*Crystal*) *Microphones*.[11,12,13]—A piezoelectric microphone is a microphone which depends upon the generation of an electromotive force by the deformation of a crystal having piezoelectric properties.

[10] Hilliard, J. K., and Noble, J. J., *Trans. IRE*, Prof. Group on Audio, Vol. AU-2, No. 6, p. 168, 1954.
[11] Sawyer, C. B., *Proc. Inst. Rad. Eng.*, Vol. 19, No. 11, p. 2020, 1931.
[12] Williams, A. L., *Jour. Soc. Mot. Pic. Eng.*, Vol. 18, No. 4, p. 196, 1934.
[13] Nicolson, U.S. Patent 1,495,429.

The voltage generated[14] due to a deformation of the crystal is proportional to the displacement. Therefore, to obtain a uniformly sensitive microphone with respect to frequency the displacement for a constant applied force must be independent of the frequency. Rochelle salt exhibits the greatest piezoelectric activity of all of the known crystals. For this reason it is used in audio-frequency microphones. There are two general classifications of crystal microphones—namely, the direct actuated and the
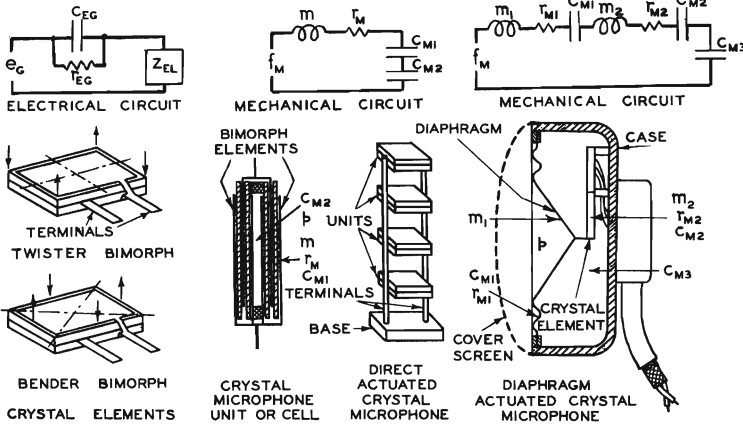

FIG. 8.9. Crystal elements and sound cells. A direct actuated crystal microphone. In the electrical network, C_{EG} = the electrical capacitance of the crystal. r_{EG} = the electrical resistance of the crystal. z_{EL} = the electrical impedance of the load. e_G = the open circuit voltage developed by the crystal. In the mechanical circuit, m, r_M, and C_{M1} = the mass, mechanical resistance, and compliance of the crystal. C_{M2} = the compliance of one half of the air chamber. f_M = the driving force, $f_M = pA$. A = the effective area of the crystal. p = the sound pressure. A diaphragm actuated crystal microphone. In the mechanical circuit, m_1, r_{M1}, and C_{M1} = the mass, mechanical resistance, and compliance of the diaphragm. m_2, r_{M2}, and C_{M2} = the mass, mechanical resistance, and compliance of the crystal. C_{M3} = the compliance due to the case volume. f_M = the driving force. $f_M = pA$. A = the effective area of the diaphragm. p = the sound pressure at the diaphragm.

diaphragm actuated. In the direct actuated, the sound pressure acts directly upon the crystal. In the diaphragm actuated, the sound pressure acts upon a diaphragm which is coupled to a crystal. The crystal element, Fig. 8.9, is made up of two crystals cut so that a voltage is generated when forces are applied as shown. The two types of bimorph elements, namely, "twisters" and "benders," are shown in Fig. 8.9. A bimorph construction has several advantages over the single crystal, as follows: it lends itself to a more efficient size and shape; it becomes more sensitive (a gain of 15 times for practical shapes); it reduces the variations of the mechanical

[14] Olson, "Dynamical Analogies," D. Van Nostrand Company, Princeton, N.J., 1943

and electrical constants of the crystal for changes in temperature. The temperature limits of bimorph crystals are from $-40°$ F. to $130°$ F. If exposed to temperatures in excess of $130°$ F. the crystal loses its piezoelectric activity permanently. The sensitivity or voltage output of the crystal varies with temperature due primarily to a change in the capacitance and in a lesser degree to a change in the developed voltage. An ADP crystal with greater temperature and humidity ranges is described in Sec. 13.12.

1. *Direct Actuated Crystal Microphone.*—In the direct actuated crystal microphone the sound pressure acts directly upon the crystal. A common form of sound cell for a direct actuated crystal microphone consists of two bimorph elements assembled as shown in Fig. 8.9. The cavity formed by the two crystal elements is completely enclosed so that the application of an external pressure causes a deformation of the crystal.

The internal voltage, e, developed by the crystal is

$$e = Kx \qquad 8.13$$

where K = constant of the crystal, and

x = effective amplitude of the deformation of the crystal by an applied force.

From the mechanical circuit of Fig. 8.9, the amplitude, in centimeters, is

$$x = \frac{f_M}{\left(r_M + j\omega m + \dfrac{1}{j\omega C_{M1}} + \dfrac{1}{j\omega C_{M2}}\right) j\omega} \qquad 8.14$$

where r_M = effective mechanical resistance of the crystal, in mechanical ohms,

m = effective mass of the crystal, in grams,

C_{M1} = effective compliance of the crystal, in centimeters per dyne,

C_{M2} = compliance of one half of the air chamber between the crystals, in centimeters per dyne,

$f_M = pA$, in dynes,

p = sound pressure at the surface of the crystal in dynes per square centimeter,

A = area of the crystal, in square centimeters,

$\omega = 2\pi f$, and

f = frequency, in cycles per second.

A consideration of equation 8.14 shows that the amplitude will be independent of the frequency in the range below the resonant frequency. Under these conditions the internal voltage developed by the crystal, as given by equation 8.13, will be independent of the frequency. The resonant frequency is placed beyond the desired response range of the microphone so that uniform response is obtained in the desired frequency range. Uniform response to 17,000 cycles can be readily obtained.

A typical direct actuated crystal microphone, shown in Fig. 8.9, consists of four cells. The internal impedance of a single cell is relatively high. This high impedance may be reduced by the use of several cells in parallel. If the crystal element is small compared to the wavelength, the individual element will be nondirectional.

2. *Diaphragm Actuated Crystal Microphone.*—In the diaphragm actuated crystal microphone the sound pressure acts upon a diaphragm which in turn drives a crystal. The output of the diaphragm actuated type is considerably higher than the direct actuated type because the diaphragm acts as a coupling unit between the relatively low impedance of the air and the high impedance of the crystal. A cross-sectional view of a diaphragm actuated crystal microphone is shown in Fig. 8.9. The response frequency characteristic for constant sound pressure on the diaphragm may be obtained from the mechanical circuit of Fig. 8.9 and equation 8.13. As shown in Fig. 1.5 the ratio of the pressure on the face of a cylinder to that in free space increases as the dimensions become comparable to the wavelength. This effect accentuates the response in the high-frequency range.

A new, miniature crystal microphone has been developed in which the diameter is $1\frac{1}{8}$ inches. The vibrating system is essentially the same as that shown in Fig. 8.9 with the addition of a mechanical resistance placed over the front of the diaphragm. The mechanical resistance controls the response at the resonant frequency of the diaphragm and crystal combination and thereby smooths out the response frequency characteristic in the high-frequency range.

3. *Diaphragm-Actuated Barium Titanate Microphones.*[15,16]—Barium titanate is a ceramic which exhibits properties similar to that of a piezo-electric crystal. Microphones employing barium titanate are constructed in a manner similar to that of the diaphragm-type crystal microphone of Fig. 8.9. A strip of barium titanate is used instead of the crystal. The performance and electrical characteristics are essentially the same as that of the crystal microphone except the sensitivity is somewhat lower. The advantage of the barium titanate microphone is that it can be operated at higher temperatures than Rochelle salt crystal microphones.

D. *Moving Conductor Microphones.*—A moving conductor microphone is a microphone in which the output results from the motion of a conductor in a magnetic field. The conductor may be in the form of a circular coil which is termed a moving coil microphone or in the form of a straight conductor which is termed an inductor microphone. These microphones are also termed dynamic microphones.

1. *Moving Coil Microphone (Dynamic Microphone).*[17,18]—A cross-sectional view of a moving coil microphone is shown in Fig. 8.10. The motion of

[15] Medill, John, *Trans. IRE*, Prof. Group on Audio., Vol. AU-1, No. 6, p. 7, 1953.

[16] Medill, John, *Jour. Acous. Soc. Amer.*, Vol. 25, No. 5, p. 864, 1953.

[17] Wente and Thuras, *Jour. Acous. Soc. Amer.*, Vol. 3, No. 1, p. 44, 1931.

[18] Wigginton, L. M., and Carroll, R. M., *Jour. Audio Eng. Soc.*, Vol. 3, No. 2, p. 77, 1955.

the diaphragm is transferred to a coil located in a magnetic field. The mechanical circuit of the mechanical system consisting of the diaphragm coil and suspension system is shown in Fig. 8.10A.

The velocity, in centimeters per second, of the voice coil is given by

$$\dot{x} = \frac{f_M}{r_{M1} + j\omega m_1 + \dfrac{1}{j\omega C_{M1}}} \qquad 8.15$$

where r_{M1} = mechanical resistance of the suspension system, in mechanical ohms,

m_1 = mass of the diaphragm and voice coil, in grams,

C_{M1} = compliance of the suspension system, in centimeters per dyne, and

f_M = driving force, in dynes.

The generated internal voltage, in abvolts, is

$$e = Bl\dot{x} \qquad 8.16$$

where B = flux density in the air gap, in gausses,

l = length of the voice coil conductor, in centimeters, and

$\dot{x}$ = velocity of the voice coil, in centimeters per second.

Equation 8.16 shows that the microphone will be uniformly sensitive with respect to frequency if the velocity is independent of the frequency. The characteristics 1 and 2 in Fig. 8.10A were computed by employing equation 8.15. These characteristics show that a uniformly sensitive dynamic microphone, with respect to frequency, must be essentially "resistance controlled."

The characteristic marked 2, Fig. 8.10A shows some falling off in velocity at the high and low frequencies. This can be corrected by the use of some additional elements (Fig. 8.10B). The major portion of the mechanical resistance is the silk cloth, $m_2 r_{M2}$. Mechanical resistance in the case of silk cloth is due to the high viscosity introduced by the small holes (see Sec. 5.5). Slits have also been used for the resistance element (see Sec. 5.4). The mass mechanical reactance of the diaphragm is reduced at the higher frequencies by the compliance, C_{M2}, formed by the volume between the silk and the diaphragm. The addition of the mechanical elements C_{M2}, r_{M2}, and m_2 changes the characteristic at the high frequencies from that marked 3 to that marked 5. An increase in response over an octave is obtained by the addition of these elements. A corresponding increase in response can be obtained at the low frequencies by means of the case volume, C_{M3}, and the addition of a tube, $m_3 r_{M3}$. The mechanical network shows the action of the additional elements in changing the response from the characteristic 3 to the characteristic 4–5.

The most common materials used for the diaphragms of pressure micro-phones are aluminum alloys, Bakelite, styrol, and paper. In order to obtain a minimum density-resistivity product, aluminum is almost universally used for the voice coil (see Table 6.1). Both edgewise wound ribbon and round wire have been used for the voice coil (see Sec. 6.27 and Fig. 6.76).
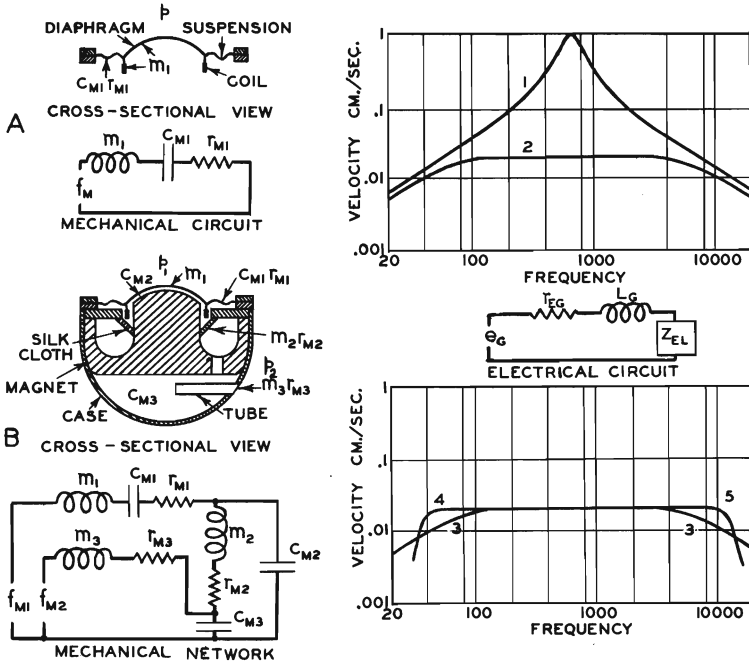

FIG. 8.10. A. Cross-sectional view and mechanical circuit of a diaphragm, coil, and suspension. In the electrical circuit, r_{EG} = the electrical resistance of the coil. L = the inductance of the coil. z_{EL} = the electrical impedance of the load. e_G = the open circuit voltage developed by the coil. In the mechanical circuit, m_1, r_{M1}, and C_{M1} = the mass, mechanical resistance, and compliance of the vibrating system. f_M = the driving force. $f_M = pA$. A = the area of the diaphragm. p = the sound pressure. The velocity frequency characteristic for a unit force and a mechanical resistance of 1 mechanical ohm is indicated as curve 1 on the graph. The same for a mechanical resistance of 60 mechanical ohms. B. Cross-sectional view and mechanical circuit of a dynamic microphone. In the mechanical network, m_1, r_{M1}, and C_{M1} = the mass, mechanical resistance, and compliance of the diaphragm and suspension. C_{M2} = the compliance of the air chamber behind the diaphragm. m_2 and r_{M2} = the mass and mechanical resistance of the silk cloth. m_3 and r_{M3} = the mass and mechanical resistance of the air in the tube. C_{M3} = the compliance of the case volume. f_{M1} and f_{M2} = the driving forces. $f_{M1} = p_1A$ and $f_{M2} = p_2A$. A = the area of the diaphragm. p_1 = the sound pressure at the diaphragm. p_2 = the sound pressure at the tube. Curve 3 on the lower graph is the same as curve 2 on the upper graph. Curve 4 is the response with the tube m_3, r_{M3} added. Curve 5 is the response with the compliance C_{M2} added.

An examination of the diffraction of sound as a function of the angle of the incident sound by various objects shows that the sphere exhibits the most uniform directional pattern. A spherical case with the diaphragm located on the surface of the sphere seems to be the logical starting point for a nondirectional pressure microphone. Referring to Fig. 1.5, it will be seen that the microphone will show excess response over the range from 0° to 60° and will be lacking in response from 120° to 160 . This nonuniform response can be corrected by placing a disk, of semi-transmitting characteristics and of diameter equal to the spherical case, directly above the diaphragm and spaced one-fourth inch. Employing this expedient, a nondirectional characteristic[19] is obtained over the response frequency range.
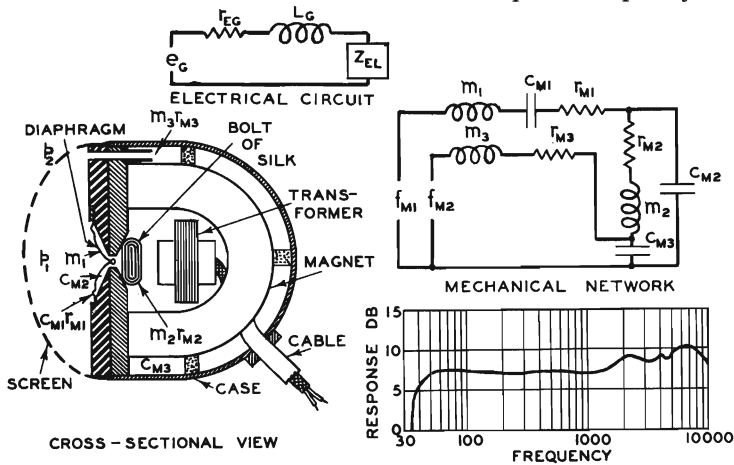

FIG. 8.11. Cross-sectional view, electrical circuit and mechanical network of an inductor microphone. In the electrical circuit, r_{EG} = the electrical resistance of the conductor. L = the inductance of the conductor. z_{EL} = the electrical impedance of the load. e_G = the open circuit voltage developed in the conductor. In the mechanical network, m_1, r_{M1}, and C_{M1} = the mass, mechanical resistance, and compliance of the diaphragm and conductor, m_2 and r_{M2} = the mass and mechanical resistance of the bolt of silk. C_{M2} = the compliance of the air chamber behind the diaphragm. m_3 and r_{M3} = the mass and mechanical resistance of the air in the tube. C_{M3} = the compliance of the case volume. f_{M1} and f_{M2} = the driving forces. $f_{M1} = p_1A$, $f_{M2} = p_2A$. A = the area of the diaphragm. p_1 = the pressure at the diaphragm. p_2 = the pressure at the tube. The graph shows the free space, open-circuit, voltage response frequency characteristic.

2. *Inductor Microphone*[20] (*Straight-Line Conductor*).—The inductor microphone is another example of a moving conductor microphone. A cross-sectional view of this microphone is shown in Fig. 8.11. The diaphragm, $r_{M1}C_{M1}m_1$, of this microphone is "V" shaped with a straight conductor

[19] Marshall and Romanow, *Bell Syst. Tech. Jour.*, Vol. 15, No. 3, p. 405, 1936.
[20] Olson, H. F., U.S. Patent 2,106,224.

located in the bottom of the "V." The mechanical network of this micro-phone is the same as that of the dynamic microphone in the preceding section and the action is the same. A transformer, housed in the magnet structure, is used to step up the low electrical impedance of the conductor to that suitable for transmission over a line of several hundred feet.

3. *Ribbon Microphone.*—The pressure ribbon microphone[21],[22] consists of a light metallic ribbon suspended in a magnetic field and freely accessible to the atmosphere on one side and terminated in an acoustical resistance on the other side. The essential elements are shown schematically in Fig. 8.12. These elements may take various forms as, for example, the pipe is usually coiled in the form of a labyrinth (see Fig. 8.14).

The acoustical network[23] of the pressure ribbon microphone is shown in Fig. 8.12.

The inertance and acoustical capacitance of the ribbon are designated by M_R and C_{AR}.

The acoustical resistance and mass of the air load upon the ribbon are designated by r_{AA} and M_{AA}. The expression for the air load upon the ribbon will now be derived. The pressure, in dynes per square centimeter, at a distance a in centimeters, from an elementary source is (see Sec. 2.2)

$$p = \frac{dS}{4\pi a} j\omega\rho u_{\max}\epsilon^{j\omega t}\epsilon^{-jka} \qquad 8.17$$

where dS = area of the source, in square centimeters,

$u_{\max}$ = maximum velocity of dS, in centimeters per second,

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

$\omega = 2\pi f$,

f = frequency, in cycles per second,

u = velocity over the surface dS, in centimeters per second,

t = time, in seconds,

$k = 2\pi/\lambda$, and

λ = wavelength, in centimeters.

The pressure at any point on the ribbon due to a velocity $u_{\max}\epsilon^{j\omega t}$ of the ribbon is

$$p = \frac{j\omega\rho}{4\pi} u_{\max}\epsilon^{j\omega t}\int\int \frac{dS}{a_1} \epsilon^{-jka_1} \qquad 8.18$$

where a_1 = radius vector having the shortest air distance from the point 1 to the surface element dS. To compute the total force, the above integra-tion must be performed and then the resulting pressure integrated over the surface of the ribbon.

[21] Olson, H. F., U.S. Patent 2,102,736.
[22] Olson, H. F., *Jour. Soc. Mot. Pic. Eng.*, Vol. 27, No. 3, p. 285, 1936.
[23] Olson, H. F., *Broadcast News*, No. 30, p. 3, May, 1939.

The total force is

$$f_{MA} = \frac{j\omega\rho u_{max}\epsilon^{j\omega t}}{4\pi} \iint dS' \iint \frac{dS}{a_1} \epsilon^{jka_1}$$ 8.19

where $dS' = $ surface element at 1.

The acoustical impedance due to the air load is

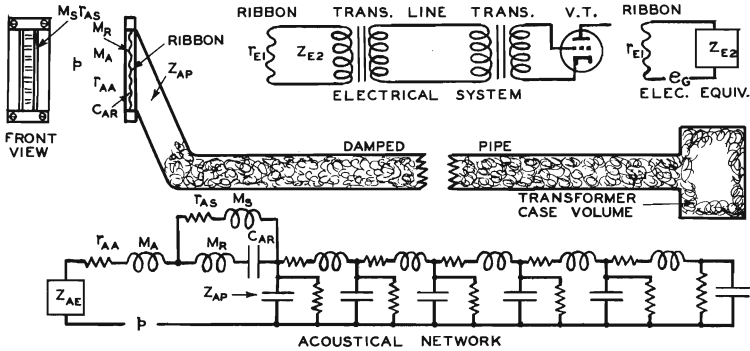$$z_{AA} = r_{AA} + jx_{AA} = \frac{f_{MA}}{A^2 u_{max}\epsilon^{j\omega t}}$$ 8.20

FIG. 8.12. Schematic view, electrical system and its equivalent and acoustical network of a pressure ribbon microphone. In the electrical circuit, $r_{E1} = $ the electrical resistance of the ribbon. $z_{E2} = $ the external electrical impedance load presented to the ribbon. $e_G = $ the open circuit voltage developed by the ribbon. In the acoustical network, M_A and $r_{AA} = $ the inertance and acoustical resistance of the air load on the ribbon. M_R and $C_{AR} = $ the inertance and acoustical capacitance of the ribbon. M_S and $r_{AS} = $ the inertance and acoustical resistance of the slit. $z_{AE} = $ the acoustical impedance due to the electrical circuit. $z_{AP} = $ the acoustical impedance of the pipe. $p = $ the sound pressure.

The ribbon is spaced from the pole pieces of the magnetic structure to allow freedom of motion. This slit or aperture, r_{AS} and M_{AS}, gives rise to an acoustical impedance (see Sec. 5.4),

$$z_{AS} = r_{AS} + j\omega M_S$$ 8.21

where $r_{AS} = $ acoustical resistance of the slit, in acoustical ohms, and

$M_S = $ inertance of the slit, in grams per (centimeter).[4]

The back of the ribbon is terminated in an acoustical resistance in the form of a finite pipe damped with tufts of felt. The acoustical network of the pipe shows that for the mid- and high-frequency range the acoustical impedance is an acoustical resistance.

The acoustical resistance of the pipe referred to the ribbon is

$$r_{AP} = \frac{42}{A_P}$$ 8.22

where $A_P = $ area of the pipe, in square centimeters.

The acoustical impedance due to the electrical circuit may influence the motion of the ribbon. The acoustical impedance due to the electrical circuit is

$$z_{AE} = \frac{(Bl)^2}{A_R{}^2 z_{ET}}$$

8.23

where z_{ET} = total electrical impedance in the ribbon circuit, in abohms,

A_R = area of the ribbon, in square centimeters,

B = flux density in gausses, and

l = length of the ribbon, in centimeters.

The acoustical impedance, z_{AE}, due to the electrical circuit, and the acoustical impedance, z_{AS}, due to the aperture between the ribbon and pole pieces, are in general small compared to the other impedances in the system save at the very low frequencies.

The acoustical impedance characteristics of the elements of a pressure ribbon microphone are shown in Fig. 8.13.

The volume current of the ribbon, in cubic centimeters per second, is given by

$$U = \frac{p}{r_{AP} + r_{AA} + jx_{AR} + jx_{AA} - jx_{AP}}$$

8.24

where r_{AP} = acoustical resistance of the pipe, in acoustical ohms,

r_{AA} = acoustical resistance of the air load upon the ribbon, in acoustical ohms,

x_{AR} = acoustical reactance of the inertance and acoustical capacitance of the ribbon, in acoustical ohms,

x_{AA} = acoustical reactance of the air load upon the ribbon, in acoustical ohms,

x_{AP} = acoustical reactance of the pipe, in acoustical ohms, and

p = sound pressure, in dynes per square centimeter.

The volume current of the ribbon and the phase angle between the volume current and pressure computed from equation 9.24 is shown in Fig. 8.13.

The velocity of the ribbon, in centimeters per second, is

$$\dot{x} = \frac{U}{A_R}$$

8.25

The voltage, in abvolts, generated in the ribbon is given by

$$e = Bl\dot{x}$$

8.26

where B = flux density, in gausses, and

l = length of the ribbon, in centimeters.

The shape of the voltage curve will be the same as that of U_V in Fig. 8.13. This assumes that the pressure is the same for all frequencies. However, due to the obstacle effect (see Sec. 1.11), the pressure on the ribbon increases at the higher frequencies and the output is practically independent of the frequency.

An example of an unobtrusive microphone[24] employing the vibrating system of Fig. 8.12 is shown in Fig. 8.14. The ribbon is terminated in a damped pipe. The upper portion of the microphone is equipped with a pipe
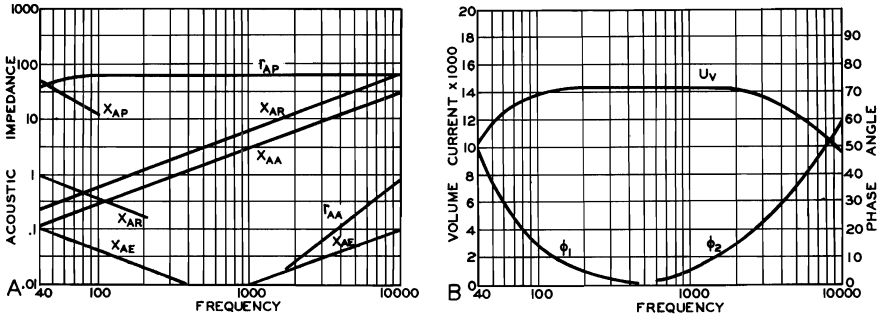

FIG. 8.13. A. The acoustical impedance characteristics of the elements of the ribbon pressure microphone. x_{AR} = the ribbon acoustical reactance. x_{AA} = the air load acoustical reactance. r_{AA} = the air load acoustical resistance. r_{AP} = the acoustical resistance of the pipe. x_{AP} = the acoustical reactance of the pipe. x_{AE} = the acoustical reactance due to the electrical system. p = the sound pressure. B. U_V = the volume current of the ribbon for a sound pressure of 1 dyne per square centimeter. ϕ = the phase angle between the ribbon volume current and the driving pressure. ϕ_1 = leading. ϕ_2 = lagging.

coupled to the ribbon. This provides a small pickup area and, therefore, a nondirectional characteristic. The surge acoustical impedance of the pipe is the same as the acoustical impedance of the damped pipe. Under these conditions a smooth response frequency characteristic is obtained because the sound flows into the pickup pipe, past the ribbon and then into the damped pipe without reflections. A small horn is used at the pickup point to accentuate the response in the frequency region above 5000 cycles.

4. *Probe Microphone.*[25,26]—In some acoustical measurements, a microphone equipped with a small sound pickup system with a high acoustical impedance so that it will not disturb the sound field is a useful tool. A probe-type microphone shown in Fig. 8.15 satisfies these requirements. The probe-type microphone consists of a tube about 4 inches in length and an inside diameter of from .020 to .1 inch coupled to a small condenser microphone. The consideration of the attenuation of sound in tubes as a function of the frequency and diameter will be found in Sec. 5.32. The

---

[24] Olson and Preston, *Audio Engineering*, Vol. 34, No. 7, p. 18, 1950.
[25] Hilliard, John K., *Trans. IRE*, Prof. Group Audio, Vol. AU-2, No. 6, p. 168, 1954.
[26] Benson, Robert W., *Jour. Acous. Soc. Amer.*, Vol. 25, No. 1, p. 128, 1953.

frequency discrimination introduced by the probe can be compensated in the microphone amplifier. The large attenuation in small tubes reduces the effects of resonance in the tube with the result that the response frequency characteristic is smooth and free of peaks and dips.
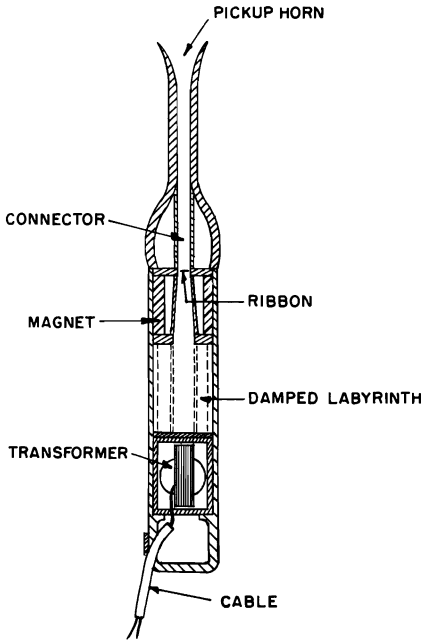

FIG. 8.14. Cross-sectional view of a ribbon-type pressure microphone.
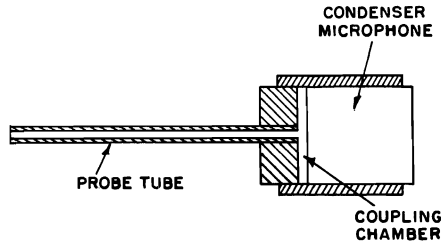
FIG. 8.15. Cross-sectional view of a probe-type microphone consisting of a tube of small bore coupled to a condenser microphone.

5. *Comparison of Electrodynamic Microphones.*—At this point it appears appropriate to digress and examine the characteristics of the two most common electrodynamic microphones, namely, the ribbon type and diaphragm-voice coil type.

The high-frequency response of any microphone is an inverse function of the mass of the vibrating system. The use of negative acoustical reactance elements to reduce the effective acoustical impedance over a limited frequency range can be applied to any system and does not alter the fundamental relationship between mass and high-frequency response. These facts can be deduced by the application of the Reactance Theorem to acoustical networks.

In a moving conductor system, there are three fundamental parameters, namely, the acoustical impedance, the flux density, and the conductor. The ultimate flux density is limited by magnetic materials and, therefore, can be made the same for any system. In the ribbon system, the entire system acts as both conductor and diaphragm. In the moving coil system, the mass

must be divided between two parts, namely, the diaphragm and the conductor. The graph of Fig. 8.16 depicts the sensitivity of a moving conductor microphone. This data shows that the highest sensitivity is obtained when the mass of the diaphragm is zero, namely, the conductor also acts as the diaphragm. Therefore, the ultimate sensitivity in the high-frequency region will always be higher in a ribbon system than in a diaphragm and moving coil system.

A diaphragm-type dynamic microphone requires a rigid diaphragm system in order to prevent spurious responses due to relative motion of different parts of the diaphragm. Since the ribbon serves a dual purpose of diaphragm and voice coil, rigidity is not required. Therefore, the response frequency characteristic of a ribbon system is smoother than the diaphragm-voice coil system.
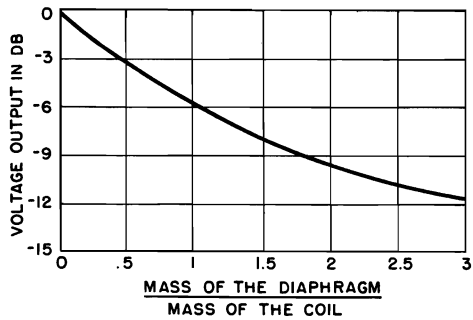
FIG. 8.16. The voltage output of a dynamic microphone as a function of the ratio of the mass of the diaphragm to the mass of the coil. The ribbon microphone corresponds to the value zero for this ratio.

Another consideration is the low-frequency response. In well-designed ribbon-type microphones, the resonant frequency of the ribbon can be placed below the audible range. Therefore, above the audible range, the diaphragm and conductor system is mass controlled. This simple mass characteristic makes it a particularly simple task to develop suitable phase shifting networks for ribbon transducers.

The transient response is another important characteristic of a microphone. It can be shown that in a well-designed system, the one with the lowest inherent mass will exhibit the most faithful response to transients. Therefore, the ribbon system will exhibit the best transient response. This has been substantiated by large- and small-scale explosion tests.

Closely allied to transient response is wind noise. In order to determine the wind response of different microphones, a wind generator, described in Sec. 10.2G, was developed. This wind machine delivers a combination of a steady and a fluxuating air stream and, therefore, simulates wind under actual conditions. Tests have been made comparing the wind-noise response of different microphones obtained with natural wind and the wind machine. These results show that there is practically a perfect correlation. The advantages of the wind machine are as follows: wind response can be obtained at any time and the wind noise produced by the machine remains constant. Using the wind machine, tests have shown that with the same screening and the same response frequency characteristic, the response to wind of all microphones is directly related to the sensitivity, which means the signal to wind noise is the same.

As regards the relationship of weight and over-all sensitivity of microphones, the following conclusion can be drawn: for the same response frequency characteristic, the weight of a ribbon microphone does not differ from that of a moving coil microphone.

E. *Magnetic Microphones.*[27],[28]—A magnetic microphone consists of a diaphragm acted upon by sound waves and connected to an armature which varies the reluctance in a magnetic field surrounded by a coil. Fig. 8.17.

Two different types of magnetic transducers used in magnetic microphones are shown in Fig. 8.17. These two transducers are the most common types used in magnetic microphones.

PERSPECTIVE VIEW   PERSPECTIVE VIEW

DIAPHRAGM   ARMATURE   DIAPHRAGM   ARMATURE

COIL   COIL

MAGNET   MAGNET

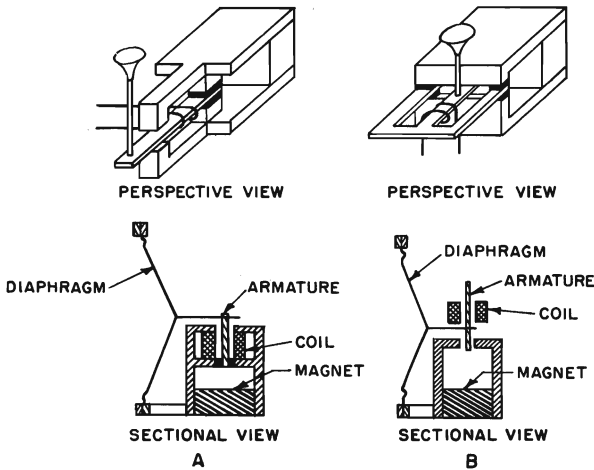SECTIONAL VIEW   SECTIONAL VIEW

A   B

FIG. 8.17. Perspective and sectional views of magnetic microphones of the balanced armature type.

In the magnetic system of Fig. 8.18, let the armature be deflected a distance Δx, in centimeters, from the center position. The flux through the armature will be

$$\Delta \phi = \frac{MA\,\Delta x}{4a^2} \qquad\qquad 8.27$$

where ϕ = flux, in maxwells,

     M = magnetomotive force of the steady field, in gilberts,

     A = area of the pole pieces in square centimeters, and

     a = spacing between the pole pieces and the armature.

In this consideration it is assumed that the area of the top and bottom pole pieces and the spacing between the armature and the top and bottom pole

[27] Bauer, B. B., *Trans. IRE*, Prof. Group on Audio, Vol. AU-1, No. 6, p. 4, 1953.
[28] Bauer, B. B., *Jour., Acous. Soc. Amer.*, Vol. 25, No. 5, p. 867, 1953.

pieces are all the same. It is also assumed that all the reluctance resides in the air gap.

The same result may be obtained from the magnetic network of Fig. 8.18. When the armature is deflected a distance Δx, there will be a change in the reluctance ΔR in the upper branches 1 and 2 of the magnetic network. The flux $\Delta \phi$ through the branch 3 which is the armature when the armature is deflected a distance ΔR is given by

$$\Delta \phi = \frac{M \Delta R}{4R^2} \qquad\qquad 8.28$$

where $R = \dfrac{a}{A}$, and

$$\Delta R = \frac{\Delta x}{A}$$

Equation 8.28 may be written

$$\Delta \phi = \frac{M A \Delta x}{4a^2} \qquad\qquad 8.29$$

Equation 8.29 is the same as equation 8.27.
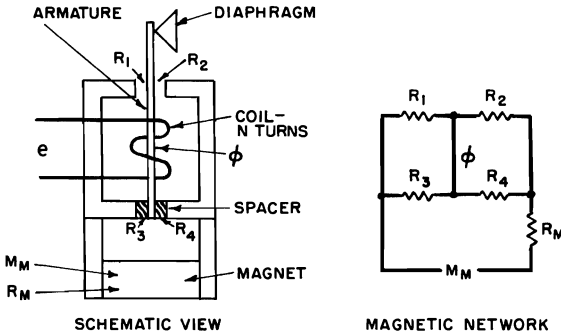

SCHEMATIC VIEW   MAGNETIC NETWORK

FIG. 8.18. Schematic view and magnetic circuit of a magnetic generator transducer. R = magnetic reluctance of each of the four air gaps between the armature and pole pieces. ΔR = the change in the reluctance R due to a displacement of the armature. M = the magnetomotive force of the magnet. ϕ = the magnetic flux in the armature.

Employing equation 8.27 or 8.29, the change in flux with respect to time is

$$\frac{\Delta \phi}{\Delta t} = \frac{M A}{4a^2} \frac{\Delta x}{\Delta t} \qquad\qquad 8.30$$

or

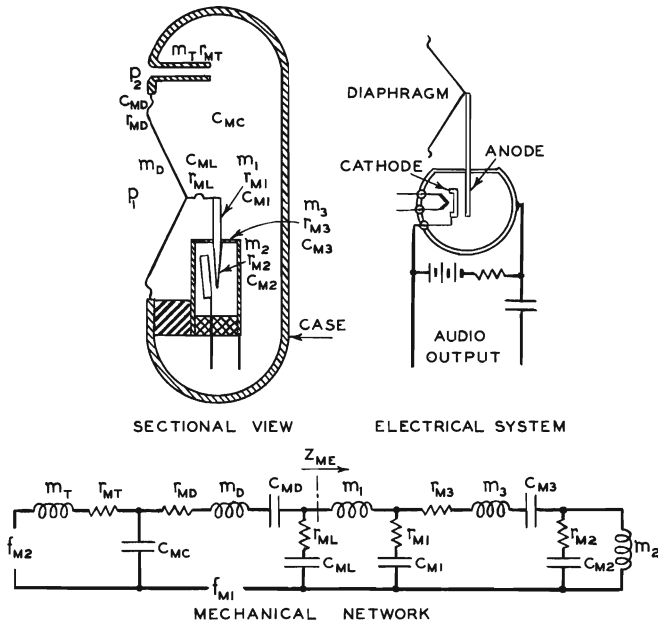$$\frac{d\phi}{dt} = \frac{M A}{4a^2} \frac{dx}{dt} \qquad\qquad 8.31$$

FIG. 8.19. Sectional view, electrical system, and mechanical network of an electronic microphone. In the mechanical network, z_{ME} = the mechanical impedance of the electronic transducer. m_1, r_{M1}, and C_{M1} = the effective mass, mechanical resistance, and compliance of the outer portion of the bar. m_2, r_{M2}, and C_{M2} = the mass, mechanical resistance, and compliance of the inner portion of the bar or anode. m_3, r_{M3}, and C_{M3} = the mass, mechanical resistance, and compliance of the diaphragm of the electronic transducer. m_D, r_{MD}, and C_{MD} = the mass, mechanical resistance, and compliance of the diaphragm and suspension. m_T and r_{MT} = the mass and mechanical resistance of the air in the tube. r_{ML} and C_{ML} = the mechanical resistance and compliance of the link connecting the diaphragm and transducer. C_{MC} = compliance of case volume. f_{M1} and f_{M2} = the driving forces. $f_{M1} = p_1 A$ and $f_{M2} = p_2 A$. A = the area of the diaphragm. p_1 = the sound pressure at the diaphragm. p_2 = the sound pressure at the tube opening. The electrical system shows the wiring diagram for a diode type electronic transducer.

The electromotive force e, in abvolts, generated in the coil is given by

$$e = N \frac{d\phi}{dt} = \frac{NMA}{4a^2} \dot{x}$$
    8.32

where $\dot{x}$ = velocity of the armature, in centimeters per second, and

N = number of turns in the coil.

Equation 8.32 shows that the open circuit voltage is proportional to the velocity of the armature. Therefore, to obtain constant output for constant

sound pressure on the diaphragm, the velocity of the armature must be independent of the frequency. Therefore, the system must be resistance controlled to obtain a constant relationship between the voltage output and the impinging sound pressure. This can be accomplished by means of an acoustical resistance behind the diaphragm similar to that of the dynamic microphone described in Sec. 8.2D1.

F. *Electronic Microphone*.[29]—An electronic microphone is a microphone in which the output results from the motion of one of the elements in a vacuum tube.

A schematic view of an electronic microphone is shown in Fig. 8.19. The voltage output of an electronic transducer is given by

$$e = Kx_3 \qquad 8.33$$

where K = constant of the system, and

x_3 = amplitude of the element.

The output of the electronic microphone may be computed from the mechanical network of Fig. 8.19.

The two driving forces f_{M1} and f_{M2}, in dynes, are equal and opposite in phase. The driving force f_{M1} is given by

$$f_{M1} = p_1 A_D \qquad 8.34$$

where A_D = area at the diaphragm, in square centimeters, and

p_1 = sound pressure at the diaphragm, in dynes per square centimeter.

The driving force f_{M2} is given by

$$f_{M2} = p_2 A_D \qquad 8.35$$

where A_D = area of the diaphragm, in square centimeters, and

p_2 = sound pressure at the port, in dynes per square centimeter.

At the high frequencies the mechanical reactance due to the compliance, C_{MC}, is small compared to the mechanical impedance of the port, m_T, r_{MT}. Under these conditions, the system is driven by f_{M1}. At the extreme low frequencies the mechanical reactance of the compliance, C_{MC}, is large compared to the mechanical impedance of the port, m_T, r_{MT}. Since f_{M1} and f_{M2} are of opposite phase, the net driving force is practically zero. In the region where the mechanical reactance due to the compliance, C_{MC}, and the mechanical reactance due to the port, r_{MT}, m_{MT}, are comparable, the addition of this mechanical network introduces a phase shift of such magnitude that both forces, f_{M1} and f_{M2}, contribute in driving the mechanical system.

[29] Olson, H. F., *Jour. Acous. Soc. Amer.*, Vol. 19, No. 2, p. 307, 1947.

The response may be obtained from a consideration of the mechanical network. The mechanical network of Fig. 8.19 may be reduced to the mechanical network of Fig. 8.20 in which

$$z_{M1} = r_{MT} + j\omega m_T \qquad 8.36$$

$$z_{M2} = \frac{1}{j\omega C_{MC}} \qquad 8.37$$

$$z_{M3} = r_{MD} + j\omega m_D + \frac{1}{j\omega C_{MD}} \qquad 8.38$$

$$z_{M4} = r_{ML} + \frac{1}{j\omega C_{ML}} \qquad 8.39$$

$$z_{M5} = z_{ME} \qquad 8.40$$

where r_{MT} = mechanical resistance of the tube, in mechanical ohms,

m_T = mass of the air in the tube, in grams,

C_{MC} = compliance of the case volume, in centimeters per dyne,

r_{MD} = mechanical resistance of the diaphragm, in mechanical ohms,

m_D = mass of the diaphragm, in grams,

C_{MD} = compliance of the diaphragm, in centimeters per dyne,

r_{ML} = mechanical resistance of the coupling link, in mechanical ohms,

C_{ML} = compliance of the coupling link, in centimeters per dyne, and

$z_{M5} = z_{ME}$ the mechanical impedance of the electronic transducer, in mechanical ohms.

The mechanical impedance of the electronic transducer is

$$z_{ME} = z_{M5}$$
$$= \frac{z_{M6}(z_{M7}+z_{M8})\,(z_{M9}+z_{M10})+z_{M9}z_{M10}(z_{M6}+z_{M7})+z_{M7}z_{M8}(z_{M9}+z_{M10})}{(z_{M7}+z_{M8})\,(z_{M9}+z_{M10})+z_{M9}z_{M10}} \qquad 8.41$$

where $z_{M6} = j\omega m_1$

$$z_{M7} = r_{M1} + \frac{1}{j\omega C_{M1}}$$

$$z_{M8} = r_{M3} + j\omega_3 + \frac{1}{j\omega C_{M3}}$$

$$z_{M9} = r_{M2} + \frac{1}{j\omega C_{M2}}$$

$z_{M10} = j\omega m_2$

The amplitude, in centimeters, due to the driving force f_{M1} is

$$x_1 = \frac{f_{M1}}{j\omega\left(z_{M3} + \dfrac{z_{M1}z_{M2}}{z_{M1}+z_{M2}} + \dfrac{z_{M4}z_{M5}}{z_{M4}+z_{M5}}\right)} \qquad 8.42$$

The amplitude, in centimeters, due to the driving force f_{M2} is

$$x_2 = \frac{-f_{M2}z_{M2}(z_{M4}+z_{M5})}{j\omega[z_{M1}(z_{M2}+z_{M3})(z_{M4}+z_{M5})+z_{M4}z_{M5}(z_{M1}+z_{M2})+z_{M2}z_{M3}(z_{M4}+z_{M5})]} \quad 8.43$$

The amplitude, in centimeters, of m_2 is

$$x_3 = \frac{f_M z_{M7} z_{M9}}{j\omega[z_{M6}(z_{M7}+z_{M8})(z_{M9}+z_{M10})+z_{M9}z_{M10}(z_{M6}+z_{M7})+z_{M7}z_{M8}(z_{M9}+z_{M10})]} \quad 8.44$$

where

$$f_M = \frac{(z_{M4}z_{M5})(x_1+x_2)}{z_{M4}+z_{M5}} j\omega$$

The amplitude response characteristic can be obtained from equation 8.44 and the constants of the system. The voltage output can be obtained from the amplitude and equation 8.33.

The electrical connections for the electronic microphone are shown in Fig. 8.19.

8.3. Velocity Microphones. — *First-Order Gradient Microphones.* —A pressure gradient microphone is a microphone in which the electrical response corresponds to the difference in pressure between two points in space. In general, when the distance between these two points is small compared to the wavelength, the
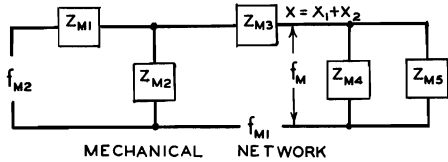

MECHANICAL NETWORK

FIG. 8.20. Mechanical network of Fig. 8.10 in terms of the following:

$$z_{M1} = r_{MT} + j\omega m_T$$

$$z_{M2} = \frac{1}{j\omega C_{MC}}$$

$$z_{M3} = r_{MD} + j\omega m_D + \frac{1}{j\omega C_{MD}}$$

$$z_{M4} = r_{ML} + \frac{1}{j\omega C_{ML}}$$

$$z_{M5} = z_{ME}$$

pressure gradient corresponds to the particle velocity. 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. The acoustical and electrical elements which form the coupling means, between the atmosphere and the electrical system, for transforming the sound vibrations into the corresponding electrical variations, may be arranged in innumerable ways to obtain pressure gradient or velocity microphones. It is the purpose of this section to consider pressure gradient and velocity microphones.

A. *Pressure Gradient Microphone.*[30,31,32,33,34]—The response of a pressure gradient microphone, as the name implies, is a function of the difference

[30] Olson, H. F., *Jour. Soc. Mot. Pic. Eng.*, Vol. 16, No. 6, p. 695, 1931.
[31] Olson, H. F., *Jour. Acous. Soc. Amer.*, Vol. 3, No. 1, p. 56, 1931.
[32] Olson, H. F., *Proc. Inst. Rad. Eng.*, Vol. 21, No. 5, p. 655, 1933.
[33] Massa, F., *Jour. Acous. Soc., Amer.*, Vol. 10, No. 3, p. 173, 1939.
[34] Anderson and Wigginton, *Audio Engineering*, Vol. 34, No. 1, p. 12, 1950.

in sound pressure between two points. Obviously, a pressure gradient microphone may be built in a number of ways. One type of pressure gradient microphone consists of two pressure actuated units, separated by a very small distance, with the electrical outputs connected in opposition. Figure 8.21 schematically depicts the essential elements of a pressure gradient microphone. A cylinder of mass m is coupled to a conductor located in a magnetic field. The cylinder is assumed to be the only portion of the system which will be influenced by sound waves. The diameter of the cylinder is assumed to be small compared to the wavelength. Therefore, the average intensity will be the same for all points on the surface of the cylinder. The vibrating system is assumed to be constrained so that the only motion possible is one in a direction parallel to the longitudinal axis of the cylinder. Under these conditions the vibrating system is driven by the difference between the forces on the two ends of the cylinder due to the impinging sound wave.

FIG. 8.21. Pressure gradient microphone.

Assume a plane sound wave, from equation 1.22, the pressure, in dynes per square centimeter, at $x = 0$ may be written

$$p = kc\rho A \sin (kct)$$
$$p = p_m \sin kct \qquad\qquad 8.45$$

where c = velocity of sound, in centimeters per second,

$k = 2\pi/\lambda$,

λ = wavelength, in centimeters,

ρ = density, in grams per cubic centimeter,

A = amplitude of ϕ,

ϕ = velocity potential, and

p_m = maximum sound pressure, in dynes per square centimeter.

The pressure at the end of the cylinder $x_1 = -\Delta x/2$ for a direction of propagation θ is

$$p_1 = p_m \sin k \left(ct + \frac{\Delta x}{2} \cos \theta\right) \qquad\qquad 8.46$$

The pressure at the other end of the cylinder $x_2 = \Delta x/2$ is

$$p_2 = p_m \sin k \left(ct - \frac{\Delta x}{2} \cos \theta\right) \qquad\qquad 8.47$$

The difference in pressure between the two ends of the cylinder is

$$\Delta p = p_1 - p_2 = 2p_m \cos (kct) \sin \left(\frac{k\Delta x}{2} \cos \theta\right) \qquad\qquad 8.48$$

The driving force, in dynes, available for driving the cylinder along the x axis is

$$f_M = S\Delta p = 2Sp_m \cos (kct) \sin \left(\frac{k\Delta x}{2} \cos \theta\right) \qquad 8.49$$

where $S =$ area of the end of the cylinder, in square centimeters.

If Δx is small compared to the wavelength the driving force is

$$f_M = S \frac{2\pi f}{c} p_m \Delta x \cos \theta \cos kct \qquad 8.50$$

A comparison of equations 8.45 and 8.50 shows that for a wave of constant sound pressure the driving force is proportional to the frequency.

The velocity of the mechanical system, for Δx small compared to the wavelength, is

$$\dot{x} = \frac{f_M}{j\omega m} = \frac{Sp_m}{jcm} \Delta x \cos \theta \cos kct = \frac{Sp_m}{cm} \Delta x \cos \theta \sin kct \qquad 8.51$$

where $m =$ mass of the cylinder, in grams, and
$\omega = 2\pi f, f =$ frequency, in cycles per second.

This quantity is independent of the frequency and as a consequence the ratio of the generated voltage to the pressure in the sound wave will be independent of the frequency.

The velocity of the mechanical system for any value of Δx is

$$\dot{x} = \frac{2Sp_m}{m\omega} \sin (kct) \sin \left(\frac{k\Delta x}{2} \cos \theta\right) \qquad 8.52$$

$$\dot{x} = \frac{2Sp_m}{m\omega} \sin (kct) \sin \left(\frac{kD}{2} \cos \theta\right) \qquad 8.53$$

where $D =$ distance between the two ends of the cylinder.

The voltage output, in abvolts, of the conductor is

$$e = Bl\dot{x} \qquad 8.54$$

where $B =$ flux density in the field in which the conductor moves, in gausses,
$l =$ length of the conductor, in centimeters, and
$\dot{x} =$ velocity of the conductor, in centimeters per second.

The response frequency characteristic of a mass controlled, dynamic pressure gradient microphone computed from equations 8.53 and 8.54 is shown in Fig. 8.22.

The directional characteristics of a pressure gradient system of the type shown in Fig. 8.21 and computed from equation 8.53 are shown in Fig. 8.23. It will be seen that when the ratio D is greater than $\lambda/4$ the directional pattern becomes progressively broader as the frequency increases. In the case of the baffle-type ribbon microphone, the directional characteristics
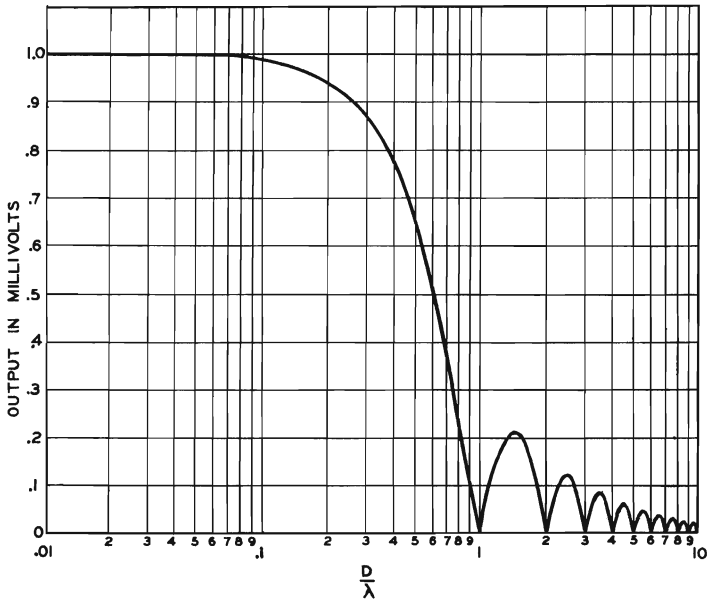
FIG. 8.22. Computed open-circuit voltage response frequency characteristic of a pressure gradient, mass-controlled, electrodynamic microphone.
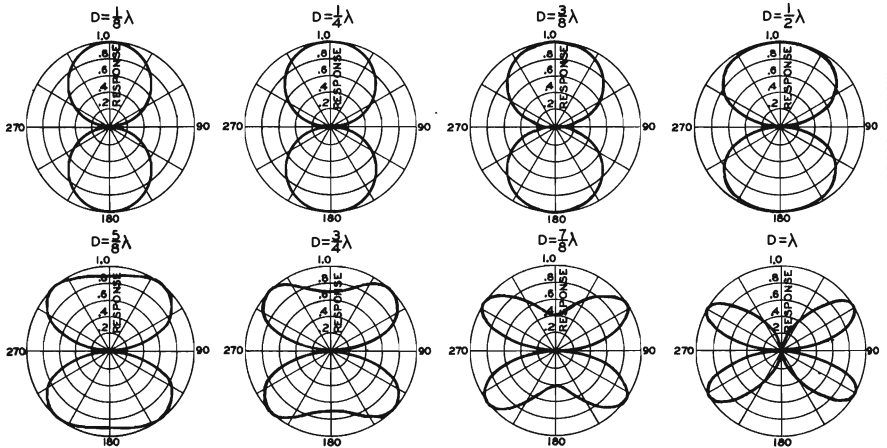

FIG. 8.23. Directional characteristics of a pressure gradient microphone as a function of the dimensions and the wavelength. The polar graph depicts the output, in volts, as a function of the angle, in degrees. The maximum response is arbitrarily chosen as unity.

first become sharper than the cosine pattern and then broader as the dimensions become comparable to the wavelength. In other words, the doublet theory is not in accord with the observed results. Of course, deviations would be expected when the dimensions of the baffle become comparable to the wavelength because of variations in both intensity and phase due to changes in the diffraction of sound by the baffle.

The above considerations have been concerned with a plane wave. From equation 1.40 the pressure component in a spherical wave is

$$p = \frac{kc\rho A}{r} \sin k(ct - r)$$ 8.55

Let the distance on the axis of the cylinder between the sources and point x_2 and x_1 on the cylinder be $r - \Delta x/2$ and $r + \Delta x/2$ (Fig. 8.12). The difference in pressure between the two ends of the cylinder is

$$\Delta p = kc\rho A \left[\frac{2r \cos k(ct - r) \sin \left(\frac{kD}{2}\right) + 2D \sin k(ct - r) \cos \left(\frac{kD}{2}\right)}{r^2 - \left(\frac{D}{2}\right)^2} \right]$$ 8.56

If D is small compared to r and kD is small compared to unity, equation 8.56 becomes approximately

$$\Delta p = kc\rho AD \left[\frac{kr \cos k(ct - r) + \sin k(ct - r)}{r^2} \right]$$ 8.57

This equation is similar to equation 1.42 for the particle velocity in a spherical sound wave. Therefore, the voltage output of this microphone corresponds to the particle velocity in a sound wave. The response of a pressure gradient microphone as a function of the distance from a point source and the frequency is shown in Fig. 8.40A.

B. *Velocity Microphone.*[35,36,37]—Free-ribbon microphones are used for all types of sound collection. Essentially these microphones consist of a loosely stretched ribbon suspended in the air gap between two pole pieces (Fig. 8.24). In addition to supplying the flux to the air gap, the pole pieces serve as a baffle for acoustically separating the two sides of the ribbon. The configuration and dimensions of the baffle determine the effective sound path between the two sides of the ribbon. Under the influence of a sound wave the ribbon is driven from its equilibrium position by the difference in pressure between the two sides. The motion of the ribbon in the magnetic field induces a voltage between the two ends of the ribbon. The electrical output of this system under certain conditions corresponds to the particle velocity in a sound wave. Accordingly, the term velocity microphone has been applied to the free-ribbon microphone. In past analysis it has been customary to treat the system as an acoustical doublet. This method is essentially accurate when the effective dimensions of the baffle are small

[35] Olson, H. F., *Jour. Soc. Mot. Pic. Eng.*, Vol. 16, No. 6, p. 695, 1931.
[36] Olson, H. F., *Jour. Acous. Soc. Amer.*, Vol. 3, No. 1, p. 56, 1931.
[37] Olson, H. F., *Proc. Inst. Rad. Eng.*, Vol. 21, No. 5, p. 655, 1933.

compared to the wavelength. When the effective dimensions are comparable to the wavelength, there is considerable discrepancy between the simple doublet theory and the actual performance. It is the purpose of
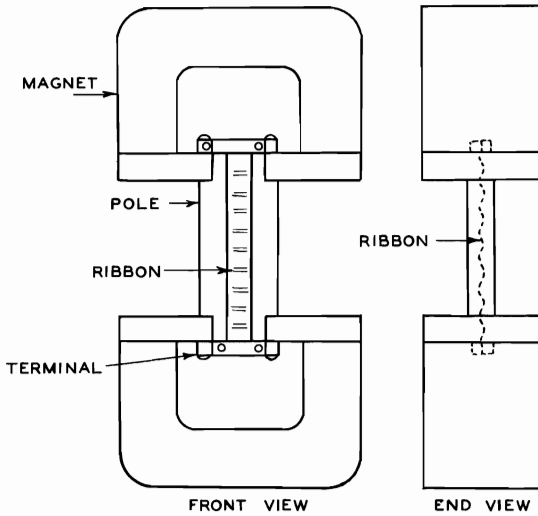

FIG. 8.24. The essential elements of a velocity microphone.

this section to develop the theory of the conventional baffle-type velocity microphone.

Approximate solutions for the diffraction of sound by a circular and square plate have been obtained.[38] These analyses may be applied to the problem of the baffle-type ribbon microphone.

The ratio of the pressure at the center of a circular plate for any angle of the incident sound is

$$\frac{p_\theta}{p} = 1 + \frac{\cos \theta}{\sqrt{1 - \sin^2 \theta}} \left[1 - \epsilon^{jkR} \sum_{u=0}^{\infty} \epsilon_u j^u \mu^u J_u \left(kR \sin \theta \right) \right] \qquad 8.58$$

where $\mu = \dfrac{1 - \sqrt{1 - \sin^2 \theta}}{\sin \theta}$,

$\epsilon_u = 1$ when $u = 0$,

$\epsilon_u = 2$ when $u \neq 0$,

$\theta = $ angle of the incidence,

$R = $ radius of the plate, in centimeters,

$k = 2\pi/\lambda$,

$\lambda = $ wavelength, in centimeters, and

$J_u = $ Bessel function, of the order u.

[38] Sivian and O'Neil, *Jour. Acous. Soc. Amer.*, Vol. 3, No. 4, p. 483, 1932.

The pressure at the center on the front and back of a circular plate for normal incidence $\theta = 0°$ or $180°$, from equation 8.58, is

$$\left|\frac{p_0}{p}\right| = \sqrt{5 - 4 \cos kR} \qquad\qquad 8.59$$

$$\left|\frac{p_{180}}{p}\right| = 1 \qquad\qquad 8.60$$

The pressure frequency characteristic on the front and back of a circular baffle for normal incidence computed from equations 8.59 and 8.60 is shown in Fig. 8.25. It will be seen that the pressure at the front rises to a value
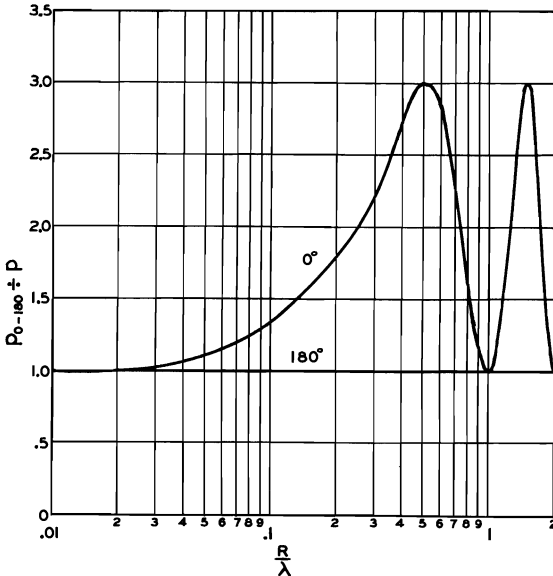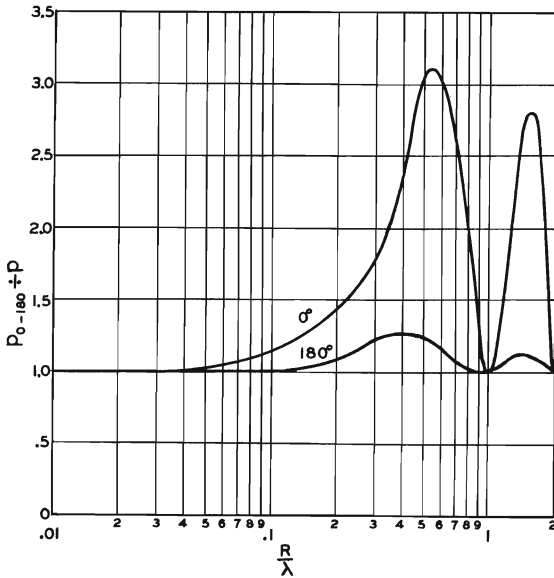

FIG. 8.25. Computed pressure frequency characteristic, at the center, on the front, and the back of a circular baffle for normal incidence of the impinging sound wave.

of three times that in free space at $R/\lambda = .5$, then falls back to the same as the free space pressure at $R/\lambda = 1$, and repeats for $R/\lambda = 1.5$ and $R/\lambda = 2$, etc. The pressure at the back is the same as the free space pressure for all frequencies. The measured pressure at the center on the front and back of a circular baffle is shown in Fig. 8.26. In order to reduce errors in measurement to a minimum, baffles of different diameters were used. In addition, several different pressure measuring arrangements were used. The results shown in Fig. 8.26 represent an average of these determinations. It will be seen that theory and experiment are in fairly good agreement. Some of the deviation may be attributed to finite size of the pressure measuring system.

The phase angles at the front and back of a circular baffle computed from equation 8.58 are shown in Fig. 8.27. A point in the plane wave corresponding to the plane of the baffle is the reference plane for the phase. It will be seen that for R/λ less than .5 the phase of the pressure at the front of the baffle leads that of the pressure in the wave. For values of R/λ less

FIG. 8.26. Measured pressure frequency characteristic at the center, on the front, and the back of a circular baffle for normal incidence of the impinging sound wave.

than .1 the phase on the front leads by the same amount as the phase on the back lags the pressure in the wave.

Equation 8.58 may be used to compute the difference in pressure between the two sides of a relatively small ribbon located in a large baffle (Fig. 8.28). The difference in pressure between the two sides of the ribbon in a circular baffle, Fig. 8.28, is

$$\Delta p = p_\theta - p_{\theta + 180} \qquad\qquad 8.61$$

where p_θ and $p_{\theta + 180}$ may be obtained from equation 8.58. The acoustical impedance of the ribbon, Fig. 8.28, is given by

$$z_{AR} = j\omega M_R - \frac{j}{\omega C_{AR}} \qquad\qquad 8.62$$

where M_R = inertance of the ribbon, and

C_{AR} = acoustical capacitance of the ribbon.

From equation 8.28 the total force of the air load upon the ribbon is

$$f_{MA} = \frac{j\omega\rho u_{max}\epsilon^{j\omega t}}{4\pi} \iint dS' \iint \frac{dS}{a_1}\epsilon^{jka_1} \qquad 8.63$$

The above integration extends over both sides of the ribbon and cognizance must be taken of the 180° difference in phase between the front
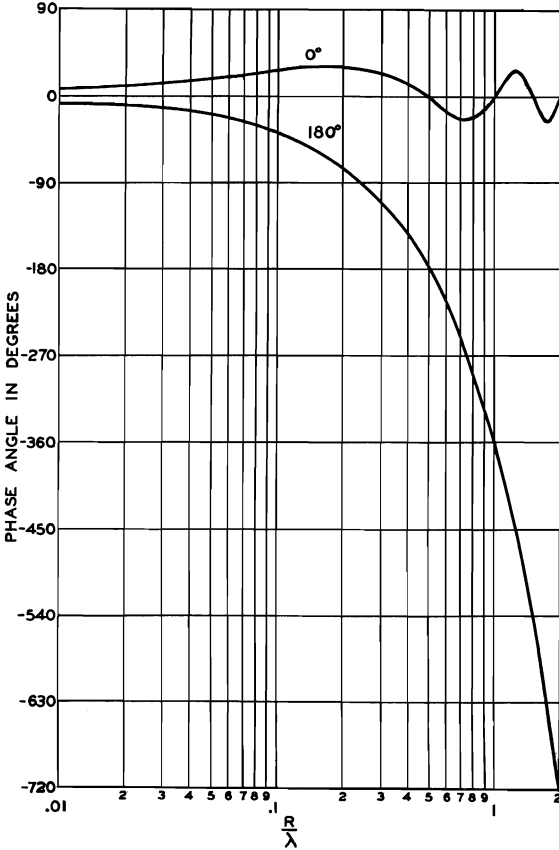

FIG. 8.27. Computed phase frequency characteristic at the center, on the front, and the back of a circular baffle for normal incidence of the impinging sound wave.

and back when integrating between the two surfaces. The integration of equation 8.63 may be carried out by dividing the ribbon into small elements and carrying out the indicated integration.

The acoustical impedance of the air load is

$$z_{AA} = r_{AA} + jx_{AA} = \frac{f_{MA}}{A_R{}^2 u_{max}\epsilon^{j\omega t}} \qquad 8.64$$

The acoustical impedance, z_{AS}, of the slit between the ribbon and pole pieces is given by equation 8.21. The acoustical impedance due to the electrical circuit is given by equation 8.23.

The resonant frequency of the ribbon is usually placed below the audible limit. Therefore, the acoustical capacitance of the ribbon may be neglected.

FIG. 8.28. A velocity microphone with a large circular baffle.

The acoustical resistance, r_{AA}, of the air load is negligible save at the very high frequencies. When the fundamental resonant frequency of the ribbon is located below the audible-frequency range, the negative reactance term in equation 8.62 may be neglected. Under these conditions the acoustical impedance of the vibrating system is

$$z_{AT} = j\omega M_R + j\omega M_{AA} \qquad 8.65$$

where M_{AA} = inertance of the air load.

The velocity, in centimeters per second, of the ribbon is

$$\dot{x} = \frac{\Delta p}{A_R z_{AT}} \qquad 8.66$$

where A_R = area of the ribbon, in square centimeters.

The voltage output in abvolts is

$$e = Bl\dot{x} \qquad 8.67$$

where B = flux density, in gausses,

   l = length of the ribbon, in centimeters, and

   $\dot{x}$ = velocity of the ribbon, in centimeters per second.

The response characteristic of a mass-controlled ribbon located in a large circular baffle, Fig. 8.28, computed from equation 8.67 is shown in Fig. 8.29. The experimental response of a ribbon microphone with a circular baffle is shown in Fig. 8.30. The agreement between the measured response and the computed response is quite good. There is some deviation between $R/\lambda = .5$ and $R/\lambda = .8$. There is also some discrepancy in this region between computed and measured pressures (Figs. 8.25 and 8.26). It is interesting to note that the theoretical response of the pressure gradient microphone and the ribbon in a baffle is practically the same, Figs. 8.22 and 8.29.

The measured directional characteristic of the ribbon microphone with a circular baffle is shown in Fig. 8.31. It will be seen that, for small values of R/λ, the directional characteristic corresponds to a cosine function. Between $R/\lambda = \frac{3}{8}$ and $\frac{5}{8}$ the directional pattern is sharper than a cosine characteristic. Then for R/λ larger than $\frac{5}{8}$ the characteristics broaden and assume irregular shapes. The theoretical directional characteristics employing equations 8.58, 8.61, 8.66, and 8.67 are shown in Fig. 8.32. It will be seen that the agreement with the experimental results of Fig. 8.31 is
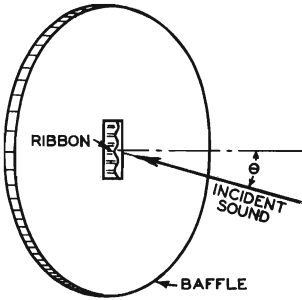
quite good. There is some deviation for $D/\lambda = \frac{3}{4}$. It is in this region that deviations occurred between the theoretical and experimental results for the pressure, Figs. 8.25 and 8.26, and for the response, Figs. 8.29 and 8.30. The theoretical directional characteristic for a doublet, Fig. 8.23, becomes progressively broader for $R/\lambda = \frac{3}{8}, \frac{1}{2},$ and $\frac{5}{8}$ and does not agree at all with the experimental results. For $R/\lambda = \frac{3}{4}, \frac{7}{8},$ and 1 the shape of the theoretical directional characteristic of the doublet does not correspond with the experimental results. Summarizing, the theoretical directional characteristics of a ribbon microphone with a circular baffle agree within a few per
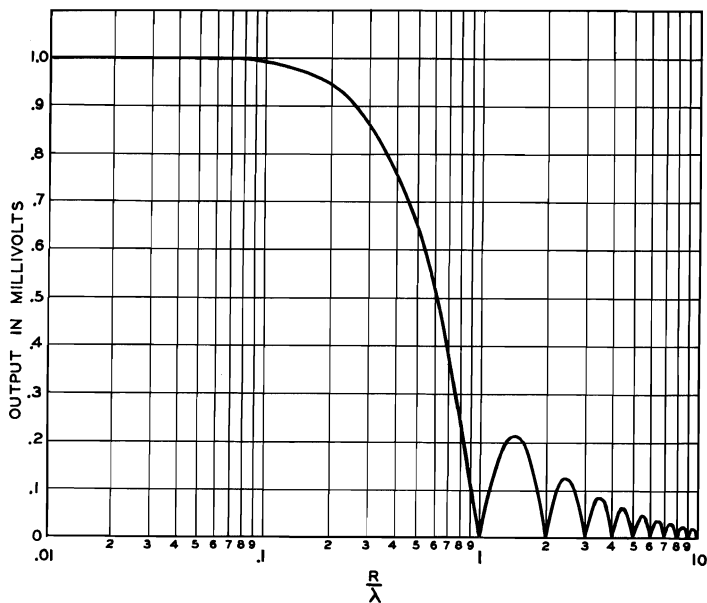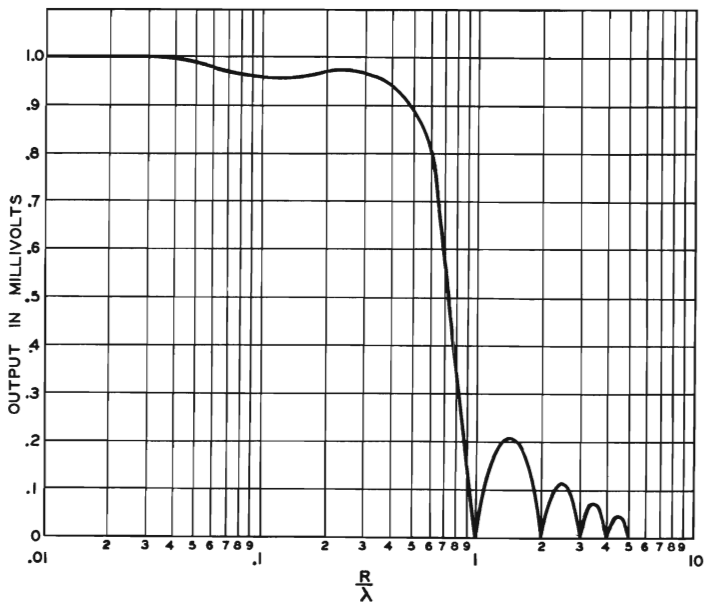

FIG. 8.29. Computed open-circuit voltage response frequency characteristic of a mass-controlled, electrodynamic ribbon located in a large circular baffle.

cent of the measured directional characteristics. However, the discrepancy between the measured directional characteristics of a ribbon in a circular baffle and the theoretical directional characteristics of a doublet or pressure gradient system is very large for values of R/λ greater than $\frac{3}{8}$.

The phase between the actuating force, equation 8.61, and the particle velocity in a plane wave, for a ribbon microphone with a circular baffle, is shown in Fig. 8.33. It will be seen that this force leads the particle velocity by 90° for small values of R/λ. The phase angle between the voltage output of the ribbon and the particle velocity is also shown in Fig. 8.33. For small values of R/λ the voltage output of a mass-controlled, dynamic ribbon microphone with a baffle corresponds to the particle velocity in the sound wave.

FIG. 8.30. Measured open-circuit voltage response frequency characteristic of a mass-controlled, electrodynamic ribbon located in a large circular baffle.
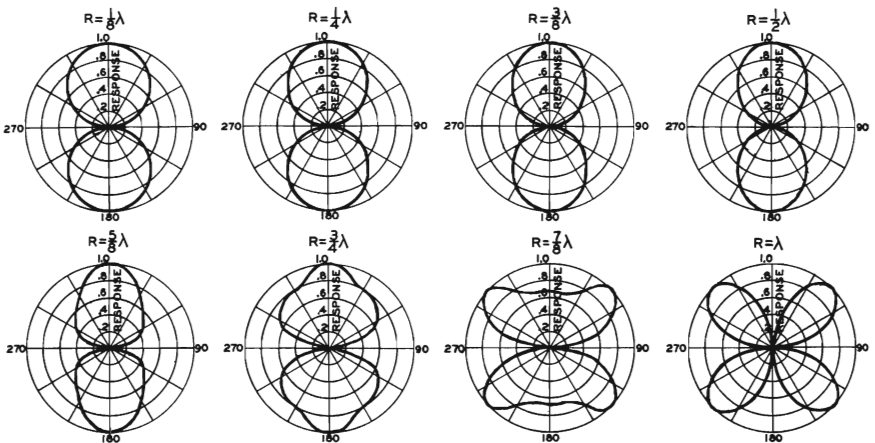

FIG. 8.31. Measured directional characteristics of a ribbon microphone with a large circular baffle (see Fig. 8.28) as a function of the radius of the baffle and the wavelength. The polar graph depicts the output, in volts, as a function of the angle, in degrees. The maximum response is arbitrarily chosen as unity.
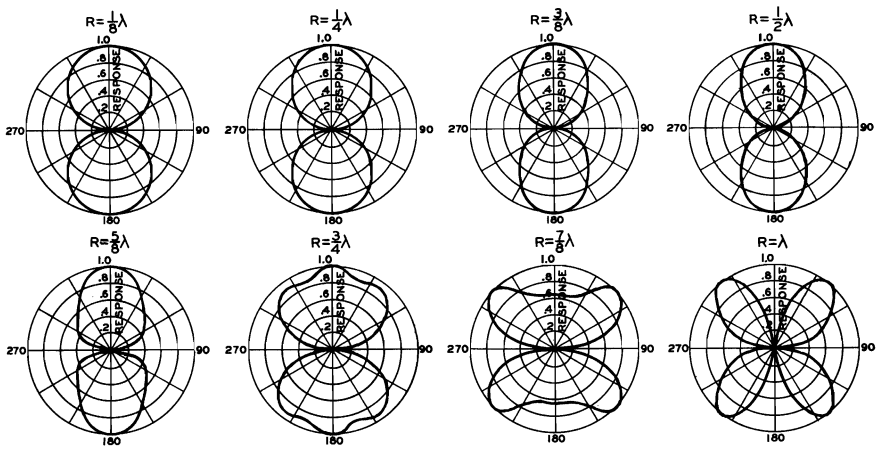
FIG. 8.32. Computed directional characteristics of a ribbon microphone with a large circular baffle (see Fig. 8.28) as a function of the radius of the baffle and the wavelength. The polar graph depicts the output, in volts, as a function of the angle, in degrees. The maximum response is arbitrarily chosen as unity.
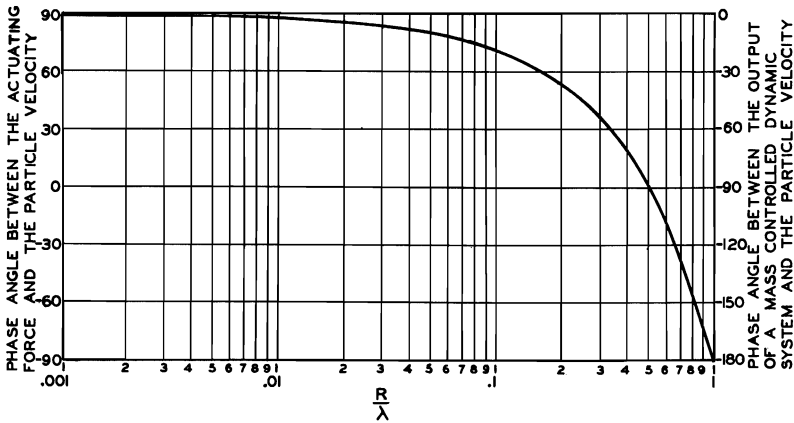

FIG. 8.33. The phase angle, in degrees, between the actuating force and the particle velocity for a mass-controlled ribbon with a circular baffle as a function of R/λ. The phase angle between the voltage output of a mass-controlled, electrodynamic ribbon located in a magnetic field as a function of R/λ.

The above analysis has been concerned with a ribbon located in a circular baffle. Irregular baffles instead of circular baffles are used in commercial microphones for two reasons: first, a suitable magnetic field results in an irregular baffle and, second, the sound path lengths between the two sides of an irregular baffle differ and, as a consequence, it is possible to obtain uniform directional response characteristics over a wide frequency

range. An analytical solution of the irregular plate is difficult. However, the graphical method may be used and is very effective.

In well-designed velocity microphones which have been built in the past, the effective sound path introduced by the baffle has been made less than one-half wavelength for all frequencies within the useful range. There are two reasons for this selection of sound path: first, the response up to this frequency is quite uniform, while above this frequency the response falls off rapidly with increase of the frequency; second, in the case of an irregular baffle the directional characteristics are of the cosine type to within a few per cent of this frequency limit. A commercial microphone is shown in Fig. 8.24. It will be seen that the effective baffle is irregular in shape. The directional characteristics of the microphone of Fig. 8.24
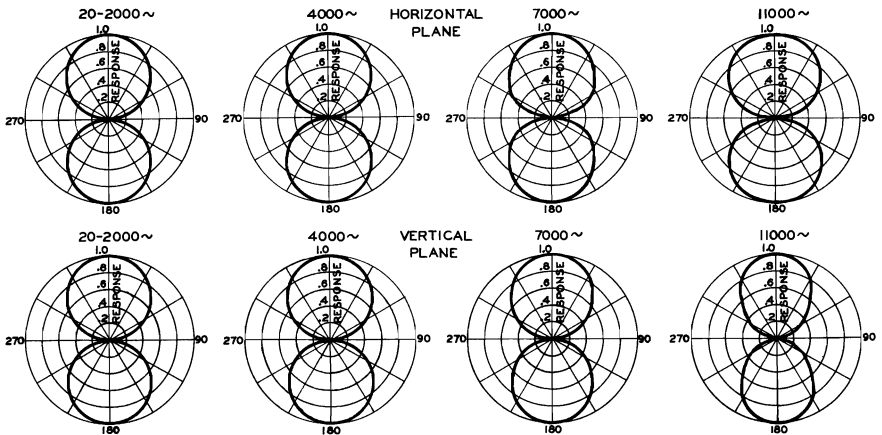

FIG. 8.34. The directional characteristics of the velocity microphone shown in Fig. 8.24. The polar graph depicts the output, in volts, as a function of the angle, in degrees. The maximum response is arbitrarily chosen as unity.

are shown in Fig. 8.34. Further, the deviation from a cosine characteristic is very small.

The above considerations have been concerned with a plane wave. As in the case of the pressure gradient microphone, it can be shown that the output of a baffle-type velocity microphone corresponds to the particle velocity in a spherical wave. The response of a baffle-type velocity microphone as a function of the distance from a point source and the frequency is shown in Fig. 8.40A.

The response of the baffle-type velocity microphone may be obtained from the acoustical network of Fig. 8.35 and the acoustical impedance characteristics of the acoustical elements of the system of Fig. 8.36. The ribbon is 2.2 inches in length and .2 inch in width. The flux density is 9000 gausses. The open-circuit voltage generated of the ribbon is given by equation 8.26. The computed voltages are indicated by the dots in
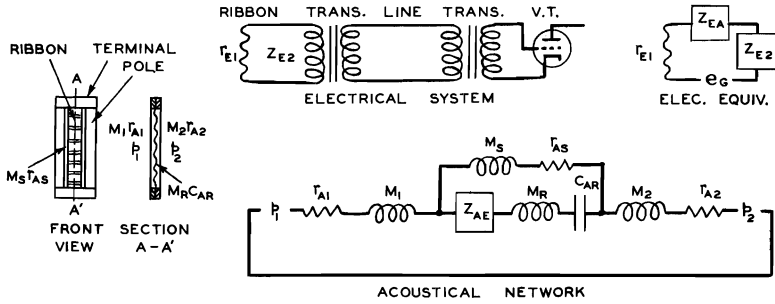
FIG. 8.35. Front and sectional views of the vibrating system of a velocity microphone. In the electrical circuit, e = the open circuit voltage. r_{E1} = the electrical resistance of the ribbon. z_{E2} = the electrical impedance of the load upon the ribbon due to the transformers and vacuum tube. z_{EA} = the electrical impedance due to the acoustical system. In the acoustical network, r_{A1}, M_1, r_{A2}, and M_2 = the acoustical resistances and inertances due to the air load on the front and back of the ribbon. r_{AS} and M_S = the acoustical resistance and inertance due to the slit between the ribbon and the pole pieces. M_R and C_{AR} = the inertance and acoustical capacitance of the ribbon. z_{AE} = the acoustical impedance due to the electrical system. p_1 and p_2 = the sound pressure at the front and back of the ribbon.
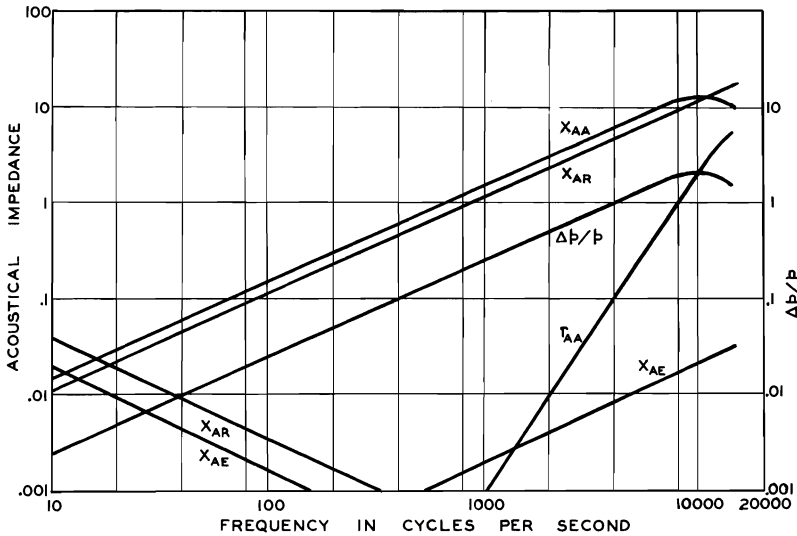

FIG. 8.36. The acoustical impedance characteristics of the velocity microphone. x_{AR} = the positive and negative acoustical reactances due to the inertance and the acoustical capacitance of the ribbon. x_{AA} = the acoustical reactance due to the air load. r_{AA} = the acoustical resistance due to the air load. x_{AE} = the positive and negative acoustical reactances due to the electrical system. $\Delta p/p$ = the ratio of the difference in pressure between the front and back of the ribbon and the free-field pressure.

Fig. 8.37.   The experimental response frequency characteristic of the micro-
phone of Fig. 8.24 is shown in Fig. 8.37.   The agreement between the
theoretical computed characteristic and the experimental determined
characteristic is very good.   A transformer is used to step up the electrical
impedance of the ribbon to 250 ohms which is suitable for transmission
over a line.   The characteristics shown in Fig. 8.37 depict the open-circuit
voltage from the terminals of the 250-ohm transformer of Fig. 8.35.   The
line electrical impedance is stepped up to 50,000 ohms at the grid of the
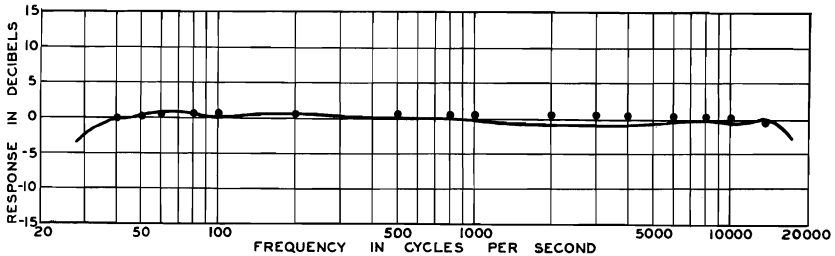vacuum tube.   The input to the vacuum tube of Fig. 8.35 is 23 db above

FIG. 8.37.   Open-circuit voltage response frequency characteristics of a velocity
microphone at the terminals of the 250-ohm output.   0 db = 600 microvolts per
dyne per square centimeter.   When the impedance is stepped up to 50,000 ohms for
the input to the grid of a vacuum tube, the voltage is 8.4 millivolts per dyne per
square centimeter.   The dots represent the computed response and the solid line, the
measured response.

the voltage in the line.   This is .0085 volt per dyne per square centimeter
at the grid of the vacuum tube.

In general, the electrical load, z_{E2}, of the transformer and vacuum tube
is large compared to the electrical resistance of the ribbon and the motional
electrical impedance, z_{EA}.   Under these conditions the voltage delivered at
the grid is the same open-circuit voltage developed by the ribbon multiplied
by the step-up ratio of the transformers.   However, if the electrical resist-
ance of the ribbon and the motional electrical impedance are comparable in
magnitude to the electrical load impedance, cognizance must be taken of the
electrical resistance of the ribbon and the motional electrical impedance
when computing the voltage developed across the load.   In the equivalent
electrical circuit of Fig. 8.35, the motional electrical impedance, in abohms,
is given by

$$z_{EA} = \frac{(Bl)}{A_R^2 \, z_{AT}} \qquad\qquad 8.68$$

where z_{AT} = the total acoustical impedance,
    A_R = area of the ribbon, in square centimeters,
    B = the flux density, in gausses, and
    l = the length of the ribbon, in centimeters.

The total acoustical impedance z_{AT} is the acoustical impedance at the
point z_{AE} of Fig. 8.35 with the acoustical impedance due to the electrical

circuit considered to be zero. In the same way, in computing the acoustical impedance z_{AE}, of the acoustical network of Fig. 8.35, from equation 8.23 and the electrical impedance at e in the electrical circuit, the electrical impedance z_{EA} due to the acoustical system is considered to be zero.

8.4. Unidirectional Microphones.—A unidirectional microphone is a microphone with a substantially unidirectional pattern over the response frequency range. Unidirectional microphones 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. It is the purpose of this section to consider combination unidirectional microphones and single-element unidirectional microphones.

A. *Combination Unidirectional Microphones.*—The combination unidirectional microphone[39,40,41] consists of a bidirectional microphone and a nondirectional microphone. A unidirectional microphone consisting of a ribbon velocity element (see Sec. 8.3B and Fig. 8.24) and a ribbon pressure element (see Sec. 8.2D3 and Fig. 8.12) is shown in Fig. 8.38. The damped pipe terminating the back of the pressure ribbon is folded in the form of a labyrinth and enclosed in a case. The velocity and pressure sections are formed from a single continuous ribbon. A common magnetic structure is used for both the velocity and pressure sections. Due to a finite length of pipe for the pressure section the velocity of the pressure ribbon leads the pressure in the sound wave at the low frequencies (see Sec. 8.2D3 and Fig. 8.13). The acoustical resistance (silk cloth) introduces a corresponding shift in the velocity section. At the high frequencies the phase shifts in the two elements are made the same by suitable geometrical configurations of the field structure.
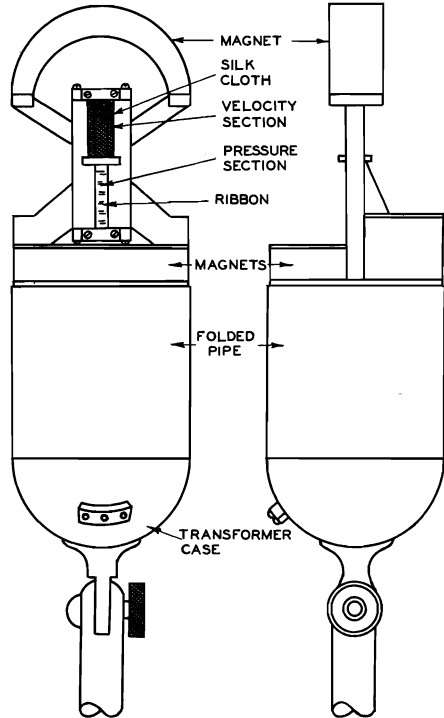

Fig. 8.38. Unidirectional microphone with the screen removed. Ribbon type pressure and velocity elements.

[39] Olson, H. F., *Jour. Acous. Soc. Amer.*, Vol. 3, No. 3, p. 315, 1932.
[40] Weinberger, Olson, and Massa, *Jour. Acous. Soc. Amer.*, Vol. 5, No. 2, p. 139, 1933.
[41] Olson, H. F., *Jour. Soc. Mot. Pic. Eng.*, Vol. 27, No. 3, p. 284, 1936.

A unidirectional microphone[42] consisting of a ribbon velocity element and a dynamic pressure element is shown in Fig. 8.39. Equalizers are used to correct the amplitude and phase of the dynamic element to conform with the velocity element.
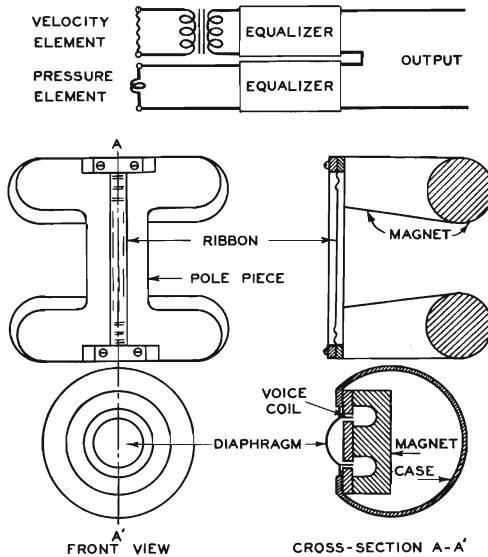

FIG. 8.39. Unidirectional microphone consisting of a ribbon-type velocity element and a dynamic-type pressure element.

1. *The Response of the Unidirectional Microphone as a Function of the Distance and the Frequency.*[43]—The low-frequency response of the velocity microphone is accentuated when the distance between the source and the microphone is less than a wavelength. The same effect occurs to a smaller extent in the unidirectional microphone. It is the purpose of this section to consider the response of the unidirectional microphone as a function of the frequency and distance from a point source.

The voltage output of a nondirectional microphone as a function of the distance, r, is given by

$$e_{ND} = \frac{R_1}{r} \sin \omega t \qquad 8.69$$

where R_1 = sensitivity constant of the microphone,

$\omega = 2\pi f$,

f = frequency, in cycles per second,

r = distance, in centimeters, from a point source of sound, and

t = time, in seconds.

[42] Marshall and Harry, *Jour. Acous. Soc. Amer.*, Vol. 12, No. 4, p. 481, 1941.
[43] Olson, H. F., *Broadcast News*, No. 30, p. 3, May, 1939.

The voltage output of the bidirectional velocity microphone as a function of the distance and the wavelength λ, in centimeters, is

$$e_{BD} = R_2 \left(\frac{1}{r} \sin \omega t - \frac{\lambda}{2\pi r^2} \cos \omega t \right) \cos \theta \qquad 8.70$$

where R = sensitivity constant of the microphone,

    r = distance, in centimeters from a point source of sound, and

    θ = angle between the direction of the incident sound and the normal to the ribbon.

If the output of the unidirectional microphone as a function of the angle θ is to be a cardioid of revolution for plane waves, then R_1 must be made equal to R_2. The ratio of the output of the unidirectional microphone as a function of the distance and frequency as compared to a pressure microphone is

$$\text{Response Ratio} = \sqrt{\frac{\left(\frac{1}{r} + \frac{\cos \theta}{r} \right)^2 + \left(\frac{\lambda \cos \theta}{2\pi r^2} \right)^2}{\left(\frac{2}{r} \right)^2}} \qquad 8.71$$

This ratio for $\theta = 0$, 30°, 60°, 90° 120°, 150°, and 180° for 1, 2, and 5 feet is shown in Fig. 8.40. The same ratio for a conventional velocity microphone for 1, 2, and 5 feet is shown in Fig. 8.40. It will be seen that the accentuation in the unidirectional microphone is smaller than in the case of the velocity microphone.

    2. *Efficiency of Energy Response to Random Sounds of the Unidirectional Microphone as a Function of the Relative Sensitivities of the Bidirectional and Nondirectional Microphones.*[44]—The unidirectional microphone consists of the combination of a bidirectional microphone, in which the output is a function of the cosine of the angle of incidence, and a nondirectional microphone. In general, it is customary to make the output of the bidirectional microphone, for $\theta = 0$, equal to the nondirectional microphone. For this condition the directional characteristic is a cardioid of revolution. In the case of both the bidirectional and the cardioid unidirectional microphones, the ratio of energy response to generally reflected sound is one-third that of a nondirectional microphone. It is interesting to investigate the efficiency of response to random sound of other ratios of sensitivity of the bidirectional to the nondirectional unit.

    The voltage output of a microphone consisting of a bidirectional and nondirectional unit is given by

$$e_{UD} = R_1 + R_2 \cos \theta \qquad 8.72$$

where R_1 = voltage putput of the nondirectional microphone, and

    R_2 = voltage output of the bidirectional unit for $\theta = 0$.

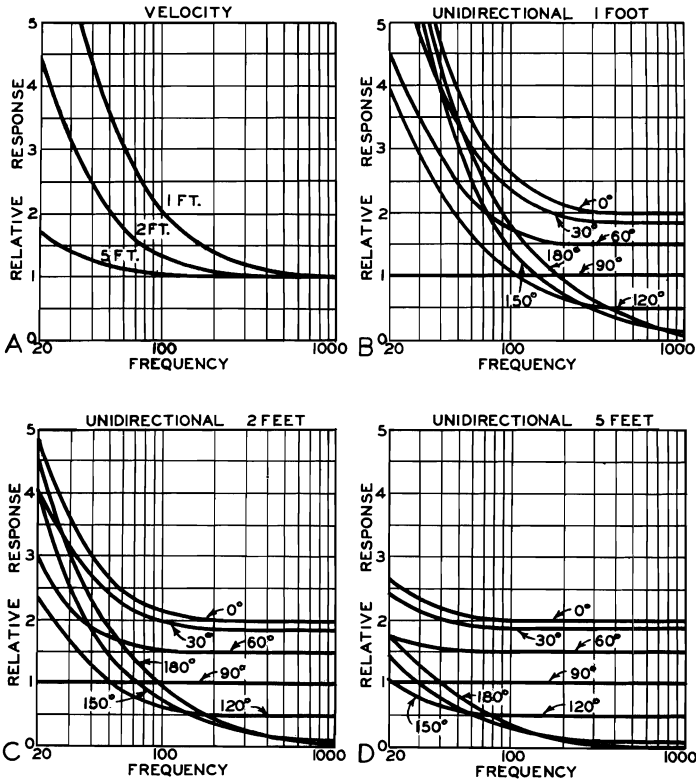[44] Olson, H. F., *Broadcast News*, No. 30, p. 3, May, 1939.

FIG. 8.40. A. The relative voltage output of a velocity (or pressure gradient) microphone as compared to a nondirectional pressure microphone for distances of 1, 2, and 5 feet. B, C, D, the relative voltage output of a unidirectional microphone as compared to a nondirectional pressure microphone for distances of 1, 2, and 5 feet and for various angles of the incident sound.

The efficiency of energy response of the unidirectional microphone as compared to a nondirectional microphone for sounds originating in random directions, all directions being equally probable, from equation 8.117, is

$$\text{Directional efficiency} = \frac{2\pi \int_0^\pi (R_1 + R_2 \cos \theta)^2 \sin \theta \, d\theta}{4\pi (R_1 + R_2)^2}$$

$$= \frac{1}{6} \frac{(R_1 + R_2)^3 - (R_1 - R_2)^3}{(R_1 + R_2)^2 R_2} \qquad 8.73$$

For the standard velocity microphone $R_1 = 0$, $R_2 = 1$, and the ratio is $\frac{1}{3}$. For the cardioid unidirectional $R_1 = 1$, $R_2 = 1$, and the ratio is $\frac{1}{3}$. However, for other values the ratio is different. For example, between $R_1/R_2 = 0$ to $R_1/R_2 = 1$ the efficiency is less than $\frac{1}{3}$ and becomes .25 for

$R_1/R_2 = .33$. The efficiency for various values of the ratio R_1/R_2 is shown in Fig. 8.41. The data in Fig. 8.41 show that it is not important that the two microphones be of the same sensitivity. It is important, however, that the ratio R_1/R_2 be equal to 1 or less than 1.

The same results are shown in Fig. 8.42 by means of polar diagrams. Fig. 8.41 shows that the energy response of the bidirectional microphone
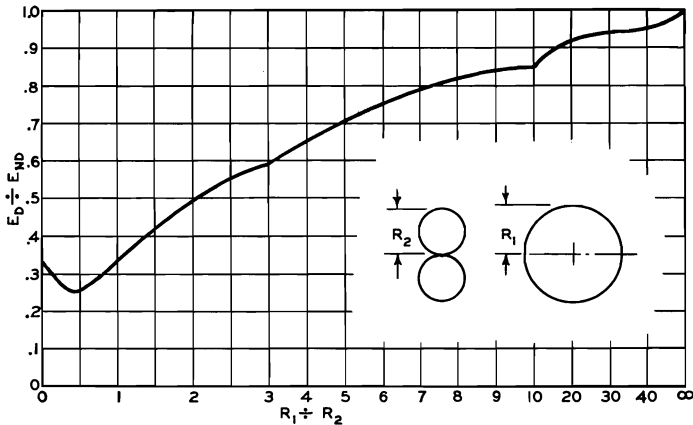

FIG. 8.41. The ratio of energy response to random sounds of a directional microphone consisting of a bidirectional and a nondirectional unit as a function of the ratio of the outputs of the elements, as compared to the nondirectional microphone. E_{ND} = energy response of a nondirectional microphone. E_D = energy response of a directional microphone. R_1 = voltage output of the nondirectional unit. R_2 = voltage output of the directional unit.

and the cardioid unidirectional is the same. However, for $0 < R_1/R_2 < 1$ the response to random sounds is less than in the case of either of these two microphones.

3. *Efficiency of Energy Response to Random Sounds of a Unidirectional Microphone as a Function of the Phase Angle between the two Units.*[45]—The preceding discussions have assumed that the phase angle between the outputs of the two units did not change with frequency. There are two principal sources of phase shift between the two units, namely, a phase shift due to a finite separation, and a phase shift due to a difference in the phase frequency characteristics.

Consider the case in which there is a phase shift ϕ between the output of the bidirectional and nondirectional units. The output of each separate unit is e_0 volts. The output of the combination is

$$e = e_0\sqrt{(\cos\theta + \cos\phi)^2 + (\sin\phi)^2} \qquad 8.74$$

[45] Olson, H. F., *Broadcast News*, No. 30, p. 3, May, 1939.

The efficiency of the energy response of the above system to that of a nondirectional microphone is

Directional efficiency

$$= \frac{2\pi e_0^2 \int_0^\pi [(\cos^2\theta + 2\cos\theta \cos\phi + \cos^2\phi) + \sin^2\phi] \sin\theta \, d\theta}{16\pi e_0^2} \qquad 8.75$$

Directional efficiency

$$= \frac{2\pi e_0^2 \int_0^\pi [\cos^2\theta + 2\cos\theta \cos\phi + 1] \sin\theta \, d\theta}{16\pi e_0^2} = \frac{1}{3} \qquad 8.76$$

The efficiency is the same as in the case of no phase angle shift.
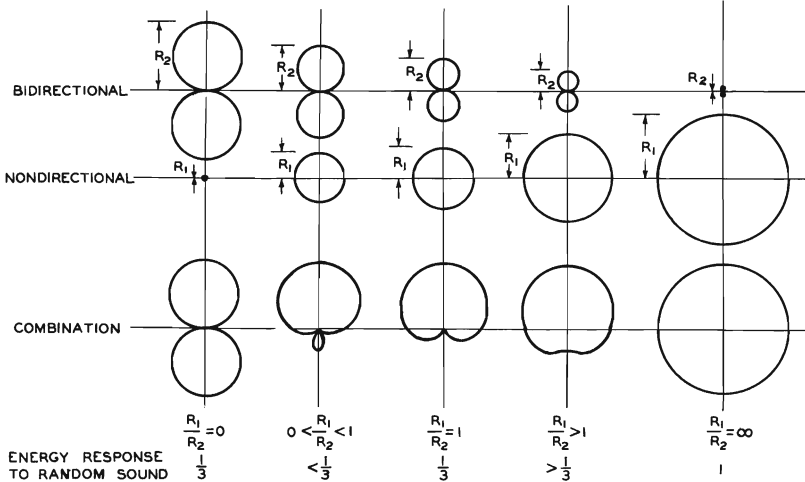
FIG. 8.42. Directional diagrams of various combinations of bidirectional and nondirectional microphones and the energy response to random sounds.

If the units are separated by a finite distance d, then there will be a phase difference between the units which is

$$\phi = \frac{d}{\lambda} 360 \cos\theta \qquad 8.77$$

where d = distance between the units, in centimeters,

λ = wavelength, in centimeters, and

θ = angle between the direction of the incident sound and the normal to the ribbon.

Note that this separation is in line with the units. Substituting $\phi = (d/\lambda) 360 \cos\theta = K \cos\theta$ in equation 8.74 the output is

$$e = e_0 \sqrt{[\cos\theta + \cos(K\cos\theta)]^2 + [\sin(K\cos\theta)]^2} \qquad 8.78$$

The efficiency of the energy response of the above system to a non-directional system is given by
Directional efficiency

$$= \frac{2\pi e_0{}^2 \int_0^\pi \{[\cos \theta + \cos (K \cos \theta)]^2 + [\sin (K \cos \theta]^2\} \sin d\theta}{16\pi e_0{}^2} = \frac{1}{3} \quad 8.79$$

That is, the efficiency is independent of the separation between the units. Of course, for very large distances the separation disturbs the response for $\theta = 0$. However, in the conventional microphone this does not occur. Therefore, the effect of finite size has no effect on the efficiency of energy response to random sounds.

4. *Distortion of the Directional Pattern in the Unidirectional Microphone.*—Deviations from the cardioid characteristic in the unidirectional microphone are due to

1. Phase shift in the velocity microphone due to deviation from a pure mass reactance
2. Phase shift in the velocity microphone due to diffraction
3. Phase shift in the pressure microphone due to deviation from resistance control
4. Phase shift in the pressure microphone due to diffraction
5. Deviation in the output from a cosine directional characteristic in the velocity microphone
6. Deviation in output with angle in the pressure microphone
7. Unequal sensitivity of the two elements

The phase angle between the output of a velocity microphone and the particle velocity in a plane wave has been considered in Sec. 8.3B. The phase shift in a pressure ribbon microphone has been considered in Sec. 8.2D3. It is possible to adjust these phase shifts and those due to diffraction so that the cancellation for 180° will be of the order of −30 db up to 10,000 cycles. In the case of the dynamic pressure unit the problem of maintaining appropriate phase shifts is more difficult.

B. *Single-Element Unidirectional Microphones.*[46]—Unidirectional microphones consisting of the combination of a nondirectional and a bidirectional microphone have been described in Sec. 8.4A. It is the purpose of this section to describe single-element unidirectional microphones in which a unidirectional pattern is obtained by combining a single-element electro-acoustic transducer with a phase-shifting network.

1. *Phase-Shifting Unidirectional Microphone.*—A unidirectional microphone consisting of a nondirectional and bidirectional microphone has been described in the preceding section. It is the purpose of this section to describe other means for obtaining directional response.

The elements of a phase-shifting microphone are shown in Fig. 8.43. The open ends of the pipes are separated by a distance D. A bend of

[46] Olson, H. F., *Jour. Soc. Mot. Pic. Eng.*, Vol. 52, No. 3, p. 293, 1949.

length d is placed in the shorter pipe. The ribbon element measures the difference in pressure between the two pipes. The difference in pressure between the two pipes is given by

$$\Delta p = 2p_0 \sin \left(\frac{d - D}{\lambda} \pi + \frac{D\pi}{\lambda} \cos \theta\right) \qquad 8.80$$

where p_0 = sound pressure, in dynes per square centimeter,

D = separation between the receiving ends of the pipes, in centimeters,

d = acoustic path introduced by the bend, in centimeters,

λ = wavelength, in centimeters, and

θ = angle the incident pencils of sound make with the axis of the system.
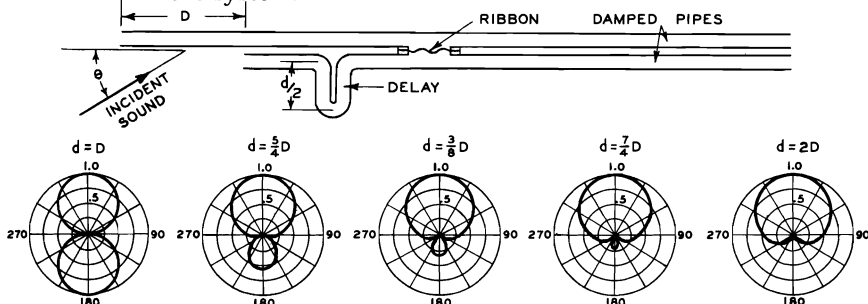
FIG. 8.43. A directional microphone employing a phase shifting system. The polar graphs show the directional characteristics for various ratios of d/D. The polar graph depicts the output, in volts, as a function of the angle, in degrees. The maximum response is arbitrarily chosen as unity.

If the distances D and d are small compared to the wavelength, Δp will be proportional to the frequency. If a mass-controlled, electrodynamic element is used, the output will be independent of the frequency.

A series of directional characteristics for various ratios of D to d is shown in Fig. 8.43

2. *Polydirectional Microphone.*—The single element polydirectional microphone[47] is shown in Fig. 8.44. The ribbon is located in the air gap formed by the pole pieces. A permanent magnet supplies the flux to the air gap. The entire one side of the ribbon is covered by the labyrinth connector. The connector, in turn, is coupled to a damped pipe or labyrinth. The type of directional characteristic is governed by the size of the aperture in the labyrinth connector.

The action of this microphone can be obtained from Fig. 8.45 which shows the schematic view of the microphone and the acoustical network. The sound pressure acting on the open side of the ribbon may be written

$$p_1 = p_{01}\epsilon^{j(\omega t + \phi_1)} \qquad 8.81$$

[47] Olson, H. F., *Proc. Inst. Rad. Eng.*, Vol. 32, No. 2, p. 77, 1944.

where p_{01} = amplitude of the pressure, in dynes per square centimeter,

$\omega = 2\pi f$,

f = frequency, in cycles per second,

t = time, in seconds, and

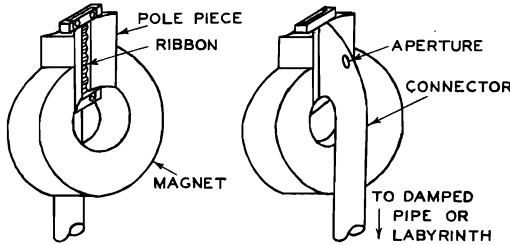ϕ_1 = phase angle with respect to a reference point, in radians.

FIG. 8.44. The elements of a single-ribbon poly-directional microphone.

The sound pressure acting on the aperture in the labyrinth connector may be written

$$p_2 = p_{02}\epsilon^{j(\omega t + \phi_2)} \qquad 8.82$$

where p_{02} = amplitude of the pressure, in dynes per square centimeter, and

ϕ_2 = phase angle with respect to a reference point, in radians.
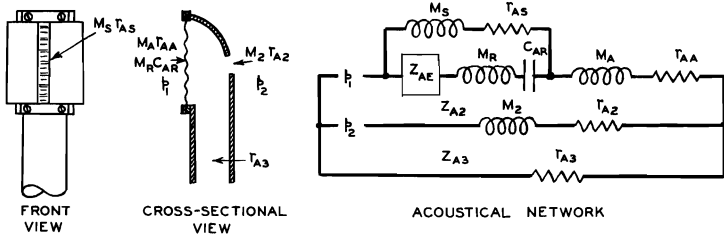
FIG. 8.45. Front view, cross-sectional view, and the acoustical network of a polydirectional microphone. In the acoustical network, M_R and C_{AR} = the inertance and acoustical capacitance of the ribbon. M_A and r_{AA} = the inertance and acoustical resistance of the air load on the front of the ribbon. M_S and r_{AS} = the inertance and acoustical resistance of the slit between the ribbon and pole pieces. M_2 and r_{A2} = the inertance and acoustical resistance of the aperture in the pipe. r_{A3} = the acoustical resistance of the damped pipe. z_{AE} = acoustical impedance due to the electrical system. p_1 = the sound pressure at the front of the ribbon. p_2 = the sound pressure at the back of the connector.

The reference point for the phase may be changed so that

$$p_1 = p_{01}\epsilon^{j(\omega t)} \qquad 8.83$$

and

$$p_2 = p_{02}\epsilon^{j(\omega t + \phi_3)} \qquad 8.84$$

The phase angle ϕ_3 is a function of the angle of the incident sound as follows:

$$\phi_3 = \phi \cos \theta \qquad 8.85$$

where θ = angle between the normal to the surface of the ribbon and the direction of the incident sound,

ϕ = phase angle for $\theta = 0$, and

ϕ = function of the frequency.

The volume current, in cubic centimeters per second, of the ribbon due to the pressure p_1 is

$$\dot{X}_1 = \frac{p_1(z_{A2} + z_{A3})}{z_{A1}z_{A2} + z_{A1}z_{A3} + z_{A2}z_{A3}} \qquad 8.86$$

where $z_{A1} = r_{AA} + j\omega M_A +$

$$\frac{r_{AS} - \omega^2 r_{AS}M_RC_{AR} + j\omega M_S - j\omega^3 M_SM_RC_{AR} - \omega M_SC_{AR}z_{AE} + j\omega C_{AR}r_{AS}z_{AE}}{1 - \omega^2 C_{AR}(M_R + M_S) + j\omega r_{AS}C_{AR} + j\omega C_{AR}z_{AE}}$$

$z_{A2} = r_{A2} + j\omega M_2$

$z_{A3} = r_{A3}$

r_{AA} = acoustical resistance of the air load on the ribbon, in acoustical ohms,

M_A = inertance of the air load on the ribbon, in grams per (centimeter)^4,

r_{AS} = acoustical resistance of the slit between the ribbon and the pole pieces, in acoustical ohms,

M_S = inertance of the slit between the ribbon and the pole pieces, in grams per (centimeter)^4,

M_R = inertance of the ribbon, in grams per (centimeter)^4,

C_{AR} = acoustical capacitance of the ribbon, in (centimeter)^5 per dyne,

r_{A2} = acoustical resistance of the aperture, in acoustical ohms,

M_2 = inertance of the aperture, in grams per (centimeter)^4,

r_{A3} = acoustical resistance of the damped pipe, in acoustical ohms, and

z_{AE} = acoustical impedance due to the electrical circuit in acoustical ohms (see equation 8.23).

Since the acoustical impedance due to the inertance and acoustical resistance of the slit between the ribbon and pole pieces is very large compared to the acoustical impedance of the ribbon, these two elements may be neglected. Further, since the resonant frequency of the ribbon is placed below the audible range, the acoustical impedance due to the acoustical capacitance of the ribbon may be neglected for the audible-frequency range. Then,

$$z_{A1} = r_{AA} + j\omega M_A + j\omega M_R = r_{A1} + j\omega M_1$$

The volume current, in cubic centimeters per second, of the ribbon due to the pressure p_2 is

$$X_2 = \frac{p_2(z_{A3})}{z_{A1}z_{A2} + z_{A1}z_{A3} + z_{A2}z_{A3}} \qquad 8.87$$

The resultant volume current, in cubic centimeters per second, $\dot{X}_R$, of the ribbon is the difference between equations 8.86 and 8.87,

$$\dot{X}_R = \dot{X}_1 - \dot{X}_2 \qquad 8.88$$

The value of the phase angle, ϕ, can be determined from the geometry of the microphone. The values of the impedance can be determined from the mass and dimensions of the ribbon, the area of the damped pipe or labyrinth, and the diameter of the aperture in the labyrinth connector.

The directional characteristics of the microphone are controlled by varying the area of the aperture in the labyrinth connector. The effect of varying the aperture can be obtained from the schematic view and the acoustical network (see Fig. 8.46).

In Fig. 8.46A the aperture is so large that the back of the ribbon is effectively open to the atmosphere. In this case the acoustical impedance z_{A2} is zero. Therefore, the acoustical resistance, r_{AS}, of the labyrinth is effectively short-circuited. The action then is exactly the same as that of the velocity microphone. From equations 8.86, 8.87, and 8.88 the volume current of the ribbon is

$$\dot{X}_R = \dot{X}_1 - \dot{X}_2 = \frac{(p_1 - p_2)}{z_{A1}} \qquad 8.89$$

If the amplitudes of p_1 and p_2 are equal, then

$$\dot{X}_R = \frac{(p_{01} - p_{01}\epsilon^{j\phi\cos\theta})\epsilon^{j\omega t}}{z_{A1}} \qquad 8.90$$

If the angle ϕ is small

$$\dot{X}_R = \frac{p_1\phi}{z_{A1}}\cos\theta = \frac{\Delta p\cos\theta}{z_{A1}} \qquad 8.91$$

where $\Delta p = p_1\phi$ the difference in pressure between the two sides of the ribbon. Equation 8.91 will be recognized as that of the velocity microphone. The directional characteristic is bidirectional.

In Fig. 8.46E the aperture is closed. In this case the acoustical impedance z_{A2} is infinite. Under these conditions the pressure p_2 is ineffective. From equations 8.86, 8.87 and 8.88 the volume current of the ribbon is given by

$$\dot{X}_R = \dot{X}_1 - \dot{X}_2 = \dot{X}_1 = \frac{p_1}{z_{A1} + z_{A3}} \qquad 8.92$$

Equation 8.92 will be recognized as that of the pressure ribbon microphone. The directional characteristic is nondirectional.

Using an aperture which may be varied, it is possible to obtain any limacon characteristic between the cosine bidirectional Fig. 8.46A and the

nondirectional characteristic Fig. 8.46E, as depicted by Fig. 8.46, parts B, C, and D. The directional characteristic of Fig. 8.46C is given by

$$e = R + R \cos \theta \qquad 8.93$$

This is a cardioid characteristic which is obtained in the two-element uni-directional microphone by making the output of the bidirectional element
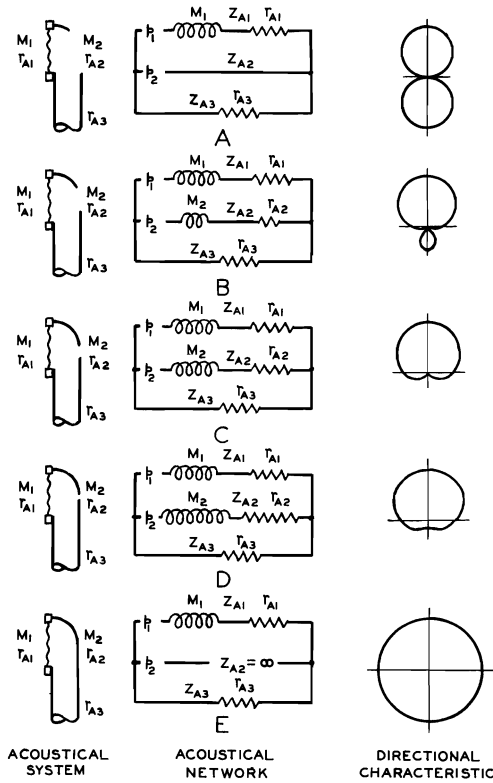

FIG. 8.46. The acoustical system, the acoustical network, and the directional characteristics of the polydirectional microphone for various values of the aperture M_2, r_{A2}.

equal to the nondirectional unit. The directional characteristic of Fig. 8.46B is given by

$$e = \frac{R}{2} + \frac{3R}{2} \cos \theta \qquad 8.94$$

For a wider directional pickup angle the characteristic of Fig. 8.46D may be more desirable. This characteristic is given by

$$e = \frac{8R}{7} + \frac{6R}{7} \cos \theta \qquad 8.95$$

The energy response to random sounds as compared to that of a non-directional microphone is $\frac{1}{3}$ for the bidirectional characteristic, Fig. 8.46A, and the cardioid characteristic, Fig. 8.46C. The energy response for the characteristic of Fig. 8.46B is $\frac{1}{4}$. This is the maximum value of discrimination obtainable in this microphone. That is, the energy response varies from $\frac{1}{3}$ to $\frac{1}{4}$ and back again to $\frac{1}{3}$ in going from the bidirectional characteristic, Fig. 8.46A, to the cardioid characteristic of Fig. 8.46C. The energy response of the characteristic of Fig. 8.46D given by equation 8.95 is 0.39. The energy response varies from $\frac{1}{3}$ to 1 in going from the cardioid characteristic of Fig. 8.46C to the nondirectional characteristic of Fig. 8.46E. The general expression[48] for the directional characteristics obtainable with this microphone is

$$e = R_1 + R_2 \cos \theta \qquad\qquad 8.96$$

The ratio of the energy response of this microphone as compared to a nondirectional microphone for any ratio of R_1 to R_2 is shown in Fig. 8.41.

3. *Uniaxial Microphone.*[49,50]—A unidirectional microphone in which the maximum directivity corresponds to the axis of the microphone is termed a uniaxial microphone. A first-order gradient uniaxial microphone is shown in Fig. 8.47. The transducer consists of a ribbon terminated in a damped pipe coupled with phase-shifting networks. The action of the system may be deduced from a consideration of the acoustical network. The two holes in the labyrinth connector form the essential portion of the phase-shifting network so that the directional pattern will be of the unidirectional type. The front face of the microphone is equipped with two lobes. The lobes perform the following functions: the reduction of the deleterious effects of diffraction, the accentuation of the high-frequency response, and the support of the blast baffles. The front of the microphone is equipped with two blast baffles and the side holes are equipped with single blast baffles. There is an additional element that contributes to increased directivity, namely, the damped cavity between the magnets.

The directivity pattern of the uniaxial microphone for a certain set of constants may be expressed as

$$e = e_0 \left(.3 + .7 \cos \theta \cos \frac{\theta}{3}\right) \qquad\qquad 8.97$$

where e_0 = sensitivity constant of the microphone, and

$\quad \theta$ = angle between the normal to the plane of the ribbon and the direction of the incident sound wave.

---

[48] A limacon is a curve defined by $e = a + b \cos \theta$. When $a = 0$, $e = b \cos \theta$, a bidirectional characteristic. When $b = 0$, $e = a$, a nondirectional characteristic. When $a = b$, $e = a + a \cos \theta$, a cardioid characteristic. For other values of a and b any type of characteristic of this family may be obtained.

[49] Olson, Preston, and Bleazey, *RCA Review*, Vol. 14, No. 1, p. 47, 1953.

[50] Olson, Preston, and Bleazey, *Trans. IRE*, Prof. Group Audio, Vol. AU-1, No. 4, p. 12, 1952.
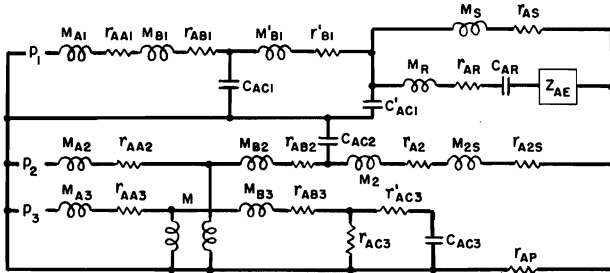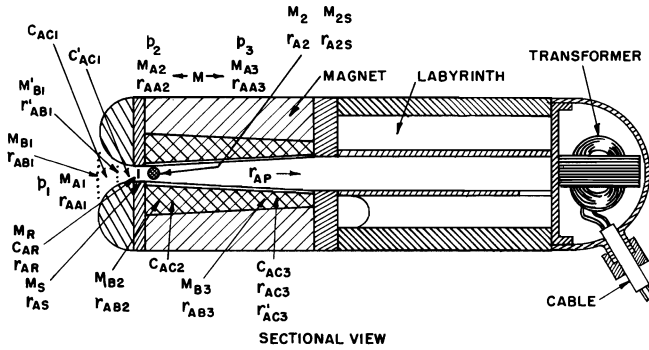
SECTIONAL VIEW

ACOUSTICAL NETWORK

FIG. 8.47. Sectional view and acoustical network of a uniaxial microphone. In the acoustical network, p_1 = the sound pressure on the front of the microphone. M_{A1} and r_{AA1} = the inertance and acoustical resistance of the air load on the front of the microphone. M_{B1}, r_{AB1}, M'_{B1}, and r'_{AB1} = the inertances and acoustical resistances of the blast baffles on the front of the microphone. C_{AC1} and C'_{AC1} = the acoustical capacitances of the volumes between the blast baffles. M_S and r_{AS} = the inertance and acoustical resistance of the slit between the ribbon and pole pieces. M_R, r_{AR}, and C_{AR} = the inertance, acoustical resistance, and acoustical capacitance of the ribbon. z_{AE} = the acoustical impedance due to the electrical circuit. p_2 = the sound pressure at the apertures in the labyrinth connector. M_{A2} and r_{AA2} = the inertance and acoustical resistance of the air load at the apertures of the labyrinth connector. M_{B2} and r_{AB2} = the inertance and acoustical resistance of the blast baffles on the side on the microphone. C_{AC2} = the acoustical capacitance of the volume behind the blast baffle. M_{2S} and r_{A2S} = the inertance and acoustical resistance of the screen covering the hole in the labyrinth connector. M_2 and r_{A2} = the inertance of the hole in the labyrinth connector. r_{AP} = the acoustical resistance of the labyrinth. p_3 = the sound pressure at the damped cavity behind the magnets. M_{A3} and r_{AA3} = the inertance and acoustical resistance of the air load upon the damped cavity. M_{B3} and r_{AB3} = the inertance and acoustical resistance of the blast baffle over the damped cavity. C_{AC3}, r_{AC3}, and r'_{AC3} = the acoustical capacitance and acoustical resistances of the cavity between the magnets. M = the coupling between the cavity and the apertures.

The directional pattern of the first-order gradient uniaxial microphone is shown in Fig. 8.48.   It will be seen that the directivity is greater than that of a cardioid pattern obtained with a conventional unidirectional microphone in that the response is down 10 db at 90° and 26 db at 180°.   This increased directivity is due to the damped cavity between the magnets.   The response of the uniaxial microphone to random sounds is $\frac{1}{5}$.   The corresponding response for a microphone with a cardioid pattern is $\frac{1}{3}$.   This means that the response of this microphone to random sounds is 60 per cent that of a microphone with a cardioid directional characteristic.   From the standpoint of sound pickup distance, the uniaxial microphone will operate at 30 per
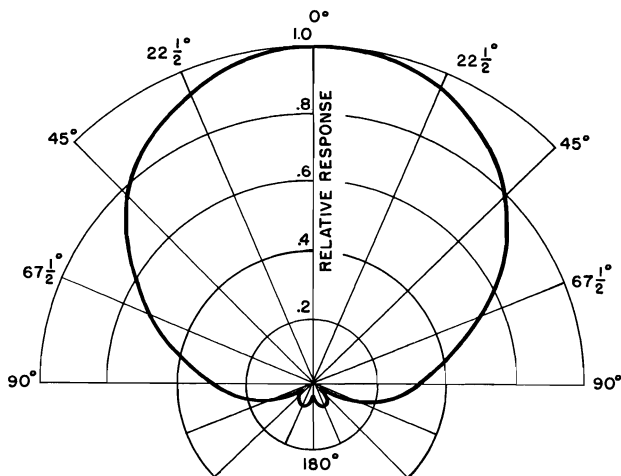
FIG. 8.48.   Directional characteristic of the uniaxial micro-
phone of Fig. 8.47.

cent greater distance than the unidirectional microphone with a cardioid directional characteristic for the same reverberation, undesirable sounds, or noise.   Tests of the blast proofing shows that the microphone will stand the firing of a 45 caliber pistol firing blanks at a distance of 5 feet indoors with the direction of firing at right angles to the microphone.   It will withstand the same firing at smaller distances outdoors.

4. *Uniphase Dynamic Microphone.*—The uniphase dynamic micro-phone[51,52] is a unidirectional microphone employing a diaphragm-voice coil transducer unit and a phase-shifting acoustical network to obtain uni-directional characteristics.   Schematic views and the acoustical network of the unidyne microphone are shown in Fig. 8.49.   The diaphragm and voice coil assembly is mounted on two spiders.   The clearance between the voice coil and pole piece is used as one of the phase-shifting elements, $M_2 r_{A2}$.

[51] Bauer, B. B., *Jour. Acous. Soc. Amer.*, Vol. 13, No. 1, p. 41, 1941.
[52] Bauer, B. B., *Electronics*, Vol. 15, No. 1, p. 31, 1942.

The volume behind the diaphragm forms an acoustical capacitance, C_{A2}. The volume in the magnet structure forms another acoustical capacitance, C_{A3}. C_{A3} is coupled to C_{A2} by means of the silk cloth which forms the acoustical resistance element, r_{A2}. The performance of the system may be determined from a consideration of the acoustical network. The constants of the acoustical network are selected so that the difference in pressure between the two sides of the diaphragm is proportional to the frequency. Under these conditions uniform response with respect to the frequency will
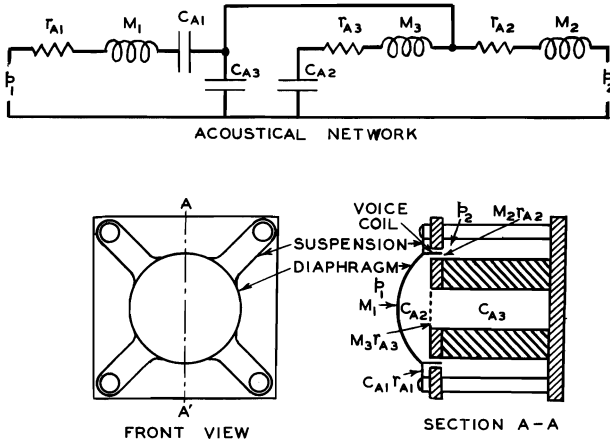
FIG. 8.49. Front view, sectional view, and the acoustical network of a unidyne unidirectional microphone. In the acoustic circuit, M_1, r_{A1}, and C_{A1} = the inertance, acoustical resistance, and acoustical capacitance of the diaphragm and suspension system. M_2 and r_{A2} = the inertance and acoustical resistance of the slit between voice coil and pole. C_{A2} = the acoustical capacitance of the air space between the diaphragm and pole. M_3 and r_{A3} = the inertance and acoustical resistance of the silk cloth. C_{A3} = the acoustical capacitance of the air space in the magnet. p_1 = the pressure at diaphragm. p_2 = the pressure at the voice coil.

be obtained if the diaphragm system is a mass reactance. In a particular model of the microphone the constants were selected so that the directional pattern is that corresponding to a ratio of $R_1/R_2 = .5$ in Fig. 8.49.

A unidirectional microphone[53] employing a ribbon conductor and essentially the same acoustical system as that of the microphone of Fig. 8.49 is shown in Fig. 8.50. The back of the ribbon is terminated in an acoustical capacitance in shunt with an acoustical resistance and inertance in series. The phase-shifting network shown in Fig. 8.50 provides unidirectional response with a single ribbon transducer.

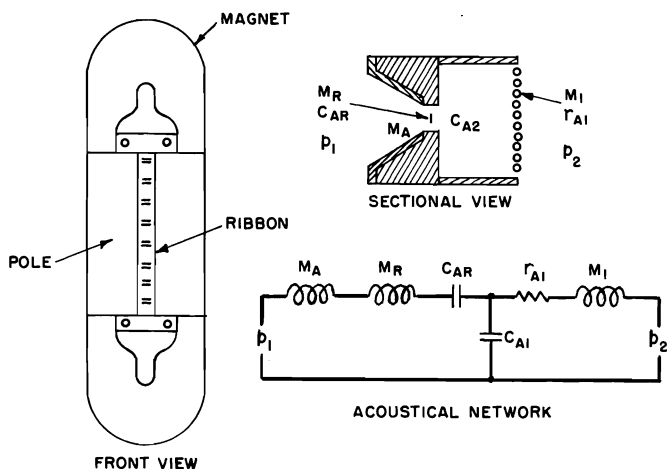[53] Bauer and Medill, *Conv. Record, IRE*, Part 6, Audio, p. 12, 1954.

FIG. 8.50. Front and sectional views and the acoustical network of a ribbon-type unidirectional microphone with lumped acoustical elements. In the acoustical network, p_1 = the sound pressure on the front of the microphone. M_A = the inertance of the air load upon the ribbon. M_R and C_{AR} = the inertance and acoustical resistance of the ribbon. C_{A1} = the acoustical capacitance of the cavity behind the ribbon. M_1 and r_{A1} = the inertance and acoustical resistance of the cloth covering the chamber. p_2 = the sound pressure on the back of the microphone.

5. *Variable-Distance Unidirectional Microphone*.[54]—A unidirectional gradient microphone in which the front to back distance varies approximately inversely as the frequency over a major portion of the audio-frequency range is shown in Fig. 8.51. There are four actuating pressures as follows: p_1 acting on the front of the diaphragm of the microphone and p_2, p_3, and p_4 acting through acoustical impedances upon the back of the diaphragm. The networks are designed so that a unidirectional pattern is obtained as outlined in the preceding sections. In addition, the effective difference in pressure at the low-frequency range involve p_1 and p_4, through intermediate steps of p_1 and p_3 in the mid-frequency range and p_1 and p_2 in the high-frequency range. With the variable type of pickup the proximity effect is reduced.

6. *Directional Condenser Microphone*.[55]—A directional microphone employing a condenser unit as the transducer is shown in Fig. 8.52. The transducer unit consists of two damped diaphragms. The vibrating system consists of the two separate and spaced diaphragms. Each diaphragm is spaced at a small distance from the back plate. The damped space provides an acoustical capacitance and acoustical resistance for the diaphragm. The

[54] Wiggins, A. M., *Jour. Acous. Soc. Amer.*, Vol. 26, No. 5, p. 687, 1954.
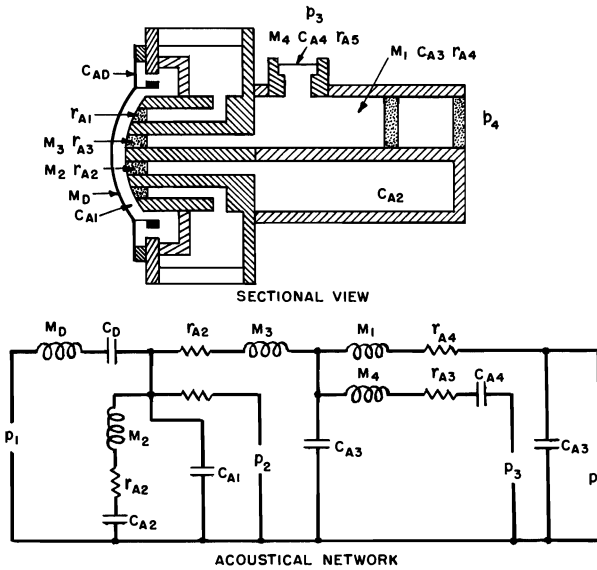[55] Bauch, F. W. O., *Jour. Audio Eng. Soc.*, Vol. 1, No. 3, p. 232, 1953.

FIG. 8.51. Sectional view and acoustical network of a variable distance microphone. p_1 = the sound pressure on the front of the microphone. p_2, p_3, and p_4 = the sound pressures acting at different parts on the back of the microphone. M_D = inertance due to the mass of the diaphragm. C_{AD} = the acoustical capacitance of the diaphragm suspension system. C_{A1} = the acoustical capacitance of the volume back of the diaphragm. r_{A1} = the acoustical resistance of the shortest path. M_4, C_{A4}, and r_{A5} = the inertance, acoustical capacitance, and acoustical resistance of the diaphragm in the circuit of the medium path. M_1, M_3, C_{A3}, r_{A3}, and r_{A4} = the inertances, acoustical capacitance, and acoustical resistances involved in the largest path. M_2, r_{A2}, and C_{A2} = the inertance, acoustical resistance, and acoustical capacitance in a side branch.

two vibrating systems are placed back to back. The cavities behind the diaphragm are interconnected by small holes. The phase shift in the vibrating system combined with the electrical polarizing system makes it possible to obtain a variety of directional characteristics as well as a non-directional characteristic. With the potentiometer at the full negative position a bidirectional pattern is obtained. With the potentiometer set at the zero position a cardioid type of directional pattern is obtained. With the potentiometer set at the full positive position a nondirectional pattern is obtained.

7. *Dipole Microphone.*[56]—A dipole microphone is a microphone in which the response is a function of the sound pressure between two distinct points.

[56] Olney, Slaymaker, and Meeker, *Jour. Acous. Soc. Amer.*, Vol. 16, No. 3, p. 172, 1945.
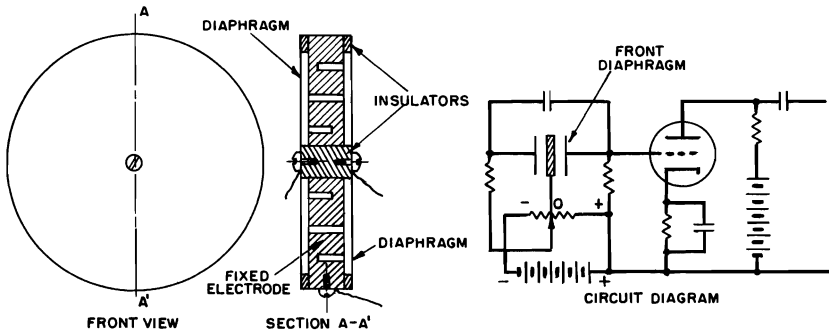
FIG. 8.52. Front and sectional views and circuit diagram of the vacuum tube amplifier of a directional condenser microphone.

A schematic view of the acoustical system is shown in Fig. 8.53. The transducer is a carbon element. The use of the two tubes makes it possible to remove the microphone transducer from a location directly in front of
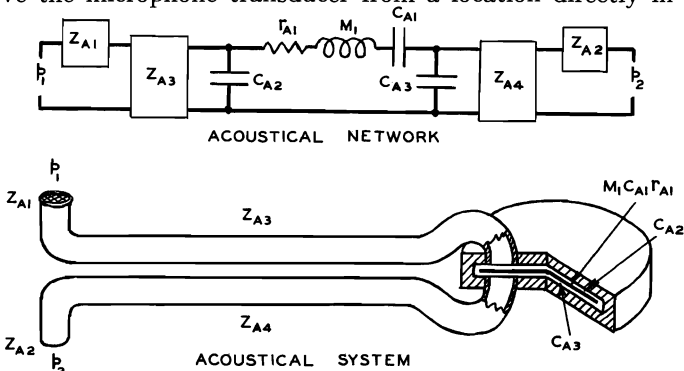

FIG. 8.53. Schematic view and acoustical network of the dipole microphone. In the acoustical circuit, z_{A1} and z_{A2} = the acoustical impedances of the silk cloth terminating the ends of the pipes. z_{A3} and z_{A4} = the acoustical quadripoles representing the cylindrical pipes. C_{A2} and C_{A3} = the acoustical capacitances of the air chambers on the two sides of the diaphragm. M_1, C_A, and r_{A1} = the inertance, acoustical capacitance, and acoustical resistance of the diaphragm and carbon button. p_1 and p_2 = the sound pressures at the pipes.

the talker's mouth and yet retain the acoustical advantage of a close-talking microphone. The microphone and telephone receiver are made an integral unit in a telephone operator's set. A disk of silk cloth covers the end of each tube. The acoustical resistance termination practically eliminates the resonance in the tubes.

The performance of the system may be determined from a consideration of the acoustical network. The two pipes are represented as acoustical

quadripoles.   The performance of a cylindrical pipe has been considered in Sec. 5.25.

The dipole microphone is a first-order gradient microphone.   The directional pattern is of the cosine type.   The performance is essentially the same as that of the phase-shifting microphone with ribbon element, save that with the carbon element the output will be proportional to the frequency in a plane wave.   However, as a close-talking microphone the output will be independent of the frequency.   It also possess the antinoise characteristics of a close-talking, first-order, gradient microphone.
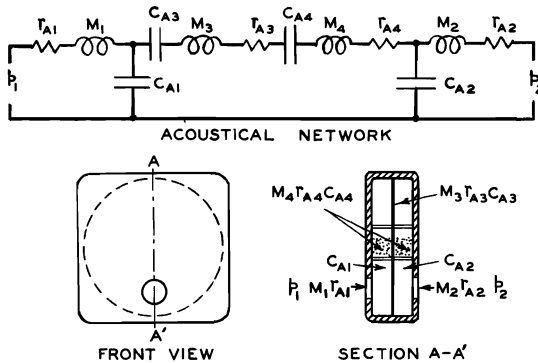
FIG. 8.54.   Front view, sectional view and acoustical network of the differential microphone.   In the acoustical network, M_1, r_{A1}, and M_2, r_{A2} = the inertances and acoustical resistances of the two holes in the case. C_{A1} and C_{A2} = the acoustical capacitances of the air chambers on the two sides of the diaphragm. M_3, r_{A3}, and C_{A3} = the inertance, acoustical resistance, and acoustical capacitance of the diaphragm. M_4, r_{A4}, and C_{A4} = the inertance, acoustical resistance, and acoustical capacitance of the carbon elements. p_1 and p_2 = the sound pressures at the two holes in the case.

8. *Differential Microphone.*[57]   *Lip Microphone.*—A differential microphone is a gradient type microphone used for close talking.   In general, it is held in place on the upper lip by a strap arrangement.   A schematic view of the acoustical system is shown in Fig. 8.54.   The transducer is a carbon element.   The performance of the system may be determined from a consideration of the acoustical network of Fig. 8.54.   The differential microphone is a first-order gradient microphone.   The directional pattern is of the cosine type.   Employing a carbon element the output will be proportional to the frequency in a plane wave.   However, as a close-talking microphone the output will be independent of the frequency.   It also possesses the anti-noise characteristics of a close-talking, first-order, gradient microphone.

[57] Ellithorn and Wiggins, *Proc. Inst. Rad. Eng.*, Vol. 34, No. 2, p. 84P, 1946.

8.5. Higher Order Gradient Microphones.[58,59]—First-order pressure gradient microphones have been described in Sec. 8.3. The response in a first-order gradient microphone corresponds to the gradient of the sound pressure. The response of higher order gradient microphones corresponds to the order of the gradient of the sound pressure. The directional characteristics of gradient microphones are cosine functions; the power of the cosine is the order of the gradient. It is the purpose of this section to consider higher order gradient microphones.

A. *Second-Order Gradient Microphones.*[60,61]—A gradient microphone of order two is a microphone in which the response corresponds to the pressure gradient of the pressure gradient.

The actuating force in a second-order gradient microphone is the difference in pressure between two two-point systems, and may be written

$$\Delta(\Delta p) =$$
$$p_M D_1 D_2 \left[\frac{- k^2 r^2 \sin k(c - r) + 2kr \cos k(ct - r) + 2 \sin k(ct - r)}{r^3} \right] \cos^2 \theta$$

$$8.98$$

where D_1 = distance between the points in the pairs of points, and
D_2 = distance between the two pairs.

A second-order gradient microphone may be made up of two oppositely phased first-order gradient microphones as shown in Fig. 8.55.

The acoustical network of the acoustical system of one of the units in the second-order microphone is shown in Fig. 8.55. The controlling element in the system is an acoustical resistance. The transducer is of the dynamic type. Therefore, in a plane wave, the voltage output of a single unit will be proportional to the frequency. Connecting two of the units in opposition the voltage output of the second-order gradient microphone will be proportional to the square of the frequency. However, as a close-talking microphone the output will be independent of the frequency. It possesses the antinoise characteristics of a close-talking, second-order, gradient microphone. The directional characteristics of the second-order gradient microphone, as equation 8.98 shows, are bidirectional and proportional to the square of the frequency.

B. *Gradient Microphones of Any Order.*[62]—The general expression for the actuating pressure for a microphone of any order n for any two points separated by a distance δr is is

$$\delta n_p = \frac{\partial n_p}{\partial r^n} \delta r^n = \frac{\partial}{\partial r^n} \left(-j \frac{p_M}{r} \epsilon^{jk(ct-r)} \right) (\delta r \cos \theta)^n \qquad 8.99$$

[58] Olson, H. F., U. S. Patent 2,301,744.
[59] Olson, H. F., *Jour. Acous. Soc. Amer.*, Vol. 17, No. 3, p. 192, 1946.
[60] Olson, H. F., U. S. Patent 2,301,744.
[61] Olson, H. F., *Jour. Acous. Soc. Amer.*, Vol. 17, No. 3, p. 192, 1946.
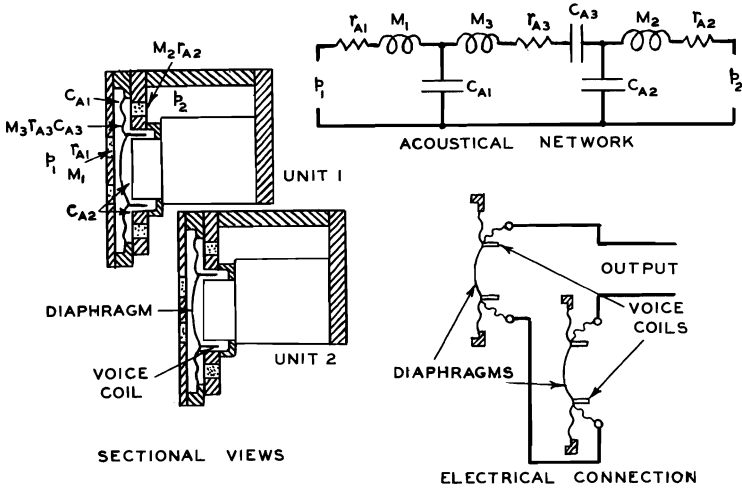[62] Olson, H. F., *Jour. Acous. Soc. Amer.*, Vol. 17, No. 3, p. 192, 1946.

FIG. 8.55. Sectional view, electrical connection, and acoustical network of a second-order gradient microphone. The acoustical network applies to a single unit. The electrical connection shows the two units connected in opposition. In the acoustical network, r_{A1} and M_1 = the acoustical resistance and inertance at the front of the unit. r_{A2} and M_2 = the acoustical resistance and inertance at the back of the unit. C_{A1} and C_{A2} = the acoustical capacitances on the two sides of the diaphragm. M_3, r_{A3}, and C_{A3} = the inertance, acoustical resistance, and acoustical capacitance of the diaphragm. p_1 and p_2 = the sound pressures at the front and back of the unit.
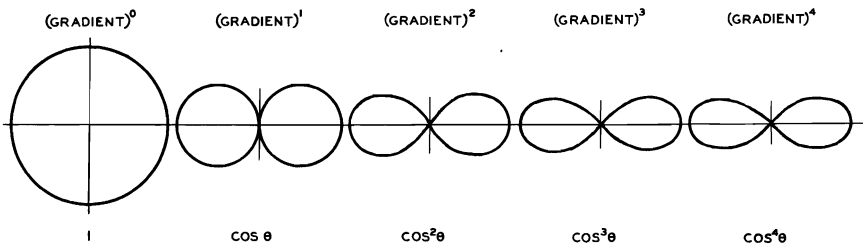
FIG. 8.56. The directional characteristics of gradient microphones of order zero, one, two, three, and four.

Equation 8.99 shows that the pressure available for driving the microphone is proportional to the nth power of the frequency. The directional characteristics are bidirectional cosine functions, the power of the cosine is the order of the gradient. The directional characteristics for gradient microphones of orders zero, one, two, three, and four are shown in Fig. 8.56.

C. *Noise Discrimination of Gradient Microphones.*[63]—Gradient microphones of order one and higher are directional. Therefore, these micro-

[63] Olson, H. F., *Jour. Acous. Soc. Amer.*, Vol. 17, No. 3, p. 192, 1946.

phones discriminate against sounds from random directions. The magnitude of the discrimination is given by the expression in equation 8.117 as follows,

$$\text{Directional efficiency} = \frac{2\pi \int_0^\pi R_n{}^2 \cos^{2n} \theta \sin \theta \, d\theta}{4\pi R_0{}^2} \qquad 8.100$$

where R_n = response of the gradient microphone on the axis,

n = order of the gradient,

θ = angle between the axis of the gradient microphone and the direction of the incident sound, and

R_0 = response of the gradient microphone of order zero.

DISTANCE							
3/4"	280	560	1120	2800	5600	11200	
1 1/2"	140	280	560	1400	2800	5600	11200
3"	70	140	280	700	1400	2800	5600

FREQUENCY IN CYCLES PER SECOND
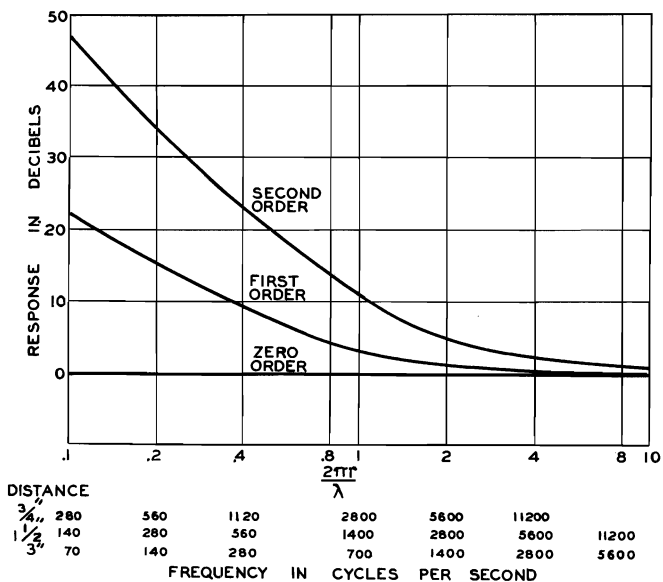
FIG. 8.57. Response of zero-, first-, and second-order gradient microphones to a small source as a function of $2\pi r/\lambda$ where r = distance and λ = wavelength. The response frequency characteristics of all three are assumed to be independent of the frequency for a plane wave, that is, $2\pi r/\lambda = \infty$. The frequency scales below the graph apply to three distances, namely, 3, 1½, and ¾ inches.

If the sensitivity of the gradient microphone of order zero is the same as that of a gradient microphone or order n, equation 8.100 becomes

$$\text{Directional efficiency} = \frac{1}{2n + 1} \qquad 8.101$$

The above equation assumes that the distance between the origin of the sound and the microphone is greater than $n\lambda$, where n is the order of the gradient, and λ is the wavelength.

A further increase in discrimination against noise and other undesired sounds may be obtained if a gradient microphone is used as a close-talking microphone.   The response of gradient microphones of order zero, one, and two to a small source as a function of the wavelength and distance from a small sound source are shown in Fig. 8.57.   The response frequency characteristics of all three are assumed to be independent of the frequency for a plane sound wave.   Referring to Fig. 8.57 it will be seen that the response of a gradient microphone is accentuated when the distance between the sound source is less than $n\lambda$.   This feature of a gradient microphone may be used to obtain high discrimination against unwanted sounds.   If the microphone is used as a close-talking microphone and the noises originate
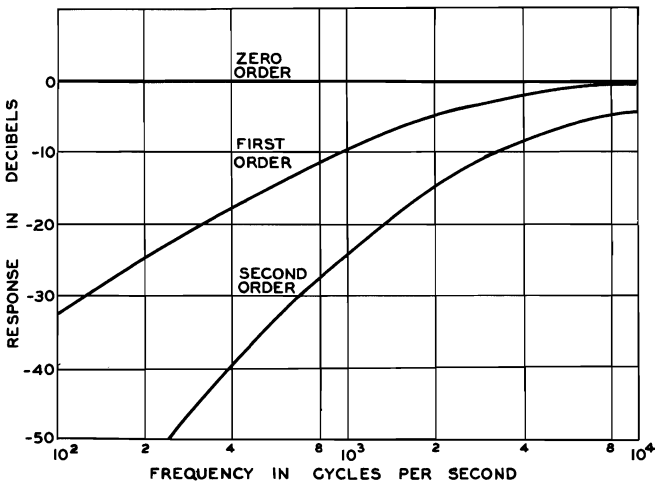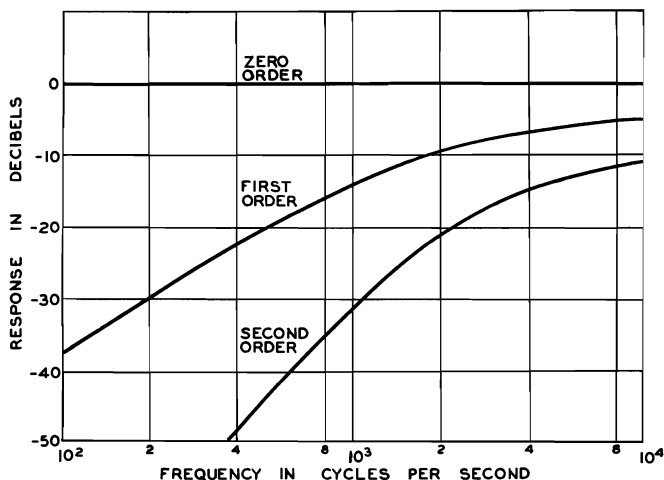

FIG. 8.58.   Response frequency characteristics of zero-, first-, and second-order gradient microphones to a plane wave.   The microphones are compensated so that the responses of all three are the same and independent of the frequency when operating at a distance of $\frac{3}{4}$ inch from a small sound source.

at a distance from the microphone, considerable discrimination against the noise can be obtained.   For example, assume that the distance between the mouth and the microphone is $\frac{3}{4}$ inch which is the average distance for a close-talking microphone, the response frequency characteristics of zero-, first-, and second-order gradient microphone, as function of the frequency are shown in Fig. 8.57.   The response of the gradient microphones is accentuated at the low frequencies.   If compensation is introduced so that the response of all three becomes uniform with respect to frequency for the $\frac{3}{4}$-inch distance from the small sound source, the response frequency characteristics for distant sounds will be as shown in Fig. 8.58.   These characteristics show the discrimination against distant axial sounds by the first- and second-order gradient microphones as compared to a pressure or

zero-order gradient microphone. These characteristics apply to all first- and second-order gradient microphones.

In general, noise and unwanted sounds originate in random directions. Under these conditions additional discrimination will be introduced by the directional pattern. The response of zero-, first-, and second-order gradient microphones, compensated for uniform response at $\frac{3}{4}$-inch distance, to distant sound originating in random directions is shown in Fig. 8.59. First-order gradient antinoise microphones have been described in Secs. 8.4B8 and 8.5B. The characteristics for a first-order gradient microphone apply to these microphones.[64] A second-order gradient microphone has

FIG. 8.59. Response frequency characteristics of zero-, first-, and second-order gradient microphones to random sounds originating at a distance. The microphones are compensated so that the responses of all three are the same and independent of the frequency when operating at a distance of $\frac{3}{4}$ inch from a small sound source.

been described in Sec. 8.5A. The discrimination of the second-order gradient microphone is tremendous. This has been substantiated by actual tests in which it is impossible to drown out speech in a second-order gradient microphone for any noise which the normal ear can withstand without pain.

D. *Higher Order Unidirectional Gradient Microphones.*[65]—It has been shown in the preceding sections that the directivity of a gradient microphone increases with increasing powers of the pressure gradient. The directional characteristics of these systems are of the bidirectional type. In many applications unidirectional characteristics are more desirable. Unidirectional microphones employing first-order gradient units have been considered

[64] Anderson and Wigginton, *Audio Engineering*, Vol. 34, No. 4, p. 16, 1950.
[65] Olson, H. F., *Jour. Acous. Soc. Amer.*, Vol. 17, No. 3, p. 192, 1946.

in Sec. 8.4.   It is the purpose of this section to consider higher order combination gradient microphones with unidirectional characteristics.

A higher order unidirectional gradient microphone may be obtained by combining two first-order gradient microphones with a delay system as shown in Fig. 8.60.   The voltage output of this system is given by

$$e_2 = e_0(D_2 + D_1 \cos \theta) \cos \theta \qquad\qquad 8.102$$

where e_0 = reference voltage output,

D_1 = distance between the first-order gradient elements, and

D_2 = path length of the delay.

Equation 8.102 holds for the frequency range in which D_1 and D_2 are small compared to the wavelength.   The reference voltage output is a function
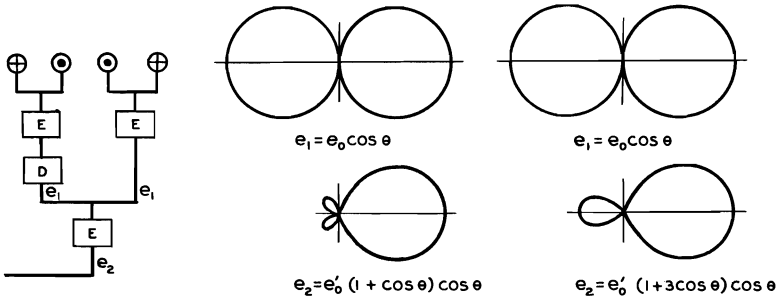
FIG. 8.60.   Higher order unidirectional microphone consisting of two similar gradient units of order one and a delay system.   The directional characteristics of the gradient microphone of order one are shown as well as the directional characteristics of the higher order gradient microphone for two different delay conditions.

of the frequency and the type of electroacoustical generating system.   The maximum discrimination against random sounds occurs when $D_2 = \frac{5}{3}D_1$. For this condition the energy response to random sounds is one-eighth that of a nondirectional microphone.   This is a very high order of directivity. The directional characteristics for two different conditions are shown in Fig. 8.60.

The system of Fig. 8.61 consists of two combination pressure and pressure gradient microphones, described in Sec. 8.4, and a delay system.   A number of combinations are possible in this system, as for example, combination units with various delays and dissimilar combination units with various delays.   The directional characteristics for two different conditions are shown in Fig. 8.61.

E. *Second-Order   Gradient   Uniaxial   Microphone.*[66]—A   second-order gradient microphone with a unidirectional directivity pattern consisting of the combination of two unidirectional microphones each consisting of two

[66] Olson and Preston, *RCA Review*, Vol. 10, No. 3, p. 339, 1949.

$$e_1 = e_0(1 + \cos\theta) \qquad e_1 = e_0(1 + 3\cos\theta)$$

$$e_2 = e_0'(1 + \cos\theta)\cos\theta \qquad e_2 = e_0'(1 + 3\cos\theta)\cos\theta$$
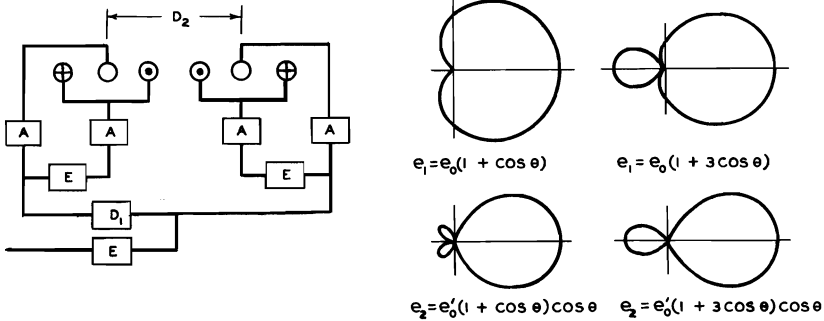
FIG. 8.61. Higher order unidirectional microphone consisting of two unidirectional elements described in Sec. 8.4 and a delay system. The directional characteristics of two different unidirectional units are shown and the combination higher order gradient for $D_2 > 0$ and $D_1 = 0$.

uniaxial, first-order gradient microphones as described in Sec. 8.4B3 is shown in Fig. 8.62.

The upper limit of the useful frequency range of a second-order gradient microphone made up of two first-order gradient microphones, as shown in Fig. 8.62 and schematically in Fig. 8.63, is determined by the distance between the units. This upper frequency limit is given by

$$f_c = \frac{c}{D} \qquad\qquad 8.103$$

where f_c = upper frequency limit, in cycles per second,

c = velocity of sound, in centimeters per second, and

D = distance between the units, in centimeters.

$$e = e_0(.3 + .7\cos\theta\cos\frac{\theta}{3})$$

$$e = e_0(.3 + .7\cos\theta\cos\frac{\theta}{3})\cos\theta$$
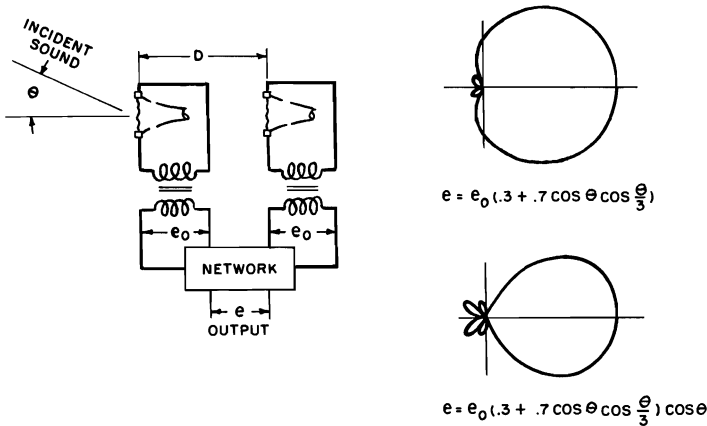
FIG. 8.62. The elements of a second-order gradient uniaxial microphone consisting of two first-order gradient uniaxial microphones.

The voltage output of a gradient microphone of the type shown in Fig. 8.62 in the low-frequency range, that is, in the range for which $D<<\lambda$ is given by

$$e = 2e_0 \frac{D}{\lambda} \pi \qquad \qquad 8.104$$

where e = voltage output of the combination, in volts,

   e_0 = voltage output of an individual unit, in volts,

   D = distance between the units, in centimeters, and

   λ = wavelength, in centimeters.

Equations 8.103 and 8.104 establish the frequency range of operation of a gradient microphone.   A consideration of the requirements for a second-order microphone indicated that a combination of two first-gradient uniaxial
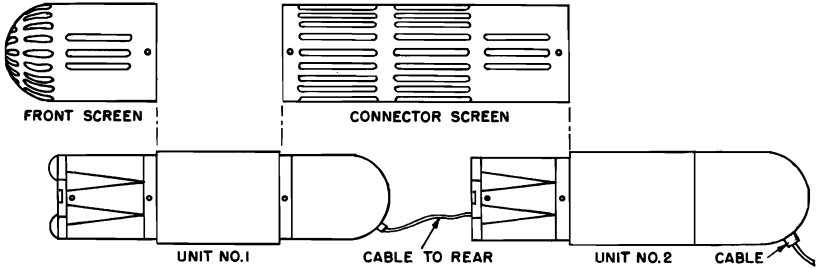
FIG. 8.63.   Schematic view of a second-order gradient microphone consisting of two first-order gradient uniaxial microphones and the front and connector screens.

microphones of the type described in the preceding section could be used for the elements.   It also appeared that a high order of directivity was not required in the high-frequency range.   The microphone which was developed operates as follows: The system is of the second-order gradient type up to 1000 cycles; there is a transition from second- to first-order gradient from 2000 to 4000 cycles, and above 4000 cycles the directivity pattern is the same as the uniaxial microphone.   As described in the preceding section, the directional pattern of the uniaxial microphone is sharper than a cardioid. Specifically, the directivity pattern up to 2000 cycles is given by

$$e = e_0 \left(.3 + .7 \cos \theta \cos \frac{\theta}{3}\right) \cos \theta \qquad \qquad 8.105$$

where e_0 = sensitivity constant of the microphone, and

   θ = angle between the axis of the microphone and the direction of the incident sound wave.

It is in the region below 2000 cycles that practically all of the difficulty due to reverberant and other undesirable sounds occurs.   Since operation

shifts from the two microphones to the single microphone in the front in the high-frequency region, it would be a comparatively simple task to develop a microphone with a sharper directivity pattern in the high-frequency region for use as the front microphone if this appeared to be desirable.

The electrical system used with the two first-order gradient uniaxial units is shown in Fig. 8.62. This consists of a network which transfers from the two units in series opposition in the frequency range below 2000 cycles to the single unit above 4000 cycles. Suitable compensation circuits are also included in the electrical system.

The polar directional patterns for 200, 1000, and 4000 cycles are shown in Fig. 8.64. These patterns show a high order of discrimination for the sides and rear hemisphere in the mid- and low-frequency range.
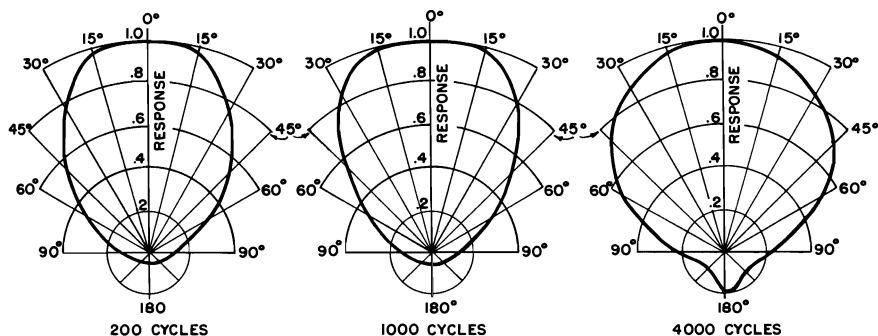
FIG. 8.64. The directional characteristics of a second-order gradient uniaxial microphone for 200, 1000, and 4000 cycles.

The application for the second-order gradient uniaxial microphone is for the pickup of sound over large distances or under acoustically difficult conditions where a high degree of directivity is desired, as for example, in sound motion pictures and television. The size and weight of the microphone are such that it may be mounted on a conventional boom. The response of the second-order gradient microphone to random sounds is $\frac{1}{9}$ that of a non-directional microphone. The increased directional efficiency makes it possible to use a pickup distance of more than 3 times that of a nondirectional microphone and 1.8 times that of a unidirectional microphone with a cardioid directional pattern.

8.6. Wave Type Microphones.—Directional microphones may be divided into two classes as follows: first, wave-type microphones which depend for directivity upon wave interference, and second, gradient-type microphones which depend for directivity upon the difference in pressure or powers of the difference in pressure between two points. In the first class of microphone, in which the directivity depends in some way upon wave interference, to obtain any semblance of directivity the dimensions of the microphone must be comparable to the wavelength of the sound wave.

Typical microphones of this classification are reflector, lens, and line microphones. The second class of microphones has been considered in Secs. 8.3, 8.4, and 8.5. The dimensions of gradient microphones, as contrasted to wave-type microphones, are small compared to the wavelength. It is the purpose of this section to consider two examples of wave microphones, namely, the parabolic reflector microphone and the line microphone.

A. *Parabolic Reflector*.[67,68,69,70,71]—Reflectors have been used for years for concentrating and amplifying all types of wave propagation. The surface of the parabolic reflector is shaped so that the various pencils of incident sound parallel to the axis are reflected to one point called the focus (Fig. 8.65). To obtain an appreciable gain in pressure at the focus, the reflector must be large compared to the wavelength of the incident sound. This requirement of size must also be satisfied in order to obtain
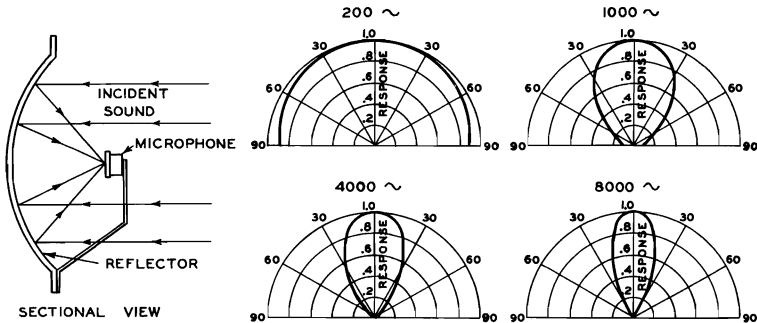
FIG. 8.65. Cross-sectional view of a parabolic reflector for a microphone. The polar graphs show the directional characteristics. The polar graph depicts the pressure, in dynes, at the microphone as a function of the angle, in degrees. The maximum response is arbitrarily chosen as unity.

sharp directional characteristics. If this condition is satisfied at the low frequencies the size of the reflector becomes prohibitive to be used with facility.

A cross-sectional view of a parabolic reflector and a pressure microphone located at the focus is shown in Fig. 8.65. When the microphone is located at the focus the gain at the high frequencies is considerably greater than at the mid-frequency range. The accentuation in high-frequency response may be overcome by moving the microphone slightly out of focus. This expedient also tends to broaden the sharp directional characteristics at the high frequencies.

The directional characteristics of a parabolic reflector 3 feet in diameter, used with a pressure microphone, are shown in Fig. 8.65. It will be seen

[67] Olson and Wolff, *Jour. Acous. Soc. Amer.*, Vol. 1, No. 2, p. 173, 1930.
[68] Hanson, O. B., *Jour. Acous. Soc. Amer.*, Vol. 3, No. 1, Part 1, p. 9, 1931.
[69] Hanson, O. B., *Jour. Acous. Soc. Amer.*, Vol. 3, No. 1, Part 1, p. 81, 1931.
[70] Dreher, Carl, *Jour. Soc. Mot. Pic. Eng.*, Vol. 16, No. 1, p. 29, 1932.
[71] Olson and Wolff, *Jour. Acous. Soc. Amer.*, Vol. 1, No. 3, p. 410, 1930.

that the directivity increases with frequency. For example, the system is practically nondirectional at 200 cycles. On the other hand, the directional characteristic is very sharp at 8000 cycles.

B. *Lens Microphone.*[72,73]—A lens microphone consists of an acoustic lens arranged so that the pencils of sound arrive in phase at a common point termed the focus, and a microphone located at focus. A schematic sectional view of a pressure microphone located at the focus of an acoustic lens is shown in Fig. 8.66. The directivity pattern is determined by a
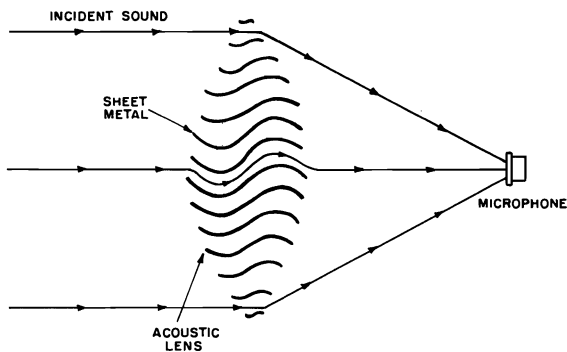
FIG. 8.66. Schematic view of a lens microphone.

relationship between the dimensions of the lens and the wavelength similar to that of the parabolic reflectormicrophone.

C. *Large-Surface Microphone.*—A large-surface microphone,[74] in the form of a large number of dynamic microphone units arranged on a spherical surface, is shown in Fig. 8.67. A curved surface source has been considered in Sec. 2.20. The microphone shown in Fig. 8.67 is based upon the fundamental principles of a curved surface radiator or receiver. The angular spread of the microphone shown in Fig. 8.67 is about 50°. The diameter is four feet. The directivity pattern follows that of a curved surface system of this diameter. The directivity pattern is reasonably uniform above 300 cycles. The low-frequency limit of uniform directivity could be extended by employing a microphone of a larger diameter.

A large-surface microphone[75] in the form of a condenser microphone consists of a nonstretched diaphragm designed to vibrate in phase over its entire surface. The operating elements consist of an aluminum foil front electrode cemented to a 3/32-inch foam rubber sheet mounted on an aluminum back plate. The whole assembly is fitted in a picture frame 18 inches square and protected by a perforated metal face plate. The directivity pattern is that of a square plate, considered in Sec. 2.17.

[72] Clark, M. A., *Jour. Acous. Soc. Amer.*, Vol. 25, No. 4, p. 829, 1953.
[73] Clark, M. A., *Trans. IRE*, Prof. Group on Audio, Vol. AU-2, No. 1, p. 5, 1954.
[74] Olson, Preston, and May, Unpublished Report.
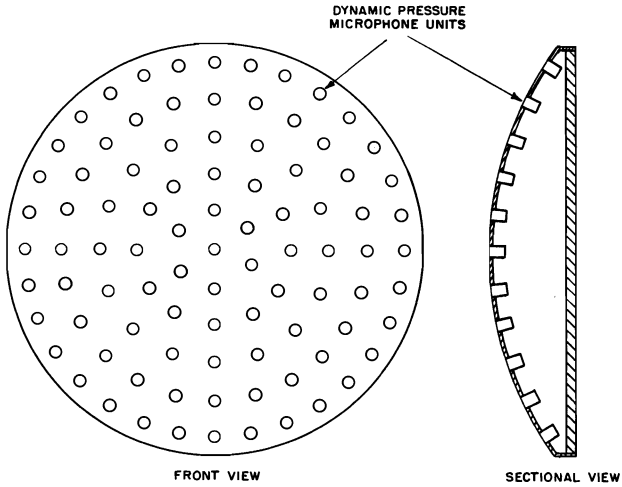[75] Aamodt and Harvey, *Jour. Acous. Soc. Amer.*, Vol. 25, No. 4, p. 825, 1953.

FIG. 8.67. Large surface microphone consisting of a large number of dynamic microphone units arranged upon a spherical surface.

D. *Line Microphones.*[76,77,78]—A line microphone is a microphone consisting of a number of small tubes with the open end, as pickup points, equally spaced along a line and the other end connected to a common junction to a transducer element for converting the sound vibrations into the corresponding electrical variations. In the line systems to be considered, the transducer will be a ribbon element located in a magnetic field and ter-
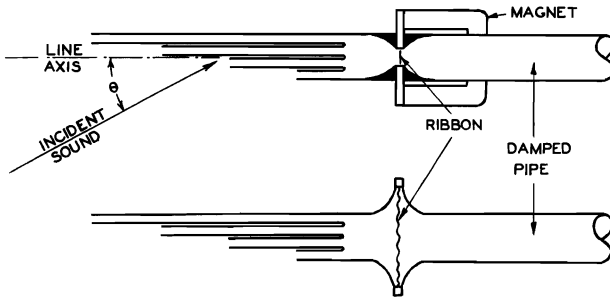
FIG. 8.68. Line microphone. Useful directivity on the line axis. This microphone consists of a large number of small pipes arranged in a line with the distance from the opening of each pipe to the common junction decreasing in equal steps. The system is terminated in a ribbon element and an acoustical resistance.

[76] Olson, H. F., *Jour. Inst. Rad. Eng.*, Vol. 27, No. 7, p. 438, 1939.
[77] Mason and Marshall, *Jour. Acous. Soc. Amer.*, Vol. 10, No. 3, p. 206, 1939.
[78] Olson, H. F., *Broadcast News*, No. 28, p. 32, July, 1938.

minated in an acoustical resistance. Under these conditions the output of the pipes can be added vectorially.

1. *Line Microphone: Useful Directivity on the Line Axis. Simple Line.*— This microphone consists of a number of small pipes with the open ends, as pickup points, equally spaced on a line and the other ends joined at a common junction decreasing in equal steps (Fig. 8.68). A ribbon element, connected to the common junction and terminated in an acoustical resistance in the form of a long damped pipe, is used for transforming the acoustical vibrations into the corresponding electrical variations.

The contribution, in dynes per square centimeter, by any element n at the common junction of the microphone may be expressed as

$$p_n = B_n \cos 2\pi \left(ft - \frac{x_n - x_n \cos \theta}{\lambda} \right)$$

$$+ jB_n \sin 2\pi \left(ft - \frac{x_n - x_n \cos \theta}{\lambda} \right) \qquad 8.106$$

$$p_n = B_n \epsilon^{2\pi jft} \epsilon^{2\pi j(x_n - x_n \cos \theta)/\lambda} \qquad 8.107$$

where f = frequency, in cycles per second,

t = time, in seconds,

x_n = distance of the element n from the center of the line, in centimeters,

λ = wavelength, in centimeters,

θ = angle between axis of the line and the incident sound, and

B_n = amplitude of the pressure due to element n, in dynes per square centimeter.

In the case of a uniform line, with the strength a constant, the resultant when all the vectors are in phase is $B_n l$, where l is the length of the line. The ratio, R_θ, of the response for the angle θ to the response for $\theta = 0$ is

$$R_\theta = \frac{1}{B_n l} \left| \int_{-l/2}^{l/2} B_n \epsilon^{2\pi j[ft + (x - x \cos \theta)/\lambda]} \, dx \right| \qquad 8.108$$

The absolute value of the term on the right is given by

$$R_\theta = \frac{1}{l} \left| \int_{-l/2}^{l/2} \epsilon^{2\pi j(x - x \cos \theta)/\lambda} \, dx \right| \qquad 8.109$$

$$R_\theta = \frac{\sin \frac{\pi}{\lambda} (l - l \cos \theta)}{\frac{\pi}{\lambda} (l - l \cos \theta)} \qquad 8.110$$

The directional characteristics of the microphone of Fig. 8.68 for various ratios of length of the line to the wavelength are shown in Fig. 8.69. These characteristics are surfaces of revolution about the line as an axis. This microphone is useful for collecting sounds arriving from directions making small angles with the microphone axis.
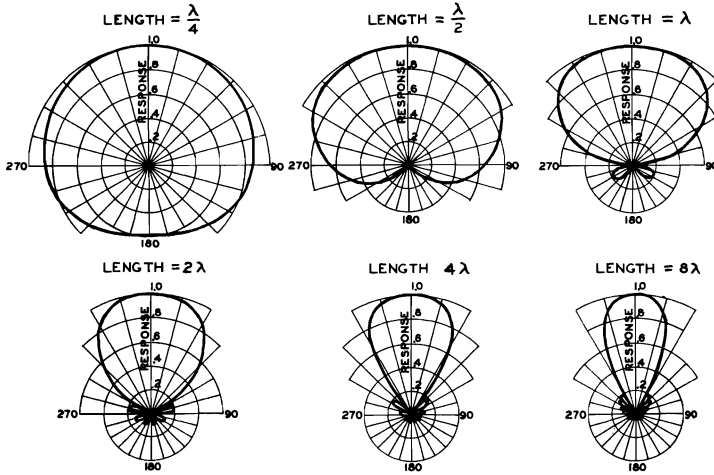
FIG. 8.69. The directional characteristics of the microphone shown in Fig. 8.68 as a function of the ratio of the length of the line to the wavelength. The polar graph depicts the output, in volts, as a function of the angle, in degrees.

2. *Line Microphone: Useful Directivity on the Line Axis. Line with Progressive Delay.*—As in the case of Fig. 8.68 this microphone consists of a number of small pipes with the open ends, as pickup points, equally spaced on a line and the other ends joined at a common junction. In addition, there is inserted a delay which is proportional to the distance from the end of the line or the pickup point nearest the common junction (Fig. 8.70).

$$R_\theta = \frac{1}{B_n l} \left| \int_{-l/2}^{l/2} B_n \epsilon^{2\pi j [(x - x \cos \theta/\lambda) + d/\lambda]} dx \right| \qquad 8.111$$
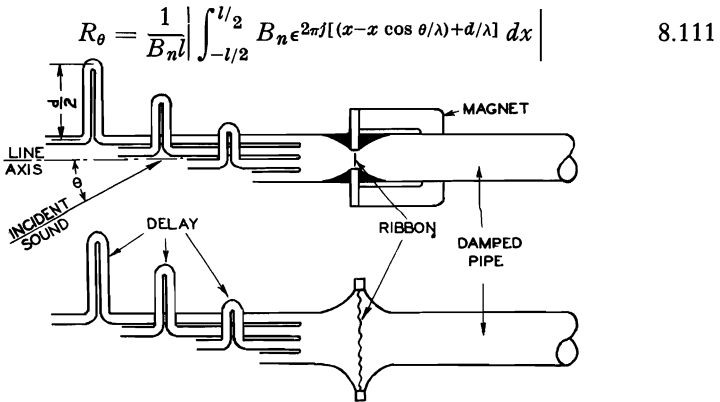
FIG. 8.70. Line microphone. Useful directivity on the line axis. This microphone differs from Fig. 8.68 in that a delay is inserted in each small pipe. The amount of delay is proportional to the distance from the pipe opening to the pickup point nearest the common junction.

where d is the path length of the delay introduced for the point furthest removed from the common junction.

$$R_\theta = \frac{\sin \frac{\pi}{\lambda} (l - l \cos \theta + d)}{\frac{\pi}{\lambda} (l - l \cos \theta + d)} \qquad 8.112$$

The directional characteristic of the microphone of Fig. 8.70 for various ratios of the length of the line to the wavelength, and for a delay path of one-fourth times the length of the line is shown in Fig. 8.71. Comparing
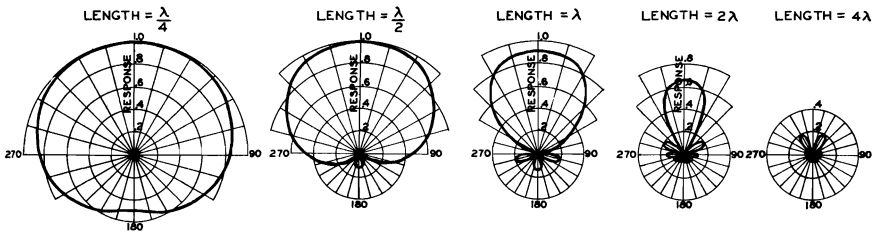
FIG. 8.71. The directional characteristics of the microphone shown in Fig. 8.70 for a time delay of one-quarter the length of the line as a function of the ratio of the length of the lines to the wavelength. The polar graph depicts the output, in volts, as a function of the angle, in degrees.

Fig. 8.71 with Fig. 8.69, it will be seen that the same directional characteristic can be obtained with a shorter line by introducing appropriate delay. In the case of a delay path comparable to the wavelength, loss in sensitivity occurs.

3. *Line Microphone: Useful Directivity on the Line Axis. Two Lines and a Pressure Gradient Element.*—This microphone consists of two lines of the type shown in Fig. 8.70 arranged so that the ribbon element measures the difference in the pressures generated in the two lines (Fig. 8.72). The centers of the two lines are displaced by a distance D. In the line nearest the element, a bend of length D is inserted between the junction and the ribbon element.

To show the action of the pressure gradient system, assume that the length of all the small pipes is the same and the openings between the two sets are separated by a distance D. Under these conditions the line systems are nondirectional.

The difference between the forces on the two sides of the ribbon, assuming

that the mass mechanical reactance of the ribbon is large compared to the mechanical resistance of the damped pipes, may be expressed as

$$f_M = A \cos (2\pi f t) \sin \left(\frac{\pi D \cos \theta}{\lambda}\right) \qquad 8.113$$

where A = constant, including the pressure of the impinging sound wave and dimensions of the microphone.

If D is small compared to the wavelength, equation 8.113 becomes

$$f_M = A \frac{\pi D}{\lambda} \cos (2\pi f t) \cos \theta \qquad 8.114$$

Equation 8.113 shows that the force available for driving the ribbon is proportional to the frequency and the cosine of the angle θ.

Employing mass-controlled ribbon of mass m_r, the velocity is given by

$$\begin{aligned}
\dot{x} &= \frac{A}{j2\pi f m_r} \left(\frac{\pi D}{\lambda}\right) \cos (2\pi f t) \cos \theta \\
&= \frac{A}{2\pi m_r} \left(\frac{\pi D}{c}\right) \sin 2\pi f t \cos \theta \qquad 8.115
\end{aligned}$$

This quantity is independent of the frequency and, as a consequence, the ratio of the generated voltage to the pressure in the sound wave will be independent of the frequency.

The above discussion assumes that the lines are nondirectional. The directional characteristics of the individual lines of Fig. 8.72 are given by
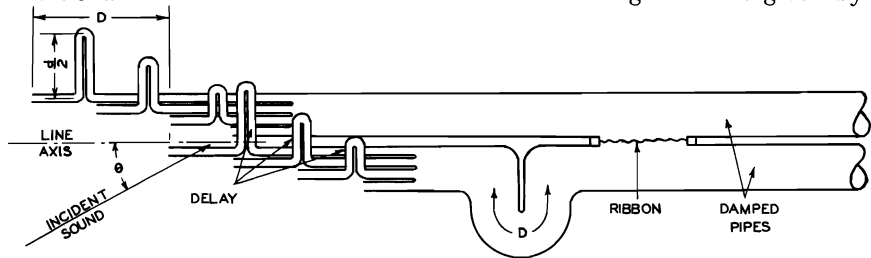
FIG. 8.72. Line microphone. Useful directivity on the line axis. This microphone consists of two lines of the type shown in Fig. 8.70 displaced by a distance D along the axis. In the line nearest the ribbon element a bend is inserted which introduces a path length D. The ribbon element measures the difference in pressure in the two lines.

equation 8.111. The directional characteristics of the microphone, shown in Fig. 8.72, for D small compared to the wavelength are the product of equations 8.112 and 8.115. The directional characteristics may be written as

$$R_\theta = \frac{\sin \frac{\pi}{\lambda} (l - l \cos \theta + d)}{\frac{\pi}{\lambda} (l - l \cos \theta + d)} \cos \theta \qquad 8.116$$

The directional characteristics of the microphone shown in Fig. 8.72 for various ratios of the length of the line to the wavelength for a delay of one-quarter times the length of the line are shown in Fig. 8.73. A measure of the value of a line with progressive delay and a pressure gradient element for improving the directivity may be obtained by comparing Fig. 8.73 with
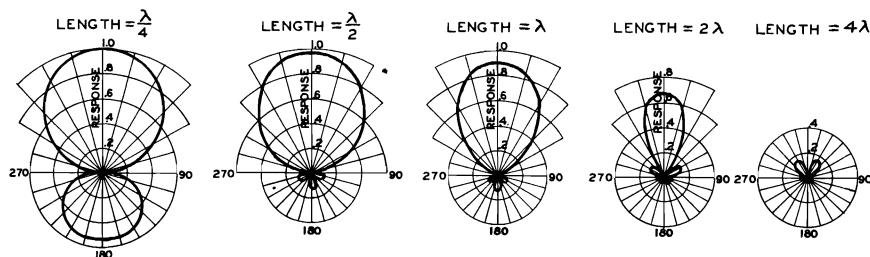

FIG. 8.73. The directional characteristics of the microphone shown in Fig. 8.72 for a time delay equivalent to one-quarter the length of the line as a function of the ratio of the length of the line to the wavelength. The polar graph depicts the output, in volts, as a function of the angle, in degrees.

Fig. 8.69. Employing these expedients approximately the same directivity can be obtained with a line of one-quarter the length of the simple line shown in Fig. 8.68.

4. *Ultradirectional Microphone.*[79]—Directional microphones employing lines of various types have been considered in the preceding section. These directional characteristics indicated considerable variation with frequency. Experience gained from work on reflectors indicated that a directional characteristic which varies with frequency is undesirable, principally due to the introduction of frequency discrimination for points removed from the axis. In addition, the response to reflected sound is a function of the frequency which alters the reverberation characteristics of received sound.

From the results of experiments upon directional systems, it appears that a microphone with a small solid angle of pickup would be useful in recording sound motion pictures, in television pickup, in certain types of sound broadcast as, for example, symphony and stage productions, and in many applications of sound reinforcing. However, the directional characteristics must be independent of the frequency. This can be accomplished by employing a number of separate lines, each covering a certain portion of the frequency range. It is the purpose of this section to describe an ultradirectional microphone consisting of five separate lines.

The ultradirectional microphone shown schematically in Fig. 8.74 consists of five units. Units 1, 2, and 3 are of the type shown in Fig. 8.68. Units 4 and 5 are of the type shown in Fig. 8.72. An electrical filter system is used to allocate the outputs of the units to their respective ranges. The response characteristics of the units with the filter systems are shown in Fig. 8.75.

[79] Olson, H. F., *Jour. Inst. Rad. Eng.,* Vol. 27, No. 7, p. 438, 1939.
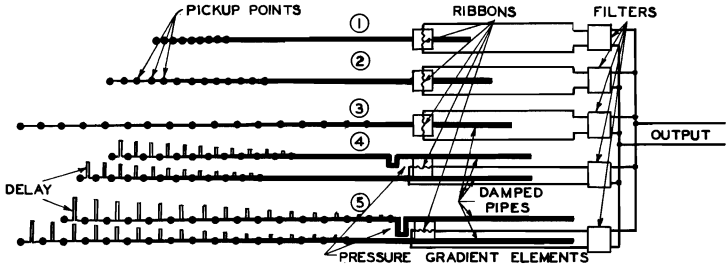
FIG. 8.74. Ultradirectional microphone consisting of five units. Units 1, 2, and 3 are of the type shown in Fig. 8.68. Units 4 and 5 are of the type shown in Fig. 8.72. An electrical filter system is used to allocate the output of the units to their respective ranges.
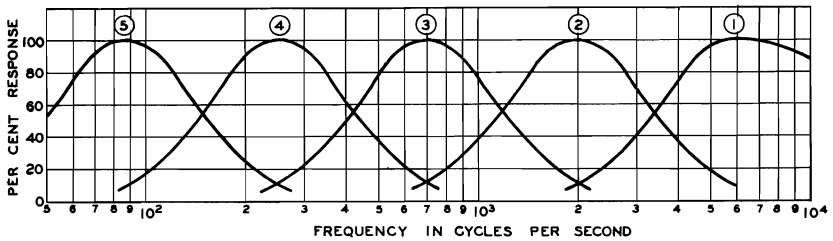

FIG. 8.75. Voltage response frequency characteristics of the units and electrical filter system shown in Fig. 8.74.

Fig. 8.76 illustrates the principles used in obtaining uniform directional characteristics. Fig. 8.76A is the directional characteristic of line 3 at 700 cycles. Fig. 8.76B shows the directional characteristics of lines 2 and 3 at 950 cycles. The resultant of these characteristics is also shown in Fig. 8.76B. The same is shown in Fig. 8.76C for 1250 cycles. In Figs. 8.76B and 8.76C the directional characteristic of line 2 is broader than Fig. 8.76A while the characteristic of line 3 is narrower. The resultant of
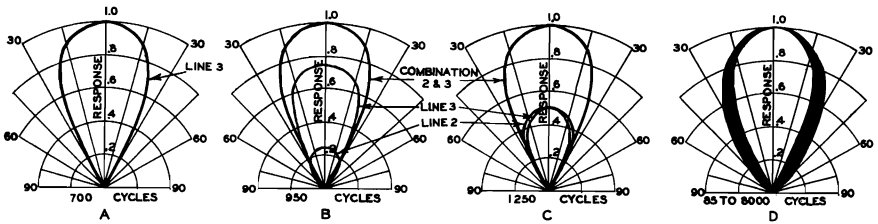
FIG. 8.76. A. The directional characteristic of line 3 of Fig. 8.74 at 700 cycles. B. The directional characteristics of lines 2 and 3 and the resultant at 950 cycles. C. The directional characteristics of lines 2 and 3 and the resultant at 1250 cycles. D. The directional characteristics of the microphone shown in Fig. 8.72 for the range from 85 to 8000 cycles fall within the shaded area.

lines 2 and 3 is a directional characteristic very close to Fig. 8.76A. The directional characteristics of the microphone shown in Fig. 8.76 for the range from 85 to 8000 cycles, except for the small lobes for angles greater than 90°, fall within the shaded area of Fig. 8.76D. Considering that this microphone has a frequency range of $6\frac{1}{2}$ octaves, it is a remarkably uniform directional characteristic.

8.7. Throat Microphone.[80,81,82,83]—The throat microphone is a microphone actuated by direct contact of the diaphragm with the throat. A perspective view and a sectional view of a carbon-type throat microphone are shown in Fig. 8.77. Since the acoustical impedance of the flesh of the

COMPLETE THROAT MICROPHONE   SECTIONAL VIEW OF UNIT
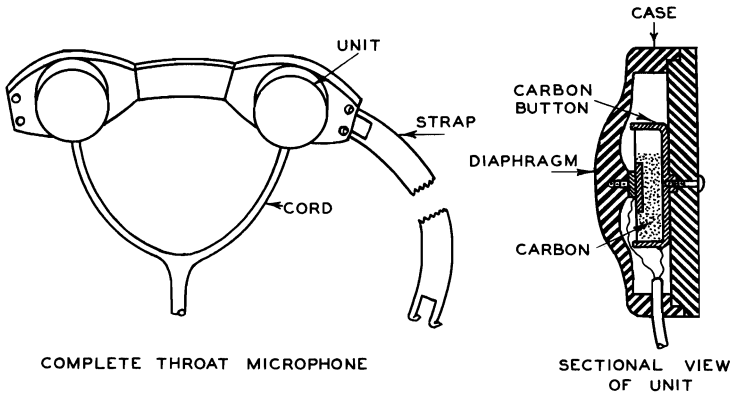
FIG. 8.77. Complete throat microphone and sectional view of the carbon-type throat microphone unit.

throat is very large compared to acoustical impedance of air, the acoustical impedance of the vibrating system of the throat microphone can be made correspondingly larger than the conventional air-type microphone. Since the vowel sounds originate in the throat and the consonants in the head the vowel sounds are predominant in the output. Furthermore, the high-frequency consonant sounds are attenuated in passing through the throat. Therefore, the high-frequency response must be accentuated to obtain intelligible speech. The units shown in Fig. 8.77 are of the carbon type. Other types of transducers as, for example, the magnetic type are also used in the throat microphone.

8.8. Lapel, Lavalier, and Boom Microphones.[84,85]—For certain applications, particularly in public address and announce systems, a microphone which can be hooked in the button-hole has been very useful. For the

[80] Shawn, J., *Communications*, Vol. 23, No. 1, p. 11, 1943.
[81] Martin, D., *Jour. Acous. Soc. Amer.*, Vol. 19, No. 1, p. 43, 1947.
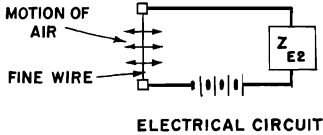[82] Greibach and Pacent, *Elec. Eng.*, Vol. 65, No. 4, p. 187, 1946.
[83] Erickson, J. R., *Bell Lab. Record*, Vol. 23, No. 6, p. 193, 1945.
[84] Olson and Carlisle, *Proc. Inst. Rad. Eng.*, Vol. 22, No. 12, p. 1354, 1934.
[85] Jones and Bell, *Jour. Soc. Mot. Pic. Eng.*, Vol. 19, No. 3, p. 219, 1932.

same applications a microphone mounted upon a small light boom supported in a variety of ways has also been used. An example of a boom-type microphone is the telephone operator's set in which the small carbon microphone is supported on a boom attached to the headband. A lavalier microphone is a term used to designate a small microphone supported by means of a small band around the neck in the form of a pendant. The principal purpose of the lapel, lavalier, and boom-type microphones is to allow the person to walk and turn freely without introducing any appreciable change in the output as would be the case if a stationary microphone were used. It also allows the talker to use his hands as contrasted to a hand-held microphone. Carbon, crystal, dynamic, and velocity microphones have been used for these applications. The general design of lapel, lavalier, and boom microphones is the same as the conventional microphones described in this chapter except that the size is smaller.

8.9. Hot-Wire Microphone.—The hot-wire microphone consists of a fine wire heated by the passage of an electrical current. Fig. 8.78. The cooling action due to the motion of air past the wire causes a change in electrical resistance of the wire. In a sound wave the particle velocity cools the wire. There are also some other minor cooling effects produced by the pressure in a sound wave. The change in resistance due to the passage of a sound wave may be used to detect the presence of a sound wave. However, the frequency of the electrical output is twice the frequency of the sound wave because the wire is cooled equally by both positive and negative particle velocities. The use of a direct current air stream for

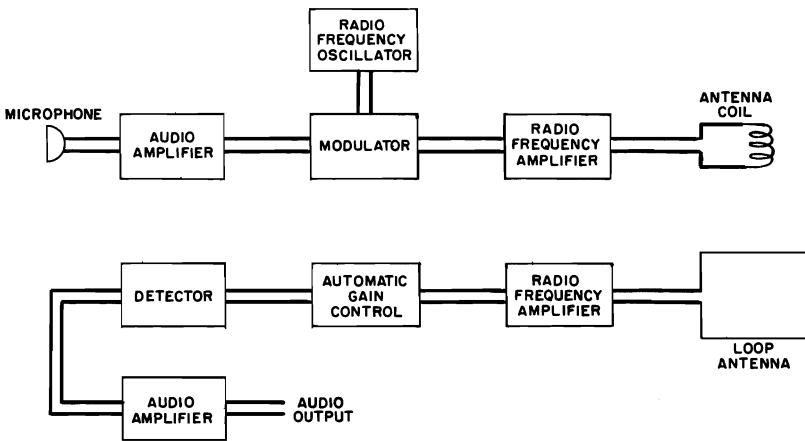FIG. 8.78. The elements of a hot-wire microphone.

FIG. 8.79. Schematic diagrams of a radio transmitter microphone and radio receiver.

polarization appears to be impractical. Therefore, this microphone cannot be used for the reproduction of sound.

8.10. Radio Microphone.[86]—A radio microphone consists of a small radio transmitter and microphone assembly and a radio receiver consisting of an antenna and receiver. The advantage of the system is the elimination of the microphone cable. Schematic diagrams of the radio transmitter microphone and the radio receiver are shown in Fig. 8.79. The radio transmitter is a conventional amplitude transmitter. The microphone, transmitter, batteries, and antenna are housed in a case about eight inches in length and one and one-half inches in diameter. In the system shown in Fig. 8.79, a loop antenna is coupled to the receiver. The loop consists of a single turn around the pickup area. A very wide range, automatic gain control system is required in the receiver to compensate for the variations in transmission due to standing waves.

8.11. Directional Efficiency of a Sound Collecting System.[87,88,89, 90,91,92]—The ratio of energy response of a directional microphone as compared to a nondirectional microphone, all directions being equally probable, is termed the directional efficiency. The directional efficiency of a microphone is given by

$$\text{Directional efficiency} = \frac{1}{4\pi} \int_0^{4\pi} f^2(\psi) \, d\Omega_\psi \qquad\qquad 8.117$$

where $f(\psi)$ = ratio of the voltage output for incidence at the angle ψ to that for $\psi = 0$, and

$d\Omega_\psi$ = element of solid angle at the angle ψ.

The directional efficiency of a microphone is a measure of the energy response to reverberation noise and other undesirable noise.

In many systems in which the directional pattern cannot be expressed in simple terms which can be integrated, the determination of the directional efficiency must be carried out by numerical integration. The directional efficiencies of cosine functions are easily determined. Directional patterns which are powers of the cosine function are plotted in Fig. 8.80. The directional efficiency for these patterns is also given. For the same signal to random noise, reverberation, etc., the directional microphone may be operated at $1/\sqrt{\text{directional efficiency}}$ times the distance of a nondirectional microphone.

[86] Phinney, Thomas A., *Trans. IRE*, Prof. Group Audio, Vol. AU-2, No. 2, p. 44, 1954.

[87] Olson, H. F., *Jour. Soc. Mot. Pic. Eng.*, Vol. 16, No. 6, p. 695, 1931.

[88] Olson, H. F., *Jour. Acous. Soc. Amer.*, Vol. 3, No. 1, p. 56, 1931.

[89] Olson, H. F., *Proc. Inst. Rad. Eng.*, Vol. 21, No. 5, p. 655, 1933.

[90] American Standards Association Sectional Committee Z-24, Report on Calibration of Microphones, *Jour. Acous. Soc. Amer.*, Vol. 7, No. 4, p. 300, 1936.

[91] Baumzweiger, B., *Jour. Acous. Soc. Amer.*, Vol. 11, No. 4, p. 447, 1940.

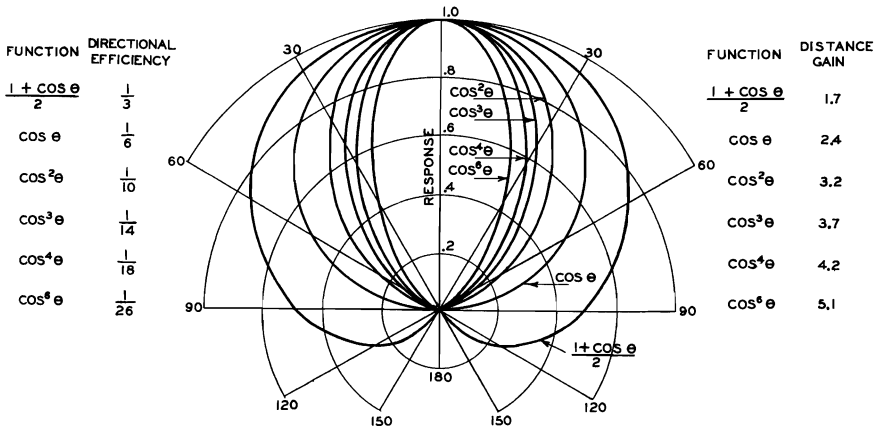[92] Bauer, B.B., *Trans. IRE*, PGA 8, July, 1952.

FIG. 8.80. The directional efficiency of microphones having directional characteristics, which are various cosine functions. The ratio of energy response of a directional microphone to the energy response of a nondirectional microphone for sounds originating in random directions is termed directional efficiency. The ratio of a distance at which a directional microphone may be operated as compared to a nondirectional microphone is also shown. All characteristics are considered to be unidirectional—that is, one lobe.

By means of the characteristics shown in Fig. 8.80, the efficiency of other characteristics may be obtained by comparing with the cosine function which has approximately the same shape.

The directional efficiency is also termed random efficiency and the inverse the directivity index.

8.12. Wind Excitation and Screening of Microphones.—There are three possible sources of excitation to which a microphone is subject when placed in a wind. There may be pressure fluctuations due to velocity fluctuations present in the wind even though the microphone is absent. There may be pressure fluctuations due to turbulence produced by the microphone in a wind otherwise free from pressure fluctuations, that is, in a wind of uniform velocity. There may be radiation from the first two sources. The effect of the first source may be reduced by screening which takes advantage of the wind pressure distribution over the microphone, the effect of the second by streamlining the microphone, and the third is minimized by reductions in the first and second sources.

The customary wind screen consists of a frame covered with silk enclosing the microphone (Fig. 8.81B). Very sheer silk reduces the response to wind without appreciable attenuation of the sound. A spherical shape has been found to offer the best shielding properties. The shielding properties increase with the volume of the shield.

In general, the response to wind is much higher for directions normal to the diaphragm by applying the principles of hydrodynamics. A wind screen has been developed which reduces the wind response of the micro-

phone. The Bernoulli[93] wind screen is shown in Fig. 8.81A. The wind pulses travel through the screen and exert a pressure on the diaphragm. These same pulses cause a reduction in pressure at the periphery 1. These two effects tend to balance each other and, therefore, the response to wind is reduced. This type of screen reduces the wind response about 12 db.
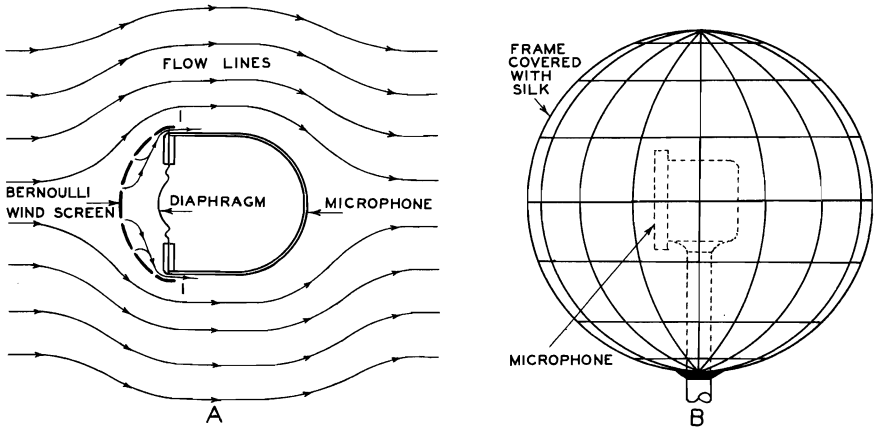

FIG. 8.81. Wind screens for microphones. A. Bernoulli wind screen applied to a dynamic microphone. B. Wind screen consisting of a wire frame covered with sheer silk.

8.13. Nonlinear Distortion in Microphones.—The sources of distortion in microphones are, in general, the same as in the case of loudspeakers. The two principal causes are due to nonlinear mechanical or acoustical elements and nonuniform magnetic field in dynamic types. The latter type of distortion can be made negligible in well-designed units. For example, in a velocity microphone the amplitude of the ribbon for a plane wave of 100 dynes per square centimeter at 30 cycles is less than a millimeter. The distortion due to a variation in the field over this distance is less than $\frac{1}{10}$ of one per cent. In the case of the velocity microphone the system is mass controlled and there are no nonlinear elements. The measured distortion (see Sec. 10.2C) in a velocity microphone for sound pressures up to 1000 dynes per square centimeter is less than $\frac{1}{3}$ of one per cent at 80 cycles. The most common source of nonlinear distortion in dynamic microphones originates in the suspension system. In some cases at the lower frequencies the harmonic distortion for a sound pressure of 100 dynes per square centimeter may be several per cent. This very high distortion is usually caused by instability of certain portions of the suspension due to dissymmetry of the corrugation and inhomogeneity of the material. As already pointed out, the distortion in carbon microphones is very high due to the nonlinear characteristics of granular contacts. Considerable improvement has been

[93] Phelps, W. D., *RCA Review*, Vol. 3, No. 2, p. 203, 1938.

made in carbon materials in recent years and the distortion, although still high, has been materially reduced.

8.14. Transient Response of Microphones.—The subject of transient response of vibrating systems, together with applications to loudspeakers has been considered in Sec. 6.25. The measurement of transient response of loudspeakers will be considered in Sec. 10.3G. The transient response of a microphone may be predicted from the mechanical or acoustical network of the vibrating system.

In the case of the vibrating system of the mass-controlled velocity microphone the response to transients is very good. The acoustical circuit of Fig. 8.35 may be reduced to the simplified acoustical circuit of Fig. 8.82A, provided the elements, M_A and M_R, the inertances due to the mass of the

ACOUSTICAL CIRCUITS
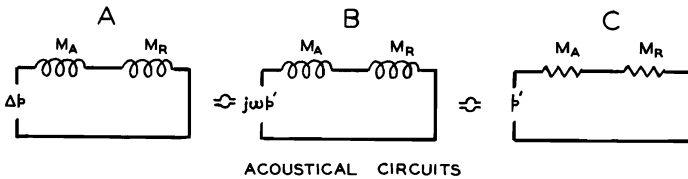
FIG. 8.82. Acoustical circuits of a velocity microphone. M_A and M_R = the inertance due to the air load and the ribbon mass. $\Delta p = j\omega p'$. Δp = the difference in pressure between the two sides of the ribbon. $\omega = 2\pi f$. f = the frequency. p' = a sound pressure proportional to the free-field sound pressure. Under the conditions depicted A is equivalent to B. B is equivalent to C.

air load and the mass of the ribbon are the controlling elements. For the audio-frequency range, the microphone may be designed so that the difference in pressure, Δp, between the two sides of the ribbon is proportional to the frequency (see Sec. 8.3). Under these conditions,

$$\Delta p = j\omega p' \qquad\qquad 8.118$$

where $\omega = 2\pi f$,

　　f = frequency in cycles per second, and

　　p' = a sound pressure proportional to the free-field sound pressure, in dynes per square centimeter.

Equation 8.118 shows that the acoustical circuit 8.82B is equivalent to acoustical circuit 8.82A. From acoustical circuit 8.82B the volume current is

$$U = \frac{j\omega p'}{j\omega M_R + j\omega M_A} = \frac{p'}{M_R + M_A} \qquad\qquad 8.119$$

Equation 8.117 shows that in Fig. 8.82 the acoustical circuit 8.82B may be reduced to acoustical circuit 8.82C. Since, in acoustical circuit 8.82C, an acoustical resistance is driven by a constant sound pressure, the response to

transients is perfect.   This has been substantiated by actual tests[93'] in which it is possible to obtain square waves from the output of a velocity microphone actuated by a loudspeaker with a very smooth, wide range response frequency characteristics.   In multiresonant systems with nonuniform response frequency characteristics it is impossible to obtain any semblance of a square wave from a loudspeaker microphone combination.

8.15. Noise in a Sound Pickup System.—Noise usually determines the lower limit of reproduction in a sound translating system.   The sources of noise in a sound pickup system, depicted in Fig. 8.83, follow: The ambient
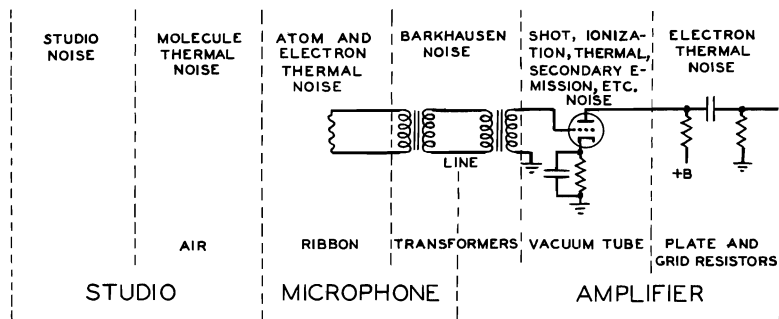
FIG. 8.83.   Sources of noise in a sound pickup system.

noise in the studio.   The noise due to the random pressures upon the diaphragm caused by the thermal agitation of the air molecules.   The noise due to the thermal agitation of the atoms in the diaphragm.   The noise due to the thermal agitation of the electrons in the conductor.   The noise due to the Barkhausen effect in the core of the transformer.   The noise due to shot effect, secondary emission, ionization, hum, etc., in the vacuum tube.   The noise due to the thermal agitation of the electrons in the plate resistor.

A. *Ambient Noise in the Studio.*—The ambient noise in the studio is usually one of the most important factors in determining the lower limit of reproduction from the standpoint of the pickup system.   The general ambient noise level in a studio varies from 10 db for a very quiet studio to 35 db for a noisy studio, as in the case of an audience.   The spectrum of room noise is shown in Fig. 12.38.   It will be seen that room noise is not uniform with respect to frequency.   In the case of thermal noise the generated voltage is proportional to the square root of the width of the frequency band regardless of the position in the frequency spectrum.

B. *Noise Due to Thermal Agitation of the Air Molecules.*—Superposed on the average atmospheric pressure there are fluctuations caused by the distribution of thermal velocities of air molecules.   The rms thermal sound

[93'] Olson and Preston, *RCA Review*, Vol. 7, No. 2, p. 155, 1946.

pressure, $\bar{p}$, in dynes per square centimeter, in the frequency interval between f_1 and f_2 may be obtained from the equation

$$\bar{p} = \sqrt{\int_{f_1}^{f_2} p_f{}^2 \, df} = \sqrt{\int_f^{f_2} 4kTr_A \, df} \qquad 8.120$$

where $p_f{}^2 \, df$ = square of the thermal acoustic pressure in the interval df.

df = frequency interval, in cycles per second,

r_A = acoustical radiation resistance, in acoustical ohms,

k = Boltzmann's constant, 1.37×10^{-16},

T = absolute temperature, in degrees Kelvin.

In the case of a diaphragm-type microphone the acoustical resistance, r_A, can be obtained from Sec. 8.2D1 and Fig. 8.10.

In the case of the velocity microphone the system is a doublet. Therefore, the acoustical radiation resistance is proportional to the fourth power of the frequency at the lower frequencies. The ultimate acoustical resistance on one side is $42/A$, where A = area of the ribbon. The acoustical resistance frequency characteristic of a velocity microphone is shown in Fig. 8.28.

C. *Noise Due to Thermal Agitation of the Atoms in the Vibrating System.*— Noise is created in the acoustical resistances in the vibrating system of a microphone. In the dynamic pressure type microphone the controlling element over a major portion of the frequency range is an acoustical resistance. The effective sound pressure generated in this element may be determined from equation 8.120 in the preceding section. This pressure is, of course, generated in the acoustical resistance and may be considered to be a generator in series with the acoustical resistance in the acoustical network.

In some instances it is more convenient to employ a mechanical network. In this case the rms thermal mechanical force, f_M, in dynes, in the frequency interval between f_1 and f_2 may be obtained from the equation

$$\bar{f}_M = \sqrt{\int_{f_1}^{f_2} f^2{}_{Mf} \, df} = \sqrt{\int_{f_1}^{f_2} 4kTr_M \, df} \qquad 8.121$$

where $f^2{}_{Mf} \, df$ = square of the thermal mechanical force in the interval df.

df = frequency interval, in cycles per second,

r_M = mechanical resistance, in mechanical ohms,

k = Boltzmann's constant, 1.37×10^{-16}, and

T = absolute temperature, in degrees Kelvin.

D. *Noise Due to Thermal Agitation of the Electrons in the Conductor.*— The thermal agitation of the electrons in the conductor of the electrical system of a microphone generates a fluctuating voltage.[94,95] The voltage,

[94] Johnson, J. B., *Phys. Rev.*, Vol. 32, No. 1, p. 97, 1928.
[95] Nyquist, H., *Phys. Rev.*, Vol. 32, No. 1, p. 110, 1928.

e, in abvolts, due to the thermal agitation of the electrons in a conductor is given by

$$e = \sqrt{4kT \, (f_2 - f_1) \, r_E} \qquad\qquad 8.122$$

where k = Boltzmann's constant, 1.37×10^{-16},

T = absolute temperature, in degrees Kelvin,

$f_2 - f_1$ = width of the frequency band, in cycles per second, and

r_E = electrical resistance of the conductor, in abohms.

E. *Noise Due to Barkhausen Effect in the Transformer.*—In the magnetization of a piece of ferromagnetic material by continuously varying magnetomotive force the resultant flux does not vary in a continuous manner but is made up of small steps. This phenomenon is termed the Barkhausen effect. In a well-designed transformer the only source of Barkhausen noise of any consequence is in the leakage reactance. Since the leakage reactance is small the Barkhausen noise will be relatively small. Furthermore in most high-grade transformer alloys the Barkhausen effect is also quite small.

F. *Noise in the Vacuum Tube.*—There are a large number of sources of noise in the vacuum tube. A few of these are shot effect, thermal noise in the plate impedance, ionization, and hum. These noises are treated at length in books[96] on vacuum tubes. The voltage generated in the plate of a well-designed triode, with an amplification of 20, from all sources except hum, is 2.8×10^{-5} volt. This is 1.4×10^{-6} volt at the grid terminals.

G. *Noise due to Thermal Agitation of the Electrons in the Plate Resistor.*— The noise voltage generated in the plate resistor can be obtained from equation 8.122 in Sec. 8.15D.

H. *Example of Noise in a Sound Pickup System.*—It is the purpose of this section to give the actual magnitude of the noise in each element of a sound pickup system. For the studio a very low level will be assumed namely, 10 db. The microphone will be the velocity type with a sensitivity of 600 microvolts per dyne per square centimeter at the 250-ohm terminals (see Sec. 8.3B). The final step-up transformer raises the impedance to 50,000 ohms at the grid of the triode vacuum tube. All noise voltages will be referred to the grid terminals of the vacuum tube. The frequency range is 30 to 15,000 cycles.

1. Ambient noise in the studio, 5.0×10^{-6} volt.
2. Noise due to thermal agitation of the air molecules, 2.5×10^{-6} volt.
3. Noise due to thermal agitation of the atoms in the ribbon vibrating system, negligible.
4. Noise due to thermal agitation of the electrons in the ribbon, 3.5×10^{-6} volt.
5. Noise due to the Barkhausen effect in the transformer, negligible.
6. Noise in the vacuum tube, 1.4×10^{-6} volt.

[96] Terman, "Radio Engineers Handbook," McGraw-Hill Book Company, New York, N.Y., 1943.
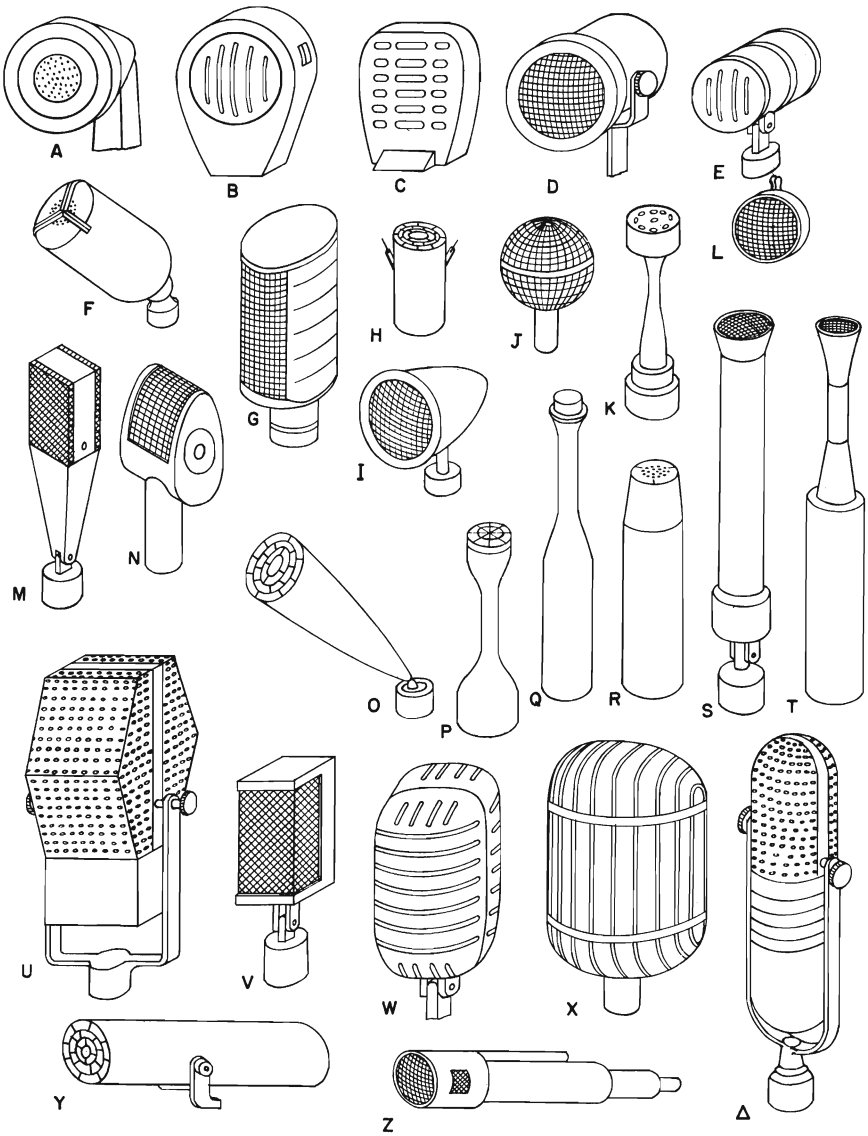
FIG. 8.84. Microphone shapes.   A. Pressure carbon.   B. Pressure carbon.   C. Pressure magnetic.   D. Pressure dynamic.   E. Pressure dynamic.   F. Pressure dynamic.   G. Pressure inductor.   H. Pressure dynamic.   I. Pressure crystal or dynamic.   J. Pressure crystal.   K. Pressure dynamic.   L. Pressure crystal. M. Velocity ribbon.   N. Velocity ribbon.   O. Pressure dynamic.   P. Pressure dynamic.   Q. Pressure condenser.   R. Pressure condenser.   S. Pressure dynamic. T. Pressure ribbon.   U. Velocity ribbon.   V. Unidirectional ribbon.   W. Uni-directional dynamic or crystal.   X. Unidirectional combination ribbon and dynamic. Y. Unidirectional ribbon.   Z. Unidirectional dynamic.   Δ. Unidirectional ribbon.

The above data show that the noises from all sources are comparable in magnitude. In a microphone of lower sensitivity the electrical noise sources in the conductor, resistor, and vacuum tube would be the limiting factors. For this reason it is very important to employ high-sensitivity microphones in wide frequency range and high-quality reproduction of sound.

8.16. Shapes of microphones.—Microphones may be classified in many different ways. One classification involves the type of response, namely, pressure, velocity, or combination of pressure and velocity. Another classification involves the type of transducer used to convert acoustical variations into the corresponding electrical variations, as for example, carbon, magnetic, dynamic, electrostatic, crystal, ribbon, etc. The configuration of the elements of a microphone is determined to a large extent by the type of response and the transducer. The outside shape in turn is largely determined by the configuration of the elements. A few typical examples of microphone shapes are shown in Fig. 8.84. The actual number of microphone shapes commercially available today is somewhere around 500. For this reason it is impossible to depict all of the different shapes. However, most of the microphones in use today follow the general patterns shown in Fig. 8.84. Some of the shapes shown are not necessarily the most common. The drawings are not replicas but reasonably accurate sketches. The pressure microphones with different types of transducers are as follows: A and B, Carbon; C, Magnetic; D, E, F, H, I, K, O, P, and S, Moving Coil, dynamic; G, Inductor dynamic; T, Ribbon; I, J, and L, Crystal; Q and R, Electrostatic. The velocity microphones with ribbon transducers are as follows: M, N, and U. Unidirectional microphones with different transducers are as follows: W and Z, Moving coil; W, Crystal; X, Combination ribbon and moving coil; V, Y, and Δ, Ribbon.

9

MISCELLANEOUS TRANSDUCERS

9.1. Introduction.—Interest in the science of sound reproduction has been stimulated during the past three decades by the almost universal use of the phonograph, radio, and the sound motion picture. The two most important acoustical elements in electrical reproduction of sound are loudspeakers and microphones. For this reason, considerable space has been given in this book to complete discussion of the most common instruments. There are innumerable electroacoustic, mechanoacoustic, and electromechanoacoustic transducers in use today for all types of applications. However, the major portion of the applications discussed in this text will be confined to sound reproduction. In addition to loudspeakers and microphones, the following transducers are in common use in various types of sound reproduction: telephone receivers, phonograph recorders and pickups, mechanical phonographs, magnetic tape or wire recorders and reproducers, sound motion-picture recorders and reproducers, sound powered phones, electrical musical instruments, and hearing aids. It is the purpose of this chapter to consider typical examples of these transducers.

9.2. Telephone Receivers.—A telephone receiver is an electroacoustic transducer actuated by energy in the electrical system and supplying energy to an acoustical system.

A. *Magnetic Telephone Receiver.*—The bipolar telephone receiver is a telephone receiver in which the alternating force, due to the alternating current in the electromagnet, operates directly upon a diaphragm armature of steel. A cross-sectional view, electrical circuit, and mechanical network of the vibrating system are shown in Fig. 9.1. The steel diaphragm is spaced a small distance from the pole pieces which are wound with insulated wire. A permanent magnet supplies the steady magnetic flux.

A schematic view and the magnetic network of the magnetic system is shown in Fig. 9.2. The elements of magnetic circuits and networks have been considered in Sec. 6.28.

The force,[1] in dynes, upon the diaphragm when an alternating current flows in the coils is

$$f_M = \frac{\phi^2}{4\pi A} = \frac{M^2_M}{4\pi R_D^2 A} + \frac{2M_M N i_{max} \sin \omega t}{R_D R_A A} + \frac{2\pi N^2 i^2_{max}}{R_A^2 A} \qquad 9.1$$
$$- \frac{2\pi N^2 i^2_{max} \cos 2\omega t}{R_A^2 A}$$

[1] Olson, "Dynamical Analogies," D. Van Nostrand Company, Princeton, N.J., 1943.

where ϕ = total flux, in maxwells, Fig. 9.2,

A = effective area of one pole, in square centimeters,

N = number of turns per coil,

R_D = reluctance of the permanent field circuit, in gilberts per maxwell,

R_A = reluctance of the alternating magnetic circuit, in gilberts per maxwell,

M_M = magnetomotive force of the magnet, in gilberts,

i_{max} = maximum current in the coil, in abamperes,

$\omega = 2\pi f$,

f = frequency, in cycles per second, and

t = time, in seconds.

The reluctance R_D of the permanent magnetic field circuit and the reluctance R_A of the alternating magnetic field circuit can be obtained from the magnetic network of Fig. 9.2.

The first and third term on the right-hand side of equation 9.1 represent a steady force, the second term represents a force of the same frequency and the last term represents a force of twice the frequency of the current in the coil.   Equation 9.1 shows the necessity for the polarizing field in order to obtain high sensitivity and to reduce second harmonic distortion.

The diaphragm in the bipolar telephone receiver is a circular plate clamped at the edge (see Sec. 3.5).   The effective mass of the diaphragm, when it is a clamped plate, is one-third the actual mass of the diaphragm.   The effective area of the diaphragm is one-third the total area of the diaphragm. The first resonant frequency is usually placed at 1000 cycles.   The effective compliance of the diaphragm can be obtained from the effective mass and the resonant frequency for the frequency region at and below the first resonant frequency of the diaphragm.   Referring to the mechanical network it will be seen that the system is stiffness controlled in the region below the resonant frequency.   This means that, for a constant driving force, f_M, the force applied to the compliance, C_{M3}, of the ear cavity will be independent of the frequency and hence the sound pressure in the ear cavity will be independent of the frequency.

The sound pressure delivered by a bipolar telephone receiver to a cavity as a function of the frequency is shown in Fig. 9.1.   In the range below the resonant frequency the response is independent of the frequency.   At the first resonant frequency of the diaphragm the response is very high. Above the resonant frequency the amplitude decreases rapidly with frequency.   The peak at 3000 cycles is the second resonant frequency of the diaphragm.

The pressure response frequency characteristic labeled A, Fig. 9.1, was obtained with no leak between the ear and the earcap.   In all hard earcaps a leak occurs between the ear and the telephone receiver and the acoustical

impedance presented to the telephone receivers is considerably more complex than that of an acoustical capacitance of a small cavity. In the case of telephone receivers worn in the customary manner the acoustical impedance has three components, namely, the resistive and inertive components due to the leak between the earcap and the ear and the acoustical capacitance due to the ear cavity. These factors will be considered in detail in the section on the testing of telephone receivers (see Sec. 10.4).

The pressure response frequency characteristic indicated as B in Fig. 9.1 was taken on an artificial ear which simulates the conditions encountered
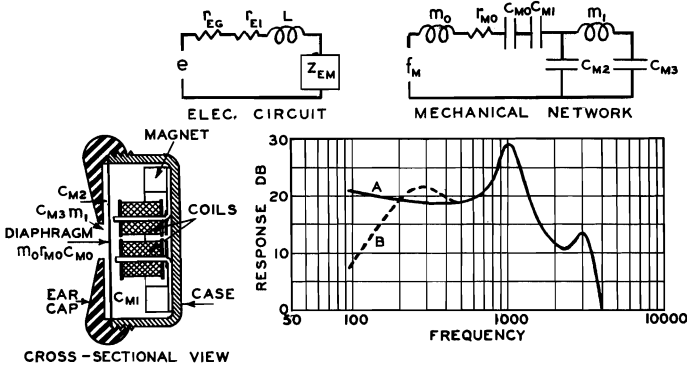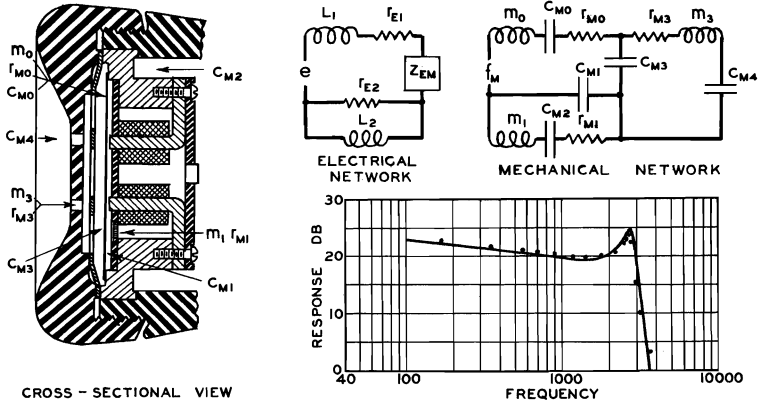

FIG. 9.1. Cross-sectional view, mechanical network, electrical circuit and response frequency characteristics of a bipolar telephone receiver. In the mechanical network, f_M = the mechanical driving force. m_0, r_{M0}, and C_{M0} = the mass, mechanical resistance, and compliance of the diaphragm. C_{M1} = the compliance due to the air in the case. C_{M2} = the compliance of the air space between the diaphragm and cover. m_1 = the mass of the air in the aperture in the cover. C_{M3} = the compliance of the ear cavity. In the electrical circuit, z_{EM} = the motional electrical impedance. L and r_{E1} = the damped inductance and electrical resistance of the coils. r_{EG} = the electrical resistance of the electrical generator. e = the voltage of the electrical generator. The graph shows the pressure response frequency characteristics. A. Receiver feeding a closed cavity. B. Receiver feeding an artificial ear.

in actual practice. The artificial ear (see Sec. 10.4B) introduces a leak which corresponds to the leak between the ear and the earcap. It will be seen that the effect of this leak is to reduce the response at the lower frequencies. Those familiar with telephone receivers have noticed that the low-frequency response is increased when the leak is reduced by pressing the telephone receivers tightly against the ears.

Since the development of the bipolar telephone receiver by Alexander Graham Bell the construction has remained essentially the same. Improvements have been made in sensitivity and response by the use of better materials. However, the clamped plate diaphragm characterized by prominent resonant peaks was retained. Referring to Fig. 9.1, it will be

seen that the peaks due to the first and second resonance fall within the response range.   These resonances not only introduce frequency distortion, but increase the intensity of reproduction of clicks due to the poor transient response (see Sec. 6.15).

A bipolar telephone receiver[2] has been designed in which all the prominent resonances within the response range have been eliminated and the response frequency characteristic improved both from the standpoint of uniformity as well as the frequency range.   The new telephone receiver is of the bipolar
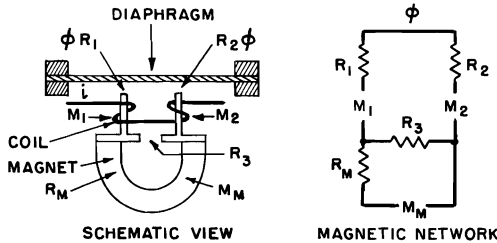
Fig. 9.2.   Schematic view and magnetic circuit of the magnetic system of a bipolar telephone receiver. In the magnetic network, M_M = the magneto-motive force developed by the permanent magnet. M_1 and M_2 = the magnetomotive forces developed by the current in the two coils. i = the current in the coils. R_M = the reluctance of the permanent magnet. R_1 and R_2 = the reluctances of the air gaps between the pole pieces and the diaphragm. R_3 = the reluctance of the shunt air gap. ϕ = the flux through the pole pieces and the diaphragm.

permanent magnet type.   The magnetic circuit consists of pole pieces of 45 per cent Permalloy, two straight bar magnets of Cobalt, and a Permandur diaphragm (see Sec. 6.19).   The use of these materials increases the efficiency of the unit.

The mechanical network of the mechanical system is shown in Fig. 9.3. The magnetic system and magnetic network of the telephone receiver of Fig. 9.3 is essentially the same as the magnetic network of Fig. 9.2.

The mass of the diaphragm is represented by m_0.   The compliance and mechanical resistance of the diaphragm are designated as C_{M0} and r_{M0}. The back of the diaphragm is enclosed, forming the compliance, C_{M1}, due to the resulting cavity.   This cavity is connected to the recess in the receiver handle by a hole in the plate.   A special silk covers this hole, forming the mechanical resistance r_{M1} and the mass m_1.   The volume due to the recess in the receiver handle forms the compliance C_{M2}.   The holes in the earcap form the mechanical resistance r_{M3} and the mass m_3.   The compliance C_{M3} is due to the cavity between the earcap and the diaphragm.

[2] Jones, W. C., *Jour. A.I.E.E.*, Vol. 57, No. 10, p. 559, 1939.

The response of this receiver was taken by measuring the pressure generated in a plain cavity. This cavity is designated by the compliance C_{M4}. The holes in the grid covering the receiver proper are large enough to have no reaction upon the response. A resilient screen of silk is mounted on the back of this grill. The mass of this screen is very small and is lumped with the diaphragm mass m_0.

The electrical portion of the circuit consists of the winding electrical resistance r_{E1} and inductance L_1. The eddy current elements are designated

CROSS – SECTIONAL VIEW

ELECTRICAL NETWORK

MECHANICAL NETWORK

FIG. 9.3. Cross-sectional view, mechanical network, electrical circuit and response frequency characteristics of an improved bipolar telephone receiver. In the mechanical network, f_M = the mechanical driving force. m_0, r_{M0}, and C_{M0} = the mass, mechanical resistance, and compliance of the diaphragm. C_{M1} = the compliance of the air in the cavity behind the diaphragm. m_1 and r_{M1} = the mass and mechanical resistance of the vent in the cavity behind the diaphragm. C_{M2} = the compliance of the air in the cavity in the handle. C_{M3} = the compliance of the air space between the diaphragm and cap. m_3 and r_{M3} = the mass and mechanical resistance of the apertures in the cap. C_{M4} = the compliance of the ear cavity. In the electrical network, z_{EM} = the motional electrical impedance. L_1 and r_{E1} = the damped inductance and electrical resistance of the coils. L_2 and r_{E2} = the inductance and electrical resistance due to eddy currents. e = the voltage of the electrical generator. The graph shows the pressure response frequency characteristic of the receiver feeding a plain cavity. The dots represent the response computed from the mechanical and electrical networks.

as r_{E2} and L_2. The electrical impedance[3] due to the mechanical system is designated by the motional electrical impedance z_{EM}. The force f_M can be obtained from equation 9.1.

The pressure response computed by means of the mechanical network is shown by the dots on the graph of Fig. 9.3. The measured pressure response is given by the curve on this graph. The agreement is very good and shows that it is possible to predetermine the response and to evaluate the effect of

[3] Olson, "Dynamical Analogies," D. Van Nostrand Company, Princeton, N.J., 1943.

changes in the constants of the component parts. Comparing the response of Figs. 9.1 and 9.3 it will be seen that large gains have been effected in uniform response over the entire range and in sensitivity from 1500 to 3000 cycles.

The new magnetic telephone receiver[4] shown in Figs. 9.4 and 9.5 differs radically from any previous commercial telephone receiver. The novel

CROSS-SECTIONAL VIEW

ELECTRICAL NETWORK
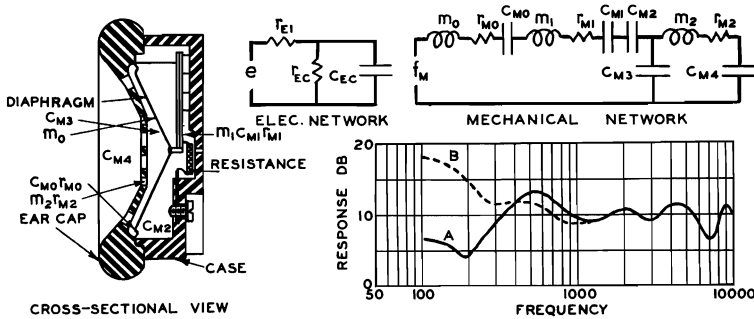
MECHANICAL NETWORK
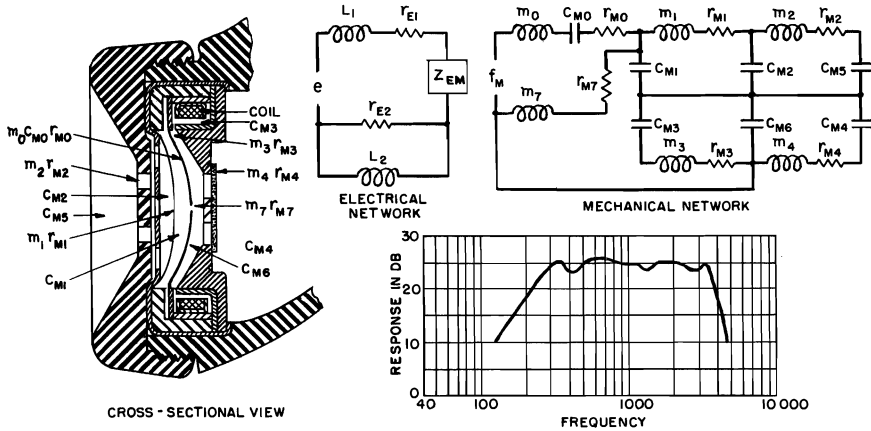
FIG. 9.4. Cross-sectional view, mechanical network, electrical circuit, and the response frequency characteristic of a ring armature telephone receiver. In the mechanical network, f_M = the mechanical driving force. m_0, r_{M0}, and C_{M0} = the mass, mechanical resistance and compliance of the diaphragm. m_1 and r_{M1} = the mass and mechanical resistance of the thin membrane. C_{M1} = the compliance of the air space between the diaphragm and the membrane. m_2 and r_{M2} = the mass and mechanical resistance of the holes in the ear cap. m_3 and r_{M3} = the mass and mechanical resistance of the air gap aperature. C_{M2} = the compliance of the air volume between the earcap and the thin membrane. C_{M3} = the compliance of the air volume in the coil space. m_4 and r_{M4} = the mass and mechanical resistance of the control mechanical resistance. C_{M4} = the compliance of the air volume in the handle. C_{M5} = the compliance of the air cavity between the earcap and the ear. C_{M6} = the compliance of the air cavity between the diaphragm and back enclosure. m_7 and r_{M7} = the mass and mechanical resistance of the small hole in the diaphragm. In the electrical network, z_{EM} = the motional electrical impedance. L_1 and r_{E1} = the damped inductance and electrical resistance of the coil. L_2 and r_{E2} = the inductance and electrical resistance due to eddy currents. e = the voltage of the electrical generator. The graph shows the pressure response frequency feeding a cavity.

features are a ring-type armature and a new magnetic and vibrating system. The driving armature is a disk of permandur. See Sec. 6.28. The action and the performance of the magnetic system may be obtained from the sectional view of the magnetic system and the magnetic network shown in Fig. 9.5. The action and performance of the vibrating system may be obtained from the mechanical network shown in Fig. 9.4. The diaphragm is a dome-shaped phenolic impregnated fabric material. A major portion

[4] Mott and Miner, *Bell Syst. Tech. Jour.*, Vol. 30, No. 1, p. 110, 1951.

of the control of the system is obtained from the mechanical resistance r_{M4} in the form of a cloth disk. See Sec. 5.1. The composite armature and diaphragm design results in a lower mechanical impedance and an appreciable increase in the ratio of effective area to effective mass of the diaphragm as compared to a simple magnetic disk-type combination diaphragm and armature shown in Figs. 9.1 and 9.3. The sensitivity is about 5 db higher

SECTIONAL VIEW   MAGNETIC NETWORK
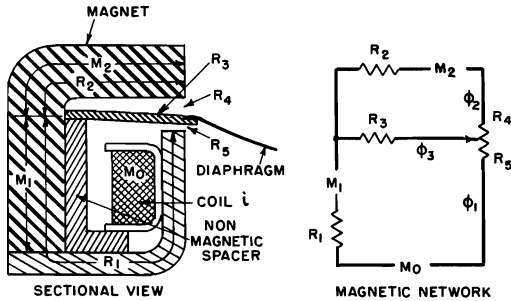
FIG. 9.5. Schematic view and magnetic network of the magnetic system of a ring-type telephone receiver. In the magnetic network, M_0 = the magnetomotive force developed by the permanent magnet. M_1 and M_2 = the magnetomotive forces developed by the current in the coil and in the lower and upper magnetic circuits. R_1 = the reluctance of the lower path in the magnetic material. R_2 = the reluctance of the upper path in the magnetic material. R_3 = reluctance of the armature. R_4 = the reluctance of the upper air gap. R_5 = the reluctance of the lower gap. ϕ_1 and ϕ_2 = the flux through the upper and lower air gaps. ϕ_3 = the flux through the armature. i = the current through the coil.

than the receiver shown in Fig. 9.3. The frequency range is extended about 700 cycles. With the lower mechanical impedance, the effect of holding the receiver off the ear does not produce as marked loss in intelligibility as in the case of the disk diaphragm armature types because the response frequency characteristic is not altered to any appreciable extent.

B. *Crystal Telephone Receiver.*—A crystal telephone receiver[5] consists of a light diaphragm connected to a Rochelle salt crystal (Fig. 9.6). The three corners of a "bender" crystal are fastened to the case. The fourth corner is connected to the diaphragm.

The electrical impedance of a crystal is primarily a capacitive electrical reactance. The electrical network of Fig. 9.6 shows that the low-frequency response can be raised relative to the high-frequency response by connecting a high electrical resistance in series with the telephone receivers. A relatively high electrical resistance must be used because the electrical impedance of the crystal is relatively high, being 80,000 ohms at 1000 cycles.

[5] Williams, A. L., *Jour. Soc. Mot. Pic. Eng.*, Vol. 32, No. 5, p. 552, 1939.

The performance of the vibrating system may be obtained from the mechanical network of Fig. 9.6.

A pressure response frequency characteristic with the telephone receiver feeding a plain cavity is indicated by B, Fig. 9.3. The pressure response frequency characteristic taken on an artificial ear is indicated by A, Fig. 9.6.
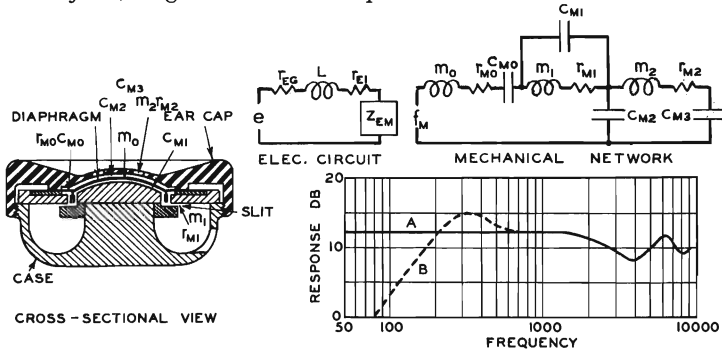

FIG. 9.6. Cross-sectional view, mechanical network, electrical network, and response frequency characteristics of a crystal telephone receiver. In the mechanical network, f_M = the mechanical driving force. m_0, r_{M0}, and C_{M0} = the mass, mechanical resistance, and compliance of the diaphragm. m_1, r_{M1}, and C_{M1} = the mass, mechanical resistance, and compliance of the crystal. C_{M2} = the compliance due to the air in the case. C_{M3} = the compliance of the air space between diaphragm and cover. m_2 and r_{M2} = the mass and mechanical resistance of the holes in the cover. C_{M4} = the compliance of the ear cavity. In the electrical network, C_{EC} and r_{EC} = the electrical capacitance and electrical resistance of the crystal. r_{E1} = the electrical resistance of the series resistor. e = the voltage of the electrical generator. The graph shows the pressure response frequency characteristics. A. Receiver feeding a closed cavity. B. Receiver feeding an artificial ear.

C. *Dynamic Telephone Receiver.*—A dynamic telephone receiver[6] consists of a light diaphragm coupled to a voice coil and a suitable mechanical network for controlling the response. A cross-sectional view of a typical dynamic telephone receiver is shown in Fig. 9.7. The mechanical network of the mechanical system is also shown in Fig. 9.7.

The electrical impedance,[7] in abohms, due to the mechanical system is given by

$$z_{EM} = \frac{(Bl)^2}{z_M}$$
9.2

where B = flux density in the air gap, in gausses,

l = length of the conductor in the voice coil, in centimeters, and

z_M = total mechanical impedance at f_M, in mechanical ohms.

In dynamic telephone receivers the flux density is relatively low and z_{EM} is small compared to r_{E1} and may be neglected.

[6] Wente and Thuras, *Jour. Acous. Soc. Amer.*, Vol. III, No. 1, p. 44, 1932.
[7] Olson, "Dynamical Analogies," D. Van Nostrand Company, Princeton, N.J., 1943.

The force, f_M, in dynes, is given by

$$f_M = Bli \qquad 9.3$$

where i, the current in abamperes, is obtained from the electrical circuit. In general, the force, f_M, is practically a constant and may be considered a constant in the mechanical network.

The pressure response frequency characteristic feeding a plain cavity is indicated by A, Fig. 9.7. The response measured on an artificial ear

FIG. 9.7. Cross-sectional view, mechanical network, electrical circuit, and response frequency characteristics of a dynamic telephone receiver. In the mechanical network, f_M = the mechanical driving force. m_0 = the mass of the diaphragm. r_{M0} and C_{M0} = the mechanical resistance and compliance of the suspension. C_{M1} = the compliance of the air space behind the diaphragm. m_1 and r_{M1} = the mass and mechanical resistance of the slit. C_{M2} = the compliance of the air space between the diaphragm and cover. m_2 and r_{M2} = the mass and mechanical resistance of the holes in the cover. C_{M3} = the compliance of the ear cavity. In the electrical circuit, z_{EM} = the motional electrical impedance. L and r_{E1} = the damped inductance and electrical resistance of the voice coil. r_{EG} = the electrical resistance of the electrical generator. e = the voltage of the electrical generator. The graph shows the pressure response frequency characteristics. A. Receiver feeding a closed cavity. B. Receiver feeding an artificial ear.

indicated by B, Fig. 9.7, shows that the response at the low frequencies is reduced due to the leak.

D. *Inductor Telephone Receiver.*—An inductor telephone receiver[8,9] is a telephone receiver in which a straight-line conductor, located in a magnetic field, drives a "V" shaped diaphragm. An acoustical network is used to compensate the response of the inductor-type telephone receiver shown in Fig. 9.8. The acoustical network compensates for the leak between the ear and the earcap. The effect of the leak between the ear and the earcap upon the response of a telephone receiver has been outlined in the preceding sections. Obviously, from a practical standpoint the performance of a

[8] Olson and Massa, *Jour. Acous. Soc. Amer.*, Vol. 6, No. 4. p. 240, 1935.

[9] Olson, H. F., *Jour. Soc. Mot. Pic. Eng.*, Vol. 27, No. 5, p. 537, 1936.

telephone receiver should be independent of the leak between the ear and the earcap. In order to design the vibrating system of the telephone receiver so that constant sound pressure will be delivered to the ear, the nature of the acoustical impedance looking through the aperture of the earcap must be considered as a part of the vibrating system. The acoustical impedance characteristic, looking through the aperture of the earcap of
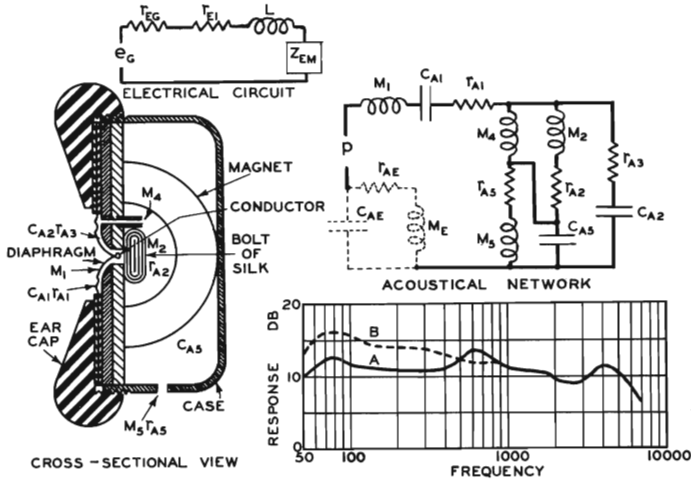
FIG. 9.8. Cross-sectional view, acoustical network and response frequency characteristic of an inductor telephone receiver. In the acoustical network, M_1 = the inertance of the diaphragm and conductor. C_{A1} and r_{A1} = the acoustical capacitance and acoustical resistance of the suspension system. M_2 and r_{A2} = the inertance and acoustical resistance of the bolt of silk. C_{A2} and r_{A3} = the acoustical capacitance and acoustical resistance of the cavity behind the diaphragm and the bolt of silk. M_4 = the inertance of the tube connecting the cavity behind the diaphragm with the case cavity. C_{A5} = the acoustical capacitance of the case volume. M_5 and r_{A5} = the inertance and acoustical resistance of the hole in the case. M_E, r_{AE}, and C_{AE} = the inertance, acoustical resistance, and acoustical capacitance of the ear. p = the driving pressure, $p = f_M/S$. f_M = the mechanical driving force. S = the area of the diaphragm. In the electrical circuit, z_{EM} = the motional electrical impedance. L and r_{E1} = the damped inductance and electrical resistance of the conductor. r_{EG} = the electrical resistance of the electrical generator. e = the voltage of the electrical generator. The graph shows the pressure response frequency characteristics. A. Receiver feeding an artificial ear. B. Receiver feeding a plain cavity.

a telephone receiver, is shown in Fig. 10.36, Sec. 10.4B. These characteristics show that the acoustical impedance is positive and increases with frequency up to 400 cycles; between 300 and 500 cycles it is practically resistive and above 400 cycles it is negative and decreases with frequency.

A generalization of the requirements for maintaining constant sound pressure in the ear cavity under these conditions is as follows: the velocity of the diaphragm below 300 cycles must be inversely proportional to the frequency, between 300 cycles and 500 cycles the velocity should be independent of the frequency, and above 500 cycles the velocity should be proportional to the frequency.

The acoustical network of a telephone receiver which delivers practically uniform sound pressure to the ear cavity in the presence of a normal leak is shown in Fig. 9.8. The acoustical network of the ear is shown dotted. The "V" shaped diaphragm is driven by a straight conductor located in the bottom of the "V."

The electrical circuit of the inductor telephone receiver is shown in Fig. 9.8. The pressure p may be considered to be independent of the frequency.

The pressure response frequency characteristic taken on an artificial ear is indicated by A, Fig. 9.8. The constants were chosen to give the smoothest response between 60 and 7000 cycles. The pressure response frequency characteristic with the receivers feeding a plain cavity is indicated by B, Fig. 9.8. The small difference between the response with and without a leak indicates the effectiveness of this type of vibrating system in minimizing the effect of the leak between the ear and the earcap.

A dynamic telephone receiver[10] employing an acoustical system similar to the inductor telephone receiver described above has also been developed. The acoustical network is similar to that of the inductor telephone receiver shown in Fig. 9.8. The essential difference between the inductor and dynamic acoustically compensated telephone receivers resides in the driving system, in the former a straight conductor is used to drive "V" diaphragm while in the latter a circular voice coil is used to drive a dome-shaped diaphragm.

9.3. Phonograph.—A phonograph is used to designate a system for recording and reproducing sound from a record. Today, a phonograph usually refers to a system in which a stylus (needle) follows the undulations in the groove of a record and transforms these undulations into the corresponding acoustical or electrical variations. The record may take the form of a cylinder or a flat disk. Today, the flat disk record is almost universally used for entertainment while the cylindrical record is used for dictographs. In the hill and dale or vertical-type record the undulations are cut in a direction normal to the surface. In the lateral-type record the undulations are cut in a direction parallel to the surface of the record. The lateral-type records are used for home reproduction. Both vertical- and lateral-type records are used for high quality reproductions as, for example, in transcriptions for broadcasting. The complete system used in the recording and processing of phonograph records will be considered in Sec. 13.16. It is the purpose of the sections which follow to consider phono-

---

[10] Anderson, L. J., *Jour. Soc. Mot. Pic. Eng.*, Vol. 37, No. 3, p. 319, 1941.

graph recorders, mechanical phonograph, record player, phonograph pickup records, and distortion in phonograph reproduction.

A. *Recording Systems.*—1. *Recorders.*—A phonograph recorder is an equipment for transforming acoustical or electrical signals into motion of approximately like form and inscribing such motion in an appropriate medium by cutting or embossing. For the recording of disk phonograph records, the electrical phonograph recorder replaced the mechanical recorder about three decades ago. An electrical phonograph recorder is shown in Fig. 9.9. The lacquer disk used in recording the master record is placed on
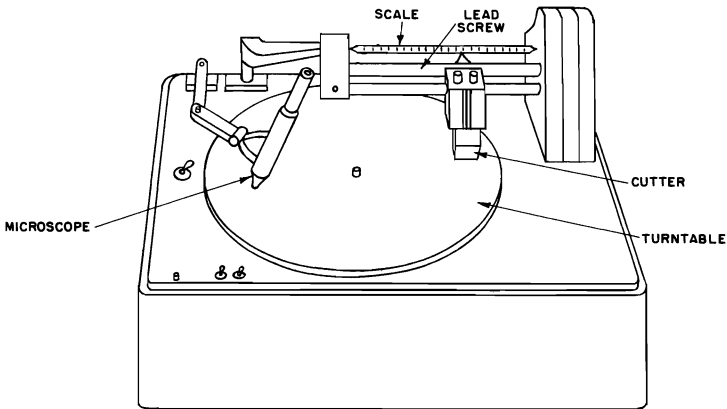
FIG. 9.9. Perspective view of a disk phonograph recorder.

the recording turntable. To insure against spurious motions, the turntable is made very heavy. A suitable mechanical filter is placed between the driving motor and the turntable so that uniform rotational motion of the turntable will be obtained. The drive system is arranged so that all the standard record speeds can be cut. In general, the recording turntable is driven with a synchronous motor to insure uniform absolute speed of rotation. The lead screw drives the cutter in a radial direction so that a spiral groove is cut in the record. Lead screws of different pitches are used ranging from 100 to 500 grooves per inch. In some recordings a variable pitch is used. In this procedure the spacing between the grooves is made to correspond to the amplitude—small spacing for small amplitudes and large spacing for large amplitudes. Under these conditions the maximum amount of information can be recorded on a record. The material which is removed in the cutting process is in the form of a fine thread. The thread is pulled into the open end of a pipe located near the cutting stylus and connected to a vacuum system. A complete phonograph recording system is described in Sec. 13.6.

2. *Lateral Cutter.*—In the lateral type of recording the undulations are cut in a direction parallel to the surface of the record and perpendicular to

the groove. Perspective and sectional views, the electrical circuit, and the mechanical network of a lateral-type magnetic phonograph cutter[11] are shown in Fig. 9.10.

A schematic view of the magnetic system and magnetic circuit of the
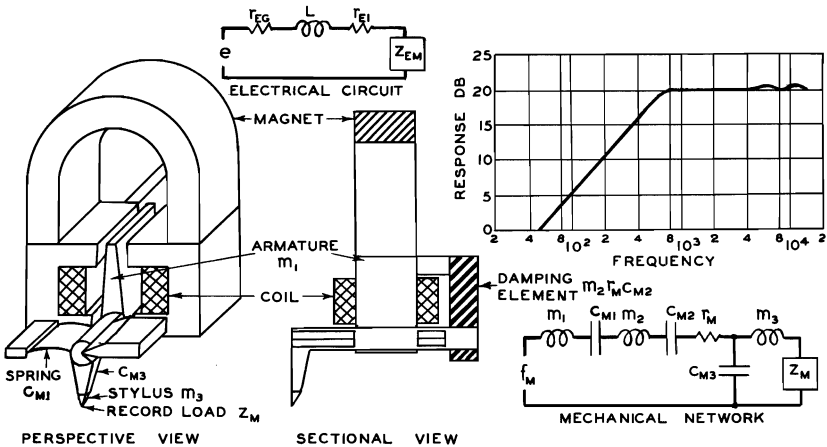
FIG. 9.10. Perspective and sectional views, mechanical network and velocity response frequency characteristic of a lateral-type phonograph recorder. In the mechanical network, f_M = the mechanical driving force. m_1 = the mass of the armature. C_{M1} = the compliance of the restoring spring. m_2, r_M, and C_{M2} = the mass, mechanical resistance, and compliance of the damping element. M_3 and C_{M3} = the mass of the stylus and holder. z_M = the mechanical impedance of the load presented to the stylus. In the electrical circuit: z_{EM} = the motional electrical impedance. L and r_{E1} = the damped inductance and electrical resistance of the coil. r_{EG} = the electrical resistance of the generator. e = the voltage of the electrical generator. The graph depicts the velocity response frequency characteristic of the recorder.

lateral-type phonograph cutter is shown in Fig. 9.11. The force f_M, in dynes, upon the armature is given by

$$f_M = \frac{(\phi_1 + \phi_2)^2}{8\pi A} - \frac{(\phi_1 - \phi_2)^2}{8\pi A} = \frac{\phi_1 \phi_2}{4\pi A} \qquad 9.4$$

where ϕ_1 = flux, in maxwells, through the reluctances R_1 and R_2 produced by the permanent magnet, and

ϕ_2 = flux, in maxwells, through the reluctances R_1 and R_2 produced by the current in the coil.

The reluctance of the armature may be neglected because it is small compared to the reluctances of the air gaps R_1, R_2, R_3, and R_4. The flux ϕ_1

[11] Hasbrouck, H. J., *Jour. Soc. Mot. Pic. Eng.*, Vol. 32, No. 3, p. 246, 1939.

due to the magnetomotive force M_D of the magnet can be obtained from the magnetic network of Fig. 9.11. The magnetomotive force M_A, in gilberts, due to a current in the coil is given by

$$M_A = 4\pi Ni \qquad 9.5$$

where N = number of turns in the coil, and
i = current in the coil, in abamperes.

The flux ϕ_2, in maxwells, due to the magnetomotive force M_A developed by the current in the coil can be computed from the magnetic circuit of Fig. 9.11. The force applied to the armature in a magnetic driving system is

SCHEMATIC VIEW   MAGNETIC NETWORK
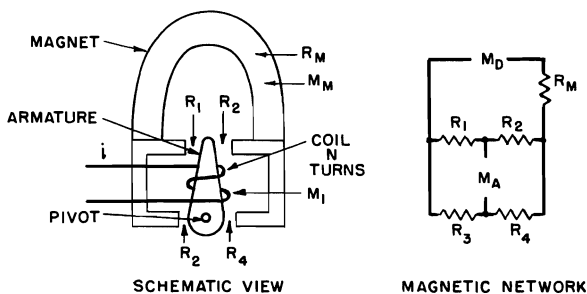
FIG. 9.11. Schematic view and magnetic network of a lateral-type phonograph recorder. In the mangetic network, M_D = the magnetomotive force developed by the permanent magnet. M_A = the magnetomotive force developed by the current in the coil. R_M = the reluctance of the path in magnetic material. R_1 and R_2 = the reluctances of the two variable air gaps. R_3 and R_4 = the reluctances of the two fixed air gaps. i = the current in the coil.

proportional to the current in the coil. The mechanical network is designed so that, for constant applied force, the amplitude will be independent of the frequency below approximately 800 cycles and the velocity will be independent of the frequency above approximately 800 cycles.

A sectional view, the mechanical circuit, and the electrical system of a feedback lateral-type phonograph cutter[12,13] is shown in Fig. 9.12. The vibrating system is of the dynamic type with two wire coils. The vibrating system is designed so that there is a single degree of freedom over the operating frequency range. The response frequency characteristic of the vibrating system shows that it is a system of one degree of freedom from 30 cycles to 16,000 cycles with the fundamental resonant frequency at 700 cycles. The output of the sensing coil is fed to the input of the amplifier. The output

[12] Davis, C. C., *Jour. Audio Eng. Soc.*, Vol. 2. No. 4, p. 228, 1954.
[13] Morgan, A. R., Unpublished Report.

of the amplifier is fed to a driving coil in an out-of-phase relationship.   The signal is fed to the input of the amplifier.   With the feedback in operation the velocity of the vibrating system is practically independent of the fre-
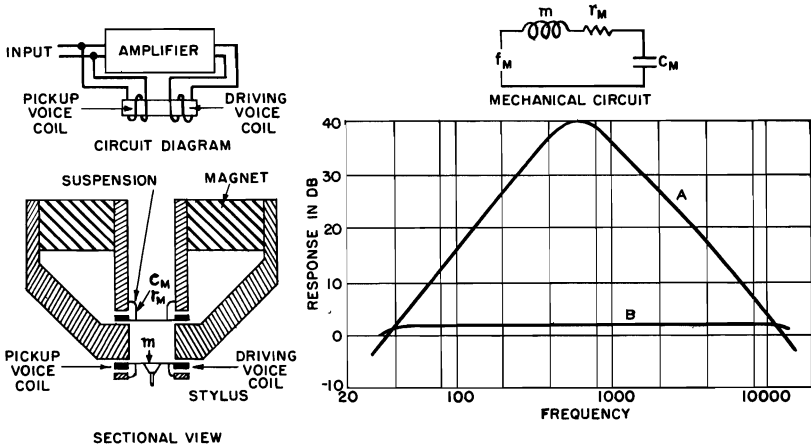
FIG. 9.12. Sectional view, mechanical circuit, electrical system, and velocity response frequency characteristic of a feedback lateral-type phonograph cutter. In the mechanical circuit, f_M = the mechanical driving force. m, r_M, and C_M = the mass, mechanical resistance and compliance of the vibrating system.   In the graph, A = the velocity frequency response characteristic without feedback. B = the velocity response frequency characteristic with feedback.
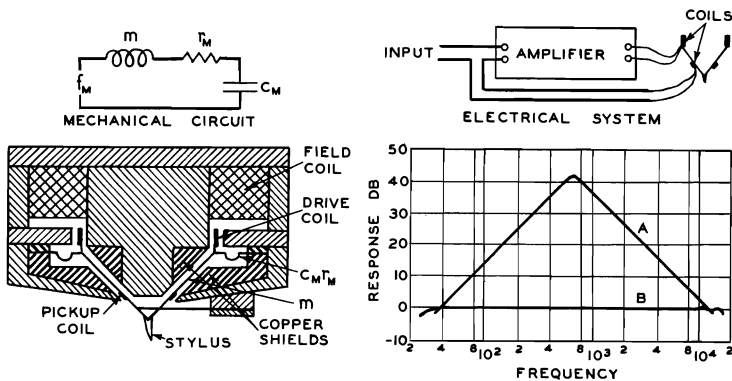
FIG. 9.13. Sectional view, mechanical circuit, electrical system, and velocity response frequency characteristic of a feedback vertical-type phonograph cutter.   In the mechanical circuit, f_M = the mechanical driving force. m, r_M, and C_M = the mass, mechanical resistance, and compliance of the vibrating system.   In the graph, A = the velocity response frequency characteristic without feedback. B = the velocity response frequency characteristic with feedback.

quency over the frequency range from 30 to 16,000 cycles. The input to the amplifier can be compensated to provide the desired recording characteristic.

3. *Vertical Cutter.*—In the vertical type of recording the undulations are cut in a direction perpendicular to the surface of the record. A sectional view, the mechanical circuit, and electrical system of a feedback type of a vertical-type phonograph cutter[14] are shown in Fig. 9.13. The mechanical system as depicted by the mechanical circuit is a system of one degree of freedom. The response frequency characteristic of the system is designated as *A* in Fig. 9.13. By feeding the output of the pickup coil in out-of-phase relationship with the input to the amplifier, the velocity response frequency characteristic with about 40 db of feedback will be as shown in Fig. 9.13B. The use of a feedback in conjunction with a simple vibrating system yields a uniform response frequency characteristic. The amplifier which drives the system can be compensated to yield the appropriate recording response frequency characteristic.

4. *Recording Characteristics.*—The velocity response frequency of one type of standard frequency record used in obtaining the response frequency characteristics of phonograph pickups and mechanical phonographs is shown in Fig. 9.14. The reason being that the characteristic shown in
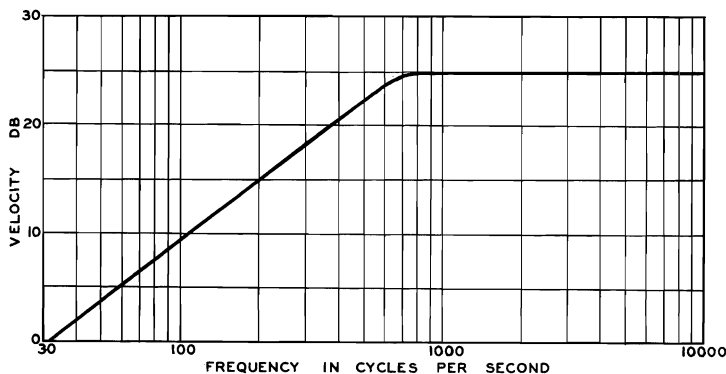
FIG. 9.14.   Typical velocity response frequency characteristic of an old type, standard frequency phonograph record.

Fig. 9.14 was used up to about two decades ago in recording phonograph records. To prevent overcutting the groove, the recording is made so that the amplitude is essentially independent of the frequency below approximately 800 cycles. The velocity under these conditions falls off 6 db per octave with decrease of the frequency. Above approximately 800 cycles the recording is made so that the velocity is independent of the frequency. The amplitude in this frequency range falls off 6 db per octave with increase of the frequency.

[14] Vieth and Wiebusch, *Jour. Soc. Mot. Pic. Eng.,* Vol. 30, No. 1, p. 96, 1938.

In radio transcription recording, the orthacoustic[15] type of recording characteristic is employed. The orthacoustic velocity frequency characteristic for constant voltage input to the microphone amplifier is shown in Fig. 9.15. This characteristic is essentially a constant amplitude frequency characteristic. In reproduction of the record, an inverse response frequency
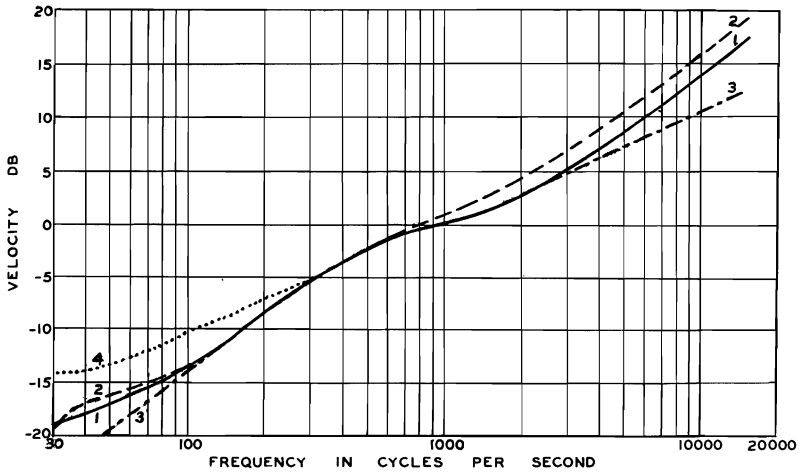
FIG. 9.15. Velocity frequency characteristics of various types of recording characteristics used in commercial phonograph records. 1. RIAA Standard. 2. Orthacoustic Standard. 2 and 3 represent the upper and lower high-frequency limits and 4 and 3 represent the upper and lower low-frequency limits of the recording characteristics in use in commercial phonograph records.

characteristic is used to obtain a uniform over-all response frequency characteristic. The use of this type of response frequency characteristic reduces ground noise and distortion.

In recording of commercial phonograph records, high-frequency accentuation is employed. The compensation used today varies over wide limits among different record manufacturers from the lowest characteristic shown in Fig. 9.15 to the orthacoustic characteristic of Fig. 9.15. The RIAA characteristic[16] is the proposed standard playback characteristic for lateral disk recordings.

Standard frequency records exhibiting the characteristics of Fig. 9.15 have replaced the record characteristic of Fig. 9.14. The use of frequency records exhibiting the characteristics of Fig. 9.15 gives the performance of the system without any further corrections. In the reproduction of commercial phonograph records, an inverse response frequency characteristic is employed to obtain a uniform over-all response frequency characteristic.

[15] Recording and Reproducing Standards, *Proc. Inst. Rad. Eng.*, Vol. 30, No. 8, p. 355, 1942.
[16] Record Industry Association of America.

The use of high-frequency accentuation, as shown in Fig. 9.15, reduces record ground noise and distortion.

5. *Heated Stylus.*[17]—In the production of phonograph records the original recording is cut in a lacquer disk by means of the cutting stylus actuated by the cutter. The walls of the resultant groove are not smooth. This lack of smoothness is due to nonlinear effects which occur in all cutting processes. It was discovered that by heating the cutting stylus the roughness of the groove wall was reduced. As a matter of fact, an improvement in signal-to-noise of 20 db was obtained by means of the heated stylus. This gain in signal-to-noise is most apparent in the high-frequency range. Two types of heating have been employed as shown in Fig. 9.16. In Fig. 9.16A, a fine wire is wound around the stylus. An electrical current is sent through the wire which heats the stylus. The mass of the fine wire is small compared to the mass of the stylus and, therefore, does not effect the vibrating performance. In Fig. 9.16B, the stylus is plated with a thin ring of copper.
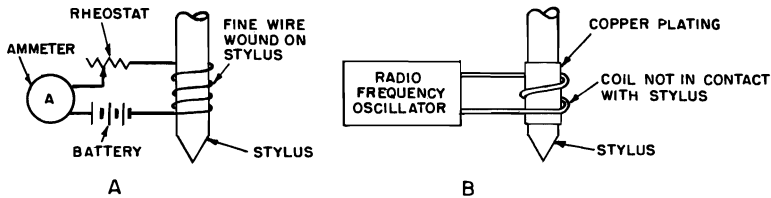
FIG. 9.16. Two systems for heating the cutting stylus of a phonograph recorder. A. Direct current system. B. Radio frequency system.

A coil fed from a radio-frequency oscillator induces a current in the copper ring which heats the stylus. The heavy coil which surrounds the stylus is not in contact with the stylus.

B. *Reproducing Systems.*—1. *Record Player.*—A phonograph record player is an equipment for transforming the undulations in a groove in a medium into the corresponding electrical or acoustical variations. In the early days of the phonograph the acoustical phonograph was used exclusively. However, about three decades ago the electrical phonograph was developed. Today, the electrical form of reproduction has almost completely displaced the mechanical phonograph.

a. *Electrical Record Player.*—An electrical record player and changer is shown in Fig. 9.17. The record is rotated by the reproducing turntable at the same angular speed as that used in recording. The turntable is rotated by means of an electric motor. The stylus or needle of the pickup follows the wavy spiral groove and generates a voltage corresponding to the undulations in the groove. Pickups for use in disk-record reproduction will be described in later sections. The record player and changer shown in Fig. 9.17 will play three rotational speeds, namely 33⅓, 45 and 78 RPM. It will also play and change a stack of eight records. The small spindle is used for

[17] Bachman, W. S., *Audio Eng.*, Vol. 34, No. 6, p. 11, 1950.

the reproduction of 33⅓ and 78 RPM records.   The large spindle is used for the reproduction of 45 RPM records.

Another type of record changer and player plays and changes a single type of record.   One of the most common is the 45 RPM record player and changer.[18]

A record player is the simplest type of disk-record reproducer.   It is manually operated.   It ranges from the simplest of all disk-record players

RECORD

PICKUP ARM

PICKUP

45 RPM
SPINDLE

SPEED
SELECTOR

START
STOP
LEVER
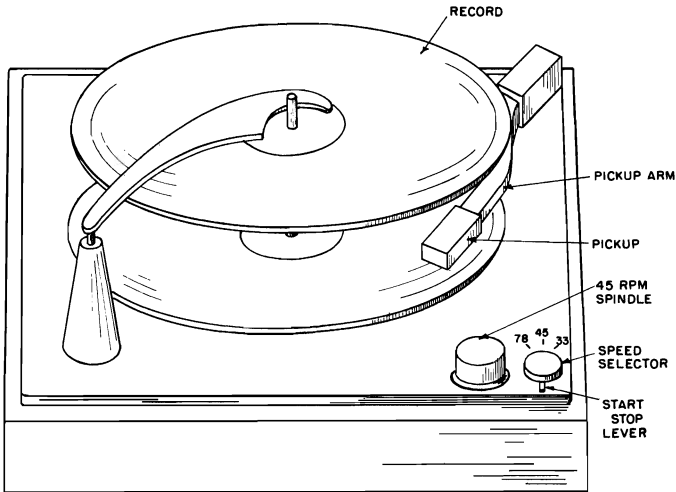
Fig. 9.17.   Perspective view of a three-speed disk phonograph player.

to elaborate transcription types with very uniform rotational velocity and high quality pickups.

b. *Mechanical Phonograph*.[19]—A mechanical phonograph is a mechano-acoustic transducer actuated by a phonograph record and by means of an acoustical system radiates acoustical energy into a room or open air.

The record is rotated by a turntable at the same angular speed as that used in recording.   The turntable is rotated either by a spring motor or by an electric motor.   A cross-sectional view of the mechanical network of the reproducing system of a mechanical phonograph is shown in Fig. 9.18.   The system consists of a diaphragm coupled to a needle which follows the wavy spiral groove and generates a sound output which corresponds to the undulations in the groove.   To improve the radiation efficiency, the diaphragm is coupled to a horn.   The record mechanical impedance is usually large compared to the mechanical impedance of the remainder of the system save at the high frequencies.   The record mechanical impedance is a function of the type of material.   Obviously, it is higher for the harder materials.

[18] Carson, Burt, and Reiskind, *RCA Review*, Vol. 10, No. 2, p. 173, 1949.
[19] Maxfield and Harrison, *Bell Syst. Tech. Jour.*, Vol. 5, No. 3, p. 493, 1926.

The generator in the mechanical network of this system is of the constant current type. That is, f_M delivers constant velocity to the mechanical network. Under these conditions the velocity is independent of the impedence of the load.

The response frequency characteristic of a mechanical phonograph of the console type is shown in Fig. 9.18.
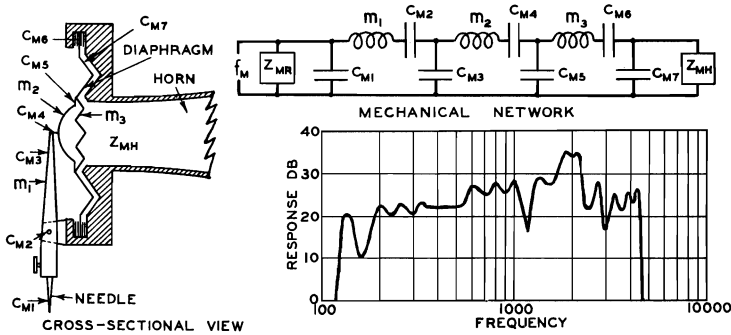
FIG. 9.18. Cross-sectional view, mechanical network and response frequency characteristic of a mechanical phonograph. In the mechanical network, z_{MR} = the mechanical impedance of the record. C_{M1}, C_{M2}, C_{M3}, C_{M4}, C_{M5}, C_{M6}, and C_{M7} = the compliances of the needle, the needle holder, the needle holder arm pivot, the needle holder arm, the connector, the spider, the diaphragm suspension, and the coupling chamber. m_1, m_2, and m_3 = the masses of the needle holder arm, the spider, and the diaphragm. z_{MH} = the mechanical impedance at the throat of the horn. f_M = the force generated by a velocity generator having the characteristic of Fig. 9.14. The graph shows the pressure response frequency characteristic of a console-type mechanical phonograph using a record having a characteristic of Fig. 9.14.

2. *Phonograph Pickups.*—A phonograph pickup is an electromechanical transducer actuated by a phonograph record and delivering energy to an electrical system, the electrical current having frequency components corresponding to those of the wave in the record. The systems for converting the mechanical vibrations in to the corresponding electrical variations are as follows: magnetic, variable resistance, condenser, electronic, dynamic, and crystal. It is the purpose of this section to consider examples of some of the most common phonograph pickups in use today.

a. *Crystal Pickup.*—A crystal pickup[20] is a phonograph pickup which depends for its operation on the piezoelectric effect. The crystal in use today is Rochelle salt. A cross-sectional view of a typical crystal pickup used in commercial phonographs employing a replaceable needle is shown in Fig. 9.19. The needle, driven by the record, is coupled to the crystal. The elements of the system and the mechanical network are shown in Fig. 9.19. The displacement of the crystal can be determined from the mechanical

[20] Williams, A. L., *Jour. Soc. Mot. Pic. Eng.*, Vol. 32, No. 5, p. 552, 1939.

network of the mechanical system and the velocity of the generator obtained from Fig. 9.15. The voltage output of the crystal is proportional to the displacement. The internal electrical impedance of the crystal increases with the decrease in frequency since the crystal is essentially an electrical capacitance. The open circuit voltage characteristic renders the low-frequency compensation problem exceedingly simple.

Referring to the mechanical network of Fig. 9.19 it will be seen that the velocity in the record, z_{MR}, is a function of the magnitude of the mechanical impedance of the pickup. As the mechanical impedance of the pickup becomes larger the vibration velocity of the record will be correspondingly greater. Vibration of the record produces radiation of sound into the air.
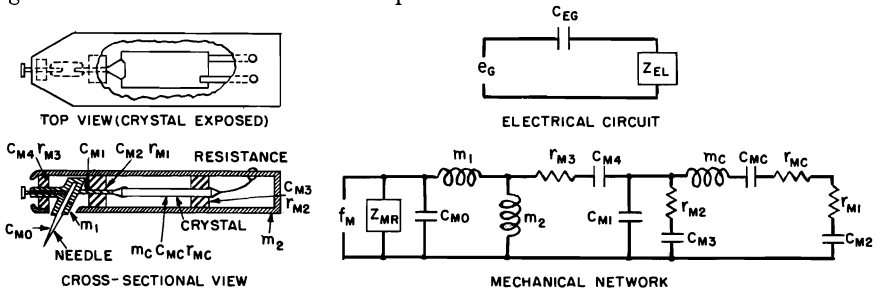
FIG. 9.19. Cross-sectional view, mechanical network, and electrical network of a crystal pickup. In the mechanical network, z_{MR} = the mechanical impedance of the record. C_{M0} = the compliance of the needle. m_1 = the mass of the needle holder. C_{M1} = the compliance of the shaft. C_{M2} and C_{M3} = the compliances of the crystal supports. r_{M1} and r_{M2} = the mechanical resistances of the crystal supports. r_{M3} and C_{M4} = the mechanical resistance and compliance of the front bearing. m_C, r_{MC}, and C_{MC} = the mass, mechanical resistance, and compliance of the crystal. m_2 = the mass of the pickup and tone arm. f_M = the force generated by a velocity generator. In the electrical circuit, e_G = the open circuit voltage developed by the crystal. C_{EG} = the electrical capacitance of the crystal. z_{EL} = the electrical impedance of the load.

Most of this radiation occurs at the high frequencies. The sound produced in this manner is termed mechanical noise. It is undesirable because it interferes with the sound from the loudspeaker and produces distortion. To overcome this, a low-noise crystal pickup[21] has been developed. The essential elements and mechanical network of a low-noise crystal pickup are shown in Fig. 9.20. A permanent sapphire stylus is used instead of a replaceable needle. The mechanical impedance of the pickup in shunt with the mechanical impedance of the record is very small. Therefore, the motion or vibration of the record due to the pickup is very small. The mechanical noise of the low-noise pickup of Fig. 9.20 is about 20 db lower than the replaceable needle pickup of Fig. 9.19. The open circuit voltage output is proportional to the amplitude of the crystal. The electrical generator may be considered to be the open circuit voltage in series with the electrical capacitance.

[21] Burt, A. D., *Electronics*. Vol. 16, No. 1, p. 90, 1943.
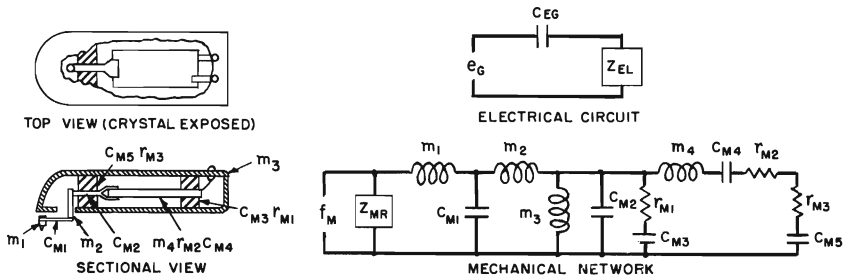
FIG. 9.20. Cross-sectional views, mechanical network and electrical circuit of a low noise crystal pickup. In the mechanical network, z_{MR} = the mechanical impedance of the record. m_1 = the mass of the stylus and holder. C_{M1} = the compliance of the stylus arm. m_2 = the mass of the vertical member. C_{M2} = the compliance of the chuck. m_4, r_{M2}, and C_{M4} = the mass, mechanical resistance and compliance of the crystal. C_{M5} and r_{M3} = the compliance and mechanical resistance of the chuck bearing. C_{M3} and r_{M1} = the compliance and mechanical resistance of the crystal support. m_3 = the mass of the pickup and tone arm. f_M = the force generated by a velocity generator. In the electrical circuit, e_G = the open circuit voltage developed by the crystal. C_{EG} = the electrical capacitance of the crystal. z_{EL} = the electrical impedance of the load.

A crystal phonograph pickup similar to the crystal phonograph pickup of Fig. 9.20 but with a reduction in mass of the vibrating system is shown in Fig. 9.21. With the design of Fig. 9.21 it is possible to obtain uniform response up to 15,000 cycles. On the other hand, if greater output over a more restricted frequency range is desired, the design of Fig. 9.20 may be used. The performance of the system may be deduced from the mechanical network of Fig. 9.21. The open circuit voltage output is proportional to
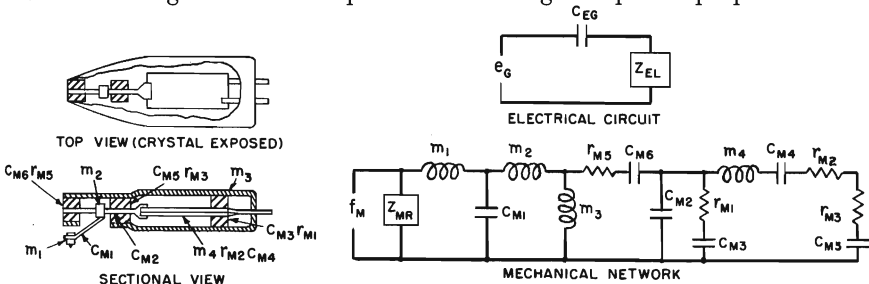
FIG. 9.21. Cross-sectional views, mechanical network, and electrical circuit of a low noise, wide range crystal pickup. In the mechanical network, z_{MR} = the mechanical impedance of the record. m_1 = the mass of the stylus and holder. C_{M1} = the compliance of the stylus arm. m_2 = the mass of the vertical member. C_{M2} = the compliance of the chuck. m_4, r_{M2}, and C_{M4} = the mass, mechanical resistance, and compliance of the crystal. C_{M3} and r_{M1} = the compliance and mechanical resistance of the crystal support. m_3 = the mass of the pickup and tone arm. r_{M5} and C_{M6} = the mechanical resistance and compliance of the front bearing. C_{M5} and r_{M3} = compliance and mechanical resistance of rear bearing. f_M = the force generated by a velocity generator. In the electrical circuit, e_G = the open circuit voltage developed by the crystal. C_{EG} = the electrical capacitance of the crystal. z_{EL} = the electrical impedance of the load.

the amplitude of the crystal. The electrical generator may be considered to be the open circuit voltage in series with the electrical capacitance.

b. *Ceramic Turnover Phonograph Pickup.*—A turnover phonograph pickup[22] employing a ceramic (barium titanate) transducer is shown in Fig. 9.22. The stylus arm contains two stylii located at the end of the stylus
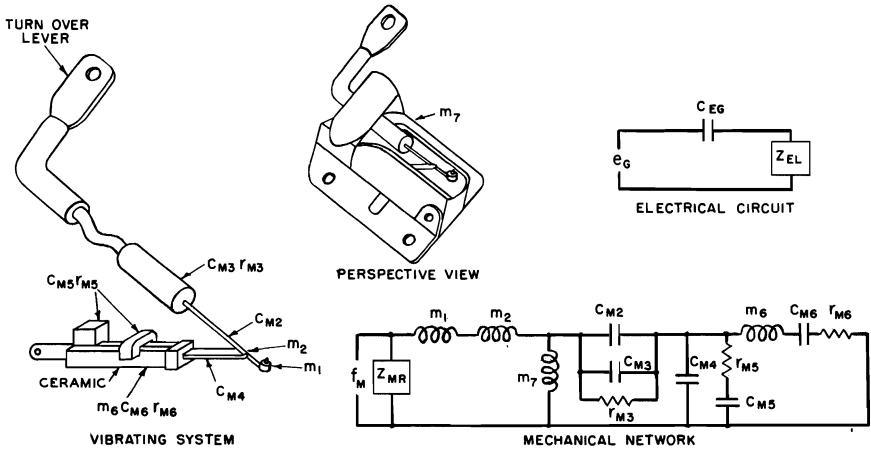
FIG. 9.22. Perspective views, mechanical network and electrical circuit of a turnover ceramic phonograph pickup. In the mechanical network, z_{MR} = the mechanical impedance of the record. m_1 = the mass of the stylii and holders. m_2 = the mass of the connecting arm. C_{M2} = the compliance of the turnover arm. r_{M3} and C_{M3} = the mechanical resistance and compliance of the turnover lever arm. C_{M4} = the compliance of the connecting arm. r_{M5} and C_{M5} = the mechanical resistance and compliance of the ceramic support. m_6, r_{M6}, and C_{M6} = the mass, mechanical resistance, and compliance of the ceramic transducer. m_7 = the mass of the pickup and tone arm. f_M = the force generated by a velocity generator. In the electrical circuit, e_G = the open circuit voltage developed by the ceramic transducer. C_{EG} = the electrical capacitance of the ceramic transducer. z_{EL} = the electrical impedance of the load.

arm and angularly displaced by 180°. A stylus with a .003″ radius at the tip is used to reproduce 78 RPM coarse groove records and a stylus with a .001″ radius at the tip used to reproduce 45 RPM and 33⅓ RPM fine groove records. By means of the turnover lever either stylus may be presented to the record. The stylus arm is coupled to the ceramic transducer by means of a cradle lever arm. The performance of the system may be deduced from the mechanical network of the system. The open circuit voltage output is proportional to the amplitude of the ceramic transducer. The internal electrical element of the ceramic transducer is an electrical capacitance. The electrical generator may be considered to be open circuit voltage in series with an electrical capacitance.

[22] Koren, Pearson, Klingener, and Sabol, *Jour. Acous. Soc. Amer.*, Vol. 26, No. 1, p. 15, 1954.

A turnover pickup[23] employing two separate stylus arms is shown in Fig. 9.23.   One stylus arm is fitted with a stylus having a tip radius of .003″ for the reproduction of 78 RPM coarse groove records and the other stylus
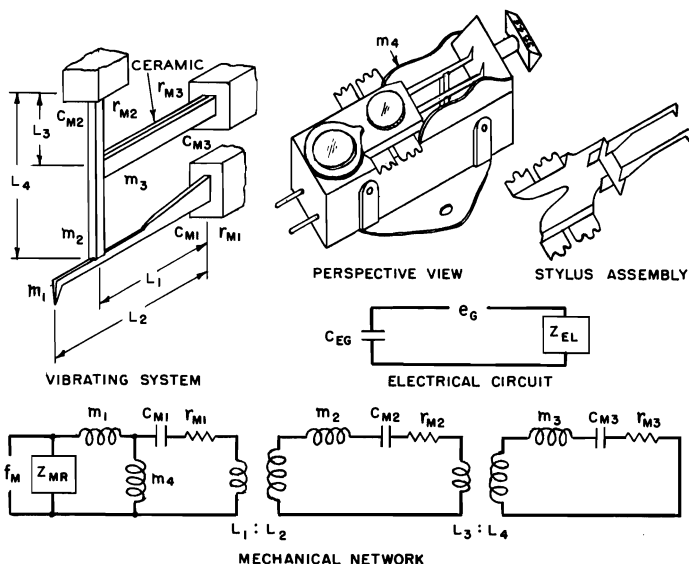
FIG. 9.23.   Perspective views, mechanical network and electrical circuit of a two stylii ceramic phonograph pickup.   In the mechanical circuit, z_{MR} = the mechanical impedance of the record. m_1 = the mass of the stylus. r_{M1} and C_{M1} = the mechanical resistance and compliance of the stylus arm. m_2, r_{M2}, and C_{M2} = the mass, mechanical resistance and compliance of the coupling arm. m_3, r_{M3}, and C_{M3} = the mass, mechanical resistance and compliance of the ceramic transducer. m_4 = the mass of the pickup and tone arm. The levers L_1, L_2, L_3, and L_4 are represented as transformers with turn ratios corresponding to the ratios of the lever arms. f_M = the force generated of a velocity generator.   In the electrical circuit, e_G = the open circuit voltage developed by the ceramic transducer. C_{EG} = the electrical capacitance of the ceramic transducer. z_{EL} = the electrical impedance of the load.

arm is fitted with a stylus having a tip radius of .001″ for the reproduction of 45 RPM and 33⅓ RPM fine groove records.   The lever located at the front of the cartridge is turned 180° to change from one stylus arm to the other.   The stylus arm under operation rests in the cradle of the lever arm connected to the ceramic transducer.   The use of the vibrating system shown in Fig. 9.23 reduces the mechanical impedance at the stylus.   The lever system provides the proper stylus for the particular application.   The performance of the system may be deduced from the mechanical network.

[23] Bauer, Gunter, and Steeler, *Jour. Audio Eng. Soc.*, Vol. 2, No. 4, p. 239, 1954.

The internal electrical element is an electrical capacitance. The electrical generator may be considered to be the open circuit voltage in series with the electrical capacitance.

c. *Magnetic Pickup*.—A magnetic pickup[24,25] is a phonograph pickup whose electrical output is generated in a coil or conductor in a magnetic field or circuit. A magnetic pickup of early design is shown in Fig. 9.24. The motion of the needle is transferred to the armature. The steady flux is furnished by a permanent magnet. The armature is of the balanced type so that in its central position there is no flux through the armature. When the armature is deflected, a flux flows through the armature which induces a voltage in the coil.

A schematic view of the magnetic system and magnetic network of a magnetic pickup is shown in Fig. 9.25. The system of Fig. 9.25 applies to Fig. 9.24 assuming the variable air gaps 1 and 3 of Fig. 9.25 are the reluctances R_1 and R_2 of Fig. 9.25. If the armature is deflected a distance Δx, there will be a change in the reluctances R_1 and R_2 and there will be a flow of flux through the armature. The reluctance of the armature is small compared to the reluctances of the air gaps R_1 and R_2 and may therefore be neglected. Furthermore, since the combined reluctance of R_1 and R_2 is large compared to the reluctance R_M, the flux through the armature due to a deflection Δx is given by

$$\Delta\phi = \frac{M_M}{R_1 + R_3 + \Delta R} - \frac{M_M}{R_2 + R_4 - \Delta R} \qquad 9.6$$

Since $R_1 = R_2 = R_3 = R_4 = R$ in a symmetrical system, equation 9.6 becomes

$$\Delta\phi = \frac{M_M \Delta R}{R^2} \qquad 9.7$$

The reluctance R is given by

$$R = \frac{a}{A} \qquad 9.8$$

where a = spacing between the armature and pole pieces, in centimeters, and
A = area of the pole piece, in square centimeters.

The incremental change in reluctance is given by

$$\Delta R = \frac{\Delta x}{A} \qquad 9.9$$

where Δx = change in distance α, in centimeters.

From equations 9.7, 9.8, and 9.9,

$$\Delta\phi = \frac{A M_M \Delta x}{\alpha^2} \qquad 9.10$$

[24] Kellogg, E. W., *Jour. A.I.E.E.*, Vol. 46, No. 10, p. 1041, 1927.
[25] Hasbrouck, H. J., *Proc. I.R.E.*, Vol. 27, No. 3, p. 184, 1939.
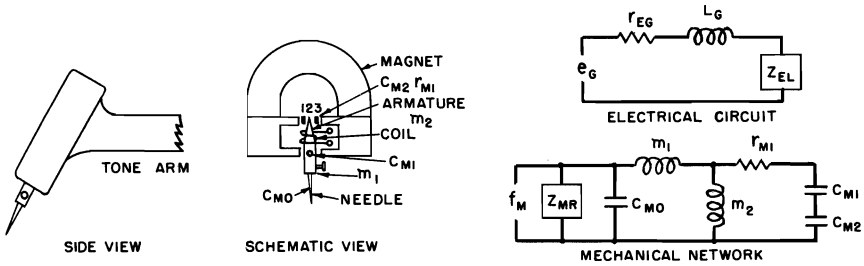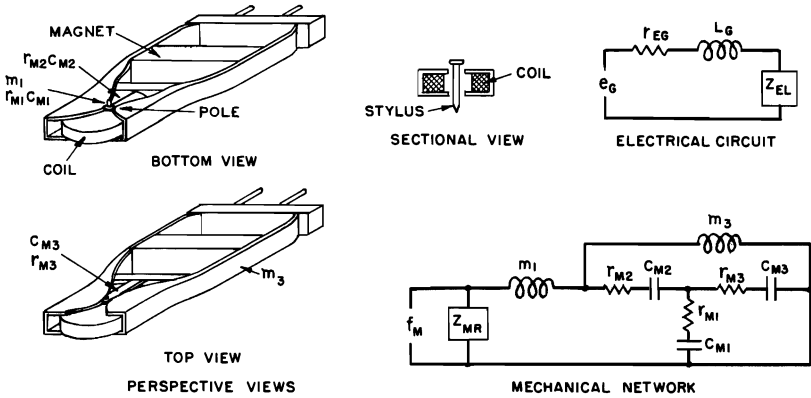
FIG. 9.24. Front and side views, mechanical network, and electrical circuit of a magnetic pickup. In the mechanical network, z_{MR} = the mechanical impedance of the record. C_{M0} = the compliance of the needle. m_1 = the mass of the needle holder and armature. C_{M1} = the compliance of the needle holder pivot. C_{M2} and r_{M1} = the compliance and mechanical resistance of the damping material. m_2 = the mass of the pickup and tone arm. f_M = the force generated by a velocity generator. In the electrical circuit, e_G = the open circuit voltage developed in the coil. L_G and r_{EG} = the inductance and electrical resistance of the coil. z_{EL} = the electrical impedance of the load.
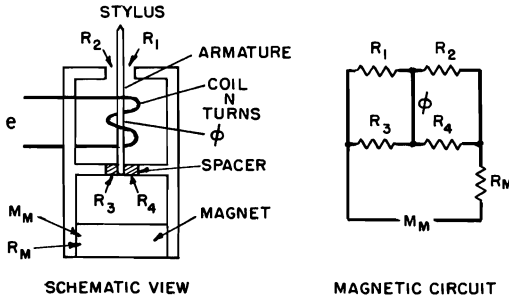
FIG. 9.25. Schematic view and magnetic circuit of the magnetic system of a magnetic phonograph pickup. In the mechanical circuit, M_M = the magnetomotive force developed by the permanent magnet. R_M = the reluctance of the path in the magnetic material. R_1 and R_2 = the reluctances of the variable air gaps. R_3 and R_4 = the reluctances of the fixed air gaps. ϕ = the flux in the armature. e = the voltage induced in the coil.

The generated voltage e, in abvolts, in the coil is given by

$$e = N \frac{d\phi}{dt} \qquad 9.11$$

where N = number of turns in the coil.

From equations 9.10 and 9.11, the generated voltage in the coil is given by

$$e = \frac{N M_M A}{\alpha^2} \dot{x} \qquad 9.12$$

where $\dot{x}$ = velocity of the armature at the pole pieces, in centimeters per second.

Equation 9.12 shows that the generated voltage will be independent of the frequency if the velocity of the armature is independent of the frequency.

The mechanical network of the mechanical system is shown in Fig. 9.24. Damping, represented by the compliance C_{M2} and the mechanical resistance r_{M1}, is furnished by a suitable material such as viscoloid.

A more recent design of magnetic pickup[26] is shown in Fig. 9.26. The horizontal stylus arm also serves as the armature. The pole pieces are located at the stylus. This design makes it possible to obtain a relatively

ARMATURE
DAMPING BLOCK

BOTTOM VIEW

ELECTRICAL CIRCUIT

SECTIONAL VIEW

MECHANICAL NETWORK
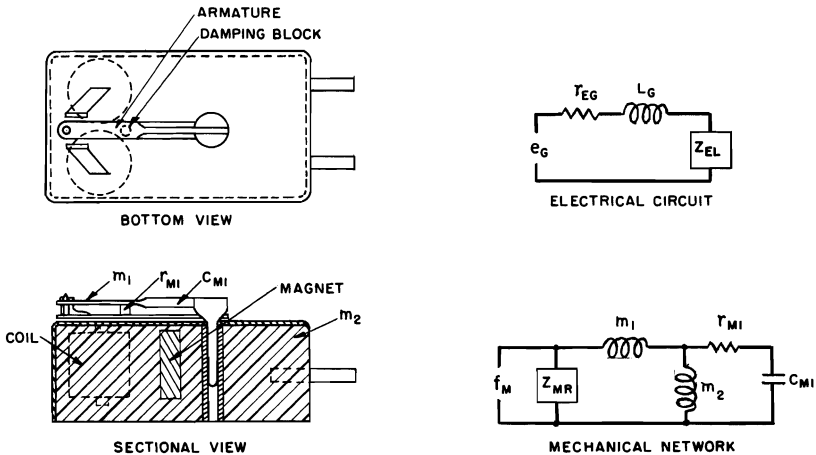
Fig. 9.26. Bottom and sectional views, mechanical network, and electrical circuit of a magnetic pickup. In the mechanical network, z_{MR} = the mechanical impedance of the pickup. m_1 = the mass of the stylus and stylus holder. r_{M1} and C_{M1} = the mechanical resistance and compliance of the armature. m_2 = the mass of the pickup and tone arm. f_M = the force generated by a velocity generator. In the electrical circuit, e_G = the open circuit voltage developed in the coil. L_G and r_{EG} = the inductance and electrical resistance of the coil. z_{EL} = the electrical impedance of the load.

low mechanical impedance. The steady flux is supplied by a small permanent magnet. The performance of the relatively simple vibrating system may be obtained from the mechanical network. The open circuit voltage output is proportional to the velocity of the armature. The electrical generator may be considered to be the open circuit voltage in series with the electrical impedance. Since the electrical impedance is primarily inductive, the electrical impedance is proportional to the frequency.

Another more recent design of magnetic pickup[27] is shown in Fig. 9.27. The vertical stylus arm also serves as the armature. The coil surrounds the armature. The armature is of the balanced type with air gaps between the armature and the pole pieces at the two ends of the armature. When

[26] Bachman, W. S., *Elec. Ind.*, Vol. 4, No. 7, p. 86, 1945.
[27] Stanton, W. O., *Jour. Audio Eng. Soc.*, Vol. 3, No. 2, p. 70, 1955.

the armature is in its central position, there is no flux in the armature. When the armature is deflected, a flux flows in the armature which induces a voltage in the coil. The performance of the relatively simple vibrating system may be obtained from the mechanical network. The open circuit voltage output is proportional to the velocity of the armature. The electrical generator may be considered to be the open circuit voltage in series with

FIG. 9.27. Top and bottom perspective views, mechanical network, and the electrical circuit of a magnetic pickup. In the mechanical circuit, z_{MR} = the mechanical impedance of the record. m_1 = the mass of the stylus, stylus holder, and armature. r_{M1} and C_{M1} = the mechanical resistance and compliance of the armature. r_{M2} and C_{M2} = the mechanical resistance and compliance of the armature longitudinal support wire. r_{M3} and C_{M3} = the mechanical resistance and compliance of the armature base support. m_3 = the mass of the pickup and tone arm. f_M = the force generated by the velocity generator. In the electrical circuit, e_G = the open circuit voltage developed in the coil. L_G and r_{EG} = the inductance and electrical resistance of the coil. z_{EL} = the electrical impedance of the load.

the electrical impedance. Since the electrical impedance is primarily inductive, the electrical impedance is proportional to the frequency.

d. *Dynamic Pickup.*—A dynamic pickup is a phonograph pickup in which the output results from the motion of a conductor in a magnetic field. Fig. 9.28 shows a cross-sectional view and mechanical network of a dynamic pickup[28] for the reproduction of hill-and-dale type records. The principal mechanical impedance is due to the mass of the needle and coil. The output of the coil is proportional to the velocity. Therefore, the response characteristic is similar to that of the magnetic pickup. The coil is practically a constant electrical resistance over the audio-frequency range.

A dynamic pickup[29] employing a stylus arm attached to a coil located in a magnetic field for the reproduction of lateral phonograph records is shown in Fig. 9.29. The performance of the system may be obtained from

[28] Frederick, H. A., *Jour. Soc. Mot. Pic. Eng.*, Vol. 18, No. 2, p. 141, 1932.
[29] Lindenberg, T. Jr., *Electronics*, Vol. 18, No. 6, p. 108, 1945.

the mechanical network of Fig. 9.29. Since the system may be made very small and light, it is possible to reproduce the entire audio-frequency range. The open circuit voltage is proportional to the velocity of the coil. The electrical generator may be considered to be the open circuit voltage in
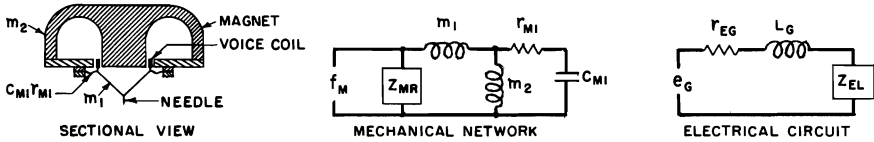
FIG. 9.28. Cross-sectional view, mechanical network, and electrical circuit of a vertical dynamic pickup. In the mechanical network, z_{MR} = the mechanical impedance of the record. m_1 = the mass of the stylus and voice coil. C_{M1} and r_{M1} = the compliance and mechanical resistance of the suspension system. m_2 = the mass of the pickup and tone arm. f_M = the force generated by a velocity generator. In the electrical circuit, e_G = the open circuit voltage developed in the voice-coil. L_G and r_{EG} = the inductance and electrical resistance of the voice-coil. z_{EL} = the electrical impedance of the load.
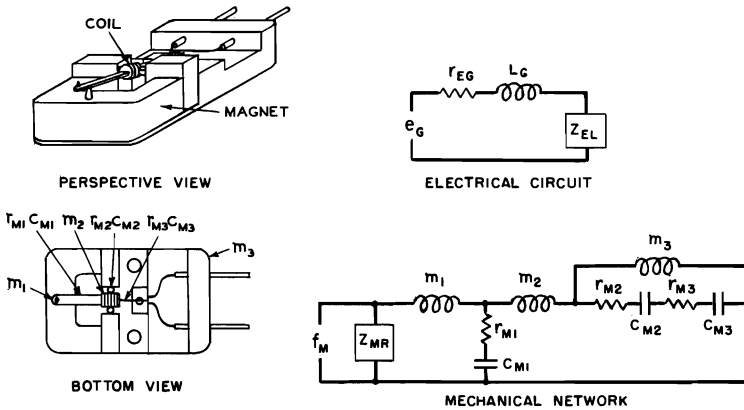
FIG. 9.29. Perspective and bottom views, mechanical network and electrical circuit of a lateral dynamic pickup. In the mechanical circuit, z_{MR} = the mechanical impedance of the record. m_1 = the mass of the stylus and stylus holder. r_{M1} and C_{M1} = the mechanical resistance and compliance of the stylus arm. m_2 = the mass of the coil. r_{M2} and C_{M2} = the mechanical resistance and compliance of the coil supports. r_{M3} and C_{M3} = the mechanical resistance and compliance of the longitudinal coil support. m_3 = the mass of the pickup and tone arm. f_M = the force generated by velocity generator. In the electrical circuit, e_G = the open circuit voltage developed in the coil. L_G and r_{EG} = the inductance and electrical resistance of the coil. z_{EG} = the electrical impedance of the load.

series with the electrical impedance of the coil. The coil is practically a constant electrical resistance over the audio-frequency range.

Another form of dynamic pickup, shown in Fig. 9.30, is capable of reproducing both lateral- and vertical-type phonograph records by merely changing the transformer connections. The vibrating system consists of two

parallel ribbons located in a magnetic field. When the stylus is actuated by a lateral-type phonograph record the ribbons rotate about the center axis. When the stylus is actuated by vertical-type phonograph record the two ribbons move together in a direction normal to the plane of the record. The direction of the currents in the two ribbons differs for the two types of motion. Each ribbon is connected to a separate transformer. In this way the outputs of the two ribbons can be brought into phase for either lateral- or vertical-cut records by merely changing the transformer connection. The open circuit voltage is proportional to the velocity of the
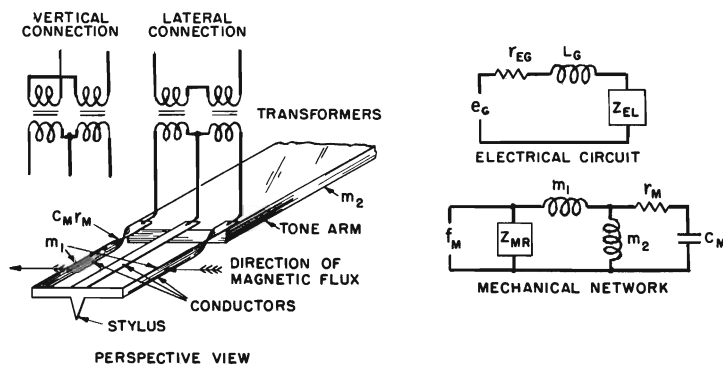
Fig. 9.30. Perspective view, electrical connection arrangement, mechanical network, and electrical circuit of a combination vertical or lateral dynamic pickup. In the mechanical network, z_{MR} = the mechanical impedance of the record. m_1, r_M, and C_M = the mass, mechanical resistance, and compliance of the stylus and conductors. m_2 = the mass of the pickup and tone arm. f_M = the force generated by a velocity generator. In the electrical circuit, e_G = the open circuit voltage developed by the conductors. L_G and r_{EG} = the inductance and electrical resistance of the conductors. z_{EL} = the electrical impedance of the load. Note: Either vertical- or lateral-type phonograph records can be reproduced by merely changing the transformer output connections.

ribbons. The electrical generator may be considered to be the open circuit voltage in series with the electrical impedance of the ribbons. This pickup is designed for wide-range reproduction of transcription phonograph records.

e. *Frequency Modulation Pickup.*—A frequency modulation pickup is a phonograph pickup in which the frequency of a high-frequency oscillator is varied by altering one of the elements in the oscillating circuit. By use of a discriminator the modulated high-frequency output is transformed to the vibration frequency of the stylus.

A perspective view, electrical diagram, mechanical network, and response frequency characteristic of a frequency modulation pickup[30] are shown in Fig. 9.31. A stretched ribbon is mounted in a plane parallel to an insulated plate and spaced by a small air gap. The stylus supporting wire is anchored

[30] Beers and Sinnett, *Proc. Inst. Rad. Eng.*, Vol. 31, No. 4, p. 138, 1943.

at its upper end. It is attached to the ribbon at approximately the mid-point of its length and the free end is bent in a plane parallel to the record groove. A sapphire stylus is attached to the end of the wire. It is evident
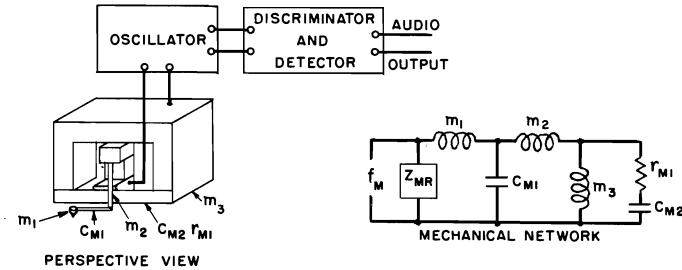
FIG. 9.31. Perspective view, electrical system, and mechanical network of a frequency modulation pickup. In the mechanical network, z_{MR} = the mechanical impedance of the record. m_1 = the mass of a stylus and holder. C_{M1} = the compliance of the stylus arm. m_2 = the mass of the vertical member. C_{M2} and r_{M1} = the compliance and mechanical resistance of the ribbon. m_3 = the mass of the pickup and tone arm. f_M = the force generated by a velocity generator.

that a lateral displacement of the stylus will produce a change in the spacing between the ribbon and insulated black plate and thus produce a change in electrical capacitance. The electrical capacitance formed by the ribbon and insulated back plate is made a part of the oscillating circuit of a 30-megacycle
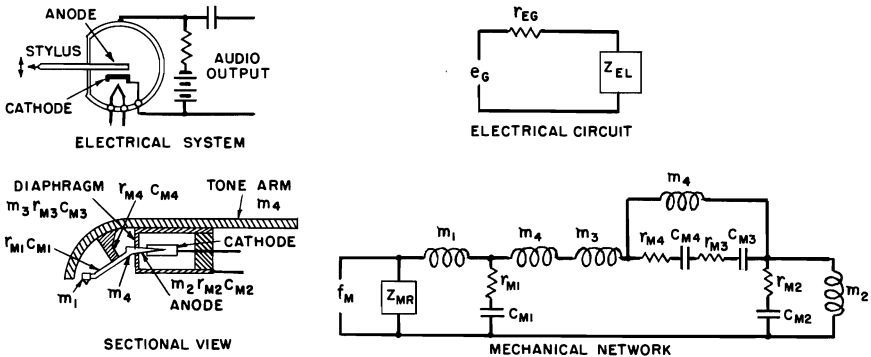
FIG. 9.32. Sectional view, electrical system, mechanical network, and electrical circuit of an electronic pickup. In the mechanical network, z_{MR} = the mechanical impedance of the record. m_1 = the mass of the stylus. r_{M1} and C_{M1} = the mechanical resistance and compliance of the stylus arm. r_{M4} and C_{M4} = the mechanical resistance and compliance of the damping member. m_2, r_{M2}, and C_{M2} = the mass, mechanical resistance, and compliance of the anode lever. m_3, r_{M3}, and C_{M3} = the mass, mechanical resistance, and compliance of the diaphragm. m_4 = the mass of the pickup and tone arm. f_M = the force generated by a velocity generator. In the electrical circuit, e_G = the open circuit voltage developed in the tube. r_{EG} = the internal electrical resistance of the tube. z_{EL} = the electrical impedance of the load.

oscillator. The change in capacity due to the motion of the stylus produces a change in the frequency of the oscillator. The output of the oscillator is impressed upon a discriminator and detector. The output of the detector corresponds to amplitude of the stylus.

f. *Electronic Pickup.*[31]—An electronic pickup is a phonograph pickup in which the output is generated by the motion of an electrode in a vacuum tube. A cross-sectional view, electrical circuit, mechanical network, and the response frequency characteristic of an electronic pickup are shown in Fig. 9.32. The voltage is generated by the change in distance between the cathode and anode. The anode is the movable element. Motion of the anode is transferred through the envelope of the tube by means of a thin metal diaphragm. A permanent sapphire stylus is used in this pickup. The voltage output is proportional to the amplitude.

g. *Variable Resistance Pickup.*—A variable resistance phonograph pickup is a pickup in which the voltage is generated in a current polarized variable electrical resistance element. The electrical resistance of the element is varied by compressions and rarefactions of the element. A schematic view of a variable resistance pickup[31a] is shown in Fig. 9.33.
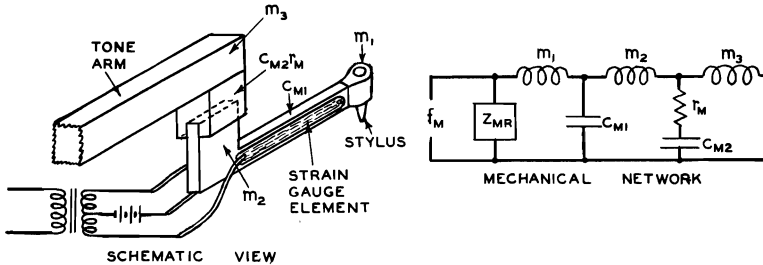
FIG. 9.33. Perspective view and mechanical network of a variable resistance pickup. In the mechanical network, z_{MR} = the mechanical impedance of the record. m_1 = the mass of the stylus. C_{M1} = the compliance of the stylus arm. m_2 = the mass at the base of the stylus arm. C_{M2} and r_M = the compliance and mechanical resistance of the damping material for the base support. m_3 = the mass of the tone arm. f_M = the force generated by the velocity generator.

A variable electrical resistance element is cemented on each side of the stylus arm. Bending of the stylus arm produces rarefactions on one side and compressions on the other side. The compressions and rarefactions produce a corresponding decrease and increase in electrical resistance of the variable resistance element. Since the element is polarized by a current, the change in electrical resistance produces a corresponding change in voltage. The electrical schematic diagram shows the polarizing battery and transformer system. The electrical system is similar to the double-button carbon microphone. The voltage output is proportional to the

[31] Olson, H. F., *Jour. Acous. Soc. Amer.*, Vol. 19, No. 2, p. 307, 1947.
[31a] Bachman, W. S., *Elec. Eng.*, Vol. 65, No. 3, p. 159, 1946.

amplitude. The base of the stylus arm is embedded in damping material. The performance of the vibrating system may be obtained from a consideration of the mechanical network.

h. *Feedback Pickup*.[32]—Feedback may be used in electrical and electromechanical systems to change such factors as the transmission and distortion characteristics, the terminal impedances and the noise levels of the systems.
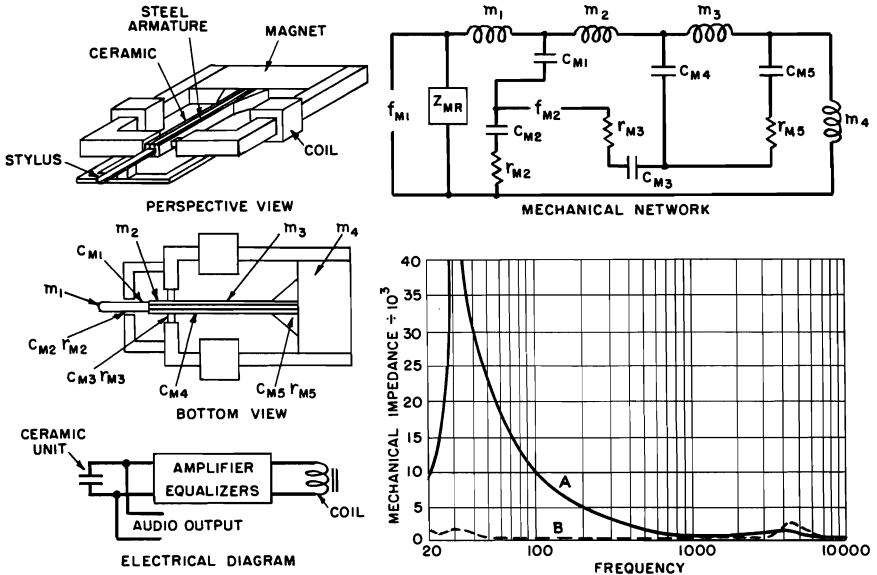
FIG. 9.34. Perspective and bottom views, mechanical network, electrical diagram, and mechanical impedance frequency characteristics of a feedback pickup. In the mechanical network, z_{MR} = the mechanical impedance of the record. m_1 = the mass of the stylus and stylus holder. C_{M1} = the compliance of the stylus arm. m_2 = the mass of the front portion of the armature and ceramic transducer. r_{M2} and C_{M2} = the mechanical resistance and compliance of the damping block under the stylus arm. r_{M3} and C_{M3} = the mechanical resistance and compliance of the damping blocks on the ceramic transducer and magnetic armature. m_3 = the mass of the rear portion of the ceramic transducer and magnetic armature. C_{M4} = the compliance of the ceramic transducer and magnetic armature. m_4 = the mass of the pickup and tone arm. r_{M5} and C_{M5} = the mechanical resistance and compliance of the support for the ceramic transducer and armature. f_{M1} and f_{M2} = the forces generated by the velocity generator and the magnetic driving system. In the graph: A. mechanical impedance characteristic without feedback. B. mechanical impedance characteristic with feedback.

Feedback has been applied to cutters, calibrators, and other electroacoustic devices. However, the application of feedback to phonograph pickups is a recent development. The problem in the phonograph pickup is to reduce the mechanical impedance at the stylus of the pickup so that the load presented to the record will be reduced. The feedback phonograph pickup shown in Fig. 9.34 employs two electromechanical transducers in the feed-

[32] Halter, J. B., Unpublished Report.

back loop. The system for driving the stylus is an electromagnetic transducer in which the stylus is attached to the armature. The sensing and reproducing system is a ceramic transducer consisting of two barium titanate strips attached to the two sides of the steel armature. The electrical diagram of the feedback phonograph pickup is shown in Fig. 9.34. The performance of the system may be deduced from the mechanical network of Fig. 9.34. In the ceramic transducer, the open circuit voltage e, in volts, is given by

$$e = K_B x \qquad 9.13$$

where x = amplitude of the transducers, and

K_B = constant of the system involving the material and construction of the transducer.

In the electromagnetic transducer the force f_M, in dynes, produced by a current i, in abamperes, in the coil is given by

$$f_M = K_I i \qquad 9.14$$

where K_I = constant involving the parameters of the electromagnetic transducer.

The problem is to adjust the amplitude and phase of the system so that a maximum displacement will be produced in C_{M4} for a minimum force f_{M1} at the stylus. The performance of the system with and without feedback is shown in Fig. 9.34. It will be seen that a tremendous reduction in the force f_{M1} is obtained with feedback.

i. *Compliance of Pickups.*—A consideration of the pickups described in the preceding sections shows that the lateral compliance at the stylus is an important element. The compliance determines the force applied to the record in the low-frequency range where the pickup system is stiffness controlled. The force is the ratio of the displacement to the compliance. See Sec. 4.6. As an indication of the magnitude of the compliance of pickups, the compliance of a pickup for fine-groove records is of the order of 1×10^{-6} centimeters per dyne.

j. *Tone Arm Resonance.*—Tone arm resonance occurs in the low-frequency range when the mechanical reactance due to the effective mass of the pickup and tone arm is equal to the mechanical reactance of the compliance of the pickup. The response at the resonant frequency is determined by the damping in the pickup and tone arm pivot. In one design of tone arm[33] additional mechanical resistance has been added by a viscous damped tone arm. The tone arm resonant frequency determines the low-frequency range because the response falls off rapidly below the tone arm resonance.

3. *Distortion in Record Reproduction.*—The recording and reproducing of a phonograph record is a complicated process and there are many sources of nonlinear distortion. The record does not present an infinite mechanical

[33] Bachman, W. S., *Trans. I.R.E.*, Prof. Group Audio, March, 1951.

impedance to the needle. As a consequence, the vibrating system of the pickup is shunted by the effective mechanical impedance of the record at the needle. Nonlinear distortion will be introduced if the record is a variable element. Another source of distortion is due to a deviation in tracking commonly termed tracking error. Still another form of distortion is known as tracing distortion, due to the finite size of the stylus. These and other types of distortion will be discussed in this section.

A source of nonlinear distortion due to a deviation in tracking,[34,35] is commonly termed tracking error. The angle between the vertical plane containing the vibration axis of the pickup and the vertical plane containing the tangent to the record is a measure of the tracking error. If the vibration axis of the pickup passes through the tone arm pivot, the tracking error can be zero for only one point on the record. The tracking error can be reduced if the vibration axis of the pickup is set at an appropriate angle with respect to the line connecting the stylus point and the tone arm pivot together with a suitable overhang distance between the stylus and the record axis. Fig. 9.35.

The amount of overhang d, in inches, is given by

$$d = \frac{R_i{}^2}{L\left[\frac{1}{4}(1 + \frac{R_i}{R_0})^2 + \frac{R_i}{R_0}\right]} \qquad 9.15$$

where $L =$ length of the tone arm from the pivot to the stylus, in inches,

$R_0 =$ radius of the start groove of the recording, and

$R_i =$ radius of the end groove of the recording.

FIG. 9.35. Geometry of a tone arm system for reducing the tracking error.

The angle α in degrees between the vertical plane containing vibration axis of the pickup and the line joining the stylus and the tone arm pivot is given by

$$\alpha = \frac{57.3\left(1 + \frac{R_i}{R_0}\right)R_i}{L\left[\frac{1}{4}\left(1 + \frac{R_i}{R_0}\right)^2 + \frac{R_i}{R_0}\right]} \qquad 9.16$$

[34] Olney, Benj., *Electronics*, Vol. 10, No. 1, p. 19, 1937.
[35] Bauer, B. B., *Electronics*, Vol. 18, No. 3, p. 110, 1945.

With the application of equations 9.15 and 9.16 in the design of a tone arm, the tracking error can be reduced to ± 5 per cent.

A form of distortion in lateral-disk record reproduction known as tracing distortion[36,37,38] is a function of the diameter of the stylus, the lateral velocity, and the groove velocity. The distortion is due to the fact that there is not a one-to-one correspondence between the shapes of the cutting and reproducing stylii. The shape of the groove produced by the cutting stylus of a lateral-disk recorder is shown in Fig. 9.36. The groove narrows as the cutting stylus approaches the center position because the cutting stylus is moving at an angle with respect to the motion of the record. A sectional view of a groove with the reproducing stylus in contact with the

TOP VIEW

A-A' B-B' C-C'
SECTIONAL VIEWS
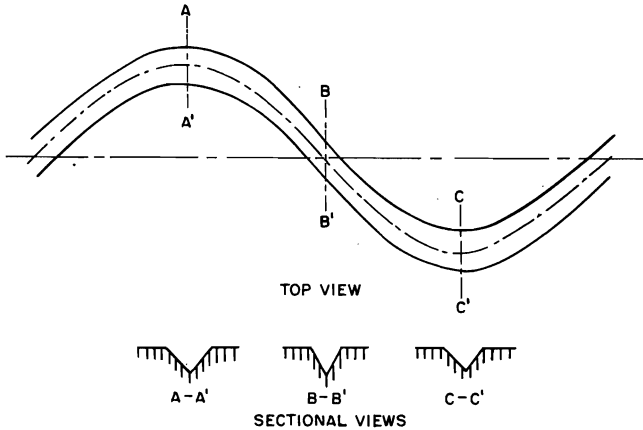
FIG. 9.36. The shape of the groove cut by the stylus of a lateral phonograph recorder in a lacquer disk.

walls of the groove is shown in Fig. 9.37. Referring to Figs. 9.36 and 9.37, it will be seen that, when the reproducing stylus moves in this groove, it will rise as the groove narrows. The frequency of the rise is twice the frequency of the modulation. The narrowing of the groove is termed the pinch effect. The two sides of the groove are symmetrical; therefore, the stylus must execute symmetrical motion about the center line which means that there should be no even harmonics. However, odd harmonics are produced. The equations for the magnitude of the harmonics have been developed.

With regard to nonlinear distortion, the two-frequency method of distortion testing has been found to agree quite well with subjective testing of nonlinear distortion.[39] The nonlinear distortion, in per cent, which occurs when two frequencies are combined is given by

[36] DiToro, M. J., *Jour. Soc. Mot. Pic. Eng.*, Vol. 29, No. 5, p. 493, 1938.
[37] Pierce and Hunt, *Jour. Acous. Soc. Amer.*, Vol. 10, No. 1, p. 14, 1938.
[38] Sepmeyer, L. W., *Jour. Acous. Soc. Amer.*, Vol. 13, No. 3, p. 276, 1942.
[39] Roys, H. E., *RCA Review*, Vol. 10, No. 2, p. 254, 1949.

$$IM = \frac{\dfrac{\pi^2 r^2 u_1{}^2 u_2{}^2}{4S^4}\left[(2f_1 + f_2)^2 + (2f_1 - f_2)^2\right]}{u_2 - \dfrac{\pi^2}{2}\dfrac{r^2 f_2{}^2}{S^4}\left(\dfrac{u_2{}^3}{2} + u_1{}^2 u_2\right)} \times 100 \qquad 9.17$$

where u_1 = peak lateral velocity of the lower frequency f_1, in centimeters
per second,

u_2 = peak lateral velocity of the upper frequency f_2, in centimeters
per second,

r = radius of the reproducing stylus, in centimeters,

f_1 = lower frequency, in cycles per second,

f_2 = upper frequency, in cycles per second, and

S_1 = groove velocity, in centimeters per second.
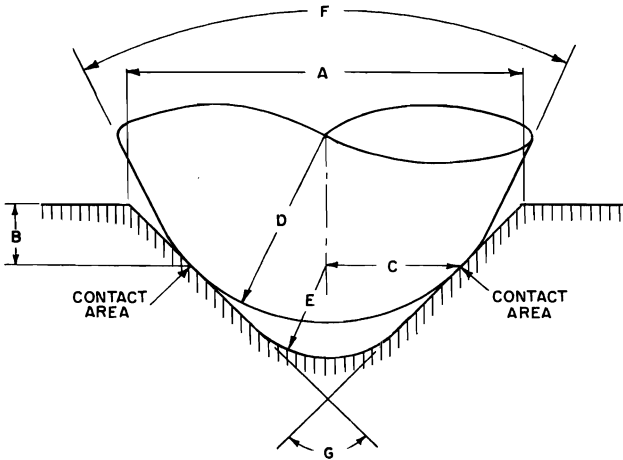
FIG. 9.37. Sectional view of a stylus in a groove. The dimen-
sions for coarse groove, fine groove, and ultra fine groove are as
follows:

Dimension	Coarse	Fine	Ultra Fine
A	·006	·0027	·0010
B	·0008	·0004	·0004
C	·0019	·0007	·00017
D	·0027	·001	·00017
E	·0023	·00027	·00015
F	45°	45°	45°
G	90°	90°	90°

Subjective tests have shown that when the intermodulation distortion
is less than 10 per cent the distortion is practically imperceptible. This then
establishes a value for the terminal velocity for a certain stylus diameter
and the peak lateral velocity.

The record does not present an infinite mechanical impedance to stylus. As a result, the vibrating system of the pickup is shunted by the mechanical impedance of the record, as depicted in the mechanical networks of pickups in this chapter. Nonlinear distortion[40,41] will occur if the record is a variable element. If the force which the stylus presents to the record is of such magnitude that it exceeds the yield point of the record material, the mechanical impedance of the record will not be a constant. The result is production of nonlinear distortion. Furthermore, if the force exceeds the yield point by a considerable amount, the record may be permanently damaged.

As the needle or stylus is worn by the groove the shape of the point changes from a spherical surface to a wedge shape. The wedge-shaped stylus[42] introduces nonlinear distortion and a loss in the high-frequency response.

A consideration of the load and needle forces at the stylus tip shows that there is force which is proportional to the tracking angle. This force is usually directed toward the center of the record and is applied to the inner boundary of the record groove. This force is known as the side thrust and it is responsible for the unequal wear[43] on the two sides of the stylus.

Another source of distortion is due to the lack of correspondence between the linear groove speed in the recording and ultimate reproduction. This type of distortion is termed "wows."[44] This may be due to a nonuniform speed of the record turntable during recording or reproduction, misplacement of the center hole or configuration distortion during the processing. In general, the major source of "wows" is due to nonuniform speed of the reproducing turntable.

The record surface noise,[45] in the absence of any signal, is one of the factors which limits the volume range and the frequency range of shellac phonograph records. The amount of surface noise for a given record is proportional to the frequency band width. In order to reduce the surface noise to a tolerable value in shellac records it is usually necessary to limit the high-frequency range in reproduction. A method of decreasing the effective surface noise consists of increasing the amplitude of the high-frequency response in recording and introducing complementary equalization in reproduction. The volume range of a phonograph record, in general, does not permit recording the full range of a symphony orchestra without some compression. To offset this compression complementary expansion may be introduced in the reproduction.

4. *Record Noise.*—When the stylus of a phonograph pickup is actuated by the groove of a phonograph record a force is developed between the stylus

[40] Begun and Lynch, *Jour. Acous. Soc. Amer.*, Vol. 13, No. 3, p. 284, 1942.
[41] Max, A. M., *Jour. Audio Eng. Soc.*, Vol. 3, No. 2, p. 66, 1955.
[42] Bauer, B. B., *Jour. Acous. Soc. Amer.*, Vol. 16, No. 4, p. 246, 1945.
[43] Bauer, B. B., *Trans. I.R.E.*, Prof. Group Audio, PGA-6, p. 11, 1952.
[44] Comerci, Frank A., *Jour. Soc. Mot. Pic. Tel. Eng.*, Vol. 64, No. 3, p. 117, 1955.
[45] Reid, J. D., *Jour. Acous. Soc. Amer.*, Vol. 13, No. 3, p. 274, 1942.

and the walls of the record. The force, in dynes, developed by the inter-
action of the pickup stylus and the record is given by

$$f_M = j \frac{z_{MR} z_{MP} x \omega}{z_{MR} + z_{MP}} \qquad\qquad 9.18$$

where $\omega = 2\pi f$,

   $f =$ frequency, in cycles per second,

   $x =$ amplitude of the groove, in centimeters,

   $z_{MR} =$ mechanical impedance of the record, in mechanical ohms, and

   $z_{MP} =$ mechanical impedance of the pickup at the stylus, in mechani-
cal ohms.

Equation 9.18 illustrates the importance of a pickup with a small mechani-
cal impedance. If the pickup mechanical impedance is comparable to
the mechanical impedance of the record, a considerable part of the amplitude
of the record groove will take place in motion of the record. This motion
or vibration of the record produces sound which is radiated into the air.
The radiated sound corresponds somewhat to the sound recorded on the
record, but it is very much distorted due to the way in which it is produced
and is, therefore, disagreeable. Furthermore, there is interference between
this sound and the sound radiated from the loudspeaker. The force which
drives the stylus is a function of the record mechanical impedance, if the
mechanical impedance of the pickup at the stylus is relatively large. This
may produce distortion in the reproduced sound because the mechanical
impedance of the record varies over wide limits from the outside to the inside
groove and is a function of the mounting of the record supporting means.
A pickup with a high mechanical impedance also produces excessive record
wear. Equation 9.18 together with the above discussion shows that
record noise and wear and distortion can be reduced by making the
mechanical impedance of the pickup small compared to the mechanical
impedance of the record. The measurement of record noise is described in
Sec. 10.5D.

C. *Selection of Rotational Speed and Record Diameter.*[46]—The playing
time, the diameters of the start and end grooves of the recording, the rota-
tional speed of the grooves per inch, and the terminal linear velocity are all
factors involved in the determination of a record. These relations are inter-
connected by the following equations:

$$P = \frac{D_0 - D_i}{2R} N \qquad\qquad 9.19$$

and

$$S_i = \frac{\pi D_i R}{60} \qquad\qquad 9.20$$

[46] Reiskind, H. I., *Trans. I.R.E.*, Prof. Group Audio, PGA-5, February, 1952.

where P = playing time, in minutes,

D_0 = diameter of the start groove of the recording, in inches,

D_t = diameter of the end groove of the recording, in inches,

R = rotational speed, in revolutions per minute,

N = grooves, per inch, and

S_t = terminal linear velocity.

Equations 9.19 and 9.20 show that there are many parameters involved in the selection of the rotational speed and the playing time. One of the most important of these is nonlinear distortion, discussed in the preceding section. The other factors such as playing time, record diameter, grooves per inch, rotational speed, etc. are determined by the particular application.
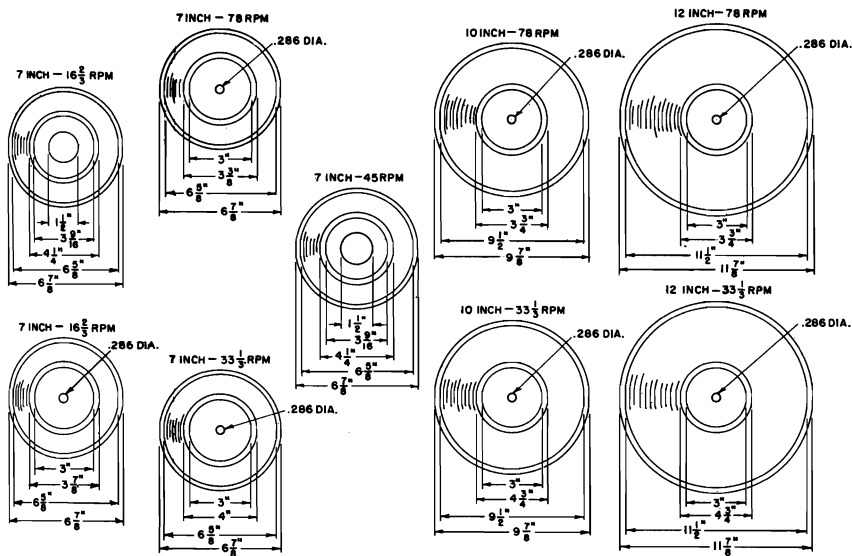
FIG. 9.38. Typical dimensions of the most common commercial-type disk phonograph records. The dimensions are the diameter of the outside of the record, the diameter of the outside and inside record groove, the diameter of the label, and the diameter of the center hole.

D. *Commercial Disk Phonograph Records.*[47,48,49]—Commercial phonograph records are made in four speeds, namely, 78, 45, $33\frac{1}{3}$, and $16\frac{2}{3}$ revolutions per minute. The 78 RPM records are made in three diameters, termed 12 inches, 10 inches, and 7 inches. The normal maximum playing times are 5, $3\frac{1}{2}$ and $2\frac{1}{2}$ minutes, respectively. The $33\frac{1}{3}$ RPM records are made in three diameters, termed 12 inches, 10 inches, and 7 inches. The nominal

[47] Goldmark, Snepvangers, and Bachman, *Proc. I.R.E.*, Vol. 37, No. 8, p. 923, 1949.
[48] Carson, Burt, and Reiskind, *RCA Review*, Vol. 10, No. 2, p. 173, 1949.
[49] Goldmark, Peter C., *Audio*, Vol. 39, No. 12, p. 15, 1955.

maximum playing times are 25, 17, and 8 minutes, respectively. The 45 RPM records are made in a diameter of 7 inches. The nominal maximum playing time is 8 minutes. The 16⅔ RPM records are made in a diameter of 7 inches. The nominal maximum playing time of the records with the large center hole is 30 minutes. The nominal maximum playing time for the small-hole records is 45 minutes for music and 60 minutes for speech. The diameter of the outside, the diameters of the first and last program grooves, the label diameters and the diameter of the center hole of the different records are shown in Fig. 9.38. It may be mentioned in passing that the specifications of Fig. 9.38 are given as representative and do not include all the variations.

The dimensions of the coarse groove, the fine groove, and the ultra-fine grooves and the dimensions of the corresponding stylii are shown in Fig. 9.37. The coarse groove is used in 78 RPM records. The fine groove is used in the 45 and 33⅓ RPM records. The ultra-fine groove is used in the 16⅔ RPM records.

The maximum nominal grooves per inch for the different size grooves are as follows: coarse groove, 125; fine groove, 275; and ultra-fine groove, 550.

The maximum amplitudes, in inches, in the frequency range 200 to 2000 cycles for the different size grooves are as follows: coarse groove, .004–.005 inch; fine groove, .0015–.002 inch; and ultra-fine groove, .0007–.001 inch.

9.4. Vibration Pickup.—Measurement and study of vibration have become an important factor in the elimination of noise in machinery, vehicles, and household appliances. Depending upon the requirements, it may be desirable to measure amplitude, velocity, or acceleration.

Direct measurement of acceleration, velocity, or displacement of vibration requires the establishment of a stationary body to serve as a reference frame against which these functions may be determined. Any type of transducer may be used to convert the motion into the corresponding electrical current. It is the purpose of this section to describe a piezoelectric inertia-type vibration pickup.

The structure of a typical inertia-type piezoelectric vibration pickup[50] is shown in Fig. 9.39. The crystal is a Rochelle salt bimorph type. With the crystal held at the three corners the output voltage is proportional to the force acting on the free corner. The crystal is enclosed in a rigid metal case. When the case is driven by a vibration normal to the plane of the crystal element, a force is developed at the unsupported section of the crystal due to inertia reaction. The mechanical network of the vibrating system is shown in Fig. 9.39. The mechanical resistance is small and does not influence the mechanical network save near the resonant frequency which occurs at about 1500 cycles. The velocity, in centimeters per second, is given by

$$\dot{x} = \frac{f_M}{j\omega m}\,(1 - \omega^2 m C_M) \qquad\qquad 9.21$$

[50] Bauer, B. B., *Jour. Acous. Soc. Amer.*, Vol. 11, No. 3, p. 303, 1940.

where f_M = driving force developed at the free edge of the crystal, in dynes,

m = effective mass of the crystal, in grams, and

C_M = effective compliance of the crystal, in centimeters per dyne.

For frequencies well below the resonant frequency the velocity is given by

$$\dot{x} \sim \frac{f_M}{j\omega m} \qquad\qquad 9.22$$

The acceleration is given by

$$\ddot{x} = j\dot{x}\omega = \frac{f_M}{m} \qquad\qquad 9.23$$

The displacement is given by

$$x = \frac{\dot{x}}{j\omega} = -\frac{f_M}{\omega^2 m} \qquad\qquad 9.24$$

Below the resonant frequency the force is proportional to and in phase with the acceleration. The voltage output of the unit then corresponds
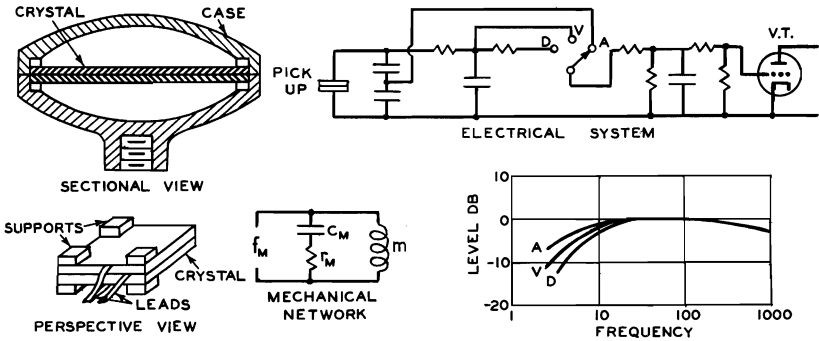
FIG. 9.39. Sectional view, perspective view of the crystal mounting arrangement, mechanical network, electrical connection, and response frequency characteristic of a vibration pickup. In the mechanical network, m = the mass of the crystal. r_M and C_M = the mechanical resistance and compliance of the crystal and supports. f_M = driving force. In the electrical system, with the switch lever on D, V, and A the response corresponds to displacement, velocity, and acceleration, respectively. The graph depicts the voltage response frequency characteristics for A, acceleration, V, velocity, D, displacement.

to the acceleration. The output of the acceleration-type pickup may be integrated once or twice by means of an electrical network as shown in Fig. 9.39 to obtain velocity and displacement.

The response frequency characteristics of the vibration pickup and the electrical system are shown in Fig. 9.39. It has been found that, above 1000 cycles, the performance of the pickup is influenced by manner of coupling to the vibration machine.

Magnetic and dynamic vibration pickups have also been developed.

In these devices two different types are used, in one the armature or voice coil is free and the field structure is coupled to the vibrating system under test, in the other the armature or voice coil is driven by the vibrating system under test. The electrical compensation in these devices differs from the crystal type because the voltage output is proportional to the velocity.

A ceramic vibration pickup[51] similar to the crystal vibration pickup has been developed. The vibrating system and the electrical characteristics are similar to those of the crystal vibration pickup. The essential difference being that the ceramic element is mounted on the four corners and the acceleration acts upon the center.

An accelerometer[52] has been developed employing a mechano-electronic transducer similar to that employed in the electronic microphone of Sec. 8.2F and the electronic pickup of Sec. 9.3B2f. A small weight is attached to the anode rod. The stiffness and restoring force are supplied by the diaphragm. The system exhibits a high order of sensitivity.

9.5. Sound-Powered Phones.—A sound-powered phone system is a point-to-point telephone communicating system employing no batteries, amplifiers or any other means of external power. The sequence of events in a sound-powered telephone system is as follows: The human voice produces a sound wave which actuates the microphone at the transmitting end. The microphone converts the acoustical energy into the corresponding electrical energy. This energy is carried by wires to the receiving end. At the receiving end the electrical variations are transformed into the corresponding sound vibrations by the receiver.

A sound-powered telephone is shown in Fig. 9.40. Cross-sectional views, mechanical and electrical networks of the microphone and receiver are shown in Fig. 9.40. In order to obtain a tolerable sound level at the receiver the over-all efficiency of the system must be quite high. This high efficiency is accomplished by the use of multi-resonant elements which reduce the mechanical impedance of the vibrating system. The transmission frequency band is made relatively narrow so that a low value of mechanical impedance can be obtained with a simple vibrating system. The response frequency characteristics of the microphone, the receiver, and the combination of the microphone and receiver are shown in Fig. 9.41. In the combination system it will be seen that there is a gain in sound pressure over the useful transmission frequency range which means that the sound pressure in the ear cavity is greater than that at the microphone. The transmission of sound without pressure loss requires a very efficient system.

9.6. Electrical Megaphone.—The electrical megaphone[53] consists of the combination of a microphone, an amplifier, and a horn loudspeaker (Fig. 9.42). The microphone and horn loudspeaker form a single unit. In use, the operator speaks into the microphone. The voice is reinforced by the amplifier and loudspeaker. The resulting power output is many

[51] Carlson, E. V., *Trans. I.R.E.*, Prof. Group on Audio, PGA-10, p. 2, 1952.
[52] Lewis, Robert C., *Jour. Acous. Soc. Amer.*, Vol. 22, No. 3, p. 357, 1950.
[53] Sanial, A. J., *Communications*, Vol. 25, No. 7, p. 33, 1945.
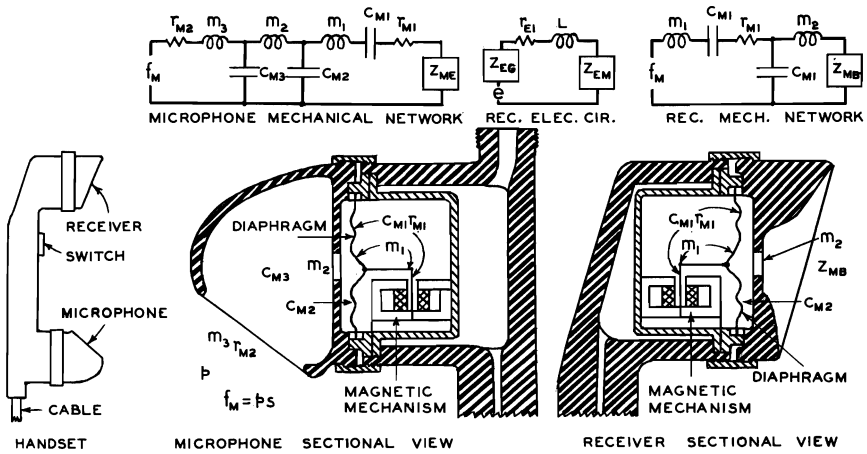
FIG. 9.40. The sound powered handset, sectional views of the microphone and receiver, mechanical networks of the microphone and receiver, and electrical circuit of the receiver. In the microphone mechanical network, z_{ME} = the mechanical impedance due to the electrical circuit. m_1, r_{M1}, and C_{M1} = the mass, mechanical resistance, and compliance of the diaphragm and armature. C_{M2} = the compliance of the air chamber in front of the diaphragm. m_2 = the mass of the air in the aperture in the diaphragm cover plate. C_{M3} = the compliance of the mouthpiece cavity. m_3 and r_{M3} = the mass and mechanical resistance of the air load upon the mouthpiece. f_M = the driving force. S = the area of the diaphragm, and p = the sound pressure. In the receiver mechanical network, m_1, r_{M1}, and C_{M1} = the mass, mechanical resistance, and compliance of the diaphragm and armature. C_{M2} = the compliance of the cavity in front of the diaphragm. m_2 = the mass of the air in the aperture of the diaphragm cover plate. z_{MB} = mechanical impedance of the external load upon the receiver. In the receiver electrical circuit, z_{EM} = the electrical motional impedance. L and r_{E1} = the damped inductance and electrical resistance of the receiver. z_{EG} = the electrical impedance of the microphone. e = the developed voltage output of the microphone.
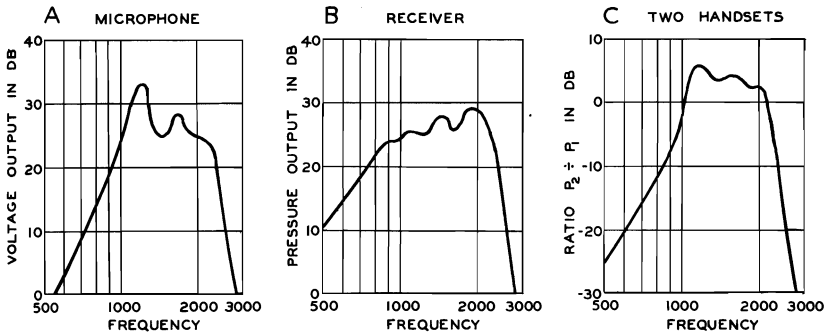
FIG. 9.41. Response frequency characteristics of a sound powered telephone. A. Voltage response frequency characteristic of the microphone. B. Voltage response frequency characteristic of the receiver. C. Over-all pressure ratio response frequency characteristic of two handsets, one used as a transmitter and the other as a receiver.

times that of the unaided voice or the voice and an acoustical megaphone. The only theoretical limitation to the amount of reinforcing is the production of continuous oscillations due to regenerative feedback from the loudspeaker to the microphone. The directional pattern of horns shows that the rear radiation is quite small compared to that directly in front when the dimensions of the mouth of the horn are comparable to the wavelength (see Sec. 2.19). By placing the microphone at the rear of the horn and
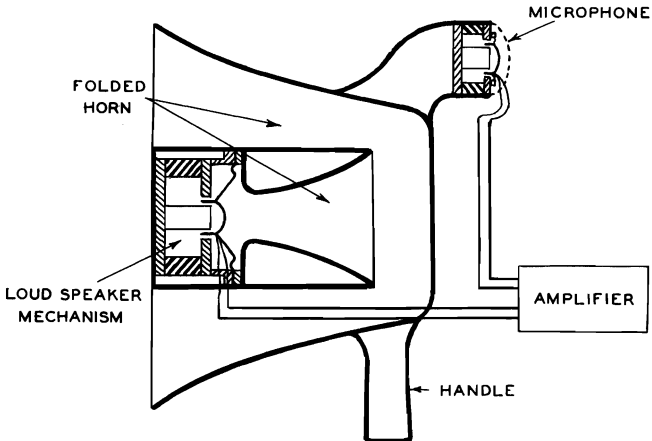
FIG. 9.42. Sectional view depicting the elements and electrical connections of an electrical megaphone.

attenuating the low-frequency range the amount of sound picked up by the microphone is small. This makes it possible to obtain a relatively large output before oscillations begin. The microphone employed for the electrical megaphone is of the close-talking type. The loudspeaker is a conventional, light-weight horn loudspeaker. Two types of amplifiers have been used—namely, a portable battery operated amplifier and a semiportable a-c line operated amplifier.

9.7. Magnetic Tape Sound Recording and Reproducing System. [54,55,56,57,58,59,60,61,62,63]—Magnetic recording and reproducing were invented

[54] Carlson and Carpenter, U.S. Patent, 1,640,881, 1927.

[55] Wetzel, W. W., *Audio Eng.*, Vol. 31, No. 12, p. 12, 1947.

[56] Camras, Marvin, *Jour. Soc. Mot. Pic. Eng.*, Vol. 48, No. 1, p. 14, 1948.

[57] Begun, "Magnetic Recording," Murray Hill Books, New York, N.Y., 1949.

[58] Frayne and Wolfe, "Elements of Sound Recording," John Wiley and Sons, New York, N.Y., 1949.

[59] Gratian, J. W., *Jour. Acous. Soc. Amer.*, Vol. 21, No. 2, p. 74, 1949.

[60] Kornei, O., *Jour. Audio Eng. Soc.*, Vol. 1, No. 1, p. 225, 1953.

[61] Joryz, Alfred, Bibliography of Magnetic Recording, *Jour. Audio Eng. Soc.*, Vol. 2, No. 3, p. 183, 1954.

[62] Camras, Marvin, *Convention Record I.R.E.*, *Part 3, Audio*, p. 16, 1953.

[63] Selsted and Snyder, *Trans. I.R.E.*, Prof. Group on Audio, Vol. AU-2, No. 5, p. 137, 1954.

more than a half century ago by Poulsen. Since that time there has been a periodic revival of magnetic recording and reproducing. During the past decade the swing has been to magnetic tape with the result that today it is universally employed for all types of magnetic reproduction. Magnetic tape consists of a plastic base with a coating of magnetic oxide as shown in Fig. 9.43. The base material in use today varies in thickness from .001 inch to .0015 inch. The magnetic coating varies in thickness from .0002 inch to .0006 inch.

MAGNETIC COATING

PLASTIC
BASE
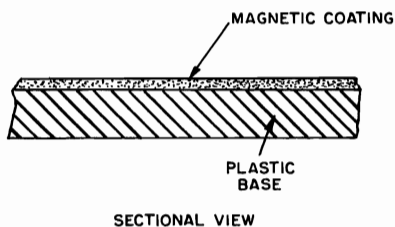
SECTIONAL VIEW

FIG. 9.43. Sectional view of magnetic recording tape.

A typical BH curve of the magnetic coating is shown in Fig. 9.44. The retentivity is of the order of 700 gausses and the coercive force is of the order of 300 gilberts.

The recording and reproducing process is depicted in Fig. 9.45. The passage of the tape past the recording head leaves a series of magnetized sections which correspond to the signal which was applied to the head when the tape was in contact with the head at each of these sections. In the reproduction process, the tape is moved past the head with the result that a change in magnetic flux is produced in the head as a magnetized section passes the head. This change in flux induces a voltage in the coil which corresponds to the voltage of the original applied signal.

B-H

H

FIG. 9.44. Typical B–H characteristic of the iron oxide coating on magnetic tape.

A schematic view and the magnetic network of a magnetic recording head and magnetic tape are shown in Fig. 9.46. The currents i_1 and i_1' produce a flux ϕ in the tape. Due to the retentivity of the coating material of the tape, a magnetized section is produced. The action and performance of the recording process can be deduced from the magnetic network of Fig. 9.46.

A schematic view and magnetic network of a magnetic reproducing head and magnetic tape is shown in Fig. 9.47. When the magnetic tape is moved past the head, the magnetomotive force of the magnetized reactions produces the magnetic flux ϕ_5 and ϕ_5' in the two coils. When this flux changes,
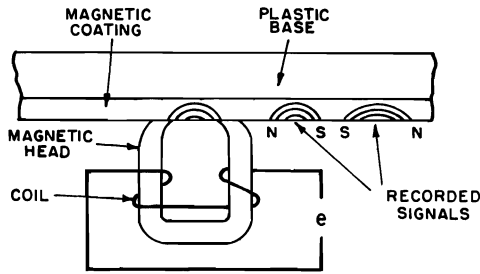
FIG. 9.45. Schematic diagram depicting the magnetic tape recording and reproducing process.
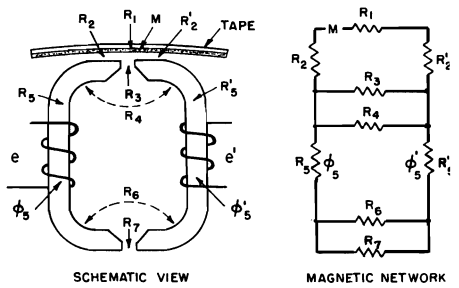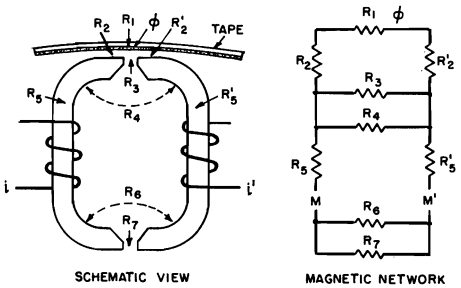
FIG. 9.46. Schematic view and magnetic network of the head and tape of a magnetic tape recording system. In the magnetic circuit, M and M' = the magnetomotive forces developed by the currents in the two coils. R_5 and R_5' = the reluctances of the magnetic material of the heads. R_3 and R_7 = the reluctances of the top and bottom air gaps. R_4 and R_6 = the reluctance of the magnetic leakage path. R_2 and R_2' = the reluctances of the air gaps between the head and the tape. R_1 = the reluctance of the magnetic tape. ϕ = the flux through the magnetic tape. i and i' = the currents in the two coils.

FIG. 9.47. Schematic view and magnetic network of the head and tape of a magnetic tape reproducing system. In the magnetic circuit M = the magnetomotive force stored in the tape. R_5 and R_5' = the reluctances of the magnetic material of the heads. R_3 and R_7 = the reluctances of the top and bottom air gaps. R_4 and R_6 = the reluctances of the magnetic leakage paths. R_2 and R_2' = the reluctances of the air gaps between the head and the tape. R_1 = the reluctance of the magnetic tape. ϕ_5 and ϕ_5' = the flux in the coils. e and e' = the induced voltages in the coils.

as the magnetomotive force changes as different magnetized sections of the tape pass over the head, a voltage e_1 and e_1' is induced in the coils. These voltages correspond to the currents i_1 and i_1' applied to the recording head.

In the recording and reproducing process there is a loss due to the finite gap length. In the recording process the gap width is of little importance because the recording process takes place from the trailing edge of the gap. It is this edge rather than the gap that is of importance in the recording head. In the reproducing process it is the length of the gap that determines

the magnetomotive force. The loss due to the finite length of the gap in the reproducing head is given by

$$R = - 20 \log \left[\frac{\sin \dfrac{\pi d}{\lambda}}{\dfrac{\pi d}{\lambda}} \right] \qquad 9.25$$

where R = loss in decibels,

  d = length of the gap, in centimeters, and

  λ = wavelength of the signal along the tape, in centimeters.

The gap loss response frequency characteristic is shown in Fig. 9.48. It
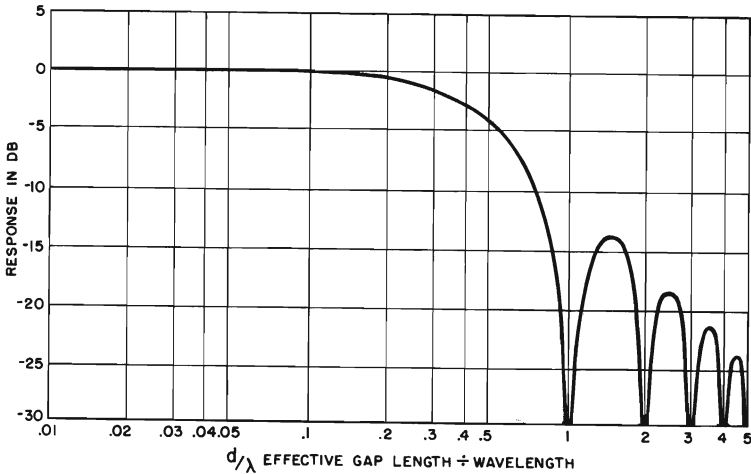
FIG. 9.48. The response loss due to the reproduction with a finite gap as a function of the ratio of the gap length to the wavelength.

will be seen that the response is zero when the gap length equals a wavelength and multiples of the wavelength.

The output of the reproducing head is

$$e = N \frac{d\phi}{dt} \qquad 9.26$$

where e = voltage output, in abvolts,

  N = number of turns in the coil,

  ϕ = flux in the coil, in gausses, and

  t = time.

If the amplitude of ϕ is a constant, the voltage will increase at the rate of 6 db per octave. However, this characteristic must be multiplied by the

gap loss. The open circuit voltage response of a magnetic reproducing system is given by

$$R_M = 20 \log \left(\sin \frac{\pi d}{\lambda} \right) \qquad 9.27$$

The open circuit voltage response frequency characteristic of a magnetic reproducing system is shown in Fig. 9.49.
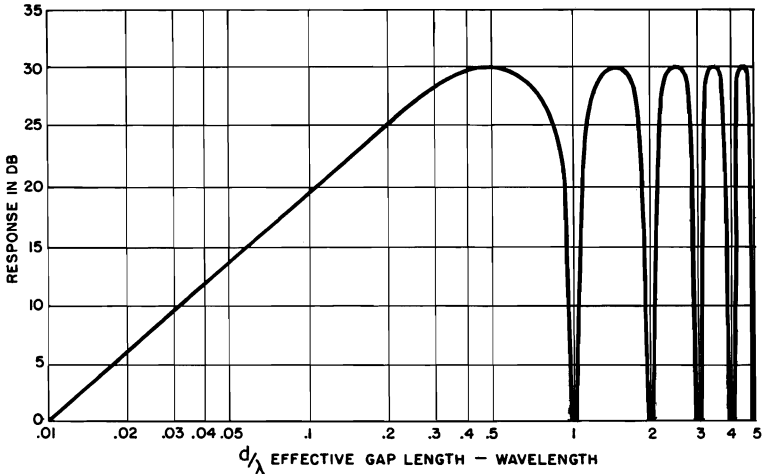
FIG. 9.49. The response of a magnetic tape reproducing system as a function of the ratio of the gap length to the wavelength.

The magnetic material used in the coating of the plastic tape is of necessity nonlinear because it must possess retentivity to retain the magnetic signal applied to the tape in recording. The characteristic which depicts the magnetomotive force or magnetizing force H produced by the recording head in the magnetic tape and the residual induction B_r after the magnetic tape leaves the head is depicted by the characteristic 1, 2, 0, 3, 4 of Fig. 9.50. The nonlinear portion in the vicinity of 0 of Fig. 9.50 will produce distortion. Various means have been developed for reducing the effects of this non-linear characteristic. The system which is universally used in sound reproduction today is the alternating current bias. The high-frequency signal 50 to 150 kilocycles is added to the audio signal in the recording head, there being no modulation of one signal by the other. The action of the addition of the high-frequency bias in reducing the effect of the nonlinear characteristic is shown in Fig. 9.50.

The system used in recording and reproducing from magnetic tape is shown in Fig. 9.51. The tape transport mechanism consists of the takeup and payoff reels and the capstan drive. Three heads are used, namely, erasing, recording, and reproducing heads. The reproducing head is used, in recording, to monitor the recording. In reproducing the erasing and

recording heads are immobilized. In recording, any previous signal is removed by the erase head which is accomplished by feeding a high-frequency signal of high intensity to the erase head.

When the recording process is carried out with constant current in the
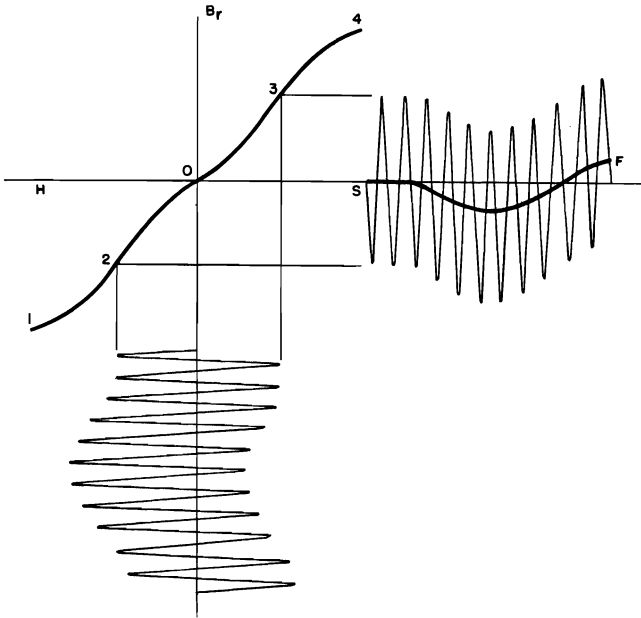
FIG. 9.50. The recording on magnetic tape with a high-frequency bias. The characteristic 1, 2, 0, 3, 4 is the residual induction Br produced by the magnetizing force H produced by the recording head. The high-frequency bias and audio signal are applied to the tape. The resultant characteristic in reproduction is the characteristic S–F.
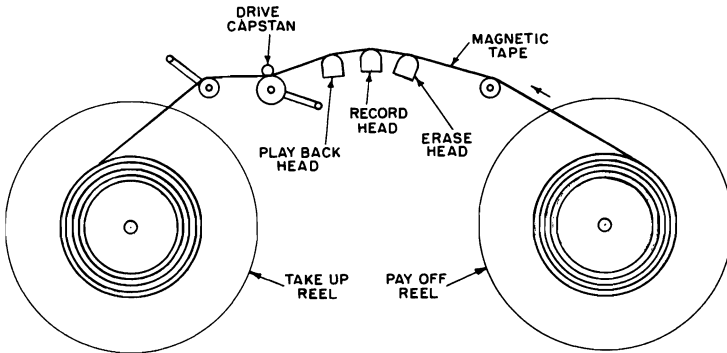
FIG. 9.51. The elements of a magnetic tape recording and reproducing system.

head and the reproducing process is carried out with an amplifier in which the response is independent of the frequency, the overall response will be given by the characteristic of Fig. 9.49. Therefore, suitable compensation must be provided in order to obtain an over-all uniform response frequency characteristic. The upper frequency limit is determined by the first dip, where the gap is equal to the wavelength. In recording, suitable high-frequency accentuation is applied in the range above $d/\lambda = .3$ so that in reproduction no additional compensation will be required in this frequency range. In reproduction, the response is accentuated 6 db per octave with decrease in frequency in the frequency range below $d/\lambda = .3$. In this way a uniform recording-reproducing characteristic is obtained. The accentuation of high-frequency response in recording increases the signal-to-noise ratio. There is no overload problem in this frequency range due to accentuation of response in recording, because the amplitude of speech and music is lower in the high-frequency range compared to the mid- and low-frequency ranges. In the frequency range below $d/\lambda = .3$ the amplitude of flux ϕ will be constant for constant current in the recording head. Therefore, in order to obviate overloading of the tape in recording, the low-frequency compensation must be supplied in the reproducing amplifier.

The standard tape speeds are as follows: 30, 15, $7\frac{1}{2}$, $3\frac{3}{4}$, and $1\frac{7}{8}$ inches per second. The higher speeds are used for high quality recording. The lower speeds are used for speech reproduction. The standard tape width is $\frac{1}{4}$ inch. Both single- and double-track recordings are used on the $\frac{1}{4}$-inch tape.

The upper frequency limit of reproduction will depend upon the air gap of the head and the tape speed. With heads in use today, and at tape speeds of 30 and 15 inches per second an upper frequency limit of 20,000 cycles can be easily achieved. At a tape speed of $7\frac{1}{2}$ inches per second an upper frequency limit of 15,000 cycles per second is possible in a well-designed system. The upper frequency limit employing tape speeds of $3\frac{3}{4}$ and $1\frac{7}{8}$ inches is correspondingly lower.

9.8. Magnetic Tape Conversion Systems.—A. *Frequency Conversion System.*—The fact that magnetic tape can be operated over a tremendous range of speeds from a fraction of an inch per second to a thousand inches per second provides a means for frequency conversion. The frequency conversion may be either to a lower or a higher frequency. The use of a magnetic tape system for providing frequency conversion is shown in Fig. 9.52. The tape is recorded with a linear speed V and a maximum frequency bandwidth of f. The system over which the signal is to be transmitted is limited to a maximum frequency band of less than f. The tape is reproduced at a slower speed V_1 so that the maximum frequency corresponds to the capabilities of the transmission system. The output of the transmission system is fed to a recorder operating at a tape speed of V_1 and the signal is recorded. The record produced in this manner is reproduced at a tape speed V. In this way the original signal is recovered. The time of transmission is increased by the ratio of $\dfrac{V}{V_1}$.
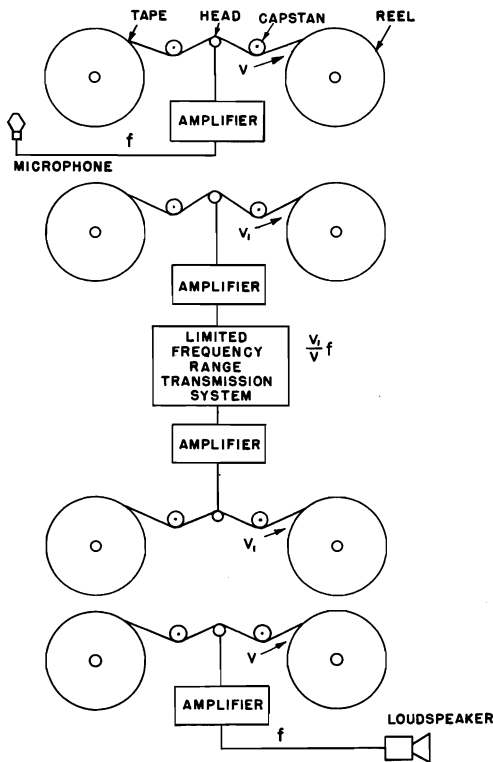
FIG. 9.52. The elements and processes in a magnetic
tape frequency conversion system.

The preceding example illustrated a reduction in the frequency band-
width. The system may also be used to step up the frequency. For
example, if the transmission takes place on a transmission system with a
wider bandwidth, the frequency may be increased and the time of trans-
mission reduced.

B. *Frequency Compression System.*—The redundancy in speech is com-
paratively large. Therefore, in the transmission of speech another method
of reducing bandwidth, termed frequency compression, may be used. The
use of magnetic tape for a frequency compression system is shown in Fig.
9.53. The original signal is recorded at a tape speed V. Then this tape is
reproduced by means of a system of rotating heads which move in the same
direction as the tape at peripheral velocity of V_1. The frequency is reduced
by the factor $\dfrac{V - V_1}{V}$. However, sections of the tape must be left out.
The amount of the signal on the original tape that is used is $\dfrac{V - V_1}{V}$. The
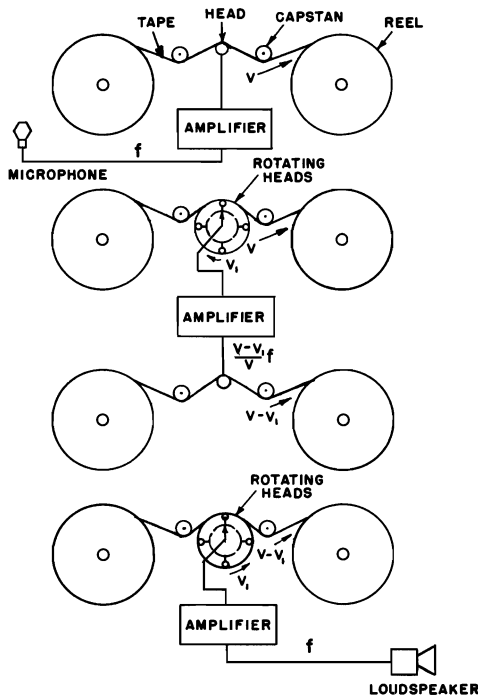
FIG. 9.53. The elements and processes in a magnetic tape frequency compression system. ·

output of the rotating heads is fed to a recorder operating with a tape speed of $V - V_1$. In reproduction, the system of rotating heads rotated in a direction opposite to the motion of the tape with a peripheral velocity of V_1. The original frequencies are restored.

The advantage of the system resides in the fact that the amount of tape required for the storage system is reduced and the frequency band required for transmission is reduced.

In an alternative reproducing system the tape is reproduced at a velocity V as shown in Fig. 9.54. The time required for the reproduction is now reduced by a factor $\dfrac{V - V_1}{V}$.

C. *Time Compression System.*—A time compression system similar to the frequency compression system may be used for reducing the time of transmission.

The use of a magnetic tape for time compression is shown in Fig. 9.55. The original signal is recorded at a tape speed V. Then this tape is repro-
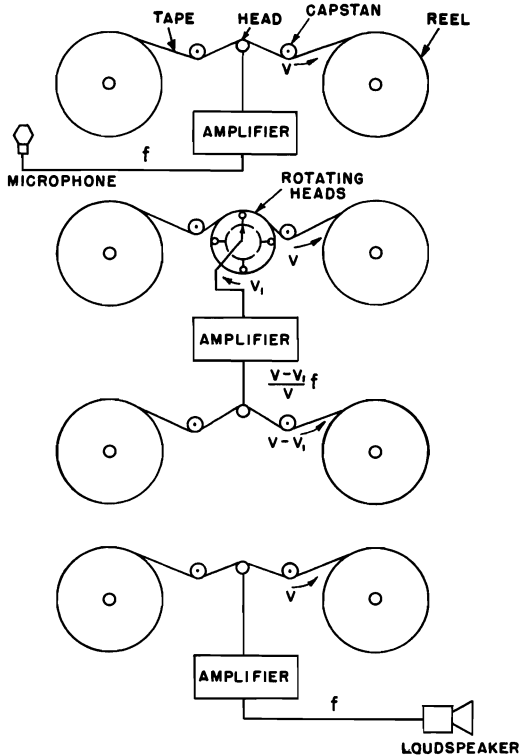
FIG. 9.54. The elements and processes in a magnetic tape frequency compression system.

duced at a tape speed $V + V_1$ by means of a system of rotating heads which move in the same direction as the tape at a peripheral velocity V_1. The frequency output is the same as the original frequency. However, sections of the original signal are left out. The amount of the signal on the original tape that is used is $\dfrac{V - V_1}{V}$. The output of the rotating heads is fed to a recorder operating with a tape speed of V. The tape is reproduced at a speed V on a conventional machine. The frequency is the same as the original. Therefore, since sections of the signal are left out, the time required for reproduction is $\dfrac{V}{V + V_1}$ times that of the original.

9.9. Sound Motion Picture Recording and Reproducing System.[64]

—A sound motion picture recording system consists of a means for producing a modulated light beam and means for moving a light sensitive film relative

[64] Kellogg, E. W., History of Motion Pictures, *Jour. Soc. Mot. Pic. Tel. Eng.*, Vol. **64**, No. 6, p. 291, 1955; Vol. 64, No. 7, p. 356, 1955; Vol. 64, No. 8, p. 422, 1955.
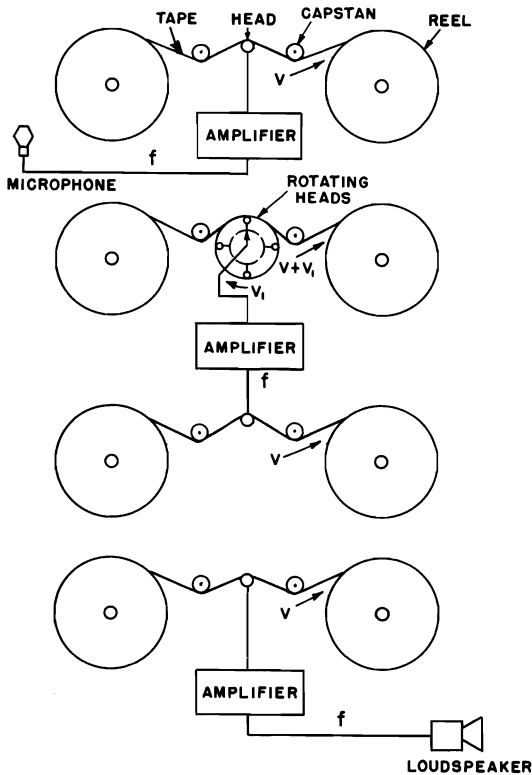
FIG. 9.55. The elements and processes in a magnetic
tape time compression system.

to the beam for recording signals derived from sound signals. A sound
motion picture reproducing system is a combination of a light source, optical
system, and photoelectric cell and a mechanism for moving a film carrying
an optical sound record by means of which the recorded variations may be
converted into electrical signals of approximately like form.   The elements
of a complete motion picture recording and reproducing system are described
in Sec. 13.7.   It is the purpose of this section to describe the film and sound
track, the modulators, and film transport used in the recording of sound on
photographic film and the optical-electronic reproducer and film transport
used in reproduction of sound from photographic film.

A. *Film and Sound Track.*—In the recording of sound motion pictures the
picture and sound are recorded on separate photographic films. See Sec.
13.7.   Therefore, the camera and sound recorder must be synchronized.
This is accomplished by the use of an interlock system between the camera
and sound recorder and the use of perforated film in the form of sprocket

holes along the two edges of the film for both the camera and sound recorder. Fig. 9.56.

The sound track on 35-millimeter film occupies a space about .1 inch wide just inside the sprocket holes as shown in Fig. 9.56. There are two types of sound track in general use today—namely, variable area and variable density. The type of sound track shown in Fig. 9.56 is termed bilateral variable area. There are also other types as, for example, unilateral, duplex, class A pushpull, and class B pushpull variable area sound tracks.
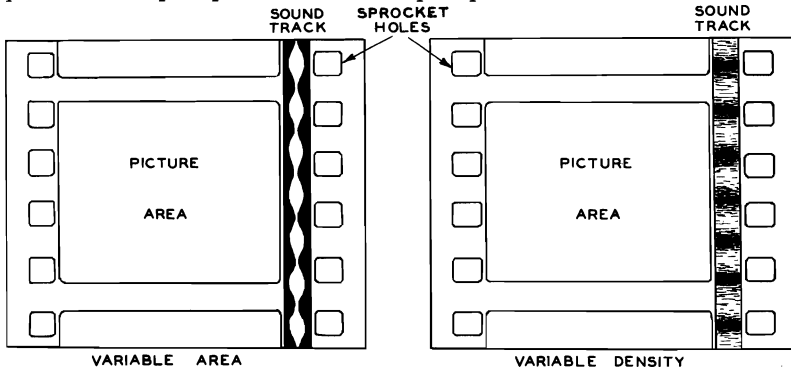
FIG. 9.56. The position of the picture and sound track in 35-millimeter sound motion-picture film. Two types of sound track are shown—namely, variable area and variable density.

In addition to the conventional variable density sound track shown in Fig. 9.56, there are other types as, for example, squeeze, class A pushpull, and class B pushpull variable density sound tracks.

B. *Recording System.*—1. *Variable area.*[65,66,67,68,69]—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

[65] Albin, F., Clark, L. E., Hill, A. P., Hilhard, J. K., Kimball, Harry, Lambert, Kenneth, and Miller, Wesley, "Motion Picture Sound Engineering," D. Van Nostrand Company, Princeton, N.J., 1938.

[66] Hardy, A. C., *Trans. Soc. Mot. Pic. Eng.*, Vol. 11, No. 31, p. 475, 1927.

[67] Dimmick, G. L., *Jour. Soc. Mot. Pic. Eng.*, Vol. 15, No. 4, p. 428, 1930.

[68] Kellogg, E. W., *Jour. Soc. Mot. Pic. Eng.*, Vol. 25, No. 3, p. 203, 1935.

[69] Sachtleben, L. T., *Jour. Soc. Mot. Pic. Eng.*, Vol. 25, No. 2, p. 175, 1935.

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[70,71,72] 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
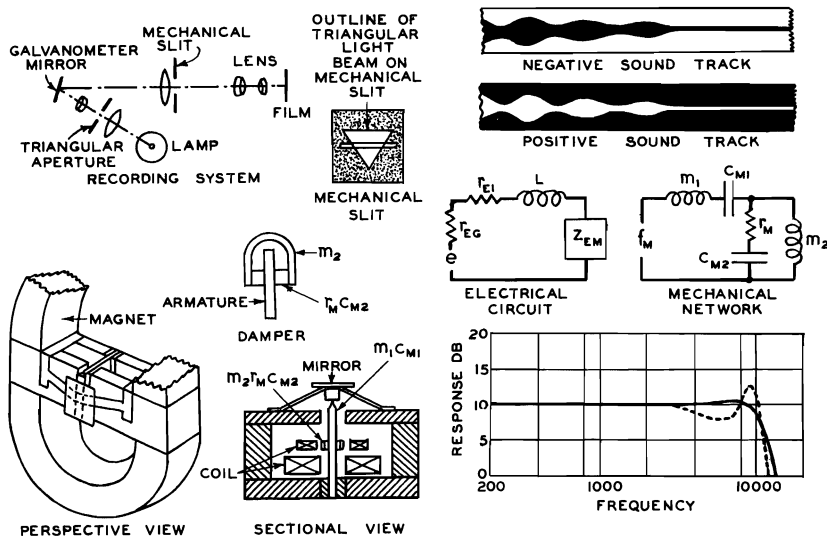
FIG. 9.57. The elements of a variable area sound motion-picture film recording system. The negative and positive sound tracks. Perspective and sectional views, mechanical network, and electrical circuit of the galvanometer. In the mechanical network, f_M = the mechanical driving force. m_1 and C_{M1} = the mass and compliance of the armature. m_2, r_M, and C_{M2} = the mass, mechanical resistance, and compliance of the damper. In the electrical circuit, z_{EM} = the electrical motional impedance. L and r_{E1} = the damped inductance and electrical resistance. r_{EG} = the electrical resistance of the generator. e = the voltage of the generator. The graph depicts the amplitude response frequency characteristic of the galvanometer. Dotted and solid lines depict the amplitude response for the galvanometer alone and with an electrical capacitance in shunt with the galvanometer, respectively.

is produced on the negative record which means a correspondingly narrow 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

[70] Kreuzer, B., *Jour. Soc. Mot. Pic. Eng.*, Vol. 16, No. 6, p. 671, 1931.

[71] Dimmick, G. L., *Jour. Soc. Mot. Pic. Eng.*, Vol. 29, No. 3, p. 258, 1937.

[72] Kellogg, E. W., *Jour. Soc. Mot. Pic. Eng.*, Vol. 36, No. 2, p. 137, 1941.

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. Perspective and sectional views, the electrical circuit, and the mechanical network of a film recording galvanometer are shown in Fig. 9.57. The controlling element in the vibrating system in the low-frequency range is the compliance C_{M1}. Under this condition the ratio of the amplitude to the applied force is independent of the frequency. A damper, m_2, r_M, C_{M2}, reduces the amplitude in the region of the resonant frequency of m_1 with C_{M1}. The amplitude response frequency characteristic is shown in Fig. 9.57. It will be seen that the rotational amplitude is uniform with respect to frequency to about 10,000 cycles.

2. *Variable Density*.[73]—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 upon the moving film. The elements of a variable density recording system are shown in Fig. 9.58. The ribbons of the light 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 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[74,75] 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 action is similar to that in the variable area system. The elements of a light valve and the electrical and mechanical circuits are shown in Fig. 9.58. Below the resonant frequency the controlling element in the mechanical circuit is the compliance C_M. Therefore, in this frequency range the ratio of the applied force and the amplitude is independent of the frequency. At the resonant frequency of the ribbons the response is accentuated. The amplitude response frequency characteristic of a light valve is shown in Fig. 9.58.

3. *Recording Film Transport*.[76,77]—The film transport mechanism used in recording sound on film consists of a positive drive of the perforated film and a constant speed drive of the film where the modulated light beam strikes the film. A film transport mechanism of this type is shown in Fig. 9.59.

[73] MacKensie, D., *Trans. S.M.P.E.*, Vol. 12, No. 35, p. 730, 1928.
[74] Silent and Frayne, *Jour. Soc. Mot. Pic. Eng.*, Vol. 18, No. 5, p. 551, 1932.
[75] Scoville and Bell, *Jour. Soc. Mot. Pic. Eng.*, Vol. 38, No. 2, p. 125, 1942.
[76] Kellogg, E. W., *Jour. Soc. Mot. Pic. Eng.*, Vol. 15, No. 5, p. 653, 1930.
[77] Collins, M. E., *Jour. Soc. Mot. Pic. Eng.*, Vol. 48, No. 2, p. 148, 1947.
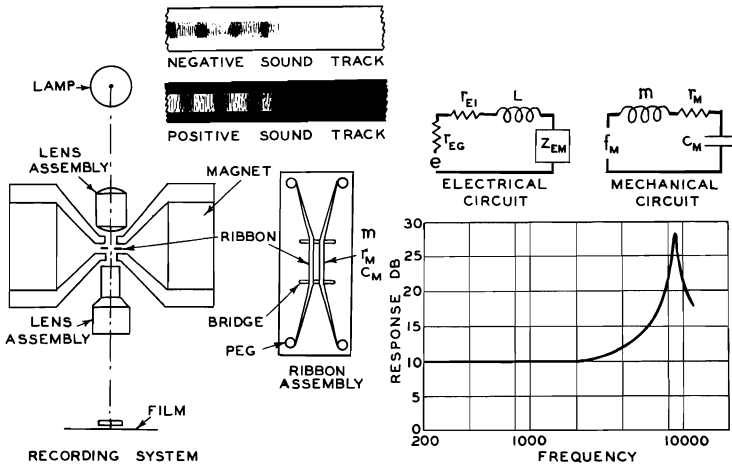
FIG. 9.58.   The elements of a variable density sound motion-picture film recording system.   The negative and positive sound tracks.   Sectional and ribbon assembly views, mechanical circuit, and electrical circuit of the light valve.   In the mechanical circuit, f_M = the mechanical driving force. m, r_M, and C_M = the mass, mechanical resistance, and compliance of the ribbons.   In the electrical circuit, z_{EM} = the motional electrical impedance. L and r_{EL} = the damped inductance and electrical resistance of the ribbons. r_{EG} = the electrical resistance of the generator. e = the voltage of the generator.   The graph depicts the amplitude response frequency characteristic of the light valve.
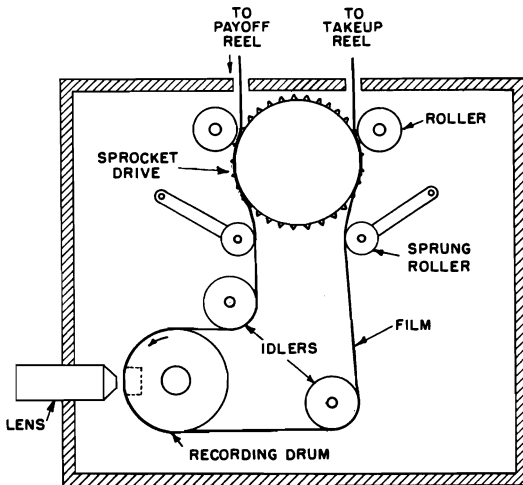
FIG. 9.59.   Schematic view of the photographic film transport mechanism of a motion-picture film sound recorder.

Positive drive of the film is obtained by the sprocket drive.   The sprocket drive is interlocked with the camera drive so that synchronism of the picture and sound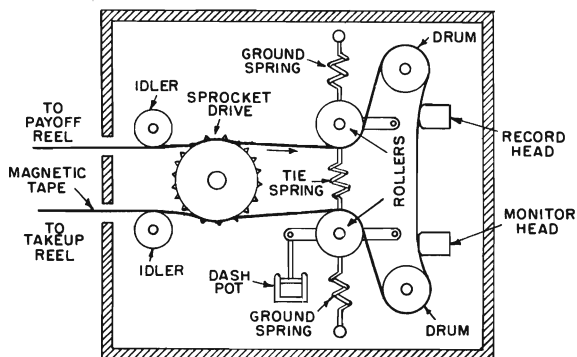 will be obtained.   See Sec. 13.7.   When the film passes over the sprocket drive, variations in the motion of the film at the sprocket hole frequency are produced.   These variations in the film speed must be removed at the recording point to eliminate spurious frequency modulation of the image on the film.   Uniform speed at the recording point is provided by the filter between the sprocket drive and recording point consisting of the inertia of the recording drum and the compliance of the film between the recording drum and the sprocket drive.   The recording drum is driven by a magnetic system from the motor which drives the sprocket which provides a slight amount of drive to the film.   This form of drive provides isolation from variations in the rotational speed of the motor drive.   The combination of the isolating filter and magnetic drive provides a system with very uniform motion at the surface of the drum.   The image of the modulator is focused on the film while it is in contact with the drum.

C. *Reproducing System.*—1. *Optical-Electronic Reproducer.*—The elements of a motion picture film sound reproducing system are shown in Fig. 9.60. The light source, in the form of an incandescent lamp, is focused upon a 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
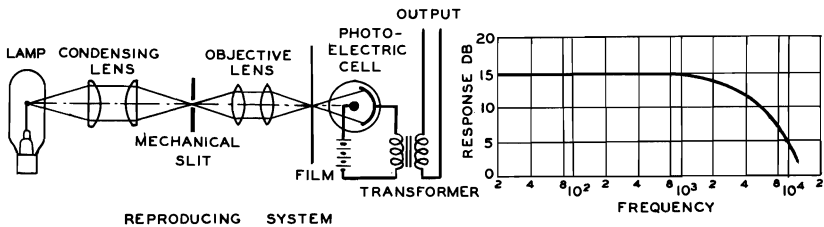
FIG. 9.60.   The elements of a motion picture film sound reproducing system and the voltage response frequency characteristic with a constant amplitude film.

usually about .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 light undulations which fall upon the photocell correspond to the voltage variations applied to the recording galvanometer.   The voltage output of the photocell is proportional to the amount of light which falls upon the cathode. The voltage output response frequency characteristic of a typical motion picture film sound reproducing system using a constant amplitude film is shown in Fig. 9.60.   The falling off in response at the high-frequency portion of the range is due to the finite dimensions of the slits in the recording and reproducing systems.   This reduction in response can be overcome by compensations in the recording and reproducing systems.

2. *Reproducing Film Transport.*[78]—The film transport used in reproducing sound on photographic film consists of a positive drive of the perforated film and a constant speed drive where the light passes through the film to the photoelectric cell. A film transport mechanism of this type is shown in Fig. 9.61. Positive drive of the film is obtained by means of the two sprocket drives. The sprocket drives are geared with the positive picture drive so that a constant loop of film is maintained between the sound head and the picture head. The positive drive also insures that the film speed in repro-
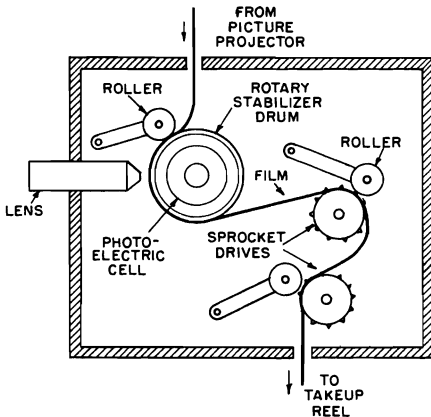
FIG. 9.61. Schematic view of the photographic film transport mechanism of a motion picture film sound reproducer.

duction will be same as that in recording. Since there is a loose loop of film between the picture head and sound head, variations in the picture drive will not be imparted to the sound head. After the film enters the sound head it passes over a drum. The light beam of the reproducing system passes through the film to the photocell located inside the drum while the film is on the drum. Since the drum is rotated at a constant speed, the film will move past the light beam at a constant speed. The drum is driven by the first sprocket drive. The second sprocket isolates the takeup reel from the reproducing system. Uniform speed at the reproducing point is provided by the filter between the sprocket drive and the reproducing point consisting of the inertia of the drum and the compliance of the film between the drum and sprocket drive. To provide damping and stabilization, the drum drives a flywheel through a fluid coupling. This is termed a rotary stabilizer. The combination of the filter and the rotary stabilizer provides a system with very uniform motion at the surface of the drum.

9.10. Motion Picture Magnetic Tape Sound Recording and Reproducing System.[79]—During the past few years the original recording of the sound directly upon motion picture film has to a large extent been replaced by recording on magnetic tape. The magnetic system for recording on magnetic tape is the same as that described in Sec. 9.7. However, in order to maintain synchronism between the picture recorded by the camera and the sound recorded on the magnetic tape, perforated film or tape is used in the sound recorder.

[78] Loomis and Reynolds, *Jour. Soc. Mot. Pic. Eng.*, Vol. 25, No. 5, p. 449, 1935.
[79] Frayne and Wolfe, "Elements of Sound Recording," John Wiley and Sons, New York, N.Y., 1949.

During the past three years, wide screen motion picture systems with stereophonic sound have been introduced on a wide scale. The information is rerecorded on the magnetic strips cemented to the positive picture release print. The magnetic system for reproducing the magnetic strips is the same as that described in Sec. 9.7. It is the purpose of this section to describe the magnetic tape, the tape transport used in recording and the tape transport used in reproducing.

A. *Magnetic Tape.*—The camera and sound recorder must be interlocked as in the case of the recording on photographic film. Therefore, a magnetic coating on a perforated plastic base is used as the recording medium. The dimensions are the same as those of 35-millimeter photographic film. Fig. 9.62A. One reason for the use of magnetic tape of these dimensions is
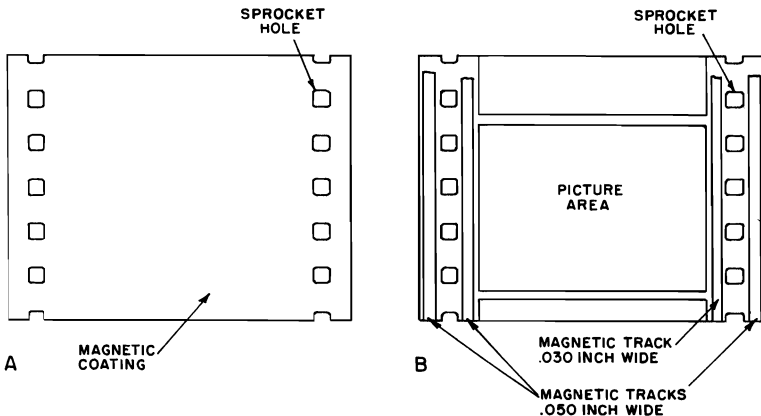
FIG. 9.62. A. Magnetic tape for original sound recording in sound motion pictures. B. Magnetic strips on a motion-picture positive film.

that if desired the existing photographic recorder may be used by the addition of magnetic heads. Special magnetic recorders have also been developed, but these also use magnetic tape of the same dimensions. Another reason for the use of the wide tape is that several tracks representing several channels may be recorded on the tape shown in Fig. 9.62A because the total width of the magnetic coating is approximately an inch. For example, a soloist may be recorded on one track and the sections of the orchestra may be recorded on other tracks. The three or more channels of stereophonic sound may be recorded on the magnetic tape. See Sec. 13.7.

The information recorded on the magnetic tape is rerecorded on the photographic film by means of the systems described in Sec. 9.8.

During the past three years wide screen motion picture systems[80] with stereophonic sound have been introduced on a wide scale. The elements

[80] Sponable, Brazz, and Grignon, *Jour. Soc. Mot. Pic. Tel. Eng.*, Vol. 63, No. 1, p. 1, 1954.

of the system are described in Sec. 13.7. The three or four channels are recorded on the tape shown in Fig. 9.62A. The information is rerecorded on the magnetic strips cemented to the positive picture release print as shown in Fig. 9.62B.

B. *Recording Tape Transport*.[81,82]—The tape transport used in recording sound on magnetic tape for sound motion pictures consists of a positive sprocket drive of the perforated film and a constant speed drive where the magnetic recording head is in contact with the tape. A magnetic tape transport mechanism of this type is shown in Fig. 9.63. Positive drive of the tape is obtained by means of the sprocket drive. The sprocket drive is interlocked with the camera drive so that synchronism of the picture and

FIG. 9.63. Schematic view of the magnetic tape transport mechanism of a magnetic sound recorder for sound motion pictures.

sound will be obtained. See Sec. 13.7. When the tape passes over the sprocket drive, variations in the motion of the film at the sprocket hole frequency are produced. These variations in the film speed must be removed at the recording point to eliminate spurious frequency modulation of the magnetic recording on the magnetic tape. Uniform speed at the recording point is provided by the filter between the sprocket drive and the recording point consisting of the inertia elements of the two rollers and the two drums and the compliance of the tape between the sprocket drive and the rollers and the drums. The drums are equipped with flywheels to provide additional inertia. Damping of the inertia and compliance system is provided by the mechanical resistance of the dash pot. The spring system maintains a tight loop for the magnetic tape. This design of mechanical system provides uniform motion at the magnetic heads. Two magnetic heads are provided, one for recording and the other for monitoring.

The film recorder of Fig. 9.59 may also be used for recording one track of magnetic tape by placing a magnetic recording head inside the drum in

[81] Hittle, C. E., *Jour. Soc. Mot. Pic. Eng.*, Vol. 58, No. 4, p. 323, 1952.
[82] Davis and Manley, *Jour. Soc. Mot. Pic. Tel. Eng.*, Vol. 62, No. 3, p. 208, 1954.

contact with overhanging portion of the tape on the drum. The magnetic head is shown dotted in Fig. 9.59.

C. *Reproducing Tape Transport*.[83]—The information on the tape in the magnetic system of sound reproduction is rerecorded on the magnetic strips of the positive film. Fig. 9.62B. The transport mechanism used in reproducing sound recorded on magnetic strips on positive film consists of a positive drive of the perforated film and a constant speed drive at the point of contact of the magnetic heads.

Fig. 9.64 Positive drive of the film is obtained by means of two sprocket drives. These two sprocket drives are geared to the positive picture drive so that constant loop of film is maintained between the sound head and the picture head. The positive drive also insures that the film speed in reproduction will be the same as in recording. The inertia drum coupled to a flywheel provides the inertia element of the filter system. The film loop tension rollers provides the compliance elements in the filter system. Damping of the loop tension rollers provides the mechanical resistance element. The filter system removes the variations

FIG. 9.64. Schematic view of the film transport mechanism of a magnetic reproducer for sound motion pictures.

introduced by the sprocket drive and provides uniform motion of the film at the magnetic heads.

9.11. Volume Limiters, Compressors, and Expanders.[84,85,86,87]— A volume compressor is a system that reduces the amplification of an amplifier when the signal being amplified is large and increases the amplification when the signal is small. Compressors are used to reduce the volume range in sound motion picture and phonograph recording, sound broadcasting, public address, and sound reinforcing systems, etc.

A volume expander is a system that increases the amplification of an amplifier when the signal is large and decreases the amplification when the signal is small. In reproduction, a volume expander is used to counteract the effect of the compressor in recording.

Volume compressors and expanders are amplifiers in which the amplification varies as a function of the general level of the signal. The elements of a compressor, limiter, or expander are shown in Fig. 9.65A. The input

[83] Phyffe and Hittle, *Jour. Soc. Mot. Pic. Eng.*, Vol. 62, No. 3, p. 215, 1954.

[84] Sinnett, C. M., *Electronics*, Vol. 8, No. 11, p. 14, 1935.

[85] Norman, N. C., *Bell Labs. Record*, Vol. 13, No. 4, p. 98, 1934.

[86] Mathes and Wright, *Bell Syst. Tech. Jour.*, Vol. 13, No. 3, p. 315, 1934.

[87] Steinberg, J. C., *Jour. Acous. Soc. Amer.*, Vol. 13, No. 2, p. 107, 1941.

signal is amplified and rectified. The rectified signal is applied to a resist-
ance condenser network. The d-c voltage across the condenser is used to
vary the bias and, as a consequence, the amplification of a push pull amplifier
employing tubes with variable transconductance. The constants of the
system can be adjusted to obtain limitation, compression or expansion.

In the limiter characteristic, shown in Fig. 9.65A, the relation between
the output and input is linear up to a certain level, beyond this point the
output remains constant regardless of the input. The limiter type is useful
for protection against a sudden overload, as, for example, in the input to a
broadcast transmitter.

In the case of the compressor characteristic, shown in Fig. 9.65A, there is
a gradual reduction in the gain with increase of the input. A reduction
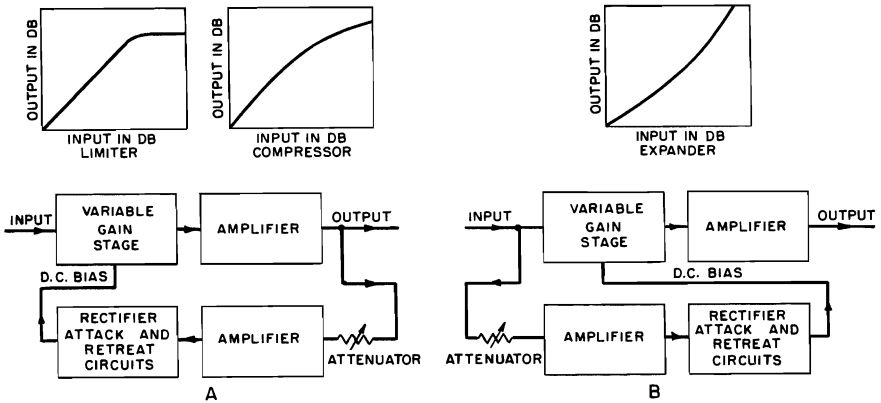
FIG. 9.65. Input vs. output characteristics of limiters, compressors, and expanders.
A. The elements of a limiter or a compressor. B. The elements of an expander.

in the volume range in radio and phonograph reproduction makes it possible
to reproduce the wide range of orchestra music in the home without excessive
top levels. It also improves the signal-to-noise ratio. It improves the
intelligibility of speech and enhances music reproduction when the ambient
noise is high, as, for example, in sound motion theater reproduction.

In the case of the expander characteristic, shown in Fig.9.65B, there is
a gradual increase in gain with increase in output. The combination of a
compressor and expander may be used to improve the signal-to-noise ratio
in sound reproduction.

The attack time for a gain reduction of 10 db, in compressors and limiters,
is of the order of a millisecond. The retreat to normal is of the order of 1
second.

9.12. Synthetic Reverberation.—The reverberation time of studios may
be changed and controlled within certain limits by varying the absorption.
The amount of control that may be obtained by varying the amount of
absorption by means of hard panels which cover the absorbing material or

other similar systems is limited.   Furthermore, the reproducing conditions may also require additional reverberation.   Where the reverberation time of reproduced sound is far below the optimum value, the reproduction may be enhanced by artificially adding reverberation.

Artificial reverberation may be added to a sound signal by means of the loudspeaker, reverberant chamber, and microphone combination shown in Fig. 9.66A.   The reverberant chamber consists of an enclosure with highly reflecting, nonparallel walls, ceiling, and floor.   The cubical content varies from 1000 to 10,000 cubic feet.   If a reduction in the reverberation time is desired, flats of absorbing material may be brought into the chamber.   A modification of the single-chamber system is the addition of a second chamber coupled to the first by means of a door.   The use of two rooms makes it possible to obtain a wide variety of reverberant effects by varying
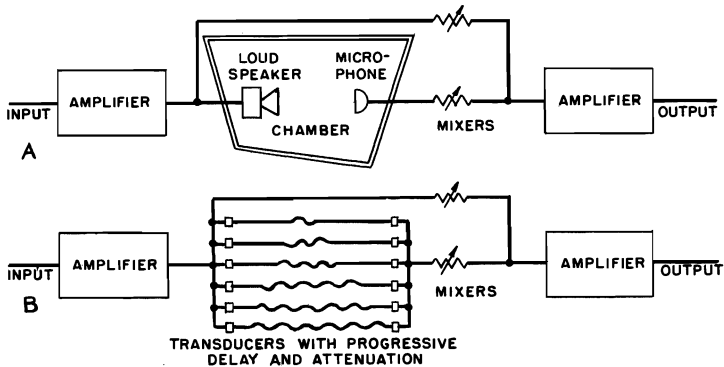
FIG. 9.66.   Schematic arrangement of systems for introducing synthetic reverberation in reproduced sound.   A. Loudspeaker, microphone, and reverberant chamber combination.   B. A system consisting of transducers with progressive delay and attenuation.

both the reverberation time of the chambers and the coupling between the chambers.   The loudspeakers, microphones, and amplifiers used for these systems should be of the highest quality.   Mixers are provided so that any ratio of the original sound to reverberant sound may be obtained.

Reverberation, in the chamber described above, consists of the multiple reflection of a large number of pencils of sound.   Each pencil of sound suffers a decrease in intensity with each reflection.   These conditions can be simulated by the system shown in Fig. 9.66B.   The amplified sound signal is passed through a number of transducers with progressive delay and attenuation.   These transducers may be a series of pipes with loudspeakers and microphones terminating the ends.   The transducer may be a recorder and a series of pickups on a phonograph record or magnetic tape[88] or phosphor wheel.[89]   The reverberation time may be varied by varying the progressive

[88] Wolf, S. K., *Proc. I.R.E.*, Vol. 27, No. 7, p. 365, 1939.
[89] Goldmark and Hendricks, *Proc. I.R.E.*, Vol. 27, No. 12, p. 747, 1939.

attenuation. Mixers are provided so that any ratio of the original sound to the reverberant sound may be obtained.

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

The first and simplest hearing aid[90] consisted of a carbon microphone, a battery, an attenuator, and a telephone receiver (Fig. 9.67A). This hearing aid gave satisfactory service where the hearing loss was about 20 db.

The hearing aid shown in Fig. 9.67B consisted of a carbon microphone, a carbon amplifier, an attenuator, a battery, and a telephone receiver. This hearing aid gave satisfactory service where the hearing loss was about 40 db.
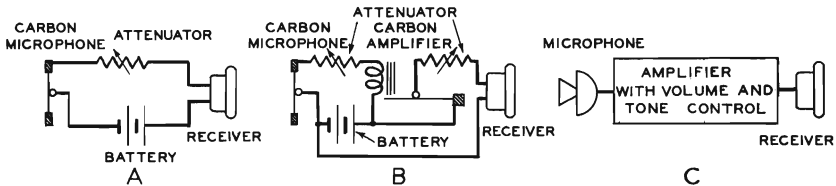
FIG. 9.67. Hearing aids. A. Simple carbon microphone hearing aid. B. Carbon microphone hearing aid with a mechanical carbon amplifier. C. Schematic diagram of a vacuum tube or transistor hearing aid.

The quality of the carbon type hearing aids was usually very poor, due to the frequency and the amplitude distortion produced by the carbon microphone and amplifier.

During the past two decades and prior to the advent of the transistor, hearing aids[91,92,93,94,95,96,97,98] employing vacuum tube amplifiers almost completely replaced the carbon types. This has been due to the development of small low-current drain vacuum tubes and small high-efficiency batteries. The quality is far superior to that of the carbon type. Furthermore, suitable compensation circuits may be introduced to complement the ear characteristics. The schematic arrangement of the components of a vacuum tube hearing aid is shown in Fig. 9.67C. The microphone used in hearing aids today is a diaphragm crystal or ceramic type similar to that described in Sec. 8.2C2 and depicted in Fig. 8.9.

[90] Tuffnell, W. L., *Bell Labs. Record*, Vol. 18, No. 1, p. 8, 1939.
[91] Ramanow, F. F., *Jour. Acous. Soc. Amer.*, Vol. 13, No. 3, p. 295, 1942.
[92] Sabine, P. E., *Jour. Acous. Soc. Amer.*, Vol. 16, No. 1, p. 38, 1944.
[93] Carlisle and Mundel, *Jour. Acous. Soc. Amer.*, Vol. 16, No. 1, p. 45, 1944.
[94] Grossman and Molloy, *Jour. Acous. Soc. Amer.*, Vol. 16, No. 1, p. 52, 1944.
[95] Hanson, W. W., *Jour. Acous. Soc. Amer.*, Vol. 16, No. 1, p. 60, 1944.
[96] LeBel, C. J., *Jour. Acous. Soc. Amer.*, Vol. 16, No. 1, p. 63, 1944.
[97] Watson, N. A., *Jour. Acous. Soc. Amer.*, Vol. 16, No. 3, p. 194, 1945.
[98] Strommen, E., *Jour. Acous. Soc. Amer.*, Vol. 15, No. 4, p. 211, 1944.

Within the past three years, hearing aid amplifiers employing transistors[99] have been developed and commercialized on a wide scale. The advantage of transistors over vacuum tubes is that only one low-voltage battery is required as contrasted to separate filament and plate batteries for vacuum tube systems. The total power required from the battery in a transistor amplifier for hearing aids is a small fraction of the battery power required for a vacuum tube amplifier for hearing aids. As a result the cost of operating a transistor hearing aid is very small compared to a vacuum tube hearing aid. Furthermore, the weight and size of transistor amplifiers with the battery are very much less than those of a corresponding vacuum tube amplifier and battery
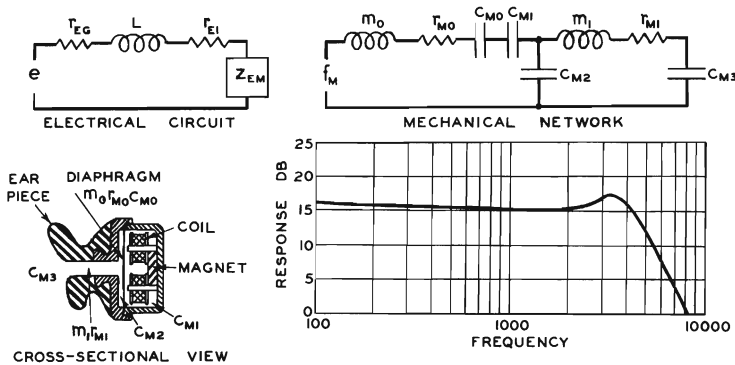
FIG. 9.68. Cross-sectional view, mechanical network, electrical circuit, and response frequency characteristic of an insert-type telephone receiver. In the mechanical network, f_M = the mechanical driving force. m_0, r_{M0}, and C_{M0} = the mass, mechanical resistance, and compliance of the diaphragm. C_{M1} = the compliance due to the air in the case. C_{M2} = the compliance of the air space between the diaphragm and the cover. m_1 and r_{M1} = the mass and mechanical resistance of the tube. C_{M3} = the compliance of the ear cavity. In the electrical circuit, z_{EM} = the electrical motional impedance. L and r_{E1} = the damped inductance and electrical resistance of the coil. r_{EG} = the electrical resistance of the coils. e = the voltage of the electrical generator. The graph depicts the pressure response frequency characteristic.

complement. Since the input electrical impedance of a transistor is of the order of a few hundred ohms, magnetic microphones are used in transistor-type hearing aids.

Two types of receivers are used—namely, the air conduction type and the bone conduction type.

A cross-sectional view, the electrical circuit, the mechanical network, and response frequency characteristic of the air conduction insert type telephone receiver is shown in Fig. 9.68. A molded plug fits the ear cavity and holds the receiver in place. Under these conditions the leak at the ear

[99] Webster, Stanley K., *Trans. I.R.E.*, Prof. Group Audio, Vol. AU-2, No. 2, p. 65, 1954.

is very small. Therefore, good response is obtained at the low frequencies. The action of the system is essentially the same as that of a bipolar telephone receiver considered in Sec. 9.2A and need not be repeated here.

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 nerves, is normal (see Sec. 12.2). Under these conditions, sound may be transmitted through the bones of the head to the inner ear by means of a bone conduction receiver.[100],[101] Usually the face of the bone conduction receiver is placed against the mastoid bone behind the ear. A cross-sectional view, the electrical circuit, mechanical network, and response frequency characteristic of a bone conduction receiver is shown in Fig. 9.69.
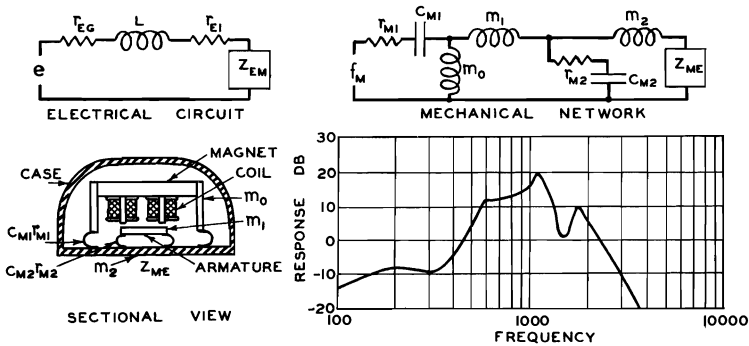
FIG. 9.69. Cross-sectional view, mechanical network, electrical circuit, and response frequency characteristic of a bone conduction receiver. In the mechanical network, $m_0 =$ the mass of the coil and magnetic structure. $m_1 =$ the mass of the armature. C_{M1} and $r_{M1} =$ the compliance and the mechanical resistance connecting the coil and magnetic structure to the case. C_{M2} and $r_{M2} =$ the compliance and mechanical resistance connecting the armature to the case. $m_2 =$ the mass of the case. $z_{ME} =$ the mechanical impedance of the mastoid bone. $f_M =$ the mechanical driving force. In the electrical circuit, $z_{EM} =$ the electrical motional impedance. L and $r_{E1} =$ the damped inductance and electrical resistance. $r_{EG} =$ the electrical resistance of the generator. $e =$ the voltage of the electrical generator. The graph depicts the force developed on an artificial mastoid.

By means of the multiple resonant system it is possible to deliver a large force to the relatively high mechanical impedance, z_{ME}, of the mastoid bone. The response frequency characteristic is quite good considering the difficult conditions under which the bone conduction receiver operates.

A modification of the insert hearing aid telephone receiver is shown in Fig. 9.70. A small plastic tube connects an ear insert earpiece and the telephone receiver. There is some attenuation and frequency discrimina-

[100] Hawley, M. S., Bell Labs. Record, Vol. 18, No. 1, p. 12, 1939.
[101] Hector, Pearson, Dean, and Carlisle, Jour. Acous. Soc. Amer., Vol. 25, No. 6, p. 1195, 1953.

tion of sound transmitted by the small plastic tube.   See Sec. 5.32.   However, this loss can be overcome by compensation in the amplifier.   The advantage of the system of Fig. 9.70 is that it is somewhat less obtrusive than the conventional system in that the earpiece is smaller and the telephone receiver can be placed under the shirt or dress.
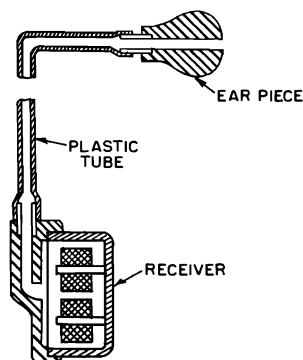
FIG. 9.70.   Insert-type hearing aid telephone receiver with tube transmission line.

9.14. Sirens.[102]—The simplest siren consists of a revolving disk perforated with a ring of equally spaced holes which interrupt a jet of air from a tube placed close to one side of the disk.   The fundamental frequency of the successive puffs of air issuing through the holes is equal to the product of the number of holes and the revolutions per second of the disk.   The wave form, of course, depends upon the shape of the holes in the disk and the shape of the projection of the air tube upon the disk.   The pressure of the air supply in large sirens is usually very high, of the order of 100 pounds per square inch.   In the smaller sirens the air pressure is supplied by a single-stage centrifugal pump and the supply pressure is of the order of a pound per square inch.   Small sirens are used by police cars, ambulances, and fire engines for signalling the approach of these vehicles.   Large power sirens are used on firehouses, lighthouses, and lightships.

A high-power siren[103] has been developed in which the blower is driven by a 95 horsepower automobile engine.   The air stream represents about 38 kilowatts.   The flow of air is interrupted by a rotary valve at a rate of 440 cycles per second and then passes into a horn.   The use of a horn provides a certain amount of directionality and contributes to the high efficiency of the siren.   The sound output from the horn is about 25 kilowatts in the fundamental.

9.15. Seismic Detectors.[104]—The variation of the velocity of sound in the various strata comprising the earth's crust forms the basis of geophysical

---

[102] Wood, "A Textbook of Sound," The Macmillan Company, New York, N.Y., 1930.
[103] Jones, Clark, *Jour. Acous. Soc. Amer.*, Vol. 18, No. 2, p. 371, 1946.
[104] Silverman, Daniel., *Jour. A.I.E.E.*, Vol. 58, No. 11, 455, 1939.

investigations in prospecting for oil.   The detonation of a charge of dynamite creates an acoustic wave which is reflected from the various strata of the earth's surface.   These reflected waves are picked up by microphones connected to recording oscillographs and located in strategic positions on the earth's surface.   From the geometrical configuration of the apparatus, the oscillograph record, and the velocity of sound in various types of strata, the conformation of the various strata may be determined.   Oil pools are located in curved strata termed by geologists as anticlines.   The presence of anticlines may be determined from seismic measurements.

Magnetic, carbon, crystal, condenser, and dynamic microphones have been used for detectors.   The large amplitude frequency components of seismic waves are usually confined to the lower frequencies.   Therefore, the response of the microphone is confined to the range below 100 cycles. For these applications a magnetic microphone has been found to be very satisfactory.   The armature is usually made massive and the stiffness small in order to obtain high sensitivity in the low-frequency range.   The microphone is placed directly upon the earth's surface.   The microphone proper then vibrates with the earth's surface.   The massive armature opposes any change from its position of rest.   As a consequence, there is relative motion between the armature and the microphone proper which results in the production of a voltage corresponding to the vibrations of the earth's surface.   By suitable orientation, the microphone can be made responsive to only vertical vibrations.   As a consequence, the wave transmitted directly through the earth is not reproduced.

9.16. Stethoscopes.[105,106,107]—The ordinary acoustical stethoscope is one of the most useful instruments which the physician uses in mediate auscultation.   By means of the stethoscope the physician is able to study sounds produced within the heart, lungs, stomach, intestines, or other portions of the body, and to determine whether normal or abnormal conditions exist as indicated by normal or abnormal sounds.   The most important sounds are normal heart sounds, heart murmurs, breathing sounds, respiratory rales or rattles, and peristaltic squeaks or groans.   Obviously, it is the structure of the sound, which involves the intensity, the fundamental frequency, the harmonic components, the duration, and the growth and decay, that makes it possible to diagnose normal or abnormal conditions by auscultation.

Since diagnosis is based on the structure of the sounds, it is very important that the stethoscope should not distort the sound by discrimination against certain frequency bands or by attenuation of the sound.   The sounds of the body range from about 40 cycles to 4000 cycles.   Fig. 9.71 shows the frequency bands of some of the most common sounds.   The fundamentals are shown as dark areas and the harmonics or overtones as cross-hatched areas. The fundamental of the systolic sound ranges from 40 to 80 cycles.   There

[105] Rappaport and Sprague, *Amer. Heart Jour.*, Vol. 21, p. 257, 1941.
[106] Frederick and Dodge, *Bell Syst. Tech. Jour.*, Vol. 3, No. 4, p. 531, 1924.
[107] Singer, C., *Electronics*, Vol. 38, No. 6, p. 66, 1939.

are some lower components but from the standpoint of ear characteristic these are very weak (see Sec. 12.6). The overtones are scattered over the remainder of the frequency band up to 4000 cycles and above. Above 4000 cycles most of the sounds in the body are so weak that they are masked by the ambient random noises generated in the body. The fundamental diastolic sounds range from 60 to 100 cycles. The overtones are scattered over the remainder of the frequency band up to 4000 cycles. The fundamental sounds of systolic and diastolic murmurs range from 300 to 800 cycles.
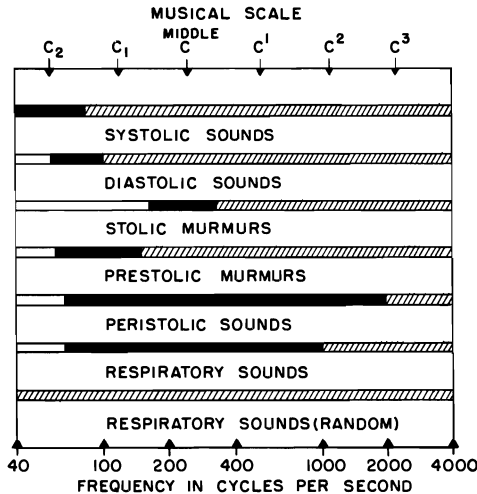
FIG. 9.71. The frequency ranges of the sounds generated in the body. The frequency ranges of the fundamental frequencies are shown as solid lines. The frequency ranges of the harmonics and overtones are shown as cross-hatched lines.

The overtones in certain cases can be observed up to 2000 or 3000 cycles. Prestolic murmurs usually range from 60 to 200 cycles. The overtones range up to about 1000 cycles. Above this frequency the overtones are masked by the body sounds. The fundamentals of peristaltic sounds have a tremendous range in both frequency and intensity. Fundamentals up to 2000 cycles are quite common. The overtones in the case of very intense sounds extend beyond 4000 cycles. The fundamental frequency of respiratory squeaks, rales, crackles, and groans ranges from 60 cycles to 1000 cycles. Respiratory sounds such as wheezes and the rushing of air are of a random nature and do not possess a true fundamental. The components of these sounds are scattered over the entire audible spectrum.

From Fig. 9.71 it is quite evident that, in order to obtain the maximum intelligence from the stethoscope all frequencies over the range from 40 to 4000 cycles should be transmitted without attenuation or discrimination.

Most acoustical and mechanical vibrating systems introduce distortion in the form of discrimination against certain frequency bands. Extreme distortion may alter the sound beyond recognition.

The two most common stethoscopes in use today are the open bell and diaphragm types shown in Fig. 9.72A and B. The response frequency characteristic of the open bell type is smoother and covers a wider frequency range than the diaphragm type. However, the tuned diaphragm type
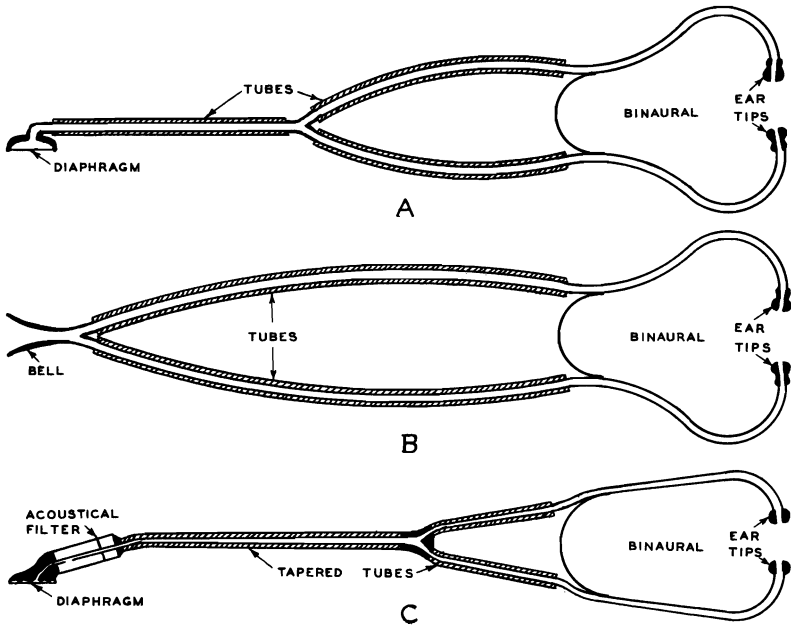
FIG. 9.72. Sectional views of stethoscopes. A. Diaphragm type. B. Open bell type. C. Wide range selective type.

delivers greater output in the frequency range from 250 to 1500 cycles. The open bell has better low-frequency response but the general output level in the mid-frequency range is lower than the diaphragm type.

There are two reasons for the use of a diaphragm instead of an open bell—namely, to exclude or attenuate external noises, and to eliminate leakage between the body and the stethoscope. The open bell stethoscope actually amplifies air-borne noises in the manner of the ear trumpet. If the effective slit between the body and the bell of the open bell stethoscope is just a small fraction of a thousandth of an inch, the low-frequency response is attenuated due to this leakage. If the bell is pressed against the body so this leak is effectively eliminated, the body stiffness represented in the acoustic impedance of the body is increased with a resultant attenuation of low frequencies.

In the existing diaphragm type stethoscopes the investigators have found that it is necessary to use a resonant diaphragm in order to obtain good output They have placed these resonances in the mid-frequency range where the ear is quite sensitive. As a consequence the stiffness of the diaphragm is quite high and the result is very high attenuation of the low-frequency response.

A wide-range acoustical stethoscope[108,109] is shown in Fig. 9.72C. The chest piece of radical design consists of a light polythene diaphragm supported by a multipyramid resilient back plate. This structure provides an efficient coupling means to the high acoustical impedance of the body. The adequate resilience of the chest piece insures uniform response to low tones. The light-weight diaphragm coupled directly to the body makes it possible to obtain output beyond 4000 cycles. The acoustical impedance of the chest piece is matched to the acoustical impedance of the tube or line at the input end. The relatively high acoustical impedance at the input end of the line is matched to the relatively low acoustical impedance of the ear by the use of a tapered tube or line. The sensitivities in the low- and high-frequency ranges are much greater than those of existing stethoscopes due to the matching of acoustical impedances. The high-frequency response is maintained to 4000 cycles while most existing stethoscopes cut off at 1500 cycles. There are certain instances in which the entire frequency range is not desired. This is particularly true when the particular sounds in question are confined to the low-, high-, or mid-frequency range. For example, in listening to high-frequency prestolic murmurs, peristaltic, and respiratory sounds, it may be desirable to eliminate the low frequencies. In other instances, it may be desirable to attenuate the high-frequency range. Therefore, to increase the usefulness of the stethoscope, an acoustical filter has been added in which it is possible to attenuate either the low- or high-frequency ranges, or both. The acoustical filter provides a system in which frequency discrimination may be introduced at will, and thereby increases the usefulness of the stethoscope by classification of the characteristic sounds in the body into frequency bands.

The electrical stethoscope consists of the combination of a microphone, amplifier, and telephone receivers. In one type the pickup device consists of a bell-shaped horn, coupled to the microphone diaphragm. The coupling system is similar to that of Fig. 8.1. Condenser, magnetic, and crystal type transducers have been used in the microphone for these applications. The amplifier is equipped with low- and high-frequency tone controls for attenuating the response in either or both the high- and low-frequency ranges. The addition of a recording system similar to the electrocardiograph may be used to obtain an oscillographic record depicting the sounds in the body. Since the output of the electrical stethoscope is greater than that of the acoustical stethoscope, noises generated by the clothing, movement of the headpiece, etc., cause considerably more interference than in the

[108] Olson, H. F., *Electronics*, Vol. 16, No. 8, p. 184, 1943.
[109] Olson, H. F., U. S. Patents 2,363,686 and 2,389,868.

acoustical stethoscope. This is due to the fact that most of these noises in the acoustical stethoscope fall below the threshold of hearing.

9.17. Ear Defenders.[110]—Ear defender is a term used to designate a device which introduces attenuation of sound between a point outside the head and the eardrum. There are two types—namely, the cushion type and the insert type. The cushion type is similar to a pair of headphones with soft cushion ear pads. The cushion type is heavy, cumbersome, and uncomfortable and for that reason it has not been used to any appreciable extent. The insert type is some form of plug which is pushed into the ear
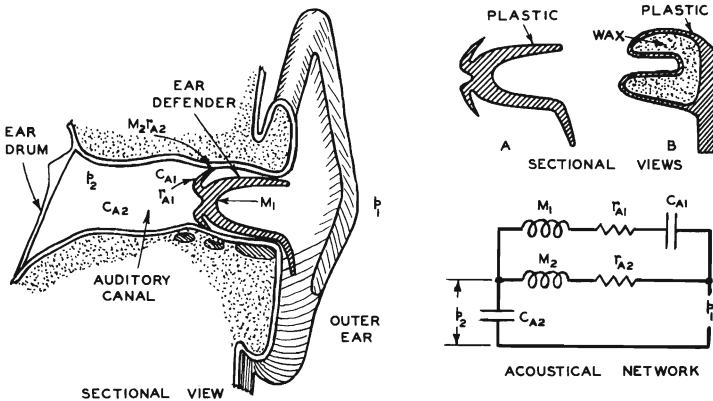
FIG. 9.73. A. Sectional view of an ear defender in the ear canal and the acoustical network of the system. In the acoustical network, $M_1 =$ the inertance due to the mass of the ear defender. C_{A1} and $r_{A1} =$ the effective acoustical capacitance and acoustical resistance of the ear defender with respect to the wall of the ear canal. M_2 and $r_{A2} =$ the inertance and acoustical resistance of the leak between the ear defender and the wall of the ear canal. $C_{A2} =$ the acoustical capacitance of the entrapped volume of the ear canal. $p_1 =$ the sound pressure outside the ear. $p_2 =$ the sound pressure in the ear canal. The separate sectional views show two different designs of ear defenders with one and two sealing flanges, respectively. B. Wax filled ear defender.

canal. One form, which was used extensively a number of years ago, consisted of a wad of cotton. The attenuation of a wad of cotton decreases with decrease of the frequency. The attenuation below 500 cycles is quite small. In order to obtain high attenuation at the low frequencies, the seal between the defender and the ear canal must be practically airtight, because a very minute hole will reduce the attenuation to a negligible amount. This fact can be deduced from a consideration of the acoustic network of the ear defender of Fig. 9.73A. A successful insert type of ear defender must be made of suitable material combined with a shape which will provide adequate attenuation, comfort, easy insertion, and positive retention.

[110] Watson and Knudsen, *Jour. Acous. Soc. Amer.*, Vol. 15, No. 3, p. 153, 1944.

Ear defenders have been developed which satisfy the above requirements. The most successful ear defenders have been made of synthetic rubbers or soft plastics, because these materials remain resilient over long periods of time and are resistant to ear wax. The shape which appears to be most successful is a skirt closed at the top and equipped with one or more thin flounces which rest against the ear canal and thereby provide the seal (Fig. 9.73A.). A tab, fastened at the bottom of the skirt, is used for inserting or removing the defender. A good ear defender will introduce an attenuation of between 30 to 35 db over the frequency range from 60 to 8000 cycles.

Another design[111] of the ear defender employs a plastic case with soft elastic walls and a viscous core of malleable wax. Fig. 9.73B. This design is more comfortable because the body heat softens the wax with the result that the defender corresponds to the ear canal in which it is placed without distortion of the ear canal. Such distortion leads to discomfort.

9.18. Electronic Sound and Vibration Reducers and Absorbers.— Existing systems for the absorption of sound and the control of vibrations are all of the passive type. Recently, active systems have been developed for the control of sound, reverberation, and vibration. These systems are in the form of combinations of electronic elements. It is the purpose of this section to describe electronic sound and vibration reducers and absorbers.

A. *Free-Field Zone-Type Sound Reducer.*[112]—The free-field, zone-type sound reducer consists of a microphone, amplifier, and loudspeaker connected in inverse fashion so as to reduce the sound pressure of any incident sound wave in the vicinity of the microphone-loudspeaker combination. A sectional view, schematic electrical diagram, and acoustical circuit of an electronic sound reducer are shown in Fig. 9.74. The system is connected and equalized for response and phase with respect to frequency, so that the sound pressure is reduced at the microphone. The driving pressure p_2, of Fig. 9.74, is given by

$$p_2 = \frac{Bli}{S} \qquad\qquad 9.28$$

where B = flux density in the air gap of the loudspeaker, in gausses,

$\quad\ l$ = length of the conductor of the voice coil, in centimeters,

$\quad\ i$ = current in the coil, in abamperes, and

$\quad\ S$ = area of the cone, in square centimeters.

The amplitude and phase relations of the sound pressures p_1 and p_2 are selected so as to make the sound pressure p_3 as small as possible. Under these conditions the operation of the system is a sound pressure reducer. The amount of sound pressure reduction is a function of the distance between the microphone and loudspeaker, the wavelength of the sound wave, the phase relation in the electronic system, and the distance from the microphone-loudspeaker combination. Typical sound reduction frequency

[111] Zwilocki, J., *Jour. Acous. Soc. Amer.*, Vol. 27, No. 3, p. 460, 1955.
[112] Olson and May, *Jour. Acous. Soc. Amer.*, Vol. 25, No. 6, p. 1130, 1953.

SECTIONAL VIEW

ACOUSTICAL NETWORK
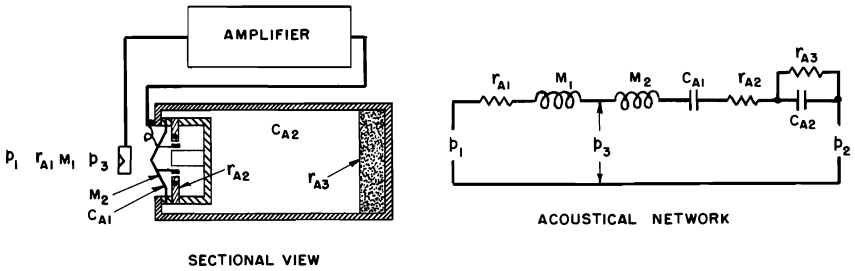
FIG. 9.74. Sectional view, schematic electrical diagram, and acoustical network of an electronic sound absorber. p_1 = the sound pressure in free space. M_1 = the inertance of the air load. r_{A1} = the acoustical resistance of the air load. M_2 = the inertance of the cone and voice coil of the loudspeaker. C_{A1} = the acoustical capacitance of the suspension system of the cone. r_{A2} = the acoustical resistance of the cloth over the apertures in the back plate. C_{A2} = the acoustical capacitance of the volume of the cabinet. r_{A3} = the acoustical resistance of the sound absorbing material in the cabinet. p_2 = the driving sound pressure in the loudspeaker. p_3 = the sound pressure at the microphone.

characteristics for various distances from the reducer are shown in Fig. 9.75. These characteristics show that the electronic sound reducer may be used to reduce undesired sounds over a limited zone of operation.

One application for the electronic noise reducer is in the form of a noise reducer in airplanes and automobiles where the noise level is relatively high in the low-frequency range. With the practical use of conventional sound absorbing materials in automobiles and airplanes, the reduction in noise level in this frequency region is relatively small. For these applications, the noise reducer may be installed on the back of the seat. There are also many other applications for the zone-type noise reducer. Other applica- tions include the reduction in noise from machines, ducts, etc.
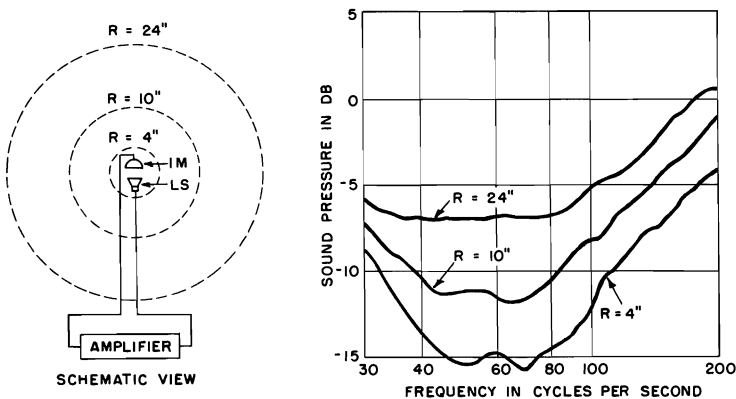
SCHEMATIC VIEW

FIG. 9.75. Schematic view of an electronic sound reducer and sound pressure reduction frequency characteristics for distances of 4, 10, and 24 inches from the microphone-loudspeaker combination.

B. *Free-Field Electronic Sound Absorber.*[113]—The free-field electronic sound absorber consists of a microphone, amplifier, loudspeaker, and acoustical resistance. Fig. 9.76. The microphone, amplifier, and loudspeaker are connected in an inverse fashion so as to provide a low acoustical impedance termination for the acoustical resistance. The principal application for the electronic sound absorber is in the low-frequency region where it is difficult to obtain high absorption due to the practical difficulty of providing a low acoustical impedance termination for passive or inactive acoustical materials. Since the application for the electronic sound absorber is in the low-frequency
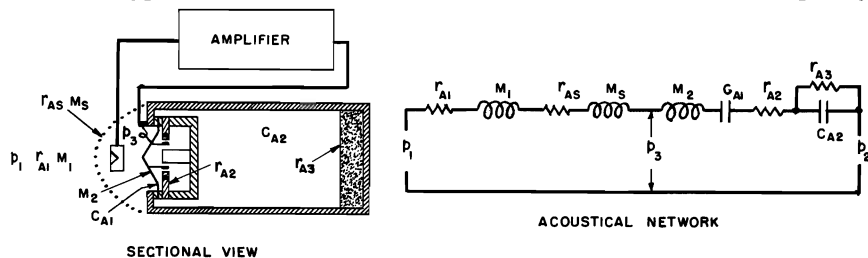
FIG. 9.76. Sectional view, schematic electrical diagram and acoustical network of an electronic sound absorber. p_1 = the sound pressure in free space. M_1 = the inertance of the air load. r_{A1} = the acoustical resistance of the air load. M_2 = the inertance of the cone and voice coil of the loudspeaker. r_{AS} = the acoustical resistance of the screen covering the microphone and cone. M_S = the inertance of the screen. C_{A1} = the acoustical capacitance of the suspension system of the cone. r_{A2} = the acoustical resistance of cloth over the apertures in the back plate. C_{A2} = the acoustical capacitance of the volume of the cabinet. p_2 = the driving sound pressure in the loudspeaker. p_3 = the sound pressure at the microphone. r_{A3} = the acoustical resistance of the sound absorbing material in the cabinet.

region, the system is operated as a diffraction absorber. The absorbing efficiency of a diffraction sound absorber with the appropriate acoustical resistance may be several hundred per cent. Therefore, for absorption of sound in the low-frequency range, the spacing between the electronic sound absorbers can be relatively large. See Sec. 9.2E and Fig. 11.8.

C. *Headphone-Type Noise Reducer.*—An application of the point-type sound and voice reducer is for the telephone receiver as shown in Fig. 9.77. The acoustic shielding of the earcap provides some reduction in the sound from the outside. The electronic sound reducer provides additional reduction in the sound pressure in the small enclosed cavity of the headphone. The telephone receiver consists of two separate diaphragm and voice coil assemblies. The useful signal is applied to the voice coil attached to the inner diaphragm and provides the driving force for reproducing the useful information. The other voice coil is connected to the output of the amplifier. The sensing microphone is connected to the input of the amplifier. This feedback system provides the noise reduction. The action is similar to the sound reducer of Sec. 9.18A.

[113] Olson, H. F., *Jour. Acous. Soc. Amer.*, Vol. 25, No. 6, p. 1130, 1953.
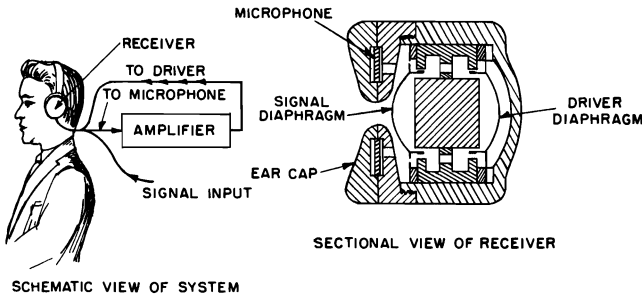
FIG. 9.77. Electronic noise reducing headset. The schematic view shows the elements of the system. The sectional view shows the elements of the headphone.

D. *Electronic Vibration Reducer.*—Reduction in the transmission of sound through structures of solid materials is usually accomplished by the addition of mass or by a compliant isolating system. The latter means is usually preferred because the addition of mass is costly and for most applications impractical. The idea is to insert an element which has a low mechanical impedance and thereby provide a shunt for the vibrations. An electronic system may be used to provide the low mechanical impedance and thereby
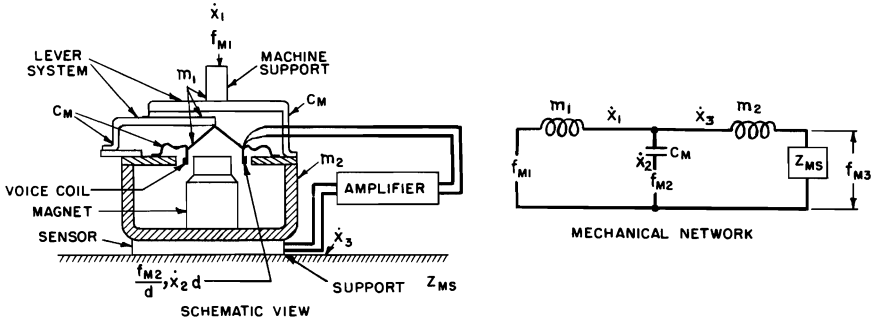
FIG. 9.78. Sectional view, schematic electrical diagram, and mechanical network of an electronic vibration reducer. $\dot{x}_1$ = the input vibrational velocity. f_{M1} = the input driving force. m_1 = the mass of the lever system and voice coil. f_{M2} = the driving force generated in the voice coil. C_M = the compliance of the voice coil and the centering spider. $\dot{x}_2$ = the vibrational velocity of the voice coil. m_2 = the mass of the magnetic structure. z_{MS} = the mechanical impedance of the support. f_{M3} = the driving force at the support. $\dot{x}_3$ = the vibrational velocity of the support.

control and isolate the vibrations. In most problems involving the control of vibrations, the amplitudes are relatively small and the mechanical impedance relatively large. Under these conditions, piezoelectric transducers may be used. For example, an electronic vibration reducer may consist of a piezoelectric driver and sensor with a suitable amplifier.

A dynamic system may also be used as a vibration reducer. Since the amplitude of the vibrations produced by most machines is small compared

to the amplitude which may be obtained from a dynamic unit, a transformer in the form of a lever may be used between the dynamic unit and the machine. A schematic view of a dynamic vibration reducer is shown in Fig. 9.78. The sensor is a piezoelectric transducer. The amplifier is similar to the amplifier used for the electronic noise reducer. The performance of the system may be obtained from the mechanical network. The mechanical impedances in the mechanical network are all referred to the input and output terminal impedances. Cognizance must be taken of the lever system, which is in effect a mechanical transformer, in referring the force f_{M2} and the masses and compliances to the input and terminal mechanical impedances. The problem in isolating the machine from the supports is to adjust the phase and magnitude of the force f_{M2} so that the resultant force f_{M3} developed in the support will be a minimum. The velocity $\dot{x}_3$ in the support may be expressed as follows:

$$\dot{x}_3 = \frac{f_{M1}z_{M2} - f_{M2}z_{M1}}{z_{M1}z_{M2} + z_{M1}z_{M3} + z_{M2}z_{M3}} \qquad 9.29$$

where z_{M1} = mechanical impedance of the machine,

z_{M2} = mechanical impedance of the driver and sensor, and

z_{M3} = mechanical impedance of the support.

It may be that a part of the mechanical impedance of the driver and sensor may be included in the mechanical impedance of the machine and/or the mechanical impedance of the support.

From a consideration of equation 9.29, it will be seen that the magnitude of the velocity $\dot{x}_3$ can be reduced by the application of the force f_{M2} in the proper magnitude and phase. For example, $\dot{x}_3 = 0$ when

$$f_{M1}z_{M2} = f_{M2}z_{M1} \qquad 9.30$$

Under these conditions, f_{M3} is also zero. That is to say, no vibrations are produced in the support. The machine is perfectly isolated from the support.

Another problem is to reduce the vibration of the machine without regard to the vibration transmitted to the support. The velocity $\dot{x}_1$ of the machine may be expressed as follows:

$$\dot{x}_1 = \frac{f_{M1}(z_{M2} + z_{M3}) - f_{M2}z_{M3}}{z_{M1}z_{M2} + z_{M1}z_{M3} + z_{M2}z_{M3}} \qquad 9.31$$

From a consideration of equation 9.31, it will be seen that the magnitude of the velocity $\dot{x}_1$ can be reduced by the application of the force f_{M2} in the proper magnitude and phase. For example, the velocity of the machine $\dot{x}_1$ will be zero if

$$f_{M1}(z_{M2} + z_{M3}) = f_{M2}z_{M3} \qquad 9.32$$

If $\dot{x}_1 = 0$, there will be no motion of the machine. There will, however, under these conditions be a larger velocity $\dot{x}_3$.

There are many applications for an electronic vibration reducer which decreases the coupling between an offending vibration producer and a

terminal location in which vibrations are undesirable. Most applications for an electronic vibration reducer will involve the isolation of the vibrations produced by machine from the foundation of the machine.

9.19. Noise Reduction Circuits.—Noise is one of the most disagreeable forms of distortion that occurs in sound reproducing systems. Therefore, any means which reduces or mitigates noise is extremely useful and important.

There are many ways of increasing the signal-to-noise ratio thereby reducing the deleterious effects of noise. A few of the systems that have been used may be listed: (1) a system in which the high-frequency response is attenuated, (2) a system with suitable precompensation and postcompensation so that the high-frequency response is accentuated in recording or transmitting and attenuated in reproducing or receiving, (3) a system using two channels—one channel is used to carry the signal and the other channel to control the amplitude of the signal in reproduction, (4) a system in which the high-frequency cutoff of the reproducing system is automatically made a function of the general level of the signal, and (5) a system employing a nonlinear element arranged so that signals below a certain threshold will be attenuated.

Attenuating the high-frequency response is the most common method for the reduction of noise.

The use of two channels, in which one is used as a volume control, has been applied in some special cases but, in general, is impractical because two channels are not available in conventional reproducing systems.

Systems in which the high-frequency response is accentuated in recording or transmitting and attenuated in reproducing or receiving are used in phonograph and sound motion-picture reproduction as well as frequency-modulation radio broadcasting. This procedure is quite effective, but in some systems it is also necessary to reduce the frequency transmission band in order to obtain a substantial reduction in noise.

A block diagram and response frequency characteristics of a system[114] in which the high-frequency response is made a function of the high-frequency level is shown in Fig. 9.79. This system is based on the fact that the masking of the noise by the signal is a function of the level of the signal. Therefore, in order to maintain masking of the noise by the signal for different signal levels, the frequency range of reproduction is made a function of the signal level. This is accomplished by means of an electronically controlled low-pass filter. The response frequency characteristics of the system are shown for different levels in Fig. 9.79. In the system shown in Fig. 9.79, high-frequency cutoff is employed. If there is noise in the low-frequency range, an electronically controlled high-pass filter may be used. If there is noise in both the high- and low-frequency ranges, an electronically controlled band filter may be used.

A block diagram, the amplitude characteristic, and the response characteristics of a threshold-type noise reducer[115] is shown in Fig. 9.80. Band-pass

[114] Scott, H. H., *Electronics*, Vol. 20, No. 12, p. 96, 1947.
[115] Olson, H. F., *Electronics*, Vol. 20, No. 12, p. 119, 1947.
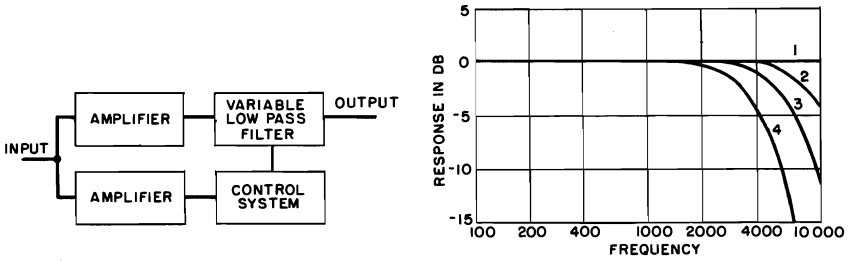
FIG. 9.79. Schematic diagram and response frequency characteristics of a noise-reducing system employing an electronically controlled low-pass filter. The response frequency characteristics labeled 1, 2, 3, and 4 depict the response characteristics with decreasing signal levels.

filters which pass frequencies over a range of an octave are used at the input and output of the nonlinear element. The amplitude characteristic of the nonlinear element is illustrated in Fig. 9.80. As will be described later, this amplitude characteristic can be obtained with a properly biased diode vacuum tube or crystal rectifier. By using this method the system will exhibit high attenuation to signals of small amplitudes.
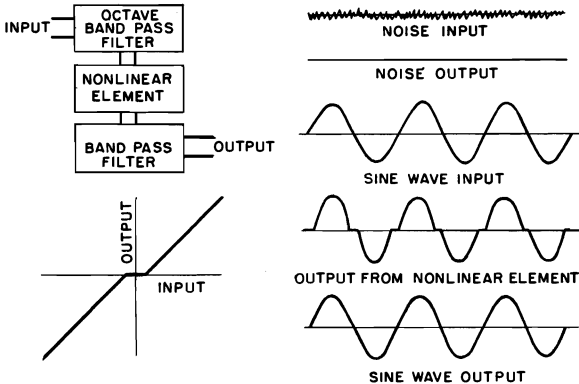
FIG. 9.80. Schematic diagram, input vs. output characteristics and the response to noise and a sine wave of a threshold noise reducer.

The response of the noise-reduction system to noise and a sine wave is depicted in Fig. 9.80. If the amplitude of the noise is kept below the response range of the noise-reduction system, the noise will not be reproduced. The response of the noise-reducing system to a sine wave signal is also shown. The output of the nonlinear element contains the fundamental, harmonics, and subharmonics of the fundamental. However, since the pass band of the input and output band-pass filters is an octave, the harmonics and subharmonics will not be transmitted by the system. The output wave,

then, is a sine wave of the same frequency as the input sine wave.   If two sine waves of different frequencies are impressed upon the system, the two frequencies must lie within the pass band octave in order to be admitted by the input band-pass filter.   The output of the nonlinear element contains harmonics and subharmonics of the two fundamental frequencies, but these are rejected by the output band-pass filter.   The output of the nonlinear element also contains the sum of the two frequencies and the difference of the two frequencies.   Since the input is confined to an octave, the band-pass output filter will reject the sum and difference frequencies.

A system with an upper cutoff of 12,000 cycles and three channels of noise reduction is depicted in the block diagram of Fig. 9.81.   This system
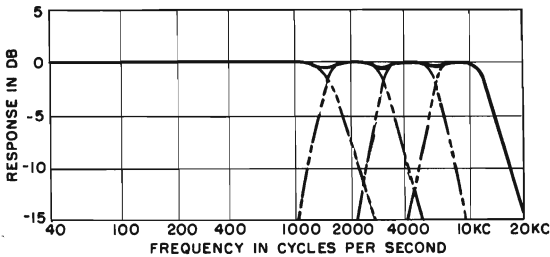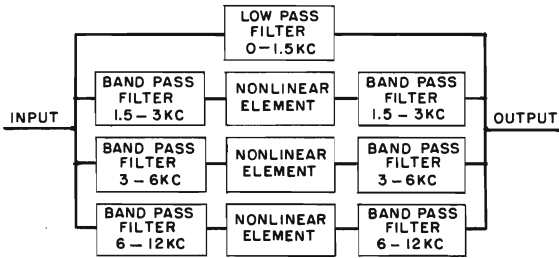
FIG. 9.81.   Schematic diagram and response frequency characteristics of a three-channel threshold noise reducer.

uses the nonlinear elements of Fig. 9.80.   Each nonlinear system is equipped with a separate bias control so that the noise can be reduced in each band without discrimination against the useful signal.   The response-frequency characteristic of the separate channels and the over-all response is shown in Fig. 9.81.   Conventional band-pass filters are used to confine the response to octave bands.   An amplifier overcomes the loss in the filters and nonlinear elements.   Noise reduction of up to 20 db can be obtained in each of the channels.

<p style="text-align:center">*10*</p>

MEASUREMENTS

10.1. Introduction.—The rapid progress made in acoustics during the past three decades has resulted in a corresponding advance in acoustical measurements.[1] In applied acoustics, as in any applied science, theoretical analysis and analytical developments are substantiated by experimental verifications. In view of the importance of acoustical measurements, it seems logical to devote a portion of this book to this phase of acoustics. It is the purpose of this chapter to consider the testing of microphones, loudspeakers and telephone receivers together with fundamental acoustical measurements.

10.2. Calibration of Microphones.[2,3,4]—A number of different measurements are required to determine the performance of a microphone. The most important characteristics which depict the performance of a microphone are as follows:

1. Response frequency characteristic
2. Directional characteristic
3. Nonlinear distortion characteristic
4. Phase distortion characteristic
5. Transient response characteristic
6. Electrical impedance characteristic

In addition to the above characteristics are such factors as the effect of temperature, humidity, and changes in atmospheric pressure upon the performance of the microphone. Carbon microphones exhibit characteristics peculiar to granular contacts such as carbon noise, packing, and breathing.

A. *Response Frequency Characteristic.*—1. *Pressure Response.*—The pressure response frequency characteristic of a microphone is the ratio e/p as a

[1] Beranek, L. L., "Acoustic Measurements," John Wiley and Sons, New York, N.Y., 1949.

[2] American Standards Association Sectional Committee z-24, Report on, Calibration of Microphones, *Jour. Acous. Soc. Amer.*, Vol. 7, No. 4, p. 330, 1936. Also American Standards Association, z-24.4, 1938.

[3] Standards on Electroacoustics of the Institute of Radio Engineers, 1933.

[4] American Standards Association, "Pressure Calibration of Laboratory Standard Pressure Microphones," z-24.4– 1949; "Laboratory Standard Pressure Microphones," z-24.8– 1949; "Free Field Secondary Calibration of Microphones," z-24.11– 1954.

function of the frequency where e is the open-circuit voltage generated by the microphone, in volts, and p is the sound pressure, in dynes per square centimeter, upon the diaphragm of the microphone. The ratio e/p is usually expressed in decibels with respect to some arbitrary reference level. The pressure upon the diaphragm may be generated by a piston-phone, thermophone, or an electrostatic actuator.

a. *Pistonphone.*[5,6,7,8]—A schematic arrangement of a pistonphone for use in calibrating a pressure type microphone having a diagram of high acoustical impedance is shown in Fig. 10.1A. The small piston is driven by a crank. The pressure generated at the diaphragm, assuming all of the walls of the enclosure to be rigid, is

$$p = \frac{rAp_0\gamma}{V}\left\{1 + \frac{(\gamma - 1)A_w}{\alpha V_0} + \frac{1}{2}\left[\frac{(\gamma - 1)A_w}{\alpha V_0}\right]^2\right\}^{-1/2} \qquad 10.1$$

where p = peak pressure, in dynes per square centimeter,

V_0 = volume of the enclosure, in cubic centimeters,

A = area of the piston, in square centimeters,

r = radius of the crank, in centimeters,

p_0 = atmospheric pressure, in dynes per square centimeter,

γ = ratio of specific heats (1.4 for air),

$\alpha = \sqrt{\dfrac{\omega\rho C_p}{2K}} = 3.9\sqrt{f}$ for air, 20° C.,

A_w = area of metallic walls, in square centimeters,

K = thermal conductivity of the enclosed gas (6.2×10^{-6} for air),

ρ = density of the gas, in grams per cubic centimeters,

C_p = specific heat of the gas at constant pressure (.24 for air),

$\omega = 2\pi f$, and

f = frequency, in cycles per second.

This method is very useful for calibrating a microphone at the low frequencies. The upper frequency limit is governed by the permissible speed of the mechanical system which is approximately 200 cycles.

Under test the output of the microphone is fed to an amplifier and output meter. For a particular value of generated pressure the output is noted. Then, the pistonphone is disconnected and a voltage of the same frequency as that generated by the pistonphone is inserted in series with the microphone and adjusted to give the same output. The response (e/p) at this frequency is the ratio of this voltage to the applied pressure.

[5] Wente, E. C., *Phys. Rev.*, Vol. 10, No. 1, p. 39, 1917.
[6] Wente, E. C., *Phys. Rev.*, Vol. 19, No. 4, p. 333, 1922.
[7] Kaye, G. W. C., *Jour. Acous. Soc. Amer.*, Vol. 7, No. 3, p. 174, 1936.
[8] Glover and Baumzweiger, *Jour. Acous. Soc. Amer.*, Vol. 10, No. 3, p. 200, 1939.

b. *Thermophone.*[9,10,11]—The thermophone consists of one or more strips of thin gold leaf mounted upon terminal blocks (Fig. 10.1B). In the usual method the thermophone strip carries a known steady current upon which a smaller sinusoidal current is superimposed. In this case, the variation of the pressure in the chamber occurs primarily at the frequency of the alternating current. The cavity of the thermophone is usually filled with hydrogen. The wavelength in hydrogen is considerably longer than in air and, as a consequence, the standing waves are shifted to a higher frequency beyond the useful response range.

The peak alternating pressure developed in the cavity is given by

$$p = \frac{.96 S i_0 i r_E}{\omega m C V A \alpha D^{1/2}}$$

10.2

where $D = \left(1 - \frac{4KS^2}{\omega CVA}\right)^2 + \left(1 + \frac{4S}{VA\alpha} + \frac{4KS\alpha}{\omega C} + \frac{4KS^2}{\omega CVA}\right)^2$

$A = \frac{T\alpha}{T_S} \frac{\gamma}{\gamma - 1} - 1$

$m = \frac{(\gamma - 1)T_S}{\gamma p_0}$

$\alpha = \sqrt{\frac{\omega C_p \rho}{2K}}$

C = total thermal capacity of the strip product of the mass in grams and the specific heat,

i_0 = steady current, in amperes,

i = peak value of the alternating current component, in amperes,

r_E = total electrical resistance of the strip, in ohms,

T_S = mean temperature of the strip, in degrees Kelvin,

$T\alpha$ = mean temperature of the gas in the enclosure, in degrees Kelvin,

K = thermal conductivity of the gas,

ρ = density of the gas, in grams per cubic centimeter,

C_V = specific heat of the gas at constant volume,

C_p = specific heat of the gas at a constant pressure,

$\gamma = C_p/C_V$,

p_0 = average pressure of the enclosure, in dynes per square centimeter,

S = total area of one side of the thermophone foil, in square centimeters,

V = volume of the enclosure, in cubic centimeters,

$\omega = 2\pi f$, and

f = frequency, in cycles per second.

[9] Arnold and Crandall, *Phys. Rev.*, Vol. 10, No. 1, p. 22, 1917.
[10] Wente, E. C., *Phys. Rev.*, Vol. 19, No. 4, p. 333, 1922.
[11] Ballantine, S., *Jour. Acous. Soc., Amer.*, Vol. III, No. 3, p. 319, 1932.

The determination of the ratio e/p is carried out in the same manner as the pistonphone.

c. *Electrostatic Actuator.*[12]—The electrostatic actuator consists of an auxiliary electrode in the form of a grill mounted in front of the microphone diaphragm, Fig. 10.1C. The actuator is perforated so that it does not
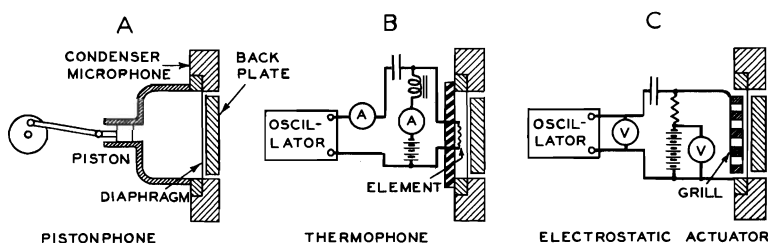
FIG. 10.1. Apparatus for obtaining the pressure-frequency characteristic of a condenser-type microphone. The pistonphone and thermophone may be used for other types of pressure microphones.

appreciably alter the mechanical impedance opposing the motion of the diaphragm. A large, steady, polarizing voltage is applied to the grill and microphone diaphragm. Then a sinusoidal voltage is applied, effectively, in series. The alternating force,[13] in dynes per square centimeter of the grill, assuming no tufting of the electrostatic lines, is

$$p = \frac{8.85 e_0 e}{d^2} \times 10^{-7} \qquad 10.3$$

where e_0 = polarizing voltage, in volts,

e = alternating voltage, in volts, and

d = spacing between the actuator and the diaphragm, in centimeters.

The force developed by the actuator is independent of the frequency. Therefore, it constitutes a simple system for obtaining the response of a condenser microphone as a function of the frequency. If the absolute response is desired this may be obtained by comparison with some known standard (thermophone or pistonphone). In the case of some actuator structures the effective area may be calculated from standard formulas which correct for tufting.

The determination of the ratio e/p is carried out in the same manner as the pistonphone.

---

[12] Ballantine, S., *Jour. Acous. Soc. Amer.*, Vol. 3, No. 3, p. 319, 1932.

[13] Olson, "Dynamical Analogies," D. Van Nostrand Company, Princeton, N.J., 1943.

d. *Reciprocity.*—The acoustical reciprocity procedure[14,15,16,17,18] may be used to obtain the pressure response frequency characteristic of a microphone. In the reciprocity procedure, three transducers are used to obtain the pressure response frequency characteristic of a microphone, namely, the microphone M to be calibrated, a reversible microphone-loudspeaker, S_1, and a loudspeaker, S_2, see Fig. 10.2.

The acoustical impedance of the microphone, M, and the reversible transducer, S_1, should be the same in order to obtain accurate results. Furthermore, the acoustical impedance of the microphone at the diaphragm should be high. Microphones satisfying these requirements are, in general, condenser and piezoelectric microphones.

The first and second experiments in the reciprocity procedure are shown in Fig. 10.2A and Fig. 10.2B. An alternating current is fed to the loudspeaker, S_2. A sound pressure, p_1, is produced in the volume having an acoustical capacitance C_A. Let the open circuit voltage, in statvolts, of S_1 used as a microphone be designated as e_S and the open circuit voltage output, in statvolts, of the microphone M be designated as e_M. Let K_S = output, in statvolts per dyne per square centimeter, of S_1 and K_M = output, in statvolts per dyne per square centimeter, of M. Since the sound pressure, p_1, in dynes per square centimeter, is the same for S_1 and M, it is evident that
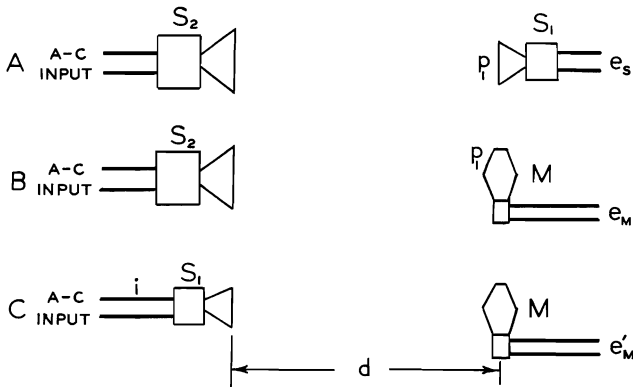

FIG. 10.2. The three experiments of the reciprocity procedure for obtaining the pressure calibration of a microphone. A. The open-circuit voltage, e_S, of the reversible microphone loudspeaker, S_1, when used as a microphone and actuated by a sound pressure, p_1. B. The open-circuit voltage, e_M, of the microphone, M, to be calibrated, when actuated by a sound pressure, p_1. C. The open-circuit voltage, e_M', of the microphone, M, to be calibrated, when actuated by a sound pressure, p_2, produced by the reversible microphone loudspeaker, S_1, used as a loudspeaker with a current input, i, and a volume coupling, C_A.

$$p_1 = \frac{e_S}{K_S} = \frac{e_M}{K_M} \qquad 10.4$$

In the experiments of Fig. 10.2, it is assumed that the acoustical impedance of the diaphragm of the units S_1 and M are large compared to the

[14] Rayleigh, "Theory of Sound," The Macmillan Company, Vol. 1, p. 145.

[15] Ballantine, S., *Proc. Inst. Rad. Engrs.*, Vol. 17, No. 6, p. 929, 1929.

[16] Cook, R. F., *Jour. Research, Natl. Bur. Standards*, Vol. 25, No. 5, p. 489, 1940.

[17] McLean, W. R., *Jour. Acous. Soc. Amer.*, Vol. 12, No. 1, p. 140, 1940.

[18] "American Standard Method for the Pressure Calibration of Laboratory Standard Pressure Microphones," American Standards Association, New York, N.Y., z-24.4–1949. This standard contains an extensive bibliography as well as the reciprocity procedure for obtaining the pressure response frequency characteristic of a microphone.

acoustical impedance of the coupling cavity. In these considerations it will be assumed that the transducers of S_1 and M are of the electrostatic or condenser type. The microphone described in Sec. 9.2B is one of the microphone types suitable for use as the units S_1 and M. In this case the transducers are reversible and the volume current[19] U_2, in cubic centimeters per second, of the diaphragm of S_1 due to a current i, in statamperes, applied to the transducer in experiment 10.2C is given by

$$U_2 = K_S i \qquad\qquad 10.5$$

The sound pressure p_2, in dynes per square centimeter, produced by the loudspeaker S_1 in experiment C of Fig. 10.2 is given by

$$p_2 = \frac{U_2}{j\omega C_A} \qquad\qquad 10.6$$

Equation 10.6 is valid provided the acoustical impedance of the microphone M is large compared to the acoustical impedance of the cavity.

The open circuit voltage e'_M, in statvolts per dyne per square centimeter, of the microphone M in experiment C of Fig. 10.2 is given by

$$K_M = \frac{e'_M}{p_2} \qquad\qquad 10.7$$

where p_2 = sound pressure, in dynes per square centimeter.

From equations 10.5, 10.6, and 10.7,

$$K_M K_S = \frac{j\omega C_A e'_M}{i} \qquad\qquad 10.8$$

From equation 10.4,

$$K_S = K_M \frac{e_S}{e_M} \qquad\qquad 10.9$$

From equations 10.8 and 10.9,

$$K_M = \sqrt{\frac{j\omega C_A e_M e'_M}{e_S i}} \qquad\qquad 10.10$$

where e_S, e_M, e'_M, and i are obtained from the three experiments A, B, and C of Fig. 10.2. The acoustical capacitance C_A is obtained from the dimensions of the coupling cavity. The voltages are in statvolts and the currents in statamperes and the acoustical capacitance in (centimeter)[5] per dyne. Equation 10.10 was derived for electrostatic or condenser transducers. However, equation 10.10 applies to other transducers provided the stipulations in the derivation are satisfied.

2. *Field Response.*—The field or free-wave response frequency characteristic of a microphone is the ratio e/p as a function of the frequency, where e is the open-circuit voltage generated by the microphone, in volts, and p

[19] Olson, "Dynamical Analogies," D. Van Nostrand Company, Princeton, N.J., 1943.

is the sound pressure, in dynes per square centimeter, in a free progressive wave prior to the introduction of the microphone.

At the present time the Rayleigh disk and the reciprocity procedure are the two most common methods in use today for obtaining the field response frequency characteristic of a microphone. It is the purpose of this section to describe the calibration of a microphone by means of the Rayleigh disk and reciprocity methods.

a. *Rayleigh Disk*.[20],[21],[22],[23]—Rayleigh observed that when a disk was suspended by a light fiber it would tend to turn at right angles to the impinging sound wave. Koenig[24] developed the formula for the turning moment of the disk as

$$M = \frac{4}{3} \rho a^3 u^2 \sin 2\theta \qquad 10.11$$

where M = turning moment acting upon the disk, in dyne centimeters,

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

a = radius of the disk, in centimeters,

θ = angle between the normal to the disk and the direction of propagation of the sound wave, in degrees, and

u = particle velocity of the sound wave, root-mean-square, in centimeters, per second.

When a sound wave falls upon the disk the angular deflection will be

$$\phi = \frac{M}{S} \qquad 10.12$$

where S = moment of torsion of the suspension, in dyne centimeter.

The moment of torsion of the suspension is given by

$$S = \frac{I}{T^2} [4\pi^2 + (\log_\epsilon \gamma)^2] \qquad 10.13$$

where T = periodic time of the suspended disk, in seconds,

I = moment of inertia of the disk,

$I = ma^2/4$,

m = mass of the disk, in grams,

a = radius of the disk, in centimeters, and

γ = damping factor, the ratio of two successive swings.

From equations 10.11, 10.12, and 10.13 it is possible to determine the particle velocity u in the sound wave.

The arrangement of a Rayleigh disk for field calibrations of microphones is shown in Fig. 10.3. The source of sound is a small direct radiator loudspeaker, with the back completely enclosed, placed halfway between the

[20] Rayleigh, *Phil. Mag.*, Vol. 14, p. 186, 1882.
[21] Ballantine, *Phys. Rev.*, Vol. 32, No. 6, p. 988, 1929.
[22] Olson and Goldman, *Electronics*, Vol. 4, No. 9, p. 106, 1931.
[23] Sivian, L. J., *Bell Syst. Tech. Jour.*, Vol. 10, No. 1, p. 96, 1931.
[24] Koenig, *Ann. d. Physik*, Vol. 43, p. 43, 1891.

disk and the microphone. A small loudspeaker is used so that a spherical
wave will be emitted. If a velocity microphone is used no correction need
be made for the spherical wave because the Rayleigh disk also measures
the particle velocity. If a pressure microphone is used the appropriate
correction for the accentuation in velocity in a spherical wave must be
made (see Sec. 1.5D and Fig. 1.3). From the geometry of the system of
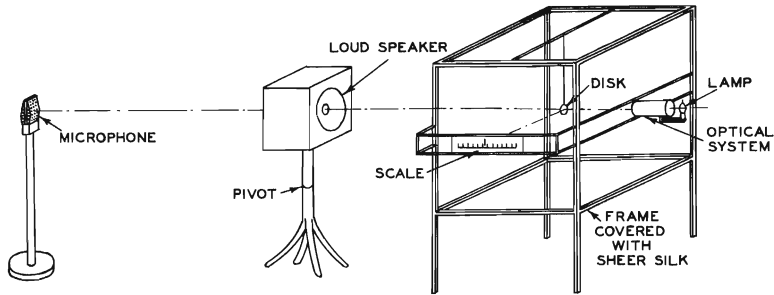
FIG. 10.3. Arrangement of apparatus for obtaining the free-field response of
a microphone by means of a Rayleigh disk.

Fig. 10.3 the deflection of the disk can be determined from the deflection of
the light beam on the scale.

b. *Reciprocity*.—The acoustical reciprocity theorem was originally
enunciated by Helmholtz and Rayleigh.[25]. Ballantine[26] established
reciprocity theorems for mechanoacoustic, electromechano, and electro-
mechanoacoustic systems. Ballantine also carried out a generalized dis-
cussion to show that a microphone may be calibrated in terms of electrical
standards by the use of the extended reciprocity relations. Later other
investigators[27,28,29] extended the applications of reciprocity in both closed
and field systems. It is the purpose of this section to outline the reciprocity
procedure for the field calibration of microphones.

For the application of the reciprocity principle to the calibration of a
microphone, three transducers are used as follows: the microphone, M, to
be calibrated, a reversible microphone loudspeaker S_1, and a loudspeaker S_2.
For the reversible microphone loudspeaker it is convenient to use a small
back-enclosed loudspeaker.

The first and second experiments are shown schematically in Fig. 10.4A
and Fig. 10.4B. An alternating current is fed to the loudspeaker S_2. A
sound pressure p_1 is produced at a distance d. Let the open-circuit voltage,
in abvolts, of S_1 used as a microphone be designated as e_S and the output of
the microphone M be designated as e_M. Let K_S = output, in abvolts per

[25] Rayleigh, "Theory of Sound," The Macmillan Company, Vol. I., p. 145.

[26] Ballantine, S., *Proc. Inst. Rad. Engrs.*, Vol. 17, No. 6, p. 929, 1929.

[27] Cook, R. K., *Jour. Research*, Natl. Bur. Standards, Vol. 25, No. 5, p. 489, 1940.

[28] McLean, W. R., *Jour. Acous. Soc. Amer.*, Vol. 12, No. 1, p. 140, 1940.

[29] Olson, H. F., *RCA Review*, Vol. 6, No. 1, p. 36, 1941.

dyne per square centimeter of S_1, and, K_M = output, in abvolts per dyne per square centimeter, of M. Since the sound pressure p_1, in dynes per square centimeter, is the same for S_1 and M_1, it is evident that

$$p_1 = \frac{e_S}{K_S} = \frac{e_M}{K_M} \qquad 10.14$$

The voltage output,[30] in abvolts, of the microphone loudspeaker S_1 used as a microphone is

$$e_S = Bl\dot{x}_1 \qquad 10.15$$

where B = flux density in the air gap, in gausses,

l = length of the conductor, in centimeters, and

$\dot{x}_1$ = velocity of the voice coil, in centimeters per second.

FIG. 10.4. The three experiments of the reciprocity procedure for obtaining the free-field calibration of a microphone. A. The open-circuit voltage, e_S, of the reversible microphone loudspeaker, S_1, when used as a microphone and actuated by a sound pressure, p_1. B. The open-circuit voltage, e_M, of the microphone, M, to be calibrated, when actuated by a sound pressure, p_1. C. The open-circuit voltage, e_M', of the microphone, M, to be calibrated when actuated by a sound pressure produced by the reversible microphone loudspeaker, S_1, used as a loudspeaker with a current input, i, and a spacial separation, d.

The velocity, in centimeters per second, of the vibrating system of S_1 as a microphone is

$$\dot{x}_1 = \frac{p_1 A}{z_M} \qquad 10.16$$

where p = actuating sound pressure, in dynes per square centimeter,

A = area of the diaphragm, in square centimeters, and

z_M = mechanical impedance of the vibrating system, in mechanical ohms.

[30] Olson, "Dynamical Analogies," D. Van Nostrand Company, Princeton, N.J., 1943.

From equations 10.14, 10.15, and 10.16

$$\frac{e_S}{p_1} = \frac{BlA}{z_M} = K_S \qquad 10.17$$

The third experiment is shown in Fig. 10.4C. The velocity, [31] in centimeters per second, of the diaphragm and voice coil of S_1 for a current i in abamperes, in the voice coil is

$$\dot{x} = \frac{Bli}{z_M} \qquad 10.18$$

The sound pressure, p, at M, in dynes per square centimeter, at a distance d, in centimeters, produced by S_1 in the range where the dimensions are small compared to the wavelength, from equations 2.1 and 2.4, is

$$p = \frac{\rho c k A \dot{x}}{4\pi d} \qquad 10.19$$

where A = area of the diaphragm, in square centimeters,

$\dot{x}$ = velocity of the diaphragm, in centimeters per second,

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

$k = 2\pi/\lambda$,

λ = wavelength, in centimeters, and

c = velocity of sound.

From equations 10.18 and 10.19,

$$p = \frac{\rho c K A B l i}{4\pi d z_M} \qquad 10.20$$

From equations 10.17 and 10.20,

$$p = \frac{\rho c k i K_S}{4\pi d} = \frac{r_A i K_S}{2d\lambda} \qquad 10.21$$

where $r_A = \rho c$.

The sound pressure, p, in dynes per square centimeter, at M in terms of the constant K_M and the open-circuit voltage e'_M, in abvolts, is

$$p = \frac{e'_M}{K_M} \qquad 10.22$$

From equations 10.21 and 10.22,

$$\frac{e'_M}{K_M} = \frac{r_A i K_S}{2d\lambda} \qquad 10.23$$

From equation 10.14,

$$K_M = \frac{e_M}{e_S} K_S \qquad 10.24$$

[31] Olson, "Dynamical Analogies," D. Van Nostrand Company, Princeton, N.J., 1943.

When K_S is eliminated from equations 10.23 and 10.24, the response of the microphone M, in abvolts per dyne per square centimeter, is

$$K_M = \sqrt{\frac{2d\lambda e_M e'_M}{r_A i e_S}} \qquad 10.25$$

where e_S, e_M, e'_M, and i are obtained from the experiments of Fig. 10.4. The units are as follows: Voltages in abvolts, currents in abamperes, distances in centimeters, wavelengths in centimeters, and $r_A = \rho c = 41.5$.

The calibration of microphones by the Rayleigh disk and reciprocity methods should be made under free-field conditions, that is, in a large room in which the reflections are negligible or outdoors at a great distance from reflecting surfaces. A free-field sound room suitable for these measurements is described in Sec. 10.3A4.

A high-quality microphone calibrated by any of the above methods may be used as a secondary standard for the calibrations of other microphones.

3. *Secondary Calibration of Microphones.*[32]—A secondary calibration of a microphone is obtained by a comparison of the response of the microphone to be calibrated with the response of a microphone that has been calibrated by primary means described in the preceding sections. Response frequency characteristics are obtained on the two microphones. The ratio response of the microphone being calibrated to the response of the calibrated microphone yields the calibrated response of the microphone being calibrated.

4. *Artificial Voice.*—The proximity of the head in close talking speech type microphones influences the response frequency characteristics. Therefore, in testing microphones of this type it is desirable to provide testing means[33] which simulate actual operating conditions. The artificial voice consisting of a small loudspeaker unit mounted in the head of a manikin, as shown in Fig. 10.5, provides a means for obtaining the response frequency characteristics of close talking microphones. Resonances in the tube connecting the loudspeaker unit and the mouth are eliminated by the introduction of series and shunt mechanical resistances. The response frequency characteristic shown in Fig. 10.5 can be obtained by a suitable choice of constants of the mechanical system.

5. *Artificial Throat.*—Throat microphones have been described in Sec. 8.7. Throat microphones are actuated by sound waves transmitted through the throat. An artificial throat[34] for testing throat microphones consists of a mass-controlled system driven by a voice coil located in magnetic field. Specifically the voice coil is coupled to a massive platform. The centering system is made very compliant to insure mass control. In order to maintain constant velocity with respect to frequency the driving oscillator and amplifier are compensated so that the current through the voice coil is proportional to the frequency. The platform system, which the voice coil

[32] American Standards Association, "Secondary Calibration of Microphones," z-24.11- 1954.

[33] Inglis, Gray, and Jenkins, *Bell Syst. Tech. Jour.*, Vol. 11, No. 2, p. 293, 1932.

[34] Greibach, E. H., *Elec. Eng.*, Vol. 65, No. 4, p. 184, 1946.

drives, is coupled to the throat microphone under test by means of a filter pad made of material with high damping—as, for example, Viscoloid.

B. *Directional Characteristic.*—The directional characteristic of a microphone is an expression of the variation of the behavior of the microphone

PERSPECTIVE VIEW   SECTIONAL VIEW   MECHANICAL NETWORK   RESPONSE DB / FREQUENCY
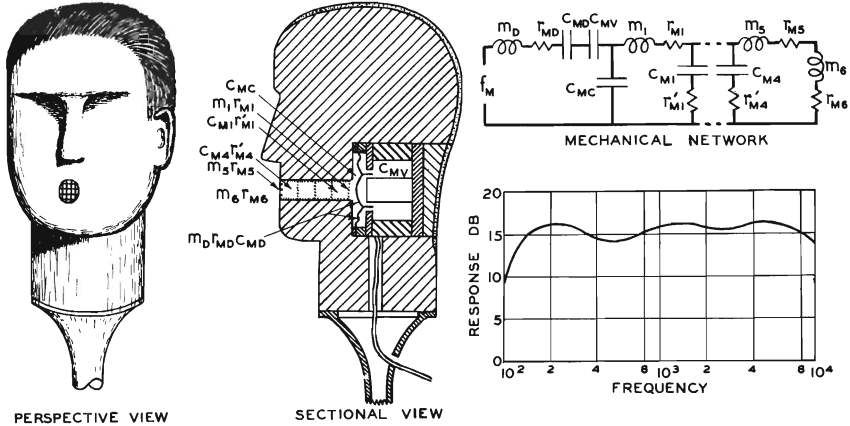
FIG. 10.5. Perspective view, sectional view, mechanical network, and response frequency characteristic of an artificial voice. In the mechanical network, m_D = the mass of the diaphragm and suspension of the small loudspeaker unit. r_{MD} and C_{MD} = the mechanical resistance and compliance of the suspension system of the small loudspeaker unit. C_{MC} = the compliance of the air chamber behind the diaphragm. $m_1, r_M \ldots m_5, r_{M5}$ = the mass and mechanical resistances of the series elements in the pipe. $r_{M1}', C_{M1} \ldots r_{M4}', C_{M4}$ = the mechanical resistances and compliances of the shunt elements of the line. m_6 and r_{M6} = the mass and mechanical resistance of the air load on the mouth. The response frequency characteristic depicts the free-field sound pressure at a distance of 2 inches.
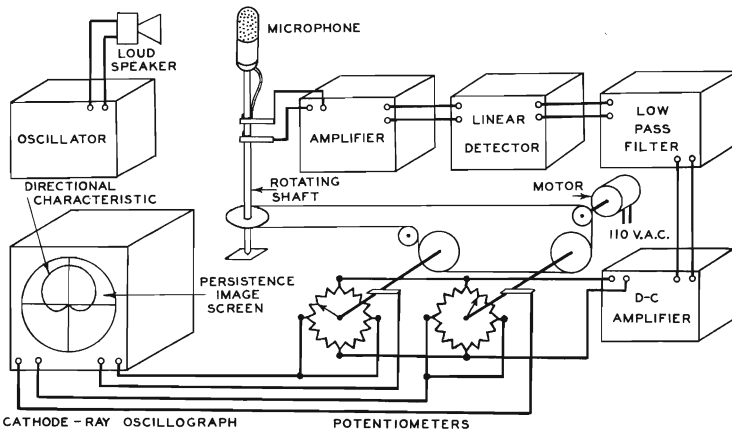
FIG. 10.6. Schematic arrangement of the apparatus employing a cathode-ray tube with a long persistence screen as a polar directional characteristic indicator and recorder.

with respect to direction. A polar diagram showing the output variation of the microphone with direction is usually employed.

The directional characteristics should be obtained at representative frequencies. In order to obviate any errors due to reflections the directional measurements should be made under free-field conditions. Obviously, very slight reflections will introduce considerable error for the angles in which the response is very low.

A cathode-ray tube with a long persistence screen may be used to obtain the directional characteristic of a microphone or loudspeaker. The apparatus of Fig. 10.6 is arranged to obtain the directional characteristic of the microphone. The directional characteristic of the loudspeaker may be obtained by placing the loudspeaker upon the rotating shaft and keeping the microphone fixed in position. The sound is picked up by a microphone and amplified. The output of the amplifier is detected and fed to a low-pass filter. The output of the filter is amplified by a d-c amplifier, the output of which is fed to two potentiometers. The arms of the potentiometers are spaced at 90°. The potentiometer arms and microphone shaft are rotated by a motor. The length of the radius vector of the spot is proportional to the output of the microphone. The angular displacement of the spot is synchronized with the microphone shaft. From this it will be seen that the cathode-ray beam traces the polar directional characteristic of the microphone. In case it is desirable to record the characteristic, this may be done photographically or by tracing the curve left upon the screen.

C. *Nonlinear Distortion Characteristic.*—The harmonic distortion tests are intended to show the spurious harmonics generated by the microphone
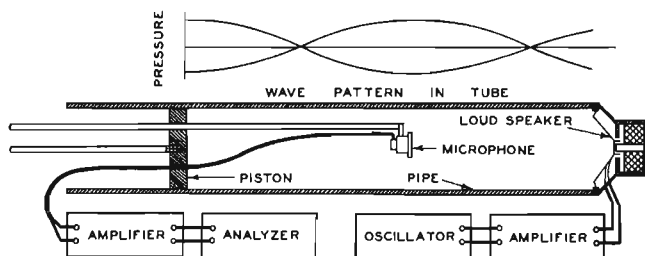
FIG. 10.7. Arrangement of apparatus for measuring the non-linear distortion generated by a microphone. (After Phelps.)

when it is actuated by a pure tone. The plot of the total distortion, in per cent of the fundamental, is termed the distortion characteristic. It is also common practice to plot the individual components in per cent as the distortion characteristics.

It is difficult to obtain a sound source which will generate an intense sound wave of very low distortion in free space. The arrangement[35] shown in Fig. 10.7 provides a simple means of obtaining a sound wave free from

[35] Phelps, W. D., *Jour. Acous. Soc. Amer.*, Vol. 11, No. 2, p. 219, 1939.

distortion. A stationary wave is obtained in the tube by moving the piston until the maximum pressure is obtained. A pressure of 1000 dynes per square centimeter can be obtained with a fraction of a watt input to the loudspeaker. For the determination of the second harmonic the microphone is placed at a second harmonic node. Under these conditions the second harmonic component at the microphone is very small. The second harmonic component is then measured by means of a harmonic analyzer (see Sec. 10.3C). For the third harmonic the microphone is placed at a third harmonic node. Either pressure or velocity microphones may be tested, the only difference being in the position in the tube.

D. *Phase Distortion Characteristic.*—The phase distortion characteristic of a microphone is a plot of the phase angle between the voltage output of the microphone with respect to some reference voltage as a function of the frequency. A microphone such as the velocity microphone (see Sec. 8.3B), in which the output is in phase with the particle velocity (its output is also in phase with the pressure in a plane sound wave), may be
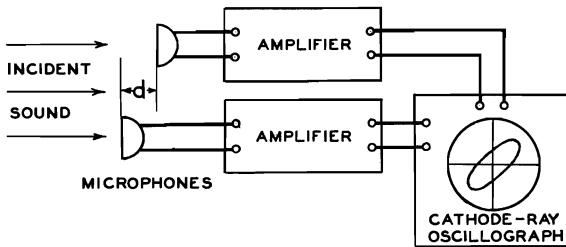
FIG. 10.8. Schematic arrangement of apparatus for measuring the phase characteristic of a microphone.

used as the reference standard. The standard microphone and the microphone to be tested may be placed side by side in a plane progressive wave in free space, Fig. 10.8. The outputs of the two microphones are amplified by separate identical amplifiers and connected to the vertical and horizontal plates of a cathode-ray oscillograph. The resultant Lissajou figure indicates the phase relations between the output of the two microphones. The two microphones are shifted relative to each other in a line parallel to the direction of propagation until the outputs of the two microphones are in phase. The phase angle, in degrees, between the output of the two microphones is

$$\phi = \frac{d}{\lambda} 360° \qquad\qquad 10.26$$

where d = distance between the two microphones in the direction of propagation, in centimeters, and

λ = wavelength of the sound, in centimeters.

Phase distortion is of importance in combination microphones such as the unidirectional microphone (see Sec. 8.4).

E. *Electrical Impedance Frequency Characteristic.*—The electrical impedance frequency characteristic of a microphone is the electrical impedance at the output terminals as a function of the frequency. Any convenient method for measuring electrical impedance may be used for determining the electrical impedance frequency characteristic.

F. *Transient Response Characteristic.*—For measurement of transient response, see Secs. 10.3G and 8.14.

G. *Measurement of Wind Response of Microphones.*—The wind response of microphones is an important characteristic when microphones are used
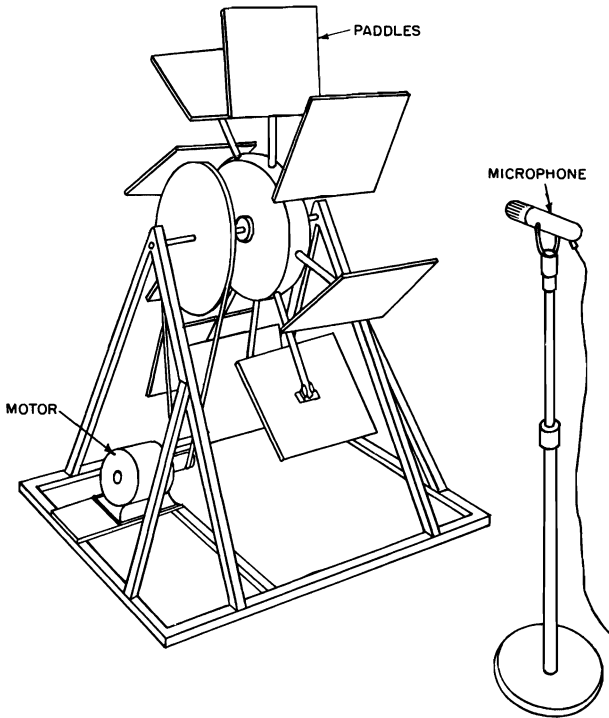
FIG. 10.9. A wind generator for obtaining the wind response of microphones.

outdoors under wind conditions. It is very difficult to test the wind response under actual wind conditions because it is impossible to find constant wind conditions. Wind consists of a steady flow of air with superimposed pulses of air. A wind generator[36] which delivers a steady flow of air with superimposed pulses is shown in Fig. 10.9. Almost any practical value of wind velocity and pulses can be obtained by adjusting the speed of rotation and

[36] Olson, Preston, and Bleazey, Unpublished Report.

the angles of the paddles. Comparison of the results obtained with actual wind with those of the wind generator shows very good correlation with respect to the distribution of the frequency components. This indicates that the wind components in the two cases are the same.

In another method[37] for obtaining the wind response of a microphone, the microphone is suspended at the end of a pendulum about ten feet in length. In this way maximum wind velocities up to 20 miles an hour can be obtained. In addition, there is a variation in velocity from zero to the maximum velocity.

10.3. Testing of Loudspeakers.[38],[39]—Many different measurements are required to determine the performance of a loudspeaker. The most important characteristics which depict the performance of a loudspeaker are as follows:

1. Response frequency characteristic
2. Directional characteristic.
3. Nonlinear distortion characteristic
4. Efficiency frequency characteristic
5. Phase distortion characteristic
6. Electrical impedance characteristic
7. Transient response characteristic

A. *Response Frequency Characteristic.*—1. *Pressure Response.*—The pressure response of a loudspeaker is a measure of the sound pressure produced at a designated position in the medium with the electrical input, frequency, and acoustic conditions specified.

Absolute response is the ratio of the sound pressure (at a specified point in space) to the square root of the apparent electrical power input. It is given by the equation

$$\text{Absolute response} = \frac{p}{\dfrac{e}{\sqrt{z_E}}} = \frac{p\sqrt{z_E}}{e} \qquad\qquad 10.27$$

where p = measured sound pressure, in dynes per square centimeter,

$\quad e$ = effective voltage applied to the voice coil, in volts, and

$\quad z_E$ = absolute value of the electrical impedance of the voice coil, in ohms (z_E is usually a function of frequency).

The absolute response characteristic is obtained by measuring the sound pressure p, as a function of frequency with constant voltage, e, on the voice coil, and measuring the electrical impedance, z_E, as a function of the frequency, and correcting the measured sound pressure for the measured electrical impedance in accordance with the equation. The resulting

[37] Carrell, R. M., *Jour. Acous. Soc. Amer.*, Vol. 3, No. 2, p. 102, 1955.
[38] American Standards Association, Loud Speaker Testing, C. 16.4, 1942.
[39] *Standards on Electroacoustics*, Institute of Radio Engineers, 1938.

characteristic represents the sound pressure as a function of the frequency which would be obtained from the speaker if fed from the generator which would automatically deliver constant apparent power, e^2/z_E, to the voice coil over the frequency range.

The response may be expressed by a value equal to the above ratio or may be expressed in decibels relative to an arbitrary value of response corresponding to 1 volt, 1 ohm, and 1 dyne per square centimeter.

$$\text{Absolute response} = 20 \log_{10} \frac{\dfrac{p}{e/\sqrt{z_E}}}{\dfrac{1}{1/\sqrt{1}}} = 20 \log_{10} \frac{p\sqrt{z_E}}{e} \qquad 10.28$$

The apparatus and methods employed for obtaining the response frequency characteristics of loudspeakers will be described in the sections which follow.

2. *Apparatus for Measuring the Sound Pressure Frequency Relationship of a Sound Source.*—An arrangement for obtaining the sound pressure frequency characteristic by the semiautomatic method[40] is shown in Fig. 10.10.

This method yields a response frequency curve on semilogarithmic paper in about 3 minutes. Rotation of a condenser governing the beat frequency
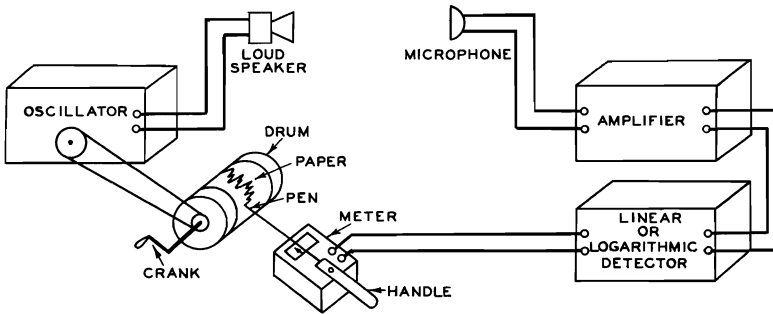
FIG. 10.10. Schematic arrangement of the apparatus for manually recording the sound pressure-frequency characteristic of a sound source. (After Wolff and Ringel.)

of the heterodyne oscillator and coupled to a drum on which the paper record is made gives the abscissas for the curves, values which are proportional to the logarithm of the frequency due to the manner in which the condenser plates are cut. The drive may be manual or by motor.

A linear or logarithmic detector[41,42] may be employed. In the former, the deflection of the meter is proportional to the sound pressure. In the

[40] Wolff and Ringel, *Proc. IRE*, Vol. 15, No. 5, p. 363, 1927.
[41] Ballantine, S., *Jour. Acous. Soc. Amer.*, Vol. 5, No. 1, p. 10, 1933.
[42] Hackley, R. A., *Broadcast News*, No. 28, p. 20, July, 1938.

latter, the deflection of the meter is a logarithmic function of the sound pressure. The resulting curve is recorded directly in decibels. A variation of this method is sometimes used in which the recording pen is coupled to a gain control in the amplifier, the operator manipulating the control in such a manner that the output indicated by the meter remains constant. Either a linear or a logarithmic coordinate scale may be obtained by suitable design of the gain control.

The acoustical level recorder[43] is an automatic device which records the gain settings required to keep the amplifier output constant as the frequency of the sound source is varied. Fig. 10.11 shows how a pressure characteristic can be made with the sound level recorder. A dark-colored tape coated
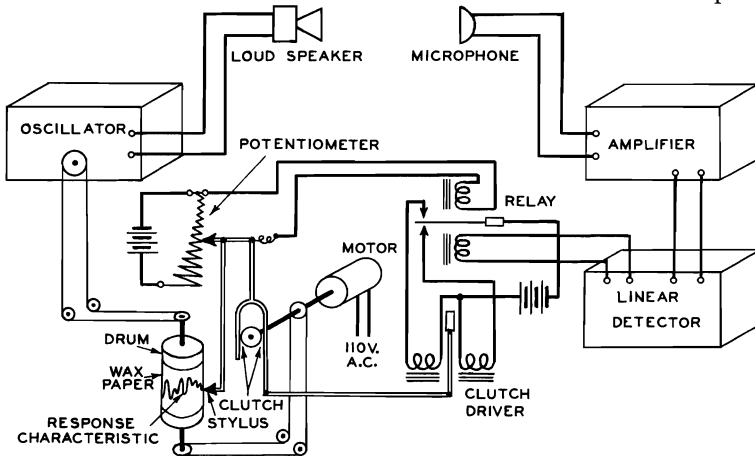
FIG. 10.11. Schematic arrangement of the apparatus used in a high-speed level recorder for automatically recording a sound pressure-frequency characteristic. (After Wente, Bedell, and Swartzel.)

with white wax is moved under a stylus by a motor which changes the value of the beat frequency generated at the same time. The loudspeaker under test is connected to the output of the beat frequency generator and the variations in response are recorded on the paper directly on a decibel scale by a stylus which scratches through the wax coating on the recording paper.

The rectifier output incorporates a control circuit which causes direct current to flow through one circuit when the rectifier current is less than a certain critical value and through a second circuit when it is greater than a second critical value. In the first case, the control circuit operates a magnetic clutch which causes a potentiometer to operate and increases the voltage. In the second case, the voltage is decreased.

The output of the rectifier is kept balanced to within the voltage change produced by a change in potentiometer corresponding to the smallest unit

[43] Wente, Bedell, and Swartzel, *Jour. Acous. Soc. Amer.*, Vol. 6, No. 3, p. 121, 1935.

of the attenuator calibration. The motion of the potentiometer is communicated to the stylus which gives a trace on the recording paper. The same motor which drives the oscillator frequency control moves the potentiometer by means of the magnetic clutches.

The speed with which changes in sound level are recorded may be varied from 10 to 560 db per second through alteration of the speed of rotation of the clutches.

In another design[44] of high-speed level recorder a thyratron actuated reversible motor drives a fountain pen and records directly on graph paper. The speed is somewhat slower than the clutch system but the conventional paper record is more convenient to use and file.

A high-speed level recorder[45] with a dynamic drive is shown in Fig. 10.12. The dynamic driving mechanism is used to drive the stylus and the contactor
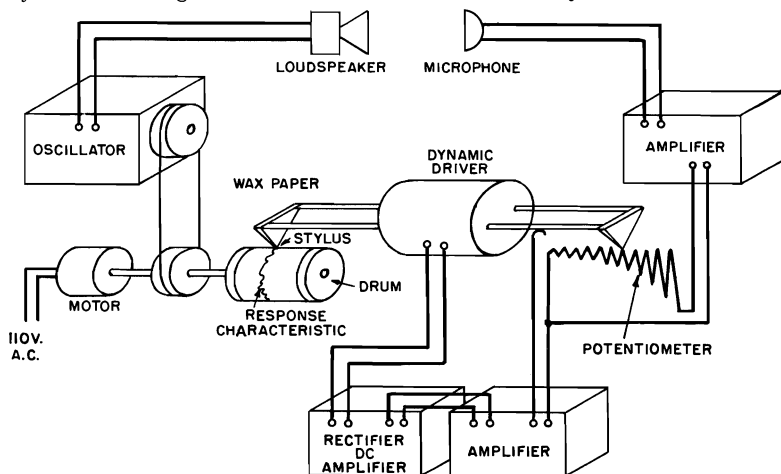
FIG. 10.12. Schematic arrangement of the apparatus used in a high-speed level recorder employing a dynamic driving system.

on the potentiometer. The dynamic driving mechanism consists of a voice coil located in a very long air gap. See Fig. 10.13. The flux for the air gap is supplied by permanent magnets. Two drive rods extending axially in both directions are attached to the voice coil. The stylus and contactor are attached to the drive rod. The useful amplitude range of the system is two inches. The potentiometer is of the logarithmic type. The drive system in conjunction with the potentiometer maintains constant input to the driving amplifier. If there is an unbalance, the amplifier supplies the proper polarity to the voice coil, and, as a result, the driving system will move to restore the balance. The loudspeaker under test is connected to the output of the beat frequency oscillator and the variations in response are recorded

[44] Clark, W. R., *A.I.E.E. Trans.*, Vol. 59, p. 957, 1940.
[45] Bruel and Kjar, *Tech. Rev.*, No. 3, 1952.

MAGNETS

VOICE COIL

POLE

DRIVE RODS FASTENED TO VOICE COIL
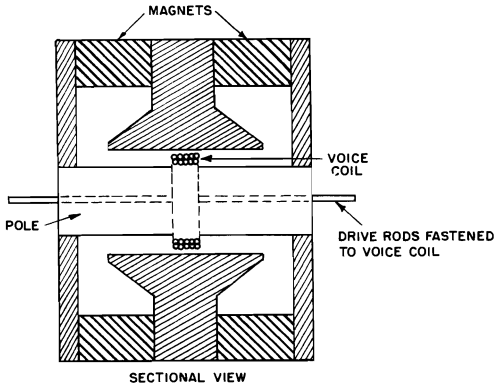
SECTIONAL VIEW

FIG. 10.13. Sectional view of the dynamic drive used in the high-speed level recorder of Fig. 10.12.

on the wax-coated paper directly on a decibel scale by the stylus which scratches through the wax coating on the paper and thereby leaves a visible
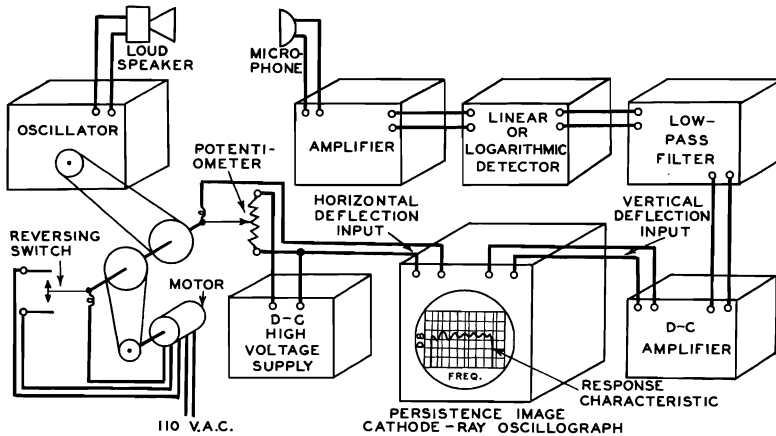
FIG. 10.14. Schematic arrangement of the apparatus employing a cathode-ray tube with a long persistence screen as a pressure response frequency indicator and recorder. (After Hackley.)

trace. The maximum recording speed is about 900 db per second with a potentiometer having a 60 db range.

A cathode-ray tube,[46],[47] with a long persistence screen, may be used as a response indicator and recorder, Fig. 10.14. A motor drives the beat frequency oscillator and a potentiometer. The potentiometer varies the

[46] Hackley, R. A., *Broadcast News*, No. 28, p. 20, July, 1938.
[47] Sherman, J. B., *Proc. IRE*, Vol. 26, No. 16, p. 700, 1938.

voltage on the horizontal deflection plate of the cathode-ray tube and thereby drives the cathode-ray beam across the tube in synchronism with the oscillator. A reversing switch changes the direction of the motor travel at the upper and lower limits of the audio-frequency range. The output of the oscillator actuates the loudspeaker. The sound is picked up by the microphone and amplified. The output of the amplifier is detected by a linear or logarithmic detector and fed to a low-pass filter. The output of the filter is amplified by a d-c amplifier, the output of which is connected to the vertical plates of the cathode-ray tube. The cathode-ray beam traces the response characteristic upon the persistence image screen. The ordinates are in decibels when the logarithmic detector is used. The ordinates are proportional to the sound pressure when the linear detector is used. The
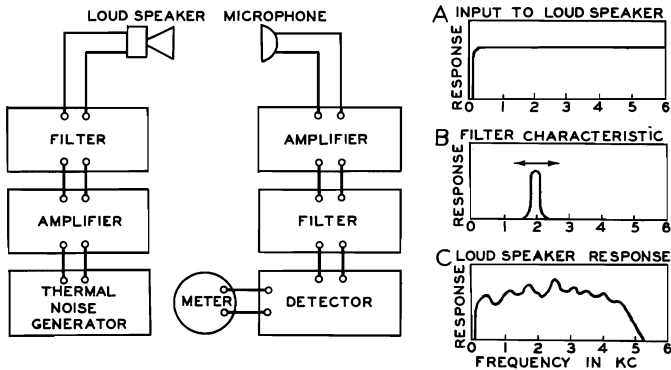
Fig. 10.15. Schematic arrangement of the apparatus employing a thermal noise generator and a band-pass filter for obtaining the response frequency characteristic of a loudspeaker. A. Input to the loudspeaker. B. The response frequency characteristic of the band-pass filter. C. Response frequency characteristic of the loud-speaker.

time required to trace a response frequency characteristic of a loudspeaker is about 30 seconds. The apparatus is very useful for development work because the motor sweeps through the range again and again. The operator is free to make changes in the equipment under test and note these changes upon the response. In case it is desirable to record the characteristic, this may be done photographically or by tracing the curve left upon the screen.

A system[48] for measuring the response of a loudspeaker employing a thermal noise generator is shown in Fig. 10.15. A diode may be used as a source of thermal noise. The output is amplified, filtered, and fed to a loudspeaker. The frequency distribution of the energy fed to the loud-speaker is shown in Fig. 10.15A. The output of the loudspeaker is picked up by the microphone, amplified, and passed through a narrow band-pass filter. The response characteristic of the filter is shown in Fig. 10.15B. The band width of the filter should be independent of the frequency. The

[48] Olney, B., *Jour. Acous. Soc. Amer.*, Vol. 13, No. 1, p. 79, 1942.

position of the band-pass filter is varied with respect to frequency. The output of the filter is detected and measured by means of a meter. The response characteristic of a loudspeaker is shown in Fig. 10.15C.

Apparatus employing thermal noise for obtaining response characteristics has not been developed to the stage where it may be used with the facility of other methods. It appears, however, that this type of measurement will become very important for all types of acoustical measurements when suitable apparatus has been developed.

3. *Calibration of the Sound Measuring Equipment.*[49]—The microphone should be calibrated in terms of the pressure in a free progressive sound wave. The microphone, amplifier, and detector should have a combined characteristic which is substantially independent of the frequency over the frequency range under consideration. If it is not substantially constant over the frequency range the data must be adjusted for known variations.
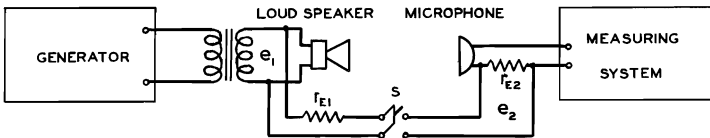
FIG. 10.16. Schematic arrangement for obtaining the factor p/e in the formula for absolute response of a loudspeaker.

A general schematic circuit arrangement showing one specific way to obtain the factor p/e in the formula for absolute response (equation 10.20) is shown in Fig. 10.16. This arrangement has the feature that it does not require an absolute calibration of the measuring system.

Referring to Fig. 10.16, the absolute response is given by

$$\text{Absolute response, in decibels,} = 20 \log_{10} \frac{p}{e} \sqrt{z_E}$$

$$= [A - B - C - D] + 10 \log_{10} z_E \qquad 10.29$$

where A = output of measuring system, in decibels, with the microphone picking up sound from the loudspeaker with S open,

B = output of measuring system, in decibels, with S closed and the microphone shielded from sound,

C = open-circuit voltage output of the microphone, in decibels above 1 volt for 1 dyne per square centimeter, in a free progressive wave,

$D = 20 \log e_1/e_2 = 20 \log (r_{E1} + r_{E2})/r_{E2}$, and

z_E = electrical impedance of loudspeaker, in ohms.

[49] *Standards on Electroacoustics*, Institute of Radio Engineers, 1938.

r_{E2} should be sufficiently small compared to the electrical impedance of the microphone or, in other words, the output of the microphone should not change when r_{E2} is short circuited. r_{E1} should be so selected as to obtain a value of B in the range of the values obtained for A.

4. *Free-Field Sound Room.*[50,51,52]—Acoustical measurements under free-field conditions are required in the development of the major portion of electroacoustic transducers. The most obvious and direct solution would seem to be to make the measurements out of doors at a great distance from all reflecting surfaces. There are several objections to outdoor testing, for example, interruptions due to wind, rain, and snow; noise, both natural and man made; difficulty in arranging experiments at sufficient distance from the earth so that reflections will be negligible. In view of the importance of free-field testing and the objections to outdoor arrangements, it is obvious that a free-field sound room is an almost indispensable part of the equipment of an acoustical laboratory. It is the purpose of this section to describe such a sound room.

The objective in the design of a free-field sound room is to reduce to a negligible amount all reflections from the boundary surfaces of the room. This is equivalent to a very small ratio of generally reflected to direct sound. The ratio of the generally reflected to the direct sound in a room is

$$E_R/E_D = 16\pi D^2 (1 - a)/aS \qquad\qquad 10.30$$

where E_R = energy density of reflected sound, in ergs per cubic centimeter,

$\quad\quad E_D$ = energy density of the direct sound, in ergs per cubic centimeter,

$\quad\quad D$ = distance from the source to the observation point, in centimeters,

$\quad\quad S$ = area of absorbing material, in square centimeters,

$\quad\quad V$ = volume of room, in cubic centimeters, and

$\quad\quad a$ = absorption coefficient (see Sec. 11.2A).

An examination of equation 10.30 shows that the ratio of reflected to direct sound may be reduced by decreasing the distance between the source and observation point, by making the absorption coefficient of the walls near unity, or by increasing the area of the walls. In other words, free-field conditions are approached by making the room large and absorption coefficient of the wall near unity. To satisfy the first requirement, the free-field room was made as large as seemed practical from an architectural and constructional standpoint. The dimensions of the free-field sound room, before acoustical treatment was applied, were as follows: 48 feet

[50] The term free-field sound room is used to designate a room in which free-field sound conditions are obtained, that is, a room in which the reflections from the boundaries are negligible. These rooms have also been termed anechoic rooms. The word anechoic is made up of the Greek prefix *an*, meaning not or without, the Greek word *echo*, meaning echo and the adjectival suffix *ic*, meaning characterized by (see Beranek, Ref. 40).

[51] Olson, H. F., *Jour. Acous. Soc. Amer.*, Vol. 15, No. 2, p. 96, 1943.

[52] Beranek, L., *Jour. Acous. Soc. Amer.*, Vol. 18, No. 1, p. 140, 1946.

long, 36 feet wide, and 36 feet high.   The next objective was to obtain an absorption coefficient as near unity as possible.   The high- and low-frequency ranges present the greatest difficulty in attaining this objective. It is a comparatively simple matter to attain high absorption in the mid-frequency range.   In the high-frequency range the principal difficulty is reflection from grills, control boxes, and test apparatus.   These reflections can be eliminated by acoustical treatment of these reflecting surfaces.   In the case of the low-frequency range it appears to be an inexorable fact that the ideal objective can be attained only in a relatively large room with correspondingly thick absorption material.   An examination of existing rooms indicates that regardless of the form of treatment it appears that absorption deviates quite rapidly from unity when the thickness of the treatment is less than a quarter wavelength.   In this statement, it is assumed that thickness of the material is measured to an outside boundary of relatively high acoustical impedance compared with the characteristic acoustical impedance of air.   It is also assumed that treatment does not involve resonant systems.

The absorbing system employed in this room is of the baffle type, that is, strips of absorbing material arranged normal to the walls of the room as shown in Fig. 10.17.   Several years ago a smaller room (22 feet long, 20 feet wide, and 13 feet high) was treated with baffles.   The performance of this room appeared to be comparable to rooms with equivalent thickness of other types of absorbing material.   The advantage of the baffle type of treatment is the relatively simple construction and lower cost as compared with more elaborate absorbing systems.

Plan and elevation views of the room are shown in Fig. 10.17.   One-inch Ozite is spaced 1 foot from the walls, ceiling, and floor.   One-inch Ozite baffles, 7 feet in length and spaced 2 feet apart, are placed normal to the walls, ceiling, and floor.   Four-foot baffles of the same material are placed between the 7-foot baffles.   The total thickness of the absorbing material, measured from the outside wall, is 8 feet.   This leaves the inside dimensions of the room 32 feet long, 20 feet wide, and 20 feet high.   A special grill, 12 feet wide and 24 feet long, is supported on vibration-isolated feet.   The ratio of open to total area in the grill is 0.87.   This is a relatively open grill when it is considered that the grill platform will carry a load of 200 pounds per square foot.   The floor level of the grill is located 11 feet above the floor level of the room.   The floor level of the grill coincides with the first floor level which makes it readily accessible to the adjoining laboratory.

The acoustical merit of the room can be expressed by the deviation in sound pressure from an inverse distance characteristic.   Pressure response frequency characteristics were obtained at various distances from a small loudspeaker.   The maximum deviation in pressure from an inverse distance characteristic for various frequencies is shown in Fig. 10.18.   It will be noted that the deviation in the mid-frequency ranges is negligible.   The deviation at the high frequencies is due to the grill, overhead trolley track, power and signal outlet boxes.   These units which if treated, will make

FREE-FIELD SOUND ROOM

MONORAIL

GRILL
PLATFORM
ENTRANCE

SECTION A-A'

C

GRILL

A

A'

C'

SECTION B-B'

0   5   10
SCALE IN FEET

1" OZITE BAFFLES

1" OZITE WALL

B

B'

SECTION C-C'
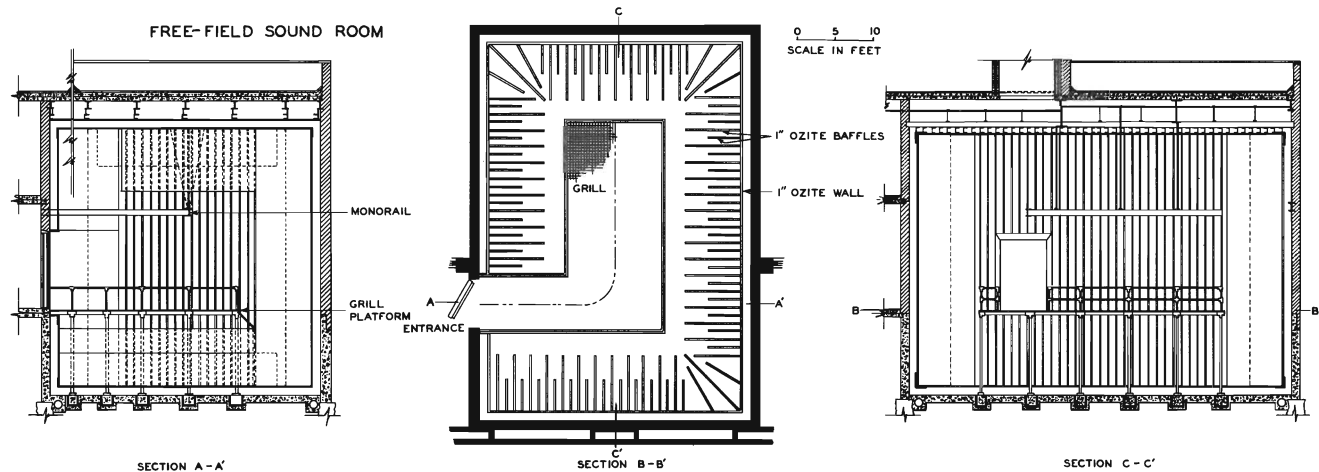
Fig. 10.17.   End elevation, plan, and side elevation of a free-field sound room.

the deviations from an inverse characteristic practically the same as the mid-frequency range. The deviation at the low frequencies begins when the thickness of the material is approximately a quarter wavelength. However, the deviation is only ± 1.7 db at 40 cycles at a distance of 8 feet. At 40 cycles the thickness of the material is 0.28 of the wavelength.

The absorption coefficient of the walls may be determined from the ratio of direct to generally reflected sound. These two components may be
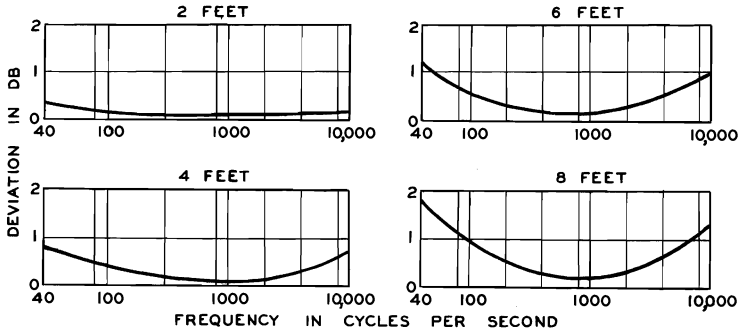
FIG. 10.18. Deviation of the pressure from an inverse distance characteristic for various distances from a sound source in the free-field sound room.

determined by employing a velocity microphone. Two measurements are made—one with the normal to the plane of the ribbon passing through the source and the other with the plane of the ribbon passing through the source. The absorption coefficient frequency characteristic of the walls of the room is shown in Fig. 10.19.

A low-noise level is another essential requirement in a free-field sound room. The noise level in the free-field sound room, when the laboratories are in normal operation, is about 10 db. At night, when the shops are
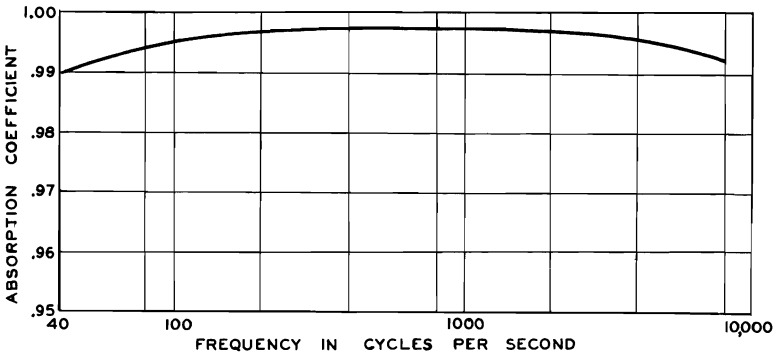
FIG. 10.19. Absorption coefficient frequency characteristic of free-field sound room.

closed down, the noise level is 0 db. This shows that the sound treatment is also quite effective in absorbing sounds generated outside the room. The free-field sound room is heated by hot air forced through 48 openings in the floor. With the blower in operation the noise level in the room is about 20 db. However, it is not necessary to operate the heater during the day because the room is very well insulated thermally as well as acoustically. For example, if the heater is operated 8 hours in every 24 hours, the temperature variation from 70° Fahrenheit is only $\pm 3°$ Fahrenheit on the coldest day.

The above data and other measurements show that it is possible to make measurements in this room under essentially free-field conditions over the
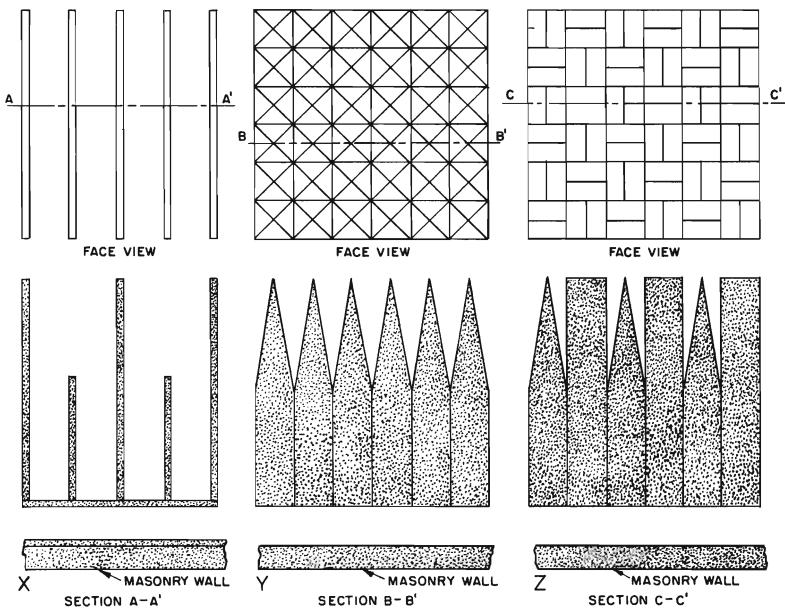
FIG. 10.20. Wall treatments for free-field rooms, X. Baffle type; Y. Pyramid type; Z. Wedge type.

frequency range above 40 cycles for distances between the source and observation up to 8 feet. This distance can be increased if either the source or the microphone, or both, are directional.

Three common types of sound absorbing systems used for free-field or anechoic rooms are shown in Fig. 10.20. Fig. 10.20, X depicts the baffle type of sound absorbing system employed in Fig. 10.17 and described in this section. In the system described in this section the spacing between the baffles is 12 inches. Each baffle is made up of two layers of $\frac{1}{2}$-inch Ozite with actual over-all thickness of 2 inches because the material was obtained in the uncompacted form. Employing twice as many baffles would improve

the absorption in the low-frequency range by a small factor. Such an improvement is not of a practical significance. In the absorbing system shown in Fig. 10.20, Y and Z are of the pyramid and wedge types made of Fibreglass. It appears that in the system of Fig. 10.20, Z exhibits the greatest absorbing efficiency. In this connection, it should be mentioned that absorbing efficiency beyond a certain value is of little practical significance under actual operating conditions when reflecting surfaces almost invariably are introduced in any measurement. In any case, the portion of the frequency range in which it is most difficult to obtain free-field conditions is in the low-frequency range. As previously stated, regardless of the form and material of the treatment employing existing nonactive materials, the absorption deviates rapidly from unity when the depth of the material is less than one-quarter wavelength. Furthermore, free-field conditions can only be obtained when the dimensions of the room are greater than a wavelength.

5. *Outdoor Response.*—If a free-field sound room is not available, free-field conditions may be obtained outdoors by locating the microphone and loudspeaker at a sufficient distance from reflecting surfaces so that the level of the direct sound striking the microphone is at least 20 db above the reflected sound. The microphone and loudspeaker may be suspended on a cable between two high towers. A velocity microphone may be used to discriminate against the reflected sound if there is only one reflecting surface, as, for example, the earth, by orienting the microphone so that the plane of the ribbon coincides with the direction of the reflected sound. Outdoor measurements have the disadvantage of being dependent upon the weather and noise conditions. For this reason, nearly all development and routine work on loudspeakers is carried on in rooms.

6. *Small and Partially Deadened Rooms.*—When only a small deadened room or a partially deadened room is available, the distance between the microphone and loudspeaker must be small in order to reduce reflection errors. A response frequency characteristic taken under these conditions is useful in determining system resonance and general smoothness of the output.

When the distance between the microphone and loudspeaker, in a partially deadened room, is large, a rotating microphone or warble tone may be used to reduce reflection errors.

In the case of the rotating microphone, the microphone is revolved in a circle about 5 feet in diameter. The plane of the circle is inclined at an angle of 30° toward the horizontal. The microphone is arranged so that it is always directed toward the source of sound.

In the case of the stationary microphone, a warble frequency (20 cycles + 10 per cent of the mean audio frequency as a maximum total band width) may be used to average out reflection errors. This method tends to average out very abrupt variations in the loudspeaker response. A check response frequency measurement taken close to the loudspeaker with no warble should be made to determine if there are any abrupt variations in its response.

7. *Arrangement of Loudspeakers for Test.*—In obtaining response frequency characteristic of loudspeakers, the systems may be divided into two classes—namely, direct radiator, loudspeaker units designed to operate in some additional structure and complete systems such as direct radiator mechanisms mounted in cabinets and horn loudspeakers.

In the test of direct radiator, loudspeaker units alone, the unit should be mounted 1 foot off center in a direction parallel to one side and 6 inches off center in a right-angle direction in a square and flat baffle 12 by 12 feet. The baffle should be of sufficient thickness so that no radiation results from vibration of the baffle. The microphone should be located on the axis of the radiator 5 feet from the surface of the baffle when the transverse dimension of the radiator is not more than $2\frac{1}{2}$ feet. For larger radiators, the distance should be the smallest integral multiple by 5 feet, which is greater than twice the maximum traverse dimension of the radiator and should be specified with the test.

Complete loudspeaker systems such as direct radiator mechanisms mounted in cabinets and horn loudspeakers are tested in the same manner as in the case of direct radiator, loudspeaker units, but without the use of additional baffles.

8. *Living Room Measurements.*[53]—The performance of a radio receiver in a living room will be discussed in Sec. 11.2R. The characteristics shown in Fig. 11.19 were obtained with the cathode-ray response measuring system described in Sec. 10.3A2. However, any of the systems described in Sec. 10.3A2 may be used. It is customary to obtain a large number of characteristics for each position of the receiver in the room.

9. *Theater Measurements.*—The performance of a loudspeaker in a theater will be discussed in Sec. 11.2G. The characteristics for the various parts of the theater may be obtained with any equipment described in Sec. 10.3A2. However, the high-speed response measuring equipments are preferable for this type of work.

10. *Automobile Measurements.*—The conditions under which an automobile radio receiver operates differ widely from those of a loudspeaker in a room. For this reason it is very important to test the performance under actual operating conditions. The response frequency characteristic should be obtained by placing the microphone at the ear position in each of the normal listening positions in the automobile. In the case of back-seat measurements persons should be seated in the front seat to simulate actual conditions. Measurements should be made with the windows open and closed. In general, the response frequency characteristics will differ widely for the front and back seats. It is customary to favor the front seats in determining the optimum response frequency characteristic. In some radio installations in automobiles an auxiliary loudspeaker is installed in the deck behind the rear seat. When this loudspeaker is used response should be obtained with both front and rear loudspeakers in operation. At high speeds, wind, road rumble, and engine noises are quite high and mask

[53] Wheeler and Whitman, *Proc. Inst. Rad. Eng.*, Vol. 23, No. 6, p. 610, 1935.

the reproduced sound. The power output should be sufficient to override these noises and give intelligible speech. In view of the fact that the sound level delivered by the loudspeaker is quite high under these conditions, it is important that the response frequency characteristic be smooth, otherwise the reproduced sound will be disagreeable.

The response frequency characteristics may be obtained with any equipment described in Sec. 10.3A2.

B. *Directional Characteristic.*—The directional characteristic of a loudspeaker is the response as a function of the angle with respect to some axis of the system. The characteristics may be plotted as a system of polar characteristics for various frequencies or as response frequency characteristics for various angles with respect to the reference axis.

The directional characteristics of a direct radiator loudspeaker in a very large baffle may be obtained at a distance of 5 feet. For a small baffle or cabinet the distance should be at least three times the largest linear dimension of the system. The directional characteristics of a horn loudspeaker should be obtained at a distance three or more times the largest dimension of the mouth.

Obviously, very slight reflections will introduce considerable error for angles in which the response is very low. For this reason, it is almost imperative that the measurements be made under free-field conditions.

Apparatus for obtaining the directional pattern of a microphone has been described in Sec. 10.2B and depicted in Fig. 10.6. The same system may be used to obtain the directional pattern of a loudspeaker. In this case the loudspeaker and microphone are interchanged, that is, the microphone is fixed and the loudspeaker rotated.

C. *Nonlinear Distortion Characteristic.*—The nonlinear distortion characteristic of a loudspeaker is a plot of the total distortion in per cent versus the frequency at a specified input power. A plot of the individual components of the distortion in per cent versus frequency is also used to depict the distortion characteristic of a loudspeaker.

The apparatus and circuit in schematic form for measuring the distortion produced by a loudspeaker are shown in Fig. 10.21. Great care must be taken to avoid appreciable harmonics in the sound generating and sound measuring equipment. To reduce the already low harmonic content in the signal generator to a negligible amount a variable cutoff low-pass electrical filter, admitting only the fundamental, should be employed. The microphone and amplifiers may be the same as those used for response measurements. The harmonic analyzer may be any of the various types employed in distortion measurements on amplifiers.

In making the test, the output of the power amplifier is connected to the loudspeaker. The sound is picked up by the microphone and then amplified and the measurement of harmonics is carried out in the conventional manner. The output switch is now thrown to the dummy load, the electrical resistance of which should be the same as the electrical impedance of the loudspeaker at the measurement frequency. The variable attenuator is adjusted until

the output of the microphone amplifier is the same as that obtained with the sound. The harmonic content under these conditions should be negligible. The purpose of this operation is to insure that no distortion is introduced by the associated measuring equipment. In the above discussion the possible distortion in the microphone has been neglected. The distortion generated by the microphone may be measured as outlined in Sec. 10.2C.
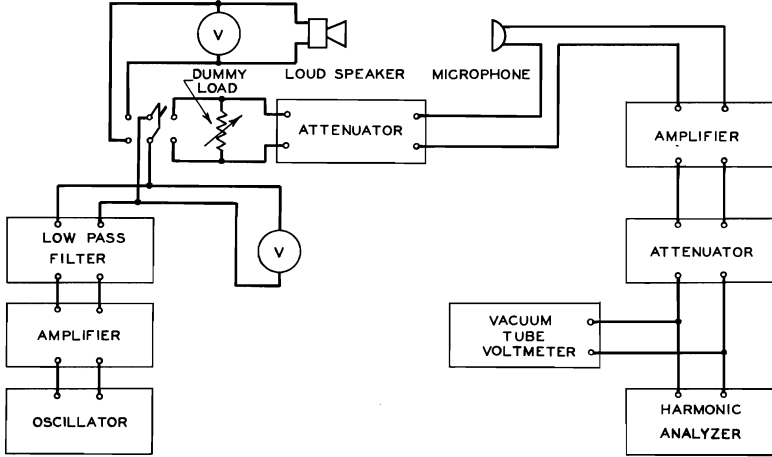
FIG. 10.21. Schematic arrangement of the apparatus for measuring the non-linear distortion of a loudspeaker.
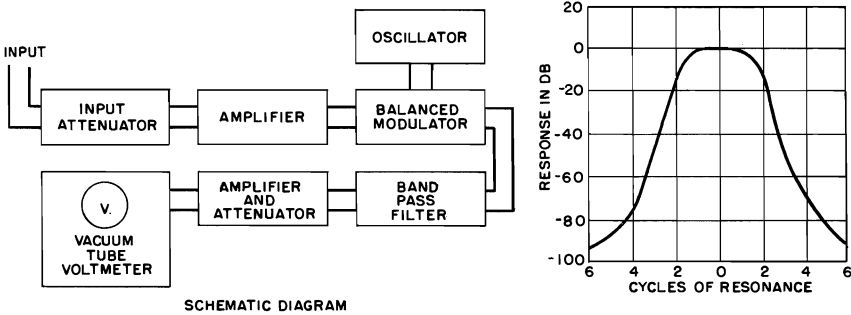
FIG. 10.22. Schematic arrangement of the elements in a heterodyne-type harmonic analyzer.

Harmonic distortion measurements should be made in a free-field sound room or outdoors to eliminate errors due to standing waves. If it is necessary to make these measurements in a room other than a free-field room they should be made under a sufficient variety of conditions with respect to frequency and microphone placements to give average values which are not appreciably affected by the errors associated with room reflections.

The heterodyne analyzer[54] is shown schematically in Fig. 10.22. The

[54] Arguimbau, L. B., *General Radio Experimenter*, No. 8, p. 1, June, July, 1933.

incoming signal, mixed with a carrier supplied by the heterodyne oscillator is fed to the modulator. A balanced modulator is usually used so that the carrier will be suppressed. The heterodyne oscillator is adjusted so that the sum of its frequency and that of one of the components of the signal equals the pass band of the highly selective tuned amplifier. The high selectivity is usually obtained by means of a quartz filter. The upper side band is passed through the selective tuned amplifier, detected and then measured on a meter.

A recording harmonic analyzer consists of a heterodyne harmonic analyzer coupled with a level recorder as shown in Fig. 10.23. The conventional harmonic analyzer is equipped with a dial calibrated in frequency. This dial can be coupled to the driving system of a high-speed level recorder. The
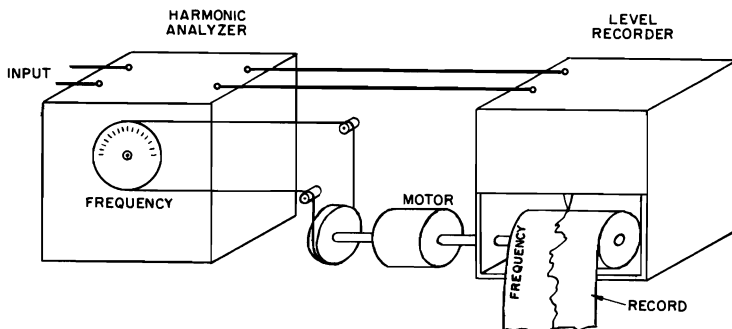
FIG. 10.23. Schematic arrangement of the apparatus used in a recording harmonic analyzer of the heterodyne type.

electrical output of the heterodyne harmonic analyzer is coupled to the input of the level recorder. In this manner the components in the complex wave input to the heterodyne harmonic analyzer may be automatically recorded by the high-speed level recorder.

A panoramic analyzer[55] is a system which presents the components of a complex wave input in the form of a response frequency characteristic on the screen of a cathode-ray tube with a persistence image screen. Fig. 10.24. The system is basically the heterodyne harmonic analyzer of Fig. 10.22. The output of a low-frequency oscillator is converted into a saw-tooth wave by the saw-tooth generator. This wave is amplified by the horizontal deflection amplifier. The output of the horizontal deflection amplifier is coupled to the horizontal deflection system of the cathode-ray tube. The saw-tooth generator is fed also to the controlled oscillator. This oscillator supplies the heterodyne signal to the modulator. The complex wave signal is amplified and fed to the modulator. The output of the modulator passes through a tuned amplifier with a narrow band-pass charac-

[55] Richard, Smith, and Stephens, *Trans. IRE*, Prof. Group Audio, Vol. AU-3, No. 2, p. 37, 1955.

teristic. The output of the tuned amplifier is amplified by the vertical deflection amplifier. The output of the vertical deflection amplifier is coupled to the vertical deflection system of the cathode-ray tube. The characteristic depicted on the cathode-ray tube represents the output as a
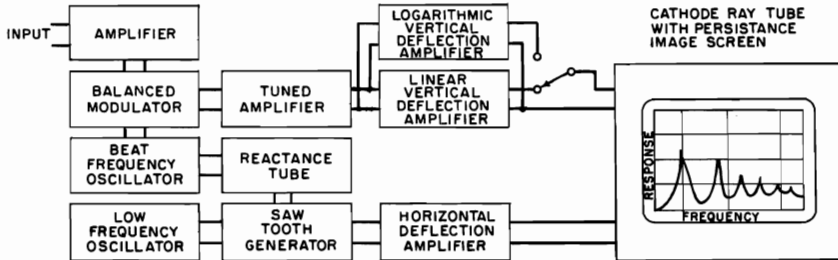
FIG. 10.24. Schematic arrangement of the elements used in a heterodyne analyzer in which the components of a complex wave are depicted as a response frequency characteristic on a cathode-ray tube.

function of the frequency. The vertical scale may be either linear or decibel. The frequency scale is logarithmic.

The balance bridge for measuring the total distortion is shown schematically in Fig. 10.25. A part of the output of the oscillator is fed to the apparatus to be tested and another part to the analyzer. The amplitude and phase relations of the fundamentals from the oscillator and apparatus
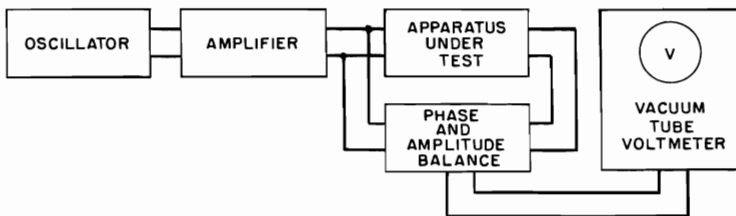
FIG. 10.25. Schematic arrangement of the elements in a balance bridge harmonic analyzer.

to be tested are adjusted by means of suitable networks so that none of the fundamentals remains. The remainder is the total harmonic generated by the system under test. This is measured by means of a root-mean-square meter.

A cathode-ray oscillograph is often used to depict the wave form and thus obtain an indication of the departure from a pure sine wave. A schematic diagram of the apparatus employing a cathode-ray oscillograph to indicate the extent to which distortion is introduced by a loudspeaker is shown in Fig. 10.26. When the switch is thrown to the right and the sine wave generated in the oscillator is sent into the amplifying system through the attenuator, the cathode-ray oscillograph should show a pure sine wave

form over the entire audio-frequency range considered. The attenuator is adjusted to give the same amplitude of the wave pattern on the oscillograph screen as is secured when the switch is thrown to the left and the power is supplied to the loudspeaker. With the switch in the latter position the microphone picks up the sound and the wave form is reproduced upon the fluorescent screen of the oscillograph. The departure from the pure sine
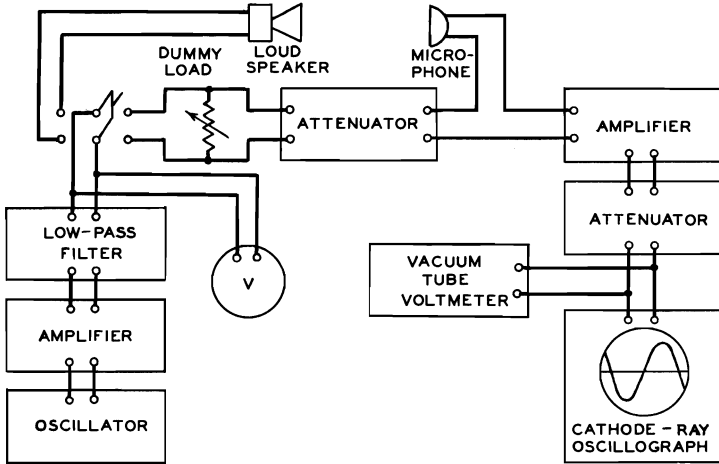
Fig. 10.26. Schematic arrangement of the apparatus employing a cathode-ray tube for indicating the nonlinear distortion of a loudspeaker.
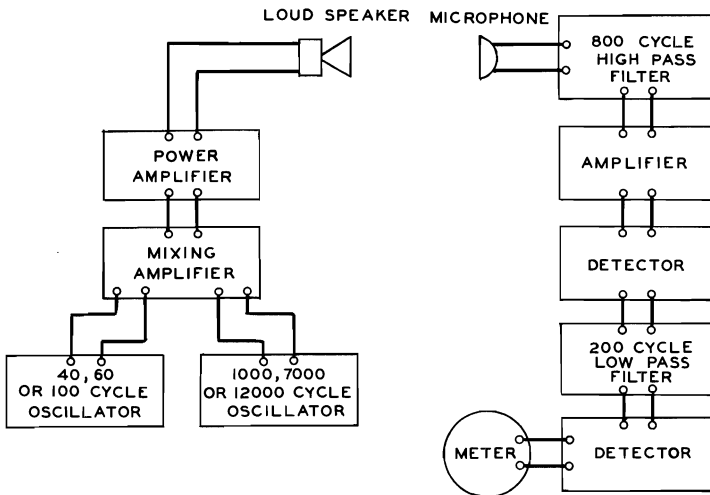
Fig. 10.27. Schematic arrangement of the apparatus for measuring the nonlinear distortion of a loudspeaker employing the intermodulation method. (After Hilliard.)

wave is indicated readily by the difference in appearance of the pattern from the pure sine wave form secured with the switch thrown to the right. The extent of introduction of harmonics by the loudspeaker can be estimated from a slight, moderate or very marked change in the wave form.
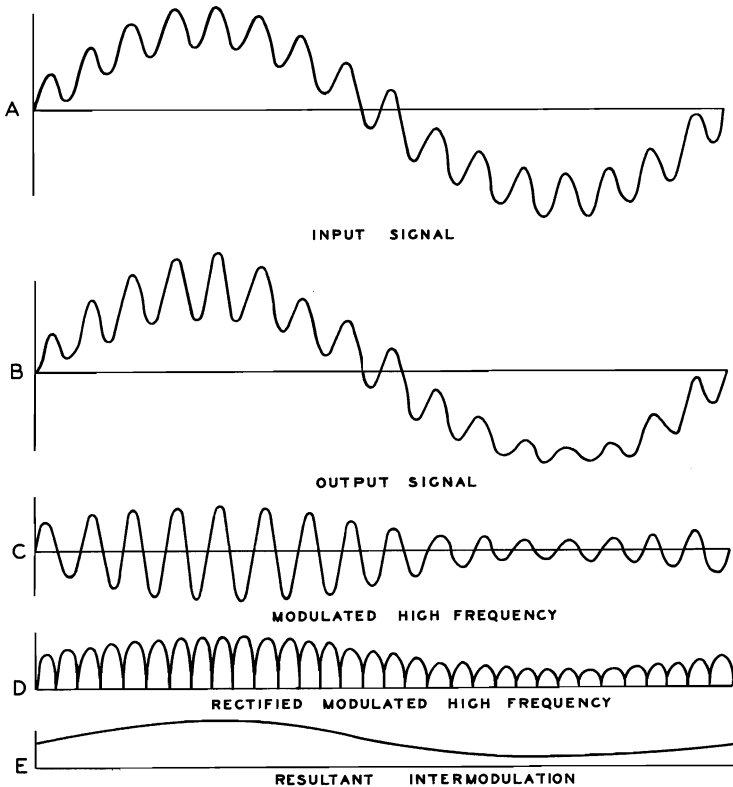
FIG. 10.28. Typical wave shapes in the various stages of the inter-modulation system of distortion measurement. A. The input signal to the loudspeaker. B. The output signal of the loudspeaker. C. The modulated high-frequency output of the band-pass filter. D. The recti-fied modulated high frequency of the detector. E. The resultant intermodulation output of the copper oxide rectifier.

A schematic diagram of the apparatus for the intermodulation method[56] of measuring nonlinear distortion is shown in Fig. 10.27. Two tones are impressed upon the loudspeaker to be tested. The low-frequency tone may be 40 or 60 or 100 cycles and the high-frequency tone may be 1000 or 7000 or 12,000 cycles. The wave shape of the input signal to the apparatus under test is shown in Fig. 10.28A. The output of the microphone is

[56] Hilliard, J. K., *Proc. Inst. Rad. Eng.*, Vol. 29, No. 12, p. 614, 1941.

shown in Fig. 10.28B.   This output is fed to an 800-cycle high-pass electrical filter.   If nonlinear distortion is produced by the equipment under test, the high-frequency output from the electrical filter will be modulated as shown in Fig. 10.28C.   Beyond the electrical filters the signal is amplified and impressed upon a full-wave detector.   The output of the detector is shown in Fig. 10.28D.   The output of the detector is passed through a 200-cycle low-pass electrical filter.   The output of the low-pass electrical filter is shown in Fig. 10.28E.   The output of the 100-cycle low-pass electrical filter is fed through an amplifier which removes the d-c electrical component.   The final resulting a-c electrical intermodulation component is measured by means of a copper oxide rectifier meter.   An approximate relation between the intermodulation and harmonic terms may be developed. It appears that, in general, the intermodulation terms are approximately four times the harmonic terms.   For example, if certain apparatus is found to have 1 per cent total distortion in harmonics, an intermodulation test will show intermodulation products of 3 to 4 per cent when the amplitude of the higher frequency is 12 db below the amplitude of the lower frequency.

A more universal method[57] for intermodulation testing consists of three radio frequency signal generators, one fixed and two variable.   The system can provide one frequency, f_1 from 20 to 20,000 cycles and a second frequency f_2 higher than f_1 by a fixed amount which may be anywhere between 0 and 10,000 cycles.   As the frequency f_1 is varied, the difference frequency remains constant.   The system can provide one fixed frequency and a variable higher frequency.   The analyzer of Fig. 10.22 may be used as the detector.

The automatic nonlinear distortion analyzer[58] consists of the conventional system for obtaining a response frequency characteristic of a loudspeaker coupled with an automatic means for suppressing the fundamental frequency, see Fig. 10.29.   The loudspeaker is supplied by a pure tone from a low distortion oscillator and power amplifier combination.   The sound output of the loudspeaker is picked up by a calibrated microphone.   Both the loudspeaker and the microphone are located in a free-field room.   The output of the microphone is amplified and fed to a recorder, and a response frequency characteristic of the loudspeaker is obtained from this system in the conventional manner.   To obtain the distortion frequency characteristic which depicts the distortion generated by the loudspeaker as a function of the frequency, the system for automatically suppressing the fundamental is connected between the microphone, amplifier, and the recorder.   Under these conditions, the voltage applied to the recorder is the root-mean-square total of the harmonic frequencies generated by the loudspeaker.

In the system described above, the major problem becomes one of attenuating the fundamental frequency in a dependable and fairly rapid manner. Of the several methods available for eliminating the fundamental, a reliable and straightforward one is shown in Fig. 10.29.   This consists of a series of

[57] Peterson, A. P. G., *Gen. Radio Exp.*, Vol. 25, No. 3, 1950.
[58] Olson and Pennie, *RCA Review*, Vol. 12, No. 1, p. 35, 1951.

high-pass filters, sequentially interposed between the microphone pickup and the recording equipment to attenuate the 40- to 15,000-cycle sweep fundamental. The primary advantage of using this method for harmonic distortion measurements is its dependability. The filters themselves may be made very rugged. Furthermore, should the filter switching circuit fail to function properly, the distortion readings will immediately go to 100 per cent, thus reading fundamental rather than harmonic, and thereby providing a positive check against a possible switching error.

The useful frequency range of each filter is determined by two frequencies, namely, f_c and f_∞. The frequency at which the response is down one
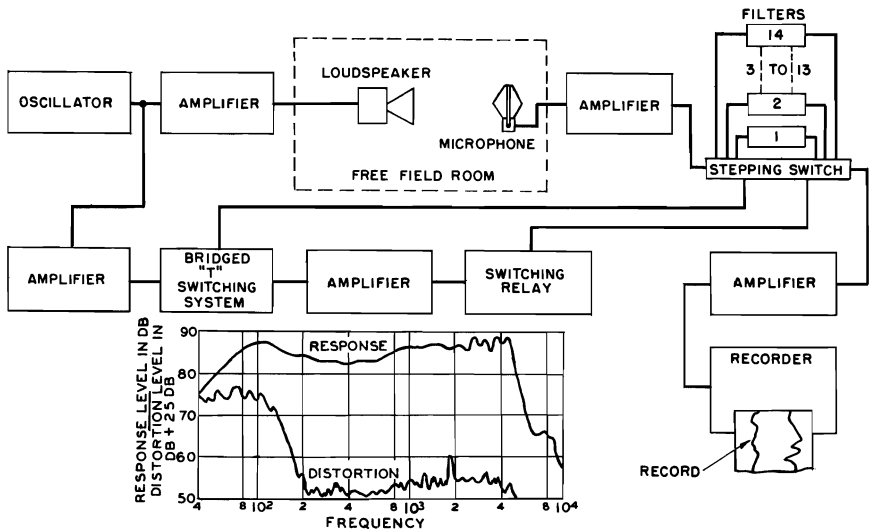
FIG. 10.29. Schematic arrangement of the apparatus employed in the automatic non-linear distortion recording system. The graph shows the response-frequency characteristic and the distortion frequency characteristic of a typical commercial 12-inch loudspeaker.

decibel is f_c. This frequency sets the lower limit of the useful pass band of each high-pass filter when recording distortion to an accuracy of 10 per cent. The frequency at which the response is down sixty decibels is f_∞. This frequency sets the upper limit of the useful rejection band of each filter when recording distortion to an accuracy of 10 per cent for a 0.3 per cent, 2nd harmonic distortion value. The response frequency characteristics of the filters are shown in Fig. 10.30. The f_c and f_∞ overlap characteristic of adjacent filters are very close at the lower frequencies; and it, therefore, becomes very important that the frequency at which a filter is switched be held to a close frequency tolerance if the full possible accuracy of the distortion analysis is to be realized. For this reason, an electronic rather than a mechanical system for the detection of the switch frequency is used. A

typical bridge-T network employed for switch frequency detection is shown in Fig. 10.31.   The response frequency characteristic of the network is also shown in Fig. 10.31.

For a loudspeaker distortion analysis a conventional amplitude versus frequency response curve is first run with the power amplifier adjusted to furnish the proper power level to the loudspeaker under test, and with the distortion analyzer step switch solenoid power turned off.   The recorder preamplifier is adjusted to a level such that the recorder will not go off scale for the response curve.   The response frequency characteristic is run with this gain setting.   This procedure is repeated with the distortion analyzer
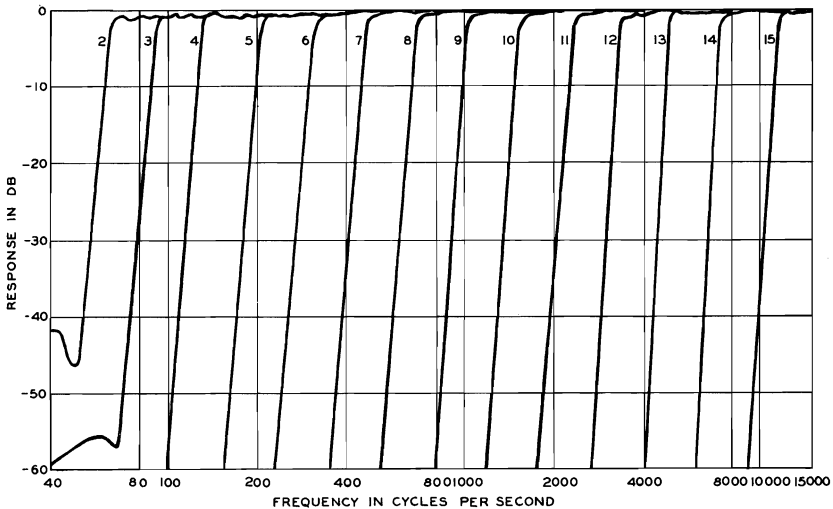
FIG. 10.30.   Response frequency characteristics of the fourteen high-pass filters used in the system of Fig. 10.29.

step switch solenoid power turned on and with the gain control of the pre-amplifier turned some 20 to 40 decibels higher.   The resultant character-istics, with due consideration for the difference in preamplifier settings, gives the relative distortion frequency characteristic of the loudspeaker under test, see Fig. 10.29.

D. *Efficiency Frequency Characteristic.*[59]—The efficiency of a loudspeaker at any frequency is the ratio of the total useful acoustical power radiated to the electrical power supplied to the load, the current wave of which exercises a controlling influence on the wave shape of the sound pressure. The plot of efficiency, in per cent, versus frequency, in cycles, is termed the efficiency frequency characteristic.

The measurement of efficiency of a loudspeaker may be divided into two methods, direct and indirect.   One direct method depends on measuring the total energy flow through a spherical surface without reflections.   Several

[59] Standards on Electroacoustics, Institute of Radio Engineers, 1938.

indirect methods have been developed. The most common of these consists in measuring the electrical impedance under various conditions of diaphragm loading. It has been found in practice that these two methods of determining efficiency are those most widely used at the present time.

1. *Direct Determination of Radiated Power.*—The sound power output from a speaker at a particular frequency may be obtained by measuring the total flow of sound power through a spherical surface of which the sound source is the center. The surface of the sphere is divided into incremental areas and the power transmitted through each area is determined from the sound pressure and the particle velocity as well as the phase displacement
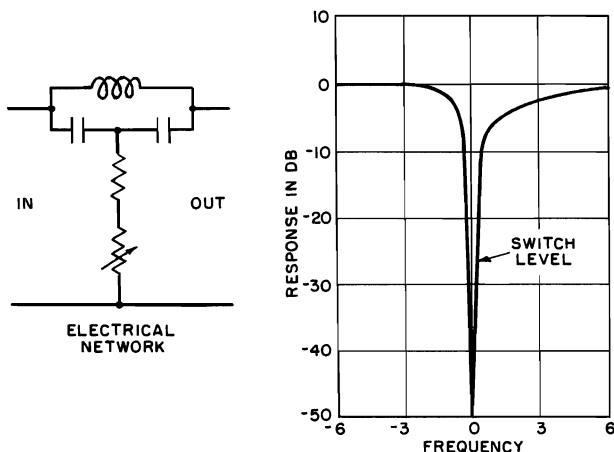
FIG. 10.31. The electrical diagram and response frequency characteristic of the bridged T network used for frequency detection in the switching system of Fig. 10.29.

between them. To simplify the process, the measurements may be made at a distance sufficiently large so that these quantities are in phase. Then, the radiated power may be determined by measuring the sound pressure or particle velocity over each incremental area (assuming the measuring equipment does not disturb the sound field and no standing wave pattern exists). The total power is equal to the summation of the power transmitted through the incremental areas and may be expressed as

$$P_A = \frac{1}{\rho c} \int \int p^2 \, dS \times 10^{-7} \qquad 10.31$$

where P_A = total acoustical power, in watts,

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

c = velocity of sound in medium, in centimeters per second,

p = root-mean-square pressure, in dynes per square centimeter, over the element of areas dS, and

dS = element of area on spherical surface, in square centimeters.

The input electrical power can be determined from the electrical current, voltage, and phase angle, while operating under the above conditions.

The efficiency, μ, in per cent, is then

$$\mu = \frac{P_A}{P_E} \times 100 \qquad\qquad 10.32$$

where P_A = total acoustical output, in watts, and

P_E = electrical input, in watts.

As previously mentioned, the loudspeaker should be located so that the reflected energy reaching the measuring equipment is negligible. This means that the measurements must be made either in a free-field sound room
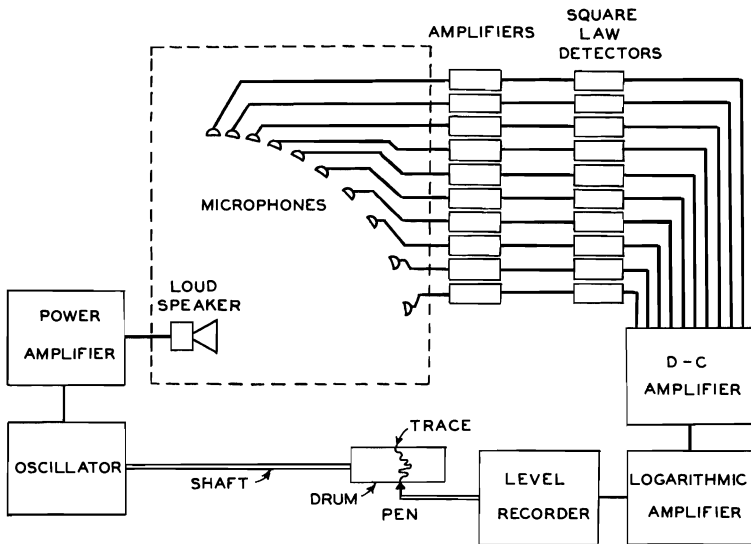
FIG. 10.32.   Schematic arrangement of the apparatus for obtaining the total sound power output frequency characteristic of a loudspeaker.

or in free space. The measurements and computations in this method are quite laborious. On the other hand, there can be no question as to the validity of results which are obtained if the test is carefully conducted. Because of its fundamental nature and validity, the direct method is usually considered standard for determining loudspeaker efficiency.

The procedure outlined above is quite laborious and time consuming. Apparatus has been developed in which the total integrated power output frequency characteristic of a loudspeaker can be obtained in a manner comparable to that of a response frequency characteristic.

The schematic arrangement of the apparatus used for obtaining the total output frequency characteristic of a loudspeaker is shown in Fig. 10.32. The total power output is depicted by a single curve on a graph sheet. The

ordinate scale is in decibels. This apparatus approximates the integration process of equation 10.31. The microphones are placed on the quadrant of a circle and arranged to intercept equal areas on the surface of a hemisphere. The measurement assumes that the directional pattern is symmetrical about the axis of the loudspeaker. If the pattern is unsymmetrical, the loudspeaker is mounted in a cradle and continuously rotated about the axis. The measurement covers one hemisphere. A similar measurement can be made in the other hemisphere if the radiation in the backward direction is of any consequence.

2. *Indirect Determination of Radiated Power.*—There are several methods for determining loudspeaker efficiency by indirect means. The most common method is to measure the electrical impedance under various conditions of diaphragm loading.

A one-to-one ratio bridge, capable of measuring the electrical impedance at the full power output of the speaker, should be used. Care should be taken that the temperature of the voice coil does not vary appreciably during the various measurements. The power supply for driving the speaker and bridge should be reasonably free from harmonic distortion.

The motional electrical impedance method[60] is generally applied to moving coil electrodynamic speakers in which the force factor is real. In case the force factor is imaginary it becomes rather complicated to employ the motional electrical impedance method.

The efficiency μ, in per cent, by the motional electrical impedance method is given by

$$\mu = \frac{r_{EM}}{r_{EN}} \times 100 \qquad 10.33$$

where $r_{EM} = r_{EN} - r_{ED}$ motional electrical resistance, in ohms,

$\quad r_{EN}$ = resistive component of the electrical impedance with the system in the normal state, in ohms, and

$\quad r_{ED}$ = damped electrical resistance with the vibrating system blocked, in ohms.

This equation describes the simplest method of determining the efficiency from motional electrical impedance measurements when the electromechanical coupling factor is real (see Chapters VI and VII). It assumes that the entire value of the motional electrical resistance may be attributed to radiation acoustical resistance. This method adds the radiation from both sides of the diaphram and, therefore, assumes that the radiation from both sides is useful. It assumes that there are no mechanical losses in the diaphragm and suspension system. These losses can be determined from the measurements of the motional electrical impedance in a vacuum. Of course, in this case, the load on the diaphragm is not normal and the losses may be quite different from those which obtain under actual operating

[60] Kennelly and Pierce, *Proc. A. A. A. S.*, Vol. 48, No. 6, 1912.

conditions.   This method also assumes that there are no losses due to viscous air friction.   Since the amplitude of the vibration of a voice coil is normally small at the higher frequencies, the problem of blocking the voice coil against motion is not a simple matter.   Obviously, any motion will introduce an error in the determination of the efficiency.

E. *Phase Distortion Characteristic.*—The phase distortion characteristic of a loudspeaker is a plot of the phase angle between the sound output and some reference sound as a function of the frequency.

Two microphones and separate amplifiers and a cathode oscillograph may be used as outlined in Sec. 10.2D, Fig. 10.8.   A reference sound may be set up by a separate loudspeaker, in which the phase shift is small, and
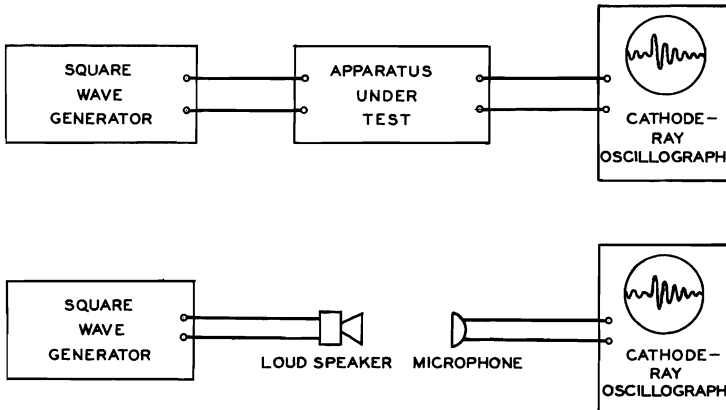
Fig. 10.33.   Schematic arrangement of the apparatus employing a square wave generator and a cathode-ray tube for indicating the transient response characteristics of acoustical apparatus such as microphones or loudspeakers.

picked up by one microphone.   A reference voltage source may be substituted for the reference microphone.   The sound from the loudspeaker to be tested may be picked up on the other microphone.   The phase difference may be determined as outlined in Sec. 10.2D.   The phase distortion is of importance in the overlap region of the multiple channel systems.   In this case the phase shift may be several hundred degrees (equivalent to a sound path difference of several feet), see Sec. 7.4B.

F. *Electrical Impedance Frequency Characteristic.*—The electrical impedance characteristic of a loudspeaker is the electrical impedance at the input terminals as a function of the frequency.   The plot of the characteristic should also include the resistive and reactive components of the electrical impedance.

A one-to-one ratio electrical impedance bridge may be used and should

be capable of measuring the electrical impedance at the full power output of the speaker. The power input should be included with every electrical impedance characteristic. If the electrical impedance characteristic varies with power input, it is desirable to show a series of electrical impedance frequency curves for various inputs. Other methods may be used as, for example, the three voltmeter and a known electrical resistance method.

G. *Transient Response Characteristic.*—The measurements in the preceding sections have been concerned with steady state conditions. In all types of sound reproduction the phenomena is of a transient character. For this reason it is important to measure the response of the system to a suddenly applied force or voltage. The Heaviside Operational Calculus is a very powerful tool for predicting the performance of a system to a suddenly applied force or voltage (see Sec. 6.25.)

The apparatus for investigating the transient response of an audio system is shown schematically in Fig. 10.33. The output of a square wave generator is fed to the apparatus to be tested. The output of the apparatus under test
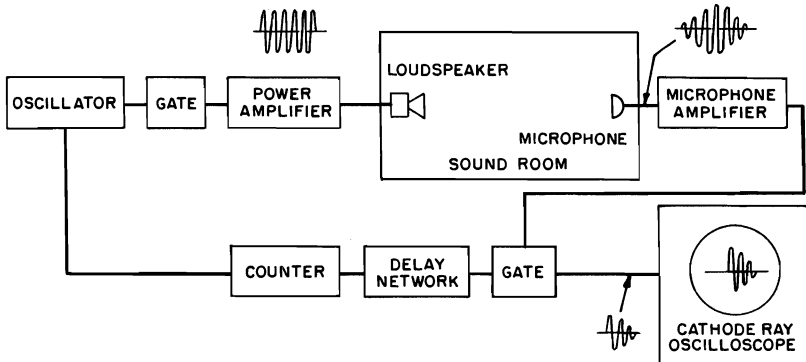
FIG. 10.34. Schematic arrangement of the apparatus used in the tone burst system for depicting the transient response of a loudspeaker.

is fed to a cathode-ray oscillograph. The deviation from the square wave is shown on the screen of the cathode-ray oscillograph. Square waves[61] offer a simple and rapid method of including both phase shift and amplitude response in a single test.

A measure of the transient response of a loudspeaker may be obtained by measuring the response to an electrical input in the form of a tone burst.[62] A tone burst is a term usually used to designate a wave with a rectangular envelope. A schematic diagram[63] of the apparatus for measuring the response of a loudspeaker to an electrical input in the form of a tone burst is shown in Fig. 10.34. An oscillator provides the sine wave signal which is

[61] Kallmann, Spencer, and Singer, *Proc. Inst. Rad. Eng.*, Vol. 33, No. 3, p. 169, 1945.
[62] Olson, H. F., *Audio Eng.*, Vol. 34, No. 10, p. 15, 1950.
[63] Corrington, M. S., *Jour. Audio Eng. Soc.*, Vol. 3, No. 1, p. 35, 1955.

fed to an electronic gate which interrupts the sine wave signal at regular intervals, thereby producing a series of tone bursts. The tone bursts are amplified and fed to the loudspeaker. The sound output from the loudspeaker is picked up on the microphone and amplified and applied to the vertical deflection system of a cathode-ray oscilloscope. The acoustical output from the loudspeaker may be compared with the electrical input to the loudspeaker. Another gate may be employed so that the residue after the electrical input has ceased may be displayed on the oscilloscope. A delay network is required to compensate for the time required for the wave to travel from the loudspeaker to microphone.

H. *Subjective Measurements.*[64]—A subjective test of a loudspeaker involves a determination of some of the performance characteristics by direct listening to the loudspeaker operating under controlled program and environment conditions. Listening tests play an important part in research, development, and commercialization of loudspeakers. Listening tests range in scope from exceedingly simple comparison tests to elaborately controlled and conducted tests. In many cases the apparatus for making all the objective tests outlined in the preceding sections are not available. Furthermore, there is always some difficulty in evaluating the objective measurements. For this reason a subjective test of efficiency, frequency response, directional characteristics, nonlinear distortion, and transient response, in which two or more loudspeakers are compared with each other or with a reference loudspeaker, is widely used. The subjective test may be used to determine the lumped effects of the following factors: loudness, frequency range, tone balance, spacial distribution, quality, and hangover. It is the purpose of this section to describe listening tests of loudspeakers.

1. *Loudspeaker Environment.*—The listening test of a loudspeaker should be conducted in the environment for which it was designed to operate. Specifically, a loudspeaker designed for home-type radio receivers, phonographs, and television receivers should be tested in a room with dimensions and acoustics similar to those of the average living room in the home. A loudspeaker for an automobile radio receiver should be tested in an automobile. A loudspeaker for a sound motion-picture system should be tested in a typical theater. A loudspeaker for a public address, sound reinforcing, or paging system should be tested under typical surroundings for these systems.

2. *Loudspeaker Housing, Placement, and Mounting.*—The horn, baffle, housing, or cabinet for loudspeaker listening tests should be similar to those used under actual operating conditions. The placement and mounting arrangement in the test invironment should correspond to those used in actual installations.

3. *Signal Sound Level.*—The signal sound level produced by a loudspeaker in a listening test should correspond to the sound level under actual operating conditions in the field. The use of the proper level is very important in

[64] Olson, H. F., *Trans. IRE*, Prof. Group on Audio, Vol. AU-1, No. 5, p. 7, 1953.

determining the balance of high, mid, and low frequencies, the distortion, the transient response, etc., under actual operating conditions. The upper sound levels in the description which follows do not necessarily represent the upper power capabilities of the systems. The signal sound level will be somewhere between 65 and 75 db for a radio receiver, phonograph, or television receiver operating in a typical or average living room. The signal sound level will be somewhere from 65 to 100 db for an automobile radio receiver. The signal sound level will be between 76 to 85 db for speech reproduction, and 75 to 95 db for music reproduction for a sound motion-picture reproducing system operating in a theater. The signal sound level for a public address, sound reinforcing, or paging system will vary over wide limits depending upon the application. To summarize, the signal sound level of the test should correspond to the level under actual operating conditions.

4. *Ambient Noise Level.*—The ambient noise under which the listening test is conducted should correspond to the ambient noise encountered under actual conditions. This involves two main factors, namely, the sound level and spectrum of the ambient noise. For example, the average ambient noise sound level in the average living room is 42 db. The average ambient noise sound level in a theater is also 42 db. In an automobile, the ambient noise sound level depends upon the speed, open or closed windows, the road, etc. In public address, sound reinforcing or paging applications, the noise sound level will vary over wide limits. This must be taken into account, and the noise conditions under which the equipment will be operated must be simulated in the listening tests. It is important that the spectrum of the noise encountered under the actual operating conditions should be simulated in the subjective tests as well as the noise level.

5. *Signal or Program Material.*—The signal or program material used in listening tests should be similar to that encountered in the field. This is not so for the material presented under "Frequency Range" and "Power Handling Capacity." A radio or television receiver should be operated from typical broadcast or television transmitters. Under certain conditions it may be necessary to use the equivalent of a radio or television transmitter, as, for example, a modulated signal generator. A phonograph should be operated from typical commercial records. A sound motion-picture reproducing system should be operated from typical sound motion-picture film. Sound reinforcing systems for use with music should be tested with musical program material. Public address and paging systems should be tested with speech as the program material.

6. *Reference Systems.*—Almost all listening tests on loudspeakers are conducted by comparing the loudspeaker under test with a reference loudspeaker. The reference system is, in general, a loudspeaker which is similar to the loudspeaker under test. The loudspeakers should be placed behind a light-opaque, sound-transparent curtain so that it is impossible to identify the loudspeakers by sight. A suitable indicator should show which loudspeaker is operating at any time. In general, the procedures in most listening

tests are not formalized because the tests are conducted to determine the engineering and commercial aspects of rather small changes in design. If a jury-type procedure is used, secret ballots should be taken of the preference. Statistical methods should be employed in planning and conducting such jury tests.

7. *Relative Loudness Efficiency*.—The relative loudness efficiency of a loudspeaker is determined from a loudness balance. High-quality transformers should be used to match each loudspeaker to the appropriate impedance. In some cases it is desirable to include the driving means in determining the efficiency, because this is important in any practical design. The input to the loudspeakers should be adjusted so that the loudness levels of all loudspeakers are the same. The attenuation required to adjust to the same loudness gives a measure of the relative loudness efficiency. In these tests, the observers should move around to different locations to insure that no advantages are given to any loudspeaker due to a better listening location. For the same reason, the locations of the loudspeakers should also be interchanged.

8. *Relative Directivity*.—The relative directivity of a loudspeaker is determined by listening at observation points removed from the axis. In order to reduce the effect of the difference in the angle during a comparison, the following precautions should be observed: Only two loudspeakers should be used at a time. The loudspeakers should be placed as close together as possible. The position of the two loudspeakers should be interchanged during the test. In determining the relative directivity, listening tests should be conducted along different angles with respect to the axis. This test indicates the loss in loudness level and frequency discrimination for observation points removed from the axis.

9. *Frequency Range*.—The approximate frequency ranges of loudspeakers may be determined from listening tests by employing program material which has a wider frequency range than the loudspeaker under test in combination with calibrated high- and low-pass filters introduced between the program source and the loudspeaker. It is very important that the program material contain adequate frequency components in both the high- and low-frequency ranges and thereby insure reliable results. The approximate frequency range can be determined by noting the settings of the filters for which there is no appreciable frequency discrimination, as determined by the quality of reproduction. The filters should have at least three cutoff steps per octave.

10. *Power Handling Capacity*.—The power handling capacity of a loudspeaker may be determined by employing a low distortion program source capable of overloading the loudspeaker without introducing distortion in the program source which is fed to the loudspeaker. The frequency range of the system which feeds the loudspeaker should be restricted by means of filters to correspond to that of the loudspeaker under test. The power level at which the distortion becomes intolerable may be considered to be the power handling capacity of the loudspeaker. In this connection intolerable

distortion depends upon the application in which the loudspeaker is to be used. This requires a high order of judgment by the listener.

The test outlined above for determining the power handling capacity may appear to be oversimplified in view of the many factors involved. For example, the power handling capacity of a loudspeaker may be determined by failure of the diaphragm, the suspension system, the voice coil structure, and heating of the voice coil. Of course, all these forms of failure will be manifested as intolerable distortion. Again it should be emphasized that the crux of this test is the determination of what is considered intolerable distortion.

11. *Response Frequency Contour.*—In most completely integrated systems, such as radio and television receivers and phonographs, there are distinct economic and technical advantages in employing components which individually do not exhibit a uniform response frequency characteristic but taken collectively do exhibit a uniform response frequency characteristic. In these applications, listening tests are very useful in checking the objective measurements for a proper balance of the frequency characteristic. This type of listening requires great skill obtained through practice. A reference system which is known to be acceptable is almost a necessity in tests of this type.

12. *Nonlinear Distortion.*—Loudspeakers are used with other components in a sound reproducing system. Therefore, in a properly integrated system the limitations upon the allowable nonlinear distortion of each element depends upon the allowable distortion of the system as a whole. For example, it would be technically and economically unsound to use a wide-range, high-quality loudspeaker in a reproducing system in which the components in the remainder of the system were of much lower quality. The quality of the loudspeaker required for the application can be determined from listening tests of loudspeakers of various degrees of quality. In this way it is possible to determine the loudspeaker which introduces distortion of such magnitude as to be perceptible above the distortion of the remainder of the system.

13. *Transient Response.*—Since all speech and music are of a transient character, the transient response is another important characteristic which depicts the performance of a loudspeaker. Poor transient response leds to fuzzy reproduction with poor definition. As a result the character of speech and music is destroyed. In objective measurements, a deviation in the sound output from the rapid growth and decay characteristic of an applied tone burst depicts the transient response of a loudspeaker. See Sec. 10.3G. A loudspeaker with a nonuniform response frequency characteristic will exhibit poor transient response. A subjective evaluation of the transient response of a loudspeaker can be obtained by listening to the reproduction of such sounds, as piano, wood blocks, and other sounds with rapid growth or decay characteristics. The growth and decay will not be clean but fuzzy when the transient response of the loudspeaker is poor. If there is a peak in the response frequency characteristic in the low-frequency range,

considerable hangover will be apparent in the reproduction of low-frequency sounds.

14. *General Aspects.*—In most conventional, mass-produced, complete sound reproducing systems, technical compromises must be made in order to obtain a product which is commercial from economic considerations. The principal factors which are involved from a subjective standpoint are frequency range, response frequency contour, directivity, nonlinear distortion, power handling capacity, and noise. For example, the objectionable effects of non-linear distortion and noise are reduced as the high-frequency cutoff is reduced. On the other hand, some of the naturalness of a restricted frequency range system can be regained by a change of the response frequency contour. Listening tests are very useful for obtaining a practical compromise between these factors.

10.4. Testing of Telephone Receivers.—The characteristic of a telephone receiver should show the performance as normally worn on the ear. The sound intensity produced in the ear by the telephone receiver should
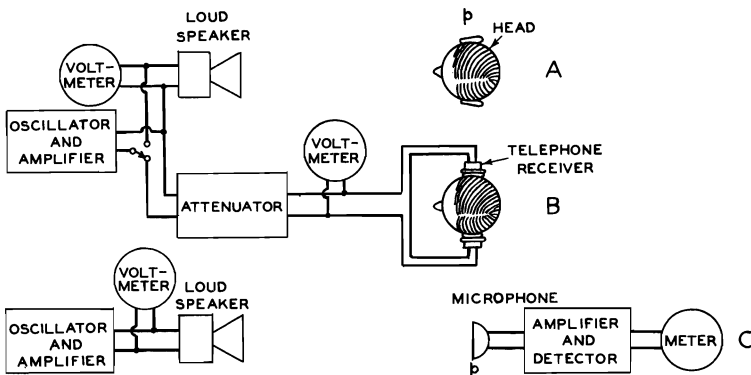
FIG. 10.35. Schematic arrangement of the apparatus for obtaining the subjective response frequency characteristic of telephone receivers. In A, the observer listens to the sound produced by the loudspeaker. In B, the observer adjusts the attenuator until the sound level produced by the telephone receivers appears to be equal to the sound level produced by the loudspeaker. In C, a sound level meter is used to measure the sound pressure produced by the loudspeaker.

be the same as the intensity produced in the ear when the head is immersed in the original sound field. There are two types of measurements upon telephone receivers, namely, subjective and objective.

A. *Subjective Measurements.*—The schematic arrangement for obtaining a subjective response frequency characteristic[65,66] of telephone receivers is shown in Fig. 10.35. A free progressive sound wave is established by means of a loudspeaker driven by an oscillator and power amplifier. The test

[65] Olson and Massa, *Jour. Acous. Soc. Amer.*, Vol. 6, No. 4, p. 250, 1935.
[66] Olson, H. F., *Jour. Soc. Mot. Pic. Eng.*, Vol. 27, No. 5, p. 537, 1936.

should be conducted in a free-field sound room (see Sec. 10.3A4). With the receivers removed the observer listens to the sound produced by the loud-speaker as shown in Fig. 10.35A. Next the observer places the receivers on his ears and the output of the oscillator and amplifier is transferred from the loudspeaker to the headphones by throwing the switch and reducing the power by means of a suitable attenuator as shown in Fig. 10.35B. The voltage across the receivers is adjusted until the intensity in the observer's ear seems to be the same as the free-wave intensity produced by the loud-speaker. The sound pressure produced by the loudspeaker at the distance of the observer is measured by means of a calibrated microphone, amplifier, and meter combination as shown in Fig. 10.35C. This procedure is repeated at several frequencies. The absolute response of the telephone receiver is given by equation 10.27, where p is the free-wave sound pressure, e is the voltage across the telephone receivers when a balance is obtained, and z_E is electrical impedance of the telephone receivers.

B. *Objective Measurements.*—1. *Artificial Ear.*[67,68,69,70,71]—The acoustical impedance frequency characteristic looking through the ear cap of a tele-phone receiver as normally worn has been investigated by Inglis, Gray, and
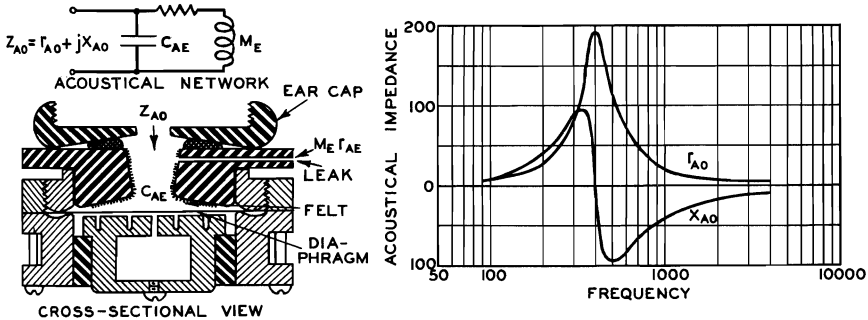
FIG. 10.36. A cross-sectional view of an artificial ear employing a standard condenser microphone and the acoustical network of the acoustical system. The graph shows the resistive, r_{A0}, and reactive x_{A0}, components of the acoustical impedance, looking into the aperture of the ear cap, as a function of the frequency. C_{AE} = acoustical capacitance of the cavity. M_E and r_{AE} = inertance and acoustical resistance of the leak.

Jenkins. This is shown in Fig. 10.36. An artificial ear and the acoustical network which yields approximately the same acoustical impedance charac-teristic are shown in Fig. 10.36. A standard condenser microphone is used to measure the pressure. A series of slits corresponding to the leak between the ear cap and the ear are represented by the inertance, M_E, and acoustical resistance, r_{AE}. The walls of the cavity, C_{AE} (4 cubic centimeters), are

[67] Inglis, Gray and Jenkins, *Bell Syst. Tech. Jour.*, Vol. 11, No. 2, p. 293, 1932.
[68] Olson and Massa, *Jour. Acous. Soc. Amer.*, Vol. 6, No. 2, p. 250, 1935.
[69] Olson, H. F., *Jour. Soc. Mot. Pic. Eng.*, Vo. 27, No. 5, p. 537, 1936.
[70] Romanow, F. F., *Jour. Acous. Soc. Amer.*, Vol. 13, No. 3, p. 294–1942.
[71] American Standards Association, "Coupler Calibration of Earphones," Z24.9, 1949.

lined with felt to reduce resonances at the high frequencies. The response frequency characteristic obtained upon the artificial ear, in general, agrees quite well with the subjective tests.

The tests outlined for loudspeakers may be performed upon telephone receivers by employing the artificial ear. The same apparatus as described in Sec. 10.3A2 may of course be used and will not be repeated here.

The pressure delivered by a telephone to a closed cavity as a function of the frequency is sometimes used to depict the response of a telephone receiver. The artificial ear shown in Fig. 10.36 may be used for this purpose by closing the slits.

2. *Artificial Mastoid.*—The artificial mastoid is a system for objectively measuring the response of a bone conduction telephone receiver. In one form the artificial mastoid[72,73] consists of a rubber block having approximately the same acoustical impedance as the human head at the mastoid bone. The velocity which the bone conduction receiver delivers to this acoustical impedance is measured by a vertical or hill-and-dale phonograph pickup (see Sec. 9.3B2d). A strain-gage type[74] of artificial mastoid has been developed. The translating element is a ceramic transducer..

10.5. Testing of Phonographs.—A. *Measurement of the Response of a Phonograph Record by the Optical Method.*—The response frequency characteristic of a lateral cut phonograph record may be obtained by means of the optical method [75,76,77] as shown in the schematic diagram of Fig. 10.37. The point source of light is placed at a distance of at least 10 feet, so that the light which strikes the grooves is practically parallel. The incident light is reflected from the sides of the groove of the record. In the case of a groove without modulation the width of the reflected light image will be a fine narrow line. If a sine wave is cut in the groove the width of the image will be proportional to the amplitude and the frequency. To make an accurate measurement of the width of the image the distance of the observer from the record should be at least 10 feet. The distance b of Fig. 10.37A can be measured accurately at this distance by employing a telescope. Under these conditions, the amplitude of the modulation, in inches, is given by

$$x_p = \frac{bu}{2f} \qquad\qquad 10.34$$

where b = total width of the image, in inches,

u = revolutions per second, of the record when it is reproduced, and

f = frequency in cycles per second when the record is rotated u revolutions per second.

[72] Hawley, M. S., *Bell Labs. Rec.*, Vol. 18, No. 3, p. 73, 1939.

[73] Romanow, F. F., *Jour. Acous. Soc. Amer.*, Vol. 13, No. 3, p. 294, 1942.

[74] Carlisle and Pearson, *Jour. Acous. Soc. Amer.*, Vol. 23, No. 3, p. 300, 1951.

[75] Buchman and Meyers, *ENT*, Vol. 7, p. 147, 1930.

[76] King, D. R., *Electronics*, Vol. 14, No. 5, p. 47, 1941.

[77] Bauer, B. B., *Jour. Acous. Soc. Amer.*, Vol. 18, No. 2, p. 387, 1946.

The peak velocity, in inches per second, is given by

$$\dot{x}_p = \pi b u \qquad 10.35$$

The rms velocity, in inches per second, is given by

$$\dot{x} = \frac{\pi}{\sqrt{2}} b u \qquad 10.36$$

The image shown in Fig. 10.37B depicts a record with constant velocity and any frequency. This shows that the radius of the groove or the frequency does not influence the width of the reflected image when the velocity
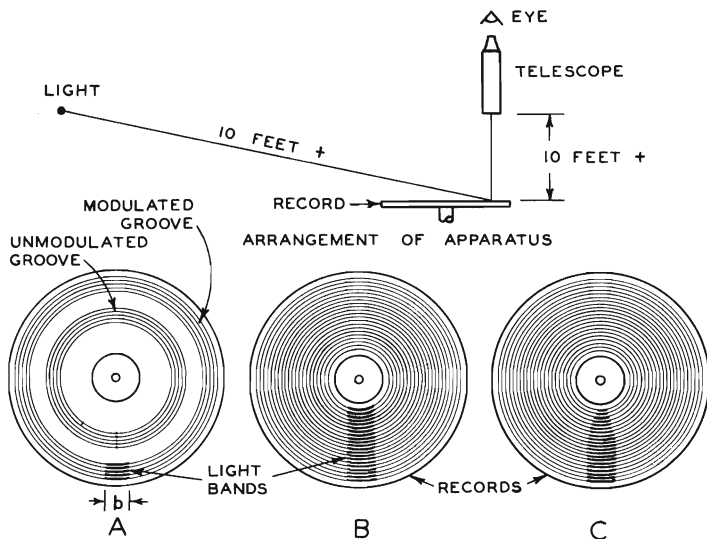
FIG. 10.37. Schematic arrangement of the apparatus for the measurement of the response of a phonograph record by the optical method. The light patterns for various types of modulation are shown in A, B, and C. A shows a record with a modulated and unmodulated groove. B shows a record with constant velocity and any frequency. C shows a record cut with sine wave modulation having the response frequency characteristic of Fig. 9.14. The frequency increases as a logarithmic function from the inside to the outside. Note: the frequency pattern at 180° is not shown.

is constant. The image in Fig. 10.37C depicts a frequency record cut with a sine wave modulation having the velocity response frequency characteristic as shown in Fig. 9.14.

It has been shown[78] that conventional light patterns are in error due to the diffraction of light. The error is proportional to the frequency. It is of the order of $1\frac{1}{2}$ decibels less than the conventional method at 20,000 cycles.

B. *Testing of Phonograph Pickups.*—Phonograph pickups are generally

[78] Bauer, B. B., *Jour. Acous. Soc. Amer.*, Vol. 27, No. 3, p. 586, 1955.

tested by employing a standard frequency record. The velocity response frequency characteristic of a standard frequency record is shown in Fig. 9.14. The velocity response frequency characteristic of the record may be determined by the optical method outlined in Sec. 10.5A. Voltage response frequency characteristics and nonlinear distortion measurements of phonograph pickups are usually made with a record of this kind. The voltage response frequency characteristic of a phonograph pickup is usually taken with the normal electrical load conditions. Frequency records are recorded either in discrete frequencies or a continuously variable frequency. The former type is used in obtaining a point-by-point response frequency characteristic. In the latter type, the record and turntable geared to the recording drum can be substituted for the oscillator and the pickup substituted for the microphone in any of the measuring systems of Sec. 10.3A2. Using this arrangement a continuous response frequency characteristic may be obtained.

The response frequency characteristics of phonograph pickups are usually measured by means of tone records. Tone records present some difficulties
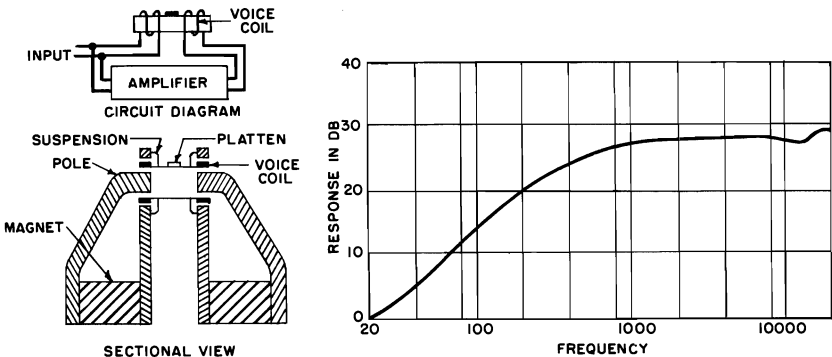
FIG. 10.38. Electrical circuit diagram, a sectional view and the response frequency characteristic of a feedback type calibrator for obtaining the response frequency characteristic of phonograph pickups.

as follows: amplitude, nonlinear distortion, and frequency variations. These factors may be small in good records. However, when the distortion in the pickups is also small, it is difficult to determine the source of the distortions. An electrodynamic system[79] has been developed for driving the stylus of a phonograph record which does not possess the distortions of a record. (Fig. 10.38.) There are two voice coils in the dynamic actuator. The one which drives the system is connected to the output of the amplifier, and the other which senses the motion of the system is connected to the input of the amplifier. The vibrating system is designed to exhibit one degree of freedom over the frequency range from 20 to 20,000 cycles. The resonant frequency of the voice coil and suspension system is placed at 700 cycles. The feedback

[79] Woodward, J. G., *RCA Review*, Vol. 11, No. 2, p. 301, 1950.

loop is adjusted in gain and phase so that the amplitude of motion of the plate is independent of the frequency for constant input below 500 cycles, and the velocity of the motion of the plate is independent of frequency above 500 cycles as shown in Fig. 10.38. This corresponds to one type of standard record response. Other response frequency characteristics may also be obtained by suitable compensation in the input and by adjusting the phase and gain of the feedback loop. Employing the pickup calibrator, response frequency characteristics of pickups can be obtained by means of the response measuring equipment described in Sec. 10.3A2. Nonlinear distortion and other characteristics may be obtained by the use of suitable equipment in conjunction with the pickup calibrator.

Nonlinear distortion characteristic of a phonograph pickup and record combination may be obtained by employing apparatus described in Sec. 10.3C. The pickup calibrator may also be used in these tests.

C. *Testing Mechanical Phonographs.*—The response frequency characteristic of a mechanical phonograph is obtained by employing a standard frequency record. The velocity response frequency characteristic of a standard frequency record is shown in Fig. 9.15. If a continuously variable frequency record is used, the record and turntable geared to the recording drum can be substituted for the oscillator in any of the measuring systems of Sec. 10.3A2. The horn or diaphragm of the mechanical phonograph is treated as a loudspeaker and the test conducted as outlined in Sec. 10.3A.

D. *Measurement of Mechanical Noise Produced by a Phonograph Pickup.*— The interaction of the phonograph pickup and the record, when the pickup is driven by undulations in the record, induces vibrations in both the record and pickup (see Sec. 9.3B4). The vibration of these parts produces direct radiation of sound into the air. If the sound level, produced by these vibrations, is comparable to the sound reproduced by the loudspeaker, undesirable distortion is created because the distortion in the mechanical noise is usually quite high. Two methods are generally employed for measuring the mechanical noise of a phonograph record and pick-up combination. In one method the complete phonograph is placed in a free-field sound room and the noise produced, at a certain distance, is measured with a standard noise meter (see Sec. 10.13). In another method, suitable for development work, the turntable, record, and pickup combination is placed in a small compartment[80] with reflecting walls. The sound level produced in the compartment is measured with a standard noise meter (see Sec. 10.13).

10.6. Measurement of Wows.[81]—In the reproduction of sound by film, tape, disk record, or other means it is important that the speed of the record in the recording and the reproducing machines be held constant. Otherwise the quality of the reproduced sound will be impaired by the frequency modulation produced by the speed variation. The term "wow" is used to designate speed variation in reproduced sound. Speed variation in reproduced sound may be detected and measured by recording a constant

[80] Burt, A. D., *Electronics*, Vol. 16, No. 1, p. 90, 1943.
[81] Kellogg and Morgan, *Jour. Acous. Soc. Amer.*, Vol. 7, No. 4, p. 271, 1936.

frequency at, for example, 1000 cycles. In reproducing this record, the output is fed to a frequency discriminating network and detector similar to that used in radio frequency modulation detectors. The magnitude of the wow is the difference between the highest and lowest speed, in percentage of the average speed.

10.7. Measurement of Acoustical Impedance.[82,83,84,85]—There are a number of methods of measuring acoustical impedance. A purely acoustical means for measuring acoustical impedance has been devised by Stewart. This method measures the change in acoustical transmission through a long uniform tube when the unknown acoustical impedance is inserted as a branch.

The acoustical impedance bridge is shown schematically in Fig. 10.39. The loudspeakers are. connected to two pipes, one of which is variable in length and the other equipped with a means for attaching the unknown impedance. At some distance beyond this point the two pipes are joined by a small pipe which, in turn, is connected to another pipe leading to a microphone and amplifier. Standing waves in the pipes are reduced by the introduction of small tufts of felt.

With the branch closed the voltage across the two loudspeaker units and the length of the variable tube are adjusted until a minimum reading is obtained in the output meter. The unknown acoustical impedance is now attached and the process repeated.

The unknown acoustical impedance can be obtained from the following equation

$$\frac{2z_{AU}}{2z_{AU} + r_A} = \frac{p_0}{p_0'} \left(\cos \theta + j \sin \theta \right) \qquad 10.37$$

where z_{AU} = unknown acoustical impedance, in acoustical ohms,

$r_A = \rho c/A$ acoustical resistance of the damped pipe, in acoustical ohms,

A = area of the pipe to which the branch is attached, in square centimeters,

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

c = velocity of sound, in centimeters per second.

The ratio p_0/p_0' can be determined from the following equation

$$\frac{p_0}{p_0'} = \frac{e_1 e_2'}{e_2 e_1'} \qquad 10.38$$

where e_1 and e_2 = the voltages applied to the loudspeaker without the branch, and

e_1' and e_2' = the voltages with the branch attached.

[82] Stewart, G. W., *Phys. Rev.*, 28, Vol. No. 5, p. 1037, 1926.

[83] Stewart and Lindsay, "Acoustics," D. Van Nostrand Company, Princeton, N.J., 1930.

[84] Flanders, P. B., *Bell Syst. Tech. Jour.*, Vol. 11, No. 3, p. 402, 1932.

[85] Morse, "Vibration and Sound," McGraw-Hill Book Company, New York, N.Y., 1936.

The phase angle θ in radians is given by

$$\theta = \frac{2\pi d}{\lambda} \qquad 10.39$$

where d = distance between the first position of the pointer without the branch to the second position with the branch in place, in centimeters. The direction toward the loudspeaker units is positive and

λ = wavelength of sound in air, in centimeters.

The recent emphasis on acoustical impedance as a means of describing the sound absorbing properties of materials has brought about a renewed interest in tube methods of acoustical impedance measurement. The
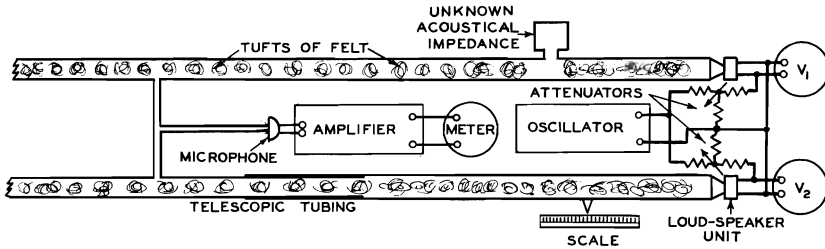
FIG. 10.39.   Schematic arrangement of an acoustical impedance bridge for measuring acoustical impedance.   (After Stewart.)

acoustical impedance bridge is quite satisfactory when the area of the acoustical impedance is relatively small.   In the case of acoustical materials the area may be as large as 144 square inches.   For these measurements the resonant tube is particularly suitable.

The arrangement of the apparatus[86,87,88,89] for the measurement of acoustical impedance by the tube method is shown in Fig. 10.40.   By means of the movable probe connected to a pressure microphone, associated amplifier, detector, and meter the difference in decibels between the maximum and minimum sound pressure in the standing wave may be measured.

The sound absorption coefficient for normal incidence is given by

$$\alpha_n = 1 - \left(\frac{\log_{10}^{-1}(L/20) - 1}{\log_{10}^{-1}(L/20) + 1}\right)^2 \qquad 10.40$$

$$= 1 - K^2$$

where L = difference, in decibels, between the maximum and minimum sound pressures in the standing wave system, and

K = pressure reflection coefficient of the material.

[86] Taylor, H. O., *Phys. Rev.*, Vol. 2, No. 4, p. 270, 1913.
[87] Hall, W. M., *Jour. Acous. Soc. Amer.*, Vol. 11, No. 1, p. 140, 1939.
[88] Bolt and Brown, *Jour. Acous. Soc. Amer.*, Vol. 12, No. 1, p. 31, 1940.
[89] Sabine, H. J., *Jour. Acous. Soc. Amer.*, Vol. 14, No. 2, p. 143, 1942.

In order to determine the acoustical impedance of the sample per unit area it is necessary to measure the distance from the surface of the material to the first minimum and the half wavelength.

The acoustical resistance, in acoustical ohms per unit area, is given by

$$r_{A1} = \rho c \frac{1 - K^2}{1 + K^2 + 2K \cos \left(\dfrac{2\pi D_1}{D_2}\right)} \qquad 10.41$$

The acoustical reactance, in acoustical ohms per unit area, is given by

$$x_{A1} = \rho c \frac{2K \sin \left(\dfrac{2\pi D_1}{D_2}\right)}{1 + K^2 + 2K \cos \left(\dfrac{2\pi D_1}{D_2}\right)} \qquad 10.42$$

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

c = velocity of sound in centimeters per second,

D_1 = distance between the material and the first pressure minimum, in centimeters,

D_2 = distance between two adjacent minima or one-half wavelength, in centimeters, and

K = pressure reflection coefficient obtained from equation 10.40.
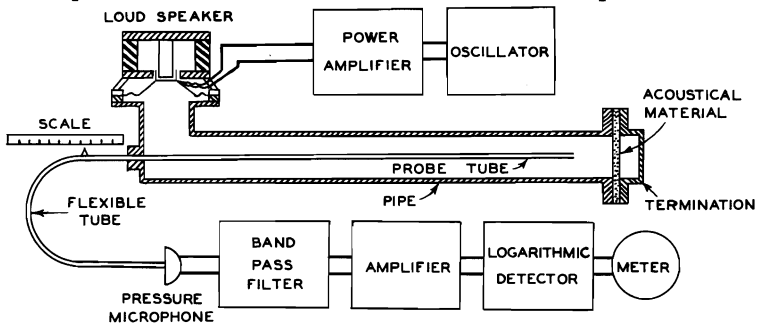
FIG. 10.40. Schematic arrangement of the apparatus for measuring the acoustical impedance by means of the standing wave pattern in a tube terminated by the unknown acoustical impedance.

The acoustical impedance, in acoustical ohms per unit area, of the acoustical material is the vector sum of the acoustical resistance and reactance as follows,

$$z_{A1} = r_{A1} + j x_{A1} \qquad 10.43$$

10.8. Measurement of Mechanical Impedance. —In certain types of mechanical vibrating systems it may be difficult to determine the constants of the mechanical network with any high degree of accuracy. Under these conditions it is sometimes desirable to measure the mechanical impedance at various frequencies. It is the purpose of this section to describe

a mechanical impedance bridge[90] which was developed for measuring the mechanical impedance of a phonograph pickup at the stylus over the entire audio-frequency range. It may also be used to determine the mechanical impedance of other vibrating systems.

If a reed, clamped at one end with the other end free, is driven at the clamped end by an alternating force, the displacement of the free end from the neutral position will be proportional to the applied force and the compliance of the reed. The force acting on the reed will be its effective mass times its acceleration. The effective mass of a reed clamped at one end is one-fourth its total mass.

The compliance of the reed, in centimeters per dyne, is given by

$$C_M = \frac{4l^3}{Qba^3}$$ 10.44

where l = length of the reed, in centimeters,

b = width of the reed, in centimeters,

a = thickness of the reed, in centimeters, and

Q = Young's modulus, in dynes per square centimeter, see Table 1.1.

The mechanical circuit of the reed is the effective mass, m (one-fourth the total mass), shunted by a compliance, C_M.

If two identical reeds are driven by the same force, the displacements of the free ends will be in phase and equal. Their relative displacement, therefore, will be zero. If one reed is then loaded with a mechanical impedance at its free end, the relative displacement will no longer be zero, but will be proportional to the driving force, the effective masses and compliances of the reeds, and the mechanical impedance of the load.

Schematic diagrams of the mechanical setup and its mechanical network are shown in Fig. 10.41. The reeds are the ground plates of condensers. The high potential plates are fastened rigidly to the driving mechanism. A high potential is connected across the plates of the condensers through a high resistance and then to amplifiers using the conventional circuit of the condenser microphone. Since the compliances of the two reeds are the same, the ratio of the two forces is equal to the ratio of the outputs of the two amplifiers. The signals from the two amplifiers are mixed 180° out of phase; so when the reeds are driven with no load applied to either reed, the two signals can be canceled. After the signals are canceled, and a load applied to the free end of one of the reeds, the mixed signal will be proportional to the vectorial difference between the force on the loaded reed and the force on the unloaded reed. The mechanical impedance of the load, in mechanical ohms, will be proportional to the ratio of the mixed signal with one reed loaded and the signal from the unloaded reed.

Two meters connected to the amplifiers read the values of the displacements of the reeds. A switch is provided so that one meter reads either

[90] Wiggins, A. M., *Jour. Acous. Soc. Amer.*, Vol. 15, No. 1, p. 50, 1943.

the mixed signal of the two reeds or the signal from the top reed alone.   The
other meter reads the signal from the bottom reed alone.   A powerful
driving system capable of handling 24 watts at the high frequencies with
the driving coil mounted on a rubber support is used.   Considerably more
power is needed at the high frequencies because of the low compliance of
the reeds used at these frequencies.

A reed whose natural frequency is somewhere above 1.5 times the fre-
quency at which the measurement is to be made is generally acceptable.

FRONT VIEW

VECTOR   DIAGRAM

SECTIONAL   VIEW   MECHANICAL   NETWORK
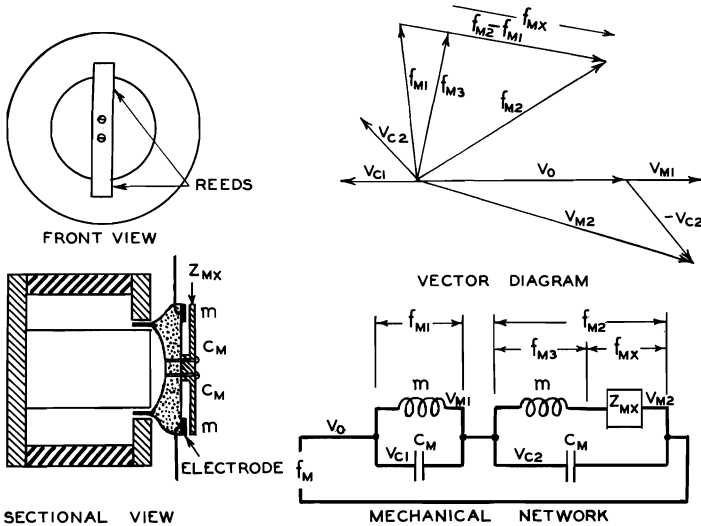
FIG. 10.41.   Front and sectional views, mechanical network, and vector
diagram of a mechanical impedance bridge.   In the mechanical network:
m = the effective mass of the reed. C_M = the compliance of the reed.
z_{MX} = the mechanical impedance being measured. f_M = the total
driving force. f_{M1} = the driving force of the unloaded reed. f_{M2} = the
driving force of the loaded reed and the mechanical impedance, z_{MX}.
f_{M3} = the driving force of the loaded reed. f_{MX} = the driving force
of the mechanical impedance, z_{MX}. v_0 = the total velocity. v_{C1} = the
velocity of the compliance of the unloaded reed. v_{M1} = the velocity
of the mass of the unloaded reed. v_{C2} = the velocity of the compliance
of the loaded reed. v_{M2} = the velocity of the mass and mechanical
impedance, z_{MX}. The vector diagram shows the magnitudes and
phases of the forces and velocities for a typical condition.

The proximity to the natural frequency at which a measurement can accu-
rately be made depends on the value of the mechanical impedance of the
load; the smaller the value of mechanical impedance the closer to resonance
the reed may be operated.   After the desired reed is selected the amplifiers
are turned on, and a signal from an oscillator and power amplifier is im-
pressed across the driving coil.   The reading on the meter for the mixed
signal is brought to zero, and the load applied.   One meter then reads the

value $f_{M2} - f_{M1}$ while the other meter reads f_{M1}. The mechanical impedance, in mechanical ohms, is given by

$$|z_{MX}| = \frac{f_{M2} - f_{M1}}{f_{M2}}\left(x_{MM} - \frac{x_{MM}^2}{x_{MC}}\right) \qquad 10.45$$

where f_{M1} = force on reed 1, in dynes,

f_{M2} = force on reed 2, in dynes,

$x_{MM} = \omega m$,

$m = \frac{1}{4}$ total mass of the reed, in grams,

$x_{MC} = \dfrac{1}{\omega C_M}$, and

C_M = compliance of the reed, given by equation 10.44.

If a vector diagram is desired, f_{M2} may be obtained by turning a switch, so the meter reads the signal from the loaded reed alone. From the values of these three meter readings, the three forces may be plotted in their proper phase relations, and a vector diagram drawn from which the mechanical impedance may be calculated. For most measurements, the quick and easy method of calculating the mechanical impedance by the formula and the ratio of the two forces will give sufficient information.

Another method[91] for measuring the mechanical impedance of phonograph pickups and other devices consists of measuring the motion of a calibrated vibrating reed with and without the load imposed by the pickup at the free end of the reed. The calibrated reed is made of steel and forms the armature of the electromagnetic vibrator, see Fig. 10.42. A set of four similar but of different dimensions covers the frequency range of 30 to 10,000 cycles.

The alternating driving force resulting from the flow of electrical

FIG. 10.42. Schematic view of the driver and pickup of a system for the measurement of mechanical impedance.

current through the coil surrounding the reed is applied to the reed at the upper pole faces near the free end of the reed. A capacity pickup electrode is situated between the reed and one of the pole pieces. This electrode is maintained at a d-c potential of about 115 volts. Motion of the reed causes a change in capacitance between the reed and the electrode and thus generates an alternating voltage component having the frequency and phase of the

[91] Woodward and Halter, *Jour. Acous. Soc. Amer.*, Vol. 25, No. 2, p. 302, 1953.

vibration of the reed and being proportional to the amplitude of vibration of the reed at the driving point. The free end of the reed is bent over at 90° to form a small platform on which the mechanical load to be measured is placed. The measurement of phonograph pickups is facilitated by V grooves in the platform in which the stylus tip may be placed.
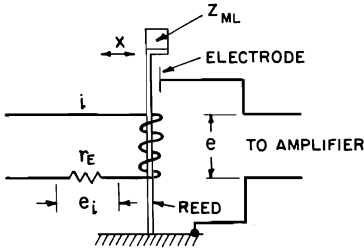
FIG. 10.43. Schematic view of the electrical and mechanical system for the measurement of mechanical impedance. In the electrical circuit: i = the input current to the coil. r_E = the electrical resistance of the calibrating resistor. e_i = the voltage across the calibrating resistor. x = the amplitude of the reed. z_{ML} = the mechanical impedance load. e = the voltage developed by the pickup electrode.

The amplitude and phase of vibration of the reed as a function of frequency, with and without the load, are measured in terms of the a-c voltage at the electrode. In order to obtain useful results the mechanical impedance of the reed must be of the same order of magnitude as the impedance of the mechanical load.

A schematic view of the electrical and mechanical system is shown in Fig. 10.43. The driving current i, in abamperes, in the coil is measured in terms of a voltage e_1, in abvolts, across a series resistor r_E, in abohms. The lateral displacement of the free end of the reed, measured from its rest position, is x, in centimeters. The alternating voltage generated at the electrode is e, in abvolts. The mechanical impedance of the load on the reed is designated by z_{ML}, in mechanical ohms. All quantities except r_E are complex. It will be assumed that the alternating quantities vary in a sinusoidal fashion at a frequency f, in cycles per second, and an angular velocity $\omega = 2\pi f$, in radians per second.

The driving force f_M, in dynes, is related to the driving current in the coil by

$$f_M = k_i \qquad\qquad 10.46$$

where k_i is, in general, a complex parameter. The force and the velocity $\dot{x}$, in centimeters per second, of motion are related by

$$f_M = z_M \dot{x} \qquad\qquad 10.47$$

where z_M is the driving point mechanical impedance, in mechanical ohms. For sinusoidal motion, $\dot{x} = j\omega x$, where x = amplitude or displacement.

$$f_M = j\omega z_M x \qquad\qquad 10.48$$

Equating equations 10.46 and 10.48 gives

$$z_M = \frac{k_i}{j\omega x} = \frac{ke_i}{j\omega r_E x} \qquad\qquad 10.49$$

The alternating voltage e, in abvolts, developed at the electrode is related to the displacement by

$$e = k'x \qquad\qquad 10.50$$

where k' is, in general, complex. Combining equations 10.49 and 10.50 gives

$$z_M = \frac{kk'}{j\omega r}\frac{e_{t0}}{e} \qquad\qquad 10.51$$

Let the mechanical impedance of the unloaded reed be designated as z_{MO}, and the corresponding measured voltages under these conditions be designated as e_{t0} and e_0, then

$$z_{MO} = \frac{kk'}{j\omega r_E}\frac{e_{t0}}{e_0} \qquad\qquad 10.52$$

When a load is placed on the reed,

$$z_{ML} + z_{MO} = \frac{kk'}{j\omega r_E}\frac{e_t}{e} \qquad\qquad 10.53$$

Combining equations 10.52 and 10.53 gives

$$z_{ML} = z_{MO}\,\frac{\dfrac{e_t}{e} - \dfrac{e_{t0}}{e_0}}{\dfrac{e_{t0}}{e_0}} \qquad\qquad 10.54$$

The voltage ratios, being complex quantities, may be written in terms of the absolute magnitudes and angles.

$$z_{ML} = z_{MO}\,\frac{\dfrac{e_t}{e}\phi - \dfrac{e_{t0}}{e_0}\phi_0}{\dfrac{e_{t0}}{e_0}\phi_0} \qquad\qquad 10.55$$

where ϕ is the phase difference between e_t and e and ϕ_0 is the phase difference between e_{t0} and e_0. In practice, the driving current for each reed is held at some constant magnitude. Therefore, $e_t = e_{t0}$, and equation (10) may be simplified to

$$z_{ML} = z_{MO}\left(\frac{e_0}{e}\,{}_E\phi - \phi_0 - 1\right) \qquad\qquad 10.56$$

Equation 10.56 is the working equation for the system. The values of e_0, e, ϕ, and ϕ_0 are measured at each frequency of interest and are substituted in equation 10.56 to compute z_{ML} in complex form. The value of z_{MO} must be known at each frequency. The converse procedure of applying known loads to the reed and computing z_{MO} serves to calibrate the reed.

A block diagram of the electrical components of the mechanical impedance measuring system is shown in Fig. 10.44. A variable-gain amplifier receiving its input signal from an audio-oscillator supplies the driving current to the vibrator. The voltage drop across a resistor in series with the vibrator coil is amplified, rectified, and used to control the gain of the driving amplifier.

In this way the driving current is automatically held constant within $\frac{1}{2}$ db for a 15 db change in the level of the input to the amplifier.
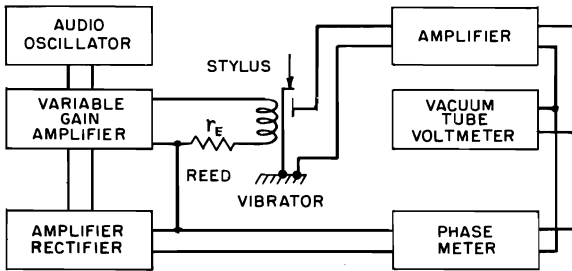
FIG. 10.44. Schematic arrangement of the apparatus used in a system for measuring mechanical impedance.

The voltage developed at the capacitor pickup electrode is amplified and its magnitude and phase are measured by a vacuum tube voltmeter

FIG. 10.45. Schematic arrangement of the apparatus for the measurement of porosity of a porous material.

and a phase meter. Phase angles are measured relative to the vibrator coil.

The calibrated vibrators and the technique described above have been used successfully in the measurement of the lateral mechanical impedance of a variety of phonograph pickups ranging from the earliest acoustic types to the most recent designs.

10.9. Measurement of Porosity. —Porosity is[92,93,94,95,96,97] a relevant mechanical property of a porous material. The porosity of a sub-stance is defined as the ratio of the volume of air in the pores to the total volume of the material.

The porosity may be measured by means of the apparatus shown in Fig. 10.45. Acoustical material of volume V_t is placed in the chamber

of volume V. The valve at the top of the chamber is opened and the level h of the water column on the two sides of the U tube measured. Then the

[92] Rettinger, M., *Jour. Acous. Soc. Amer.*, Vol. 6, No. 3, p. 188, 1935.

[93] Gemant, A., *Jour. App. Phys.*, Vol. 12, No. 10, p. 725, 1941.

[94] Morse, Bolt and Brown, *Jour. Acous. Soc. Amer.*, Vol. 12, No. 2, p. 217, 1940.

[95] Brown and Bolt, *Jour. Acous. Soc. Amer.*, Vol. 13, No. 4, p. 337, 1942.

[96] Beranek, L., *Jour. Acous. Soc. Amer.*, Vol. 13, No. 3, p. 248, 1942.

[97] Morse and Bolt, *Rev. Mod. Phys.*, Vol. 16, No. 2, p. 69, 1944.

valve is closed and the free side of the U tube is elevated until the levels, in centimeters, have changed from h to h_1 on one side and h_2 on the other side of the U tube. The pressure change Δp_0, in dynes per square centimeter, equals $(h_2 - h_1)$ 980. The reduction of volume ΔV_0, in cubic centimeters, in the chamber is $(h_1 - h)S$, where S is the cross-sectional area of the tube, in square centimeters. The porosity is given by

$$P = \frac{p_0 \Delta V_0}{V_t \Delta p_0} + 1 - \frac{V}{V_t} \qquad 10.57$$

where p_0 = atmospheric pressure, in dynes per square centimeter,

V_t = volume of the material, in cubic centimeters,

Δp_0 = change in pressure, in dynes per square centimeter,

ΔV_0 = change in volume, in cubic centimeters, and

V = volume of the chamber, in cubic centimeters.

10.10. Measurement of d-c Acoustical Resistance (Flow Resistance). —The relation between the mechanical properties of a sound absorbing material and its acoustical properties have been studied for some time. One of the important characteristics of a sound absorbing material which is an important factor in predicting the action is the d-c acoustical resistance, sometimes termed the flow resistance.[98,99,100,101,102,103]

The d-c acoustical resistance, r_{ADC}, in acoustical ohms, may be obtained from the volume current and the pressure as follows:

$$r_{ADC} = \frac{p}{U} \qquad 10.58$$

where p = difference in pressure between the two surfaces of the material, in dynes per square centimeter, and

U = volume current through the material, in cubic centimeters per second.

The d-c acoustical resistance may be measured by maintaining a steady flow of air through the material and measuring the pressure drop across the sample. Apparatus for measuring the d-c acoustical resistance is shown in Fig. 10.46. The difference in pressure between the two sides is measured by means of an inclined manometer. The volume current or ratio of volume displacement may be obtained from the ratio of the volume displacement and the time.

[98] Rettinger, M., *Jour. Acous. Soc. Amer.*, Vol. 6, No. 3, p. 188, 1935.
[99] Gemant, A., *Jour. App. Phys.*, Vol. 12, No. 10, p. 725, 1941.
[100] Morse, Bolt, and Brown, *Jour. Acous. Soc. Amer.*, Vol. 12, No. 2, p. 217, 1940.
[101] Brown and Bolt, *Jour. Acous. Soc. Amer.*, Vol. 13, No. 4, p. 337, 1942.
[102] Beranek, L., *Jour. Acous. Soc. Amer.*, Vol. 13, No. 3, p. 248, 1942.
[103] Morse and Bolt, *Rev. Mod. Phys.*, Vol. 16, No. 2, p. 69, 1944.

The d-c acoustical resistance or flow resistance is usually specified as the acoustical resistance per unit cube as follows:

$$r_{ADC1} = \frac{pA}{Ud} \qquad\qquad 10.59$$

where $A =$ area of the material, in square centimeters, and

$d =$ thickness of the material, in centimeters.
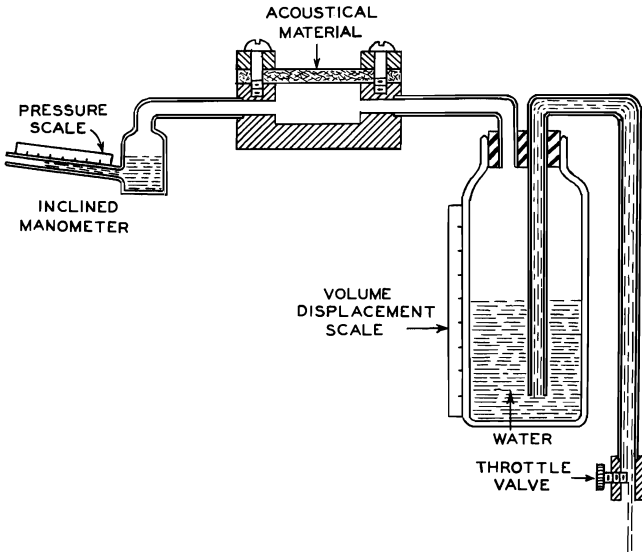
Fig. 10.46. Schematic arrangement of the apparatus for the measurement of flow resistance or d-c acoustical resistance of an acoustical material.

10.11. Measurement of Reverberation Time.—The reverberation time for a given frequency is the time required for the average sound energy density, initially in a steady state, to decrease after the source is stopped to one millionth of its initial value. The unit is the second.

Many systems[104] have been developed for the measurement of reverberation time. Sabine used an organ pipe as a source of sound, the ear as a detector, and a stop watch for measuring the time. Since that time various types of chronographs, reverberation bridges, commutators, relays, etc., have been developed to measure the reverberation time of an enclosure.

The high-speed level recorder, Fig. 10.11, and the high-speed level indicator, Fig. 10.14, have been found to be useful means for measuring reverberation time because the trace of the entire decay of the sounds may be examined.

[104] Olson and Massa, "Applied Acoustics," 2nd Ed., P. Blakiston's Son and Co., Philadelphia, 1939.

A schematic diagram depicting the apparatus for measuring the reverberation time is shown in Fig. 10.47.   The trace of the sound decay is depicted on the screen of a cathode-ray tube with a persistence image screen.   The spot is driven at a constant rate from left to right and then returned.   Decay is observed over a range of 48 db.   A transparent time scale over the front of the tube is used to read the reverberation time. The commutator interrupts the power to the loudspeaker.   The decay sequence is repeated every 9 seconds.   The beat frequency oscillator is warbled to reduce the effects of standing wave systems and thereby obtain
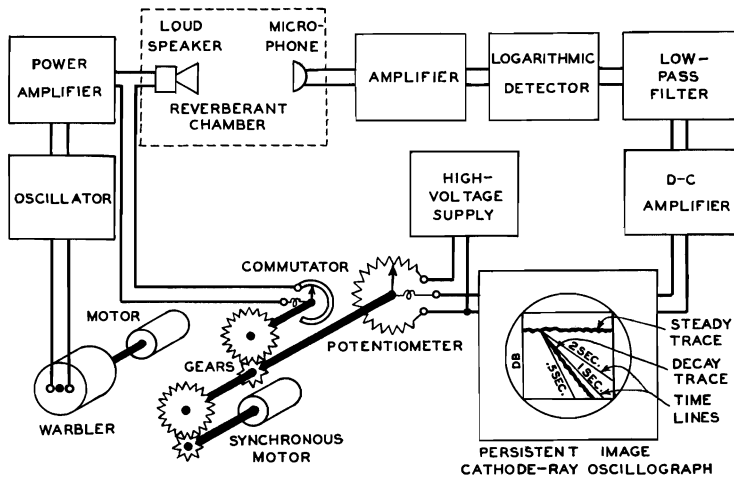
FIG. 10.47.   Schematic arrangement of the apparatus for obtaining the reverberation time of a reverberation chamber or room employing a cathode-ray oscillograph with persistent image screen.

a smoother decay trace.   As a further aid in smoothing the decay response multiple loudspeakers and microphones may be used.

   10.12. Measurement of Absorption Coefficient.[105,106,107,108]—The acoustical absorption coefficient of a surface is the ratio of the rate of sound energy absorbed by the surface to the incident rate of flow.   All directions of incidence are assumed to be equally probable.   The sabin is a unit of equivalent absorption and is equal to the equivalent absorption of 1 square foot of a surface of unit absorptivity, that is, 1 square foot of surface which absorbs all incident sound energy.

   [105] Watson, F. R., "Acoustics of Buildings," John Wiley and Sons, New York, N.Y., 1923.
   [106] Bagenal and Wood, "Planning for Good Acoustics," Methuen, 1931.
   [107] Knudsen, V. O., "Architectural Acoustics," John Wiley and Sons, New York, N.Y., 1932.
   [108] Sabine, P. E., "Acoustics and Architecture," McGraw-Hill Book Company, New York, N.Y., 1932.

The total absorption in a room may be obtained from equations 11.2 or 11.3, if the reverberation time and the dimensions of the room are known. This method of obtaining the absorption coefficient of materials has been considered to yield the most reliable results.

Specialists in the measurement of absorption coefficients have used large chambers (volume of 4000 to 20,000 cubic feet) for determining the absorption coefficient of materials from the reverberation time. The reverberation time of these chambers, when empty, is from 5 to 10 seconds. In chambers of this kind the absorption coefficients of very small samples may be determined. The absorption coefficients of representative materials obtained under the above conditions by various investigators are shown in Table 11.1, Sec. 11.2A.

In this connection it is interesting to note that there is considerable discrepancy between the values of absorption coefficients obtained in reverberant chambers and those obtained in actual use in rooms, studios, and theaters. For this reason the values given in Table 11.1, Sec. 11.2A, must be modified by a factor in computing the reverberation time of a room. It may be said, however, that these data indicate the relative efficiency of the various materials.

10.13. Measurement of Noise.—Due to the complexity of the human hearing mechanism and to the various types of sounds and noises it is
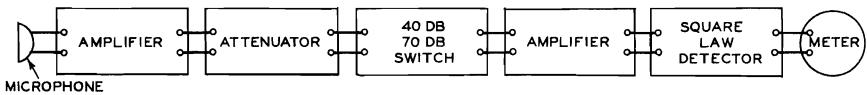
FIG. 10.48.   Schematic arrangement of the components of a noise meter.

impossible, at the present time, to build a noise meter which will show the true loudness level. The discrepancies can be determined by actual use and suitable weighting factors applied to the results. Objective measurements are almost indispensable in any scientific investigation. The noise meter or sound level meter provides a system for measuring the sound level of a sound.

A schematic diagram of a sound level or noise meter is shown in Fig. 10.48. The microphone should be calibrated in terms of a free wave. The directional characteristics of the microphone should be independent of the frequency. The attenuator and meter should be calibrated in decibels. A sound meter reading 60 db means a sound level of 60 db above the reference level. The reference point of the decibel scale incorporated in a sound meter shall be the reference sound intensity at 1000 cycles in a free progressive wave—namely, 10^{-16} watts per square centimeter. The response frequency characteristic of the human ear shows less sensitivity for frequencies above and below 3000 cycles, Fig. 12.11. The over-all frequency response of an ideal noise meter should be the reciprocal of the ear response frequency characteristics. This would make the noise meter unduly com-

plicated. The response frequency characteristics recommended for the noise meters by the American Standards Association are [109,110,111,112] shown in Fig. 10.49. Curve *A* is recommended for measurements at the lower levels and curve *B* for measurements around 70 db above the threshold. For very loud sounds (80 to 100 db), the flat characteristic of curve *C* should be used.

The noise meter may be used for noise analysis in offices, factories, restaurants, etc. In these measurements a large number of observations
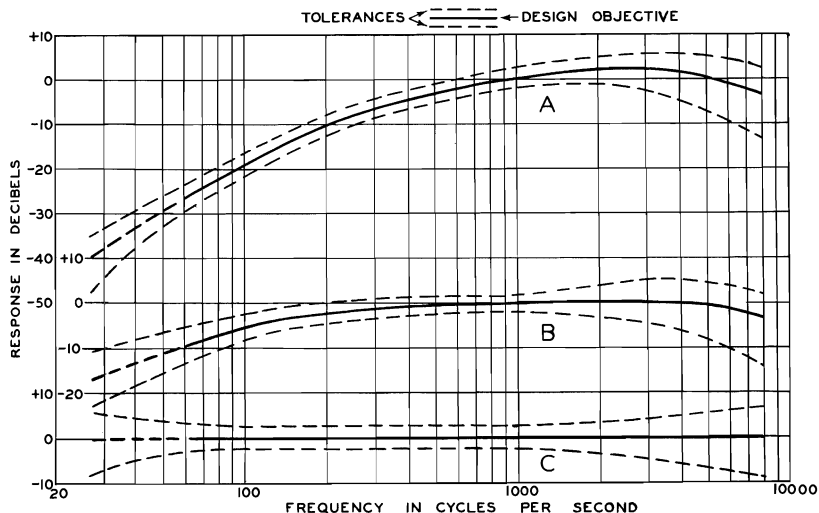
FIG. 10.49. Recommended characteristics for sound level meters. (American Standards Association.)

should be made in various positions. The noise meter may also be used to measure the transmission or attenuation by walls, floor, ceilings, and doors (see Secs. 10.13 and 11.2U).

The noise meter is also a useful tool in work on the quieting of machinery. Since the radiation pattern of machinery noise is very complex, a large number of measurements should be made in various directions relative to some axis of the system. For these investigations a frequency analyzer of the heterodyne type (see Sec. 10.3C) is a useful adjunct for determining

[109] Amer. Tent. Standards for Sound Level Meters Z24.3, American Standards Association, New York, N. Y., 1936; or *Jour. Acous. Soc. Amer.*, Vol. 8, No. 2, 1936.

[110] American Standard for Noise Measurement Z24.2–1942, or *Jour. Acous. Soc. Amer.*, Vol. 14, No. 1, p. 102, 1942. Also American Standards Association, Z24.2–1942.

[111] American Standard for Sound Level Meters for Measurement of Noise and other Sounds, Z24.3, American Standards Association, New York, N.Y., 1944. Also American Standards Association, Z24.3–1944.

[112] American Standards Association, "Test Code for Apparatus Noise Measurement," Z24.7–1949.

the nature of the noise. For routine tests in manufacturing it is customary to establish passable limits together with fixed geometrical configurations and procedures. For routine tests it is absolutely necessary that the calibrations remain correct within a decibel, i.e., $\pm\frac{1}{2}$ db.
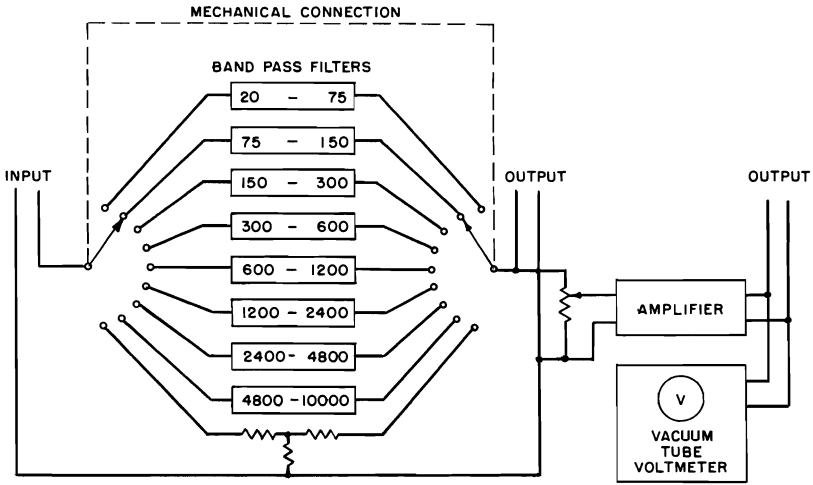
FIG. 10.50. Schematic arrangement of the apparatus used in an octave-band analyzer.
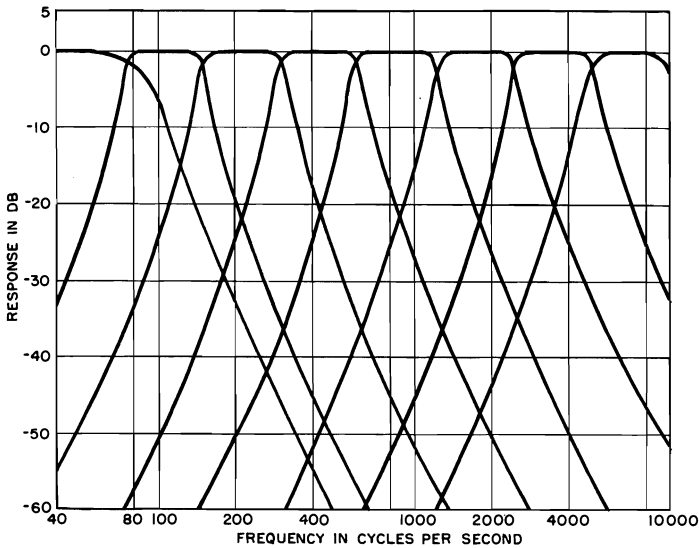
FIG. 10.51. Response frequency characteristics of the band-pass filters used in the octave-band analyzer of Fig. 10.49.

The octave band analyzer is useful for the measurement of noise of machinery, aircraft, vehicles, offices, factories, and rooms. For these applications it is operated from the output of a sound level or noise meter. The octave band noise analyzer[113,114] consists of a set of octave band pass filters with selection by means of a dual rotary switch followed by an attenuator and an amplifier which drives an indicating meter and a monitoring output. (See Fig. 10.50.) The filter characteristics are independent
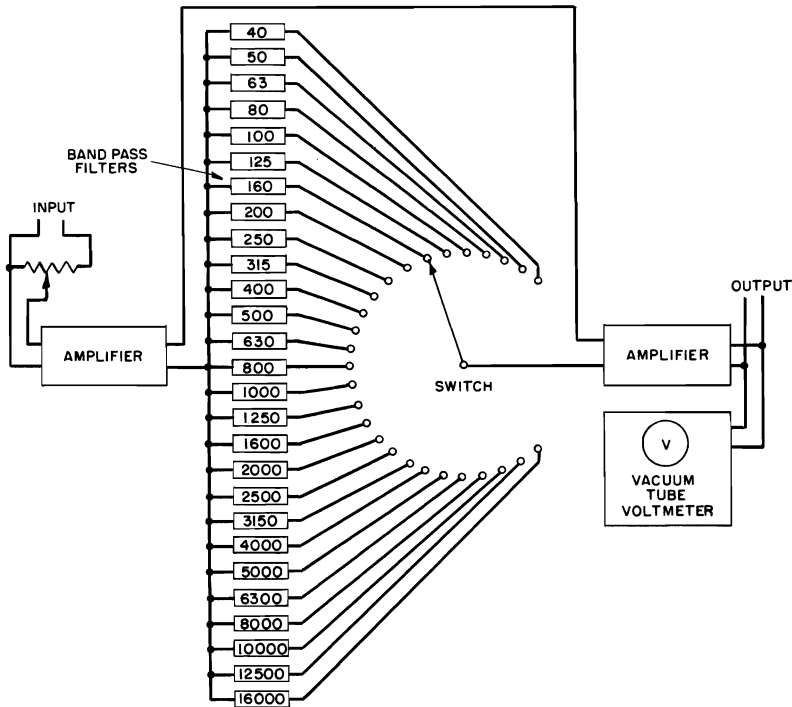
FIG. 10.52. Schematic arrangement of the apparatus used in an audio spectrometer employing third-octave-band filters.

of the source, if the source impedance is small compared to the input impedance or constant over the audio-frequency range. The output of the filter selecting switch is connected to a calibrated attenuator. The output of the attenuator is connected to an amplifier. The output of the amplifier is used to drive the indicating meter and the monitoring output. The response frequency characteristics of the filters are shown in Fig. 10.51.

[113] American Standards Association, "Octave-Band Filter Sets," Z24.10–1953.
[114] Beranek, L., "Acoustic Measurements," John Wiley and Sons, New York, N.Y., 1949.

The audio-frequency spectrometer[115] consists of a preamplifier followed by 27 third-octave filters with selection by means of a rotary switch. (See Fig. 10.52.) The switch is followed by an output amplifier. The output of the amplifier can be used to drive the indicating meter or an external instrument such as an automatic level recorder. The switch can be operated either manually or from the motor drive of a level recorder when the output is automatically recorded. The response frequency characteristics of the filters are shown in Fig. 10.53. There are many applications for an analyzer
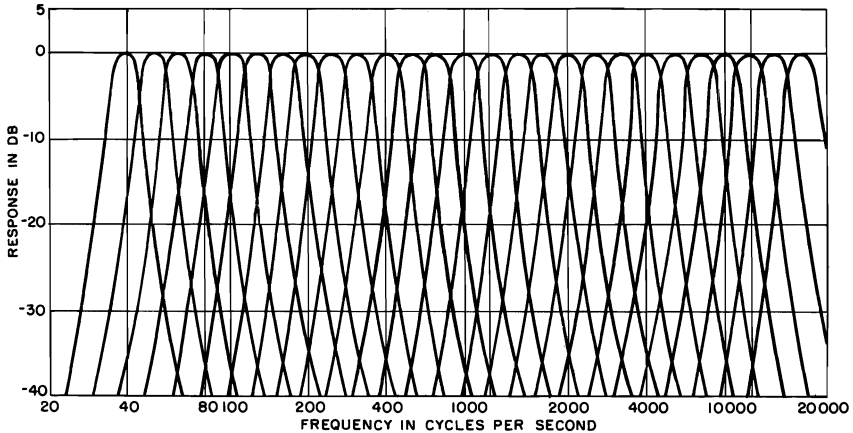
FIG. 10.53. Response frequency characteristics of the band-pass filters used in the audio spectrometer of Fig. 10.52.

of this type. These include the measurement of the noise spectrums of machinery, aircraft, vehicles, offices, factories, and rooms. The instrument may also be used to measure distortion in sound reproducing equipment.

10.14. Measurement of the Components of a Complex Wave.— In acoustic research it is necessary to analyze complex waves. If these waves do not change, it is possible to determine the components without great difficulty employing analyzers of the type described in Sec. 10.3C. If the wave is complex in frequency composition and varies with time, the problem is difficult. A system[116] for analyzing a complex wave is shown in Fig. 10.54. The complex wave is recorded on magnetic tape. Then a small section is analyzed by means of a series of rotating heads. Four heads are used in the system of Fig. 10.54. The wrap of the tape around the rotating head assembly is 90°. Thus it will be seen as one head moves in, the section of tape on the rotating head is scanned over and over again. The output of the heads is fed to an analyzer and the components measured.

An automatic system[117] for analyzing a complex wave is shown in Fig.

115 Bruel and Kjar, *Tech. Rev.*, No. 4, 1953.
116 Olson and Belar, Unpublished Report.
117 Koenig, Dunn, and Lecy, *Jour. Acous. Soc. Amer.*, Vol. 18, No. 1, p. 19, 1946.

10.55.   A section of the signal is recorded on a magnetic drum.   The recorded signal is then reproduced over and over as the variable frequency analyzer sweeps through the frequency range under consideration.   The output of the analyzer is compressed and amplified and fed to the recording
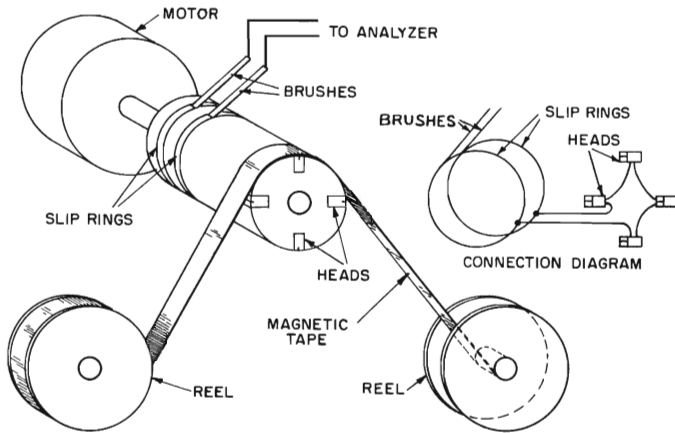
FIG. 10.54.   A perspective view of a rotating magnetic head scanner for reproducing the output from a section of recorded magnetic tape.
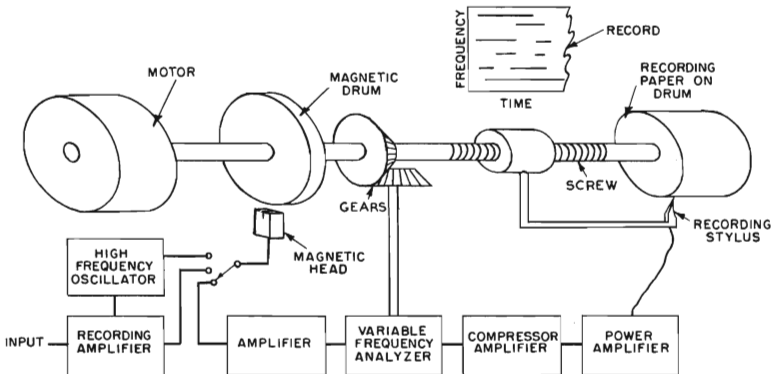
FIG. 10.55.   Schematic arrangement of the apparatus for analyzing a complex wave.

stylus.   Intensity is recorded as proportional to the density or darkness of the record.   Frequency is recorded as the ordinate.   Time is recorded as abscissa.

10.15. Measurement of Transmission Coefficient.—The transmission coefficient of a partition or wall is defined as the ratio of the transmitted sound energy to the rate of the incident flow of sound energy.   The sound insulating properties of a partition consist of a determination of its

transmission coefficient.   In a general way, the noise reduction caused by a
particular structure may be obtained by measuring the difference in level
of a sound source with and without the intervening partition.   For defini-
tion and equations relating to transmission loss and reduction factor, see
Sec. 11.2U.   The sound source should be fed with a warbled frequency or
rotated in a circle to average out reflection errors.   The noise meter (Sec.
10.13) or, as a matter of fact, any of the sound measuring systems (Sec.
10.3A2) may be used to measure the sound reduction.   In some cases it may
be desirable to use two or more samples of different shapes and areas.
Whenever possible the measurements should be made under operating
conditions.   For the transmission characteristics of various structures, see
Table 11.3, Sec. 11.2U.

   10.16.—Audiometry.[118],[119],[120]—The acuity of hearing is measured by
an audiometer.   The audiometer consists of an audio oscillator for generat-
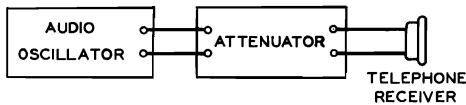
FIG. 10.56.   Schematic arrangement of the com-
ponents of an audiometer.

ing pure tones, an attenuator calibrated in decibels and a telephone receiver
(Fig. 10.56).   The usual range of test tones are 128, 256, 512, 1024, 2048,
4096, 8192 cycles per second.   The tone generated in the earphone should
be reasonably free from harmonics.   The telephone receiver is calibrated
as outlined in Sec. 10.4.   The reference level is the normal threshold of
audibility (Fig. 12.11).   This level is the zero level of the audiometer.   The
person to be tested wears the earphone in the normal manner and the level
at which the sound is no longer audible is noted on the attenuator.   A person
with normal hearing will show no hearing loss, while the person who is hard
of hearing will show a hearing loss.   These results are plotted on a graph
with the hearing loss in decibels as the ordinates and the frequency as the
abscissa.

   10.17. Articulation Measurements.—[121],[122],[123]—In the case of speech
transmission the primary object is the realization of conditions which will
result in the maximum intelligibility.   Intelligibility is used to signify the

[118] Fletcher, H., "Speech and Hearing," D. Van Nostrand Company, Princeton, N.J.,
1929.
[119] Proposed Specifications for Audiometers for General Diagnostic purposes, *Jour.
Acous. Soc. Amer.*, Vol. 9, No. 1, p. 72, 1937.
[120] American Standards Association, "Audiometers for General Diagnostic Purposes,"
Z24.5–1951.   "Pure Tone Audiometers for Screening Purposes," Z24.12–1952.
[121] Fletcher, "Speech and Hearing," D. Van Nostrand Company, Princeton, N.J.,
1929.
[122] Fletcher, "Speech and Hearing in Communication," D. Van Nostrand Company,
Princeton, N.J., 1955.
[123] Fletcher and Steinberg, *Jour. Acous. Soc. Amer.*, Vol. 1, No. 2, Part 2, p. 1, 1930.

accuracy and ease with which the articulated sounds of speech are recognized. (See Sec. 12.21.)

Many methods and tests are used to determine the person's ability to recognize the sounds of speech. Fundamentally, these methods consist of pronouncing speech sounds into one end of a transmission system and having the observer write the sounds which are heard at the receiving end. The comparison of the called sounds with those observed shows the number and kind of errors which are made. The system may be the air between the mouth and the ear in a room or it may be a telephone system, or a sound reproducing system such as a phonograph, radio, or sound motion picture.

Speech material of various kinds may be used. The percentage of the total number of speech sounds which are correctly observed is called the sound articulation. The terms vowel articulation and consonant articulation refer to the percentages of the total number of spoken vowels or consonants which are correctly observed. If a syllable is used as a unit, the per cent correctly received is termed syllable articulation.

The discrete sentence intelligibility is the percentage of the total number of spoken sentences which are correctly understood. The discrete word intelligibility is the percentage of the total number of spoken words which are correctly understood. Lists have been prepared for use in articulation testing. These may be used to determine the performance of a system as outlined above.

10.18. Testing of Hearing Aids.[124,124a]—Apparatus for testing the over-all performance of a hearing aid is depicted in Fig. 10.57. A sound
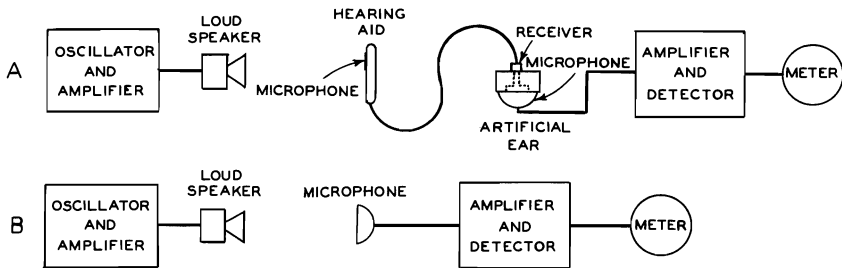
FIG. 10.57. Schematic arrangement of the apparatus for obtaining the response frequency characteristic of a hearing aid.

wave is produced by the oscillator, amplifier, and loudspeaker combination in a space free of reflections, as, for example, a free-field sound room. The sound wave is picked up by the hearing-aid microphone, amplified and reproduced by the telephone receiver. The output of the receiver is measured

[124] Tentative Code for Measurements of Performance of Hearing Aids, *Jour. Acous. Soc. Amer.*, Vol. 17, No. 2, p. 44, 1945.
[124a] American Standards Association, "Characteristics of Hearing Aids," Z24.14—1953.

by means of an artificial ear, amplifier, and meter. Any of the response measuring systems described in Sec. 10.3A2 may be employed instead of the point-by-point system depicted in Fig. 10.57. The artificial ear consists of a cylindrical tube, 1.80 centimeters in length and .305 centimeter in diameter, connected to a cavity of 2 cubic centimeters. The insert type hearing-aid receiver, without the molded ear insert, is coupled to the open end of the tube. A pressure microphone is used to measure the sound pressure delivered to the cavity. After the response frequency characteristic of the system of Fig. 10.57A has been obtained, the sound pressure response frequency characteristic in free space is obtained by placing the microphone of the system of Fig. 10.57B at the point occupied by the hearing-aid. The sound pressure is measured by means of the calibrated microphone amplifier and meter. If the same microphone is used to measure the sound pressure in the artificial ear and the sound pressure in free space produced by the loud-speaker a calibrated system is not required because the amplification of the hearing aid is the ratio between the sound pressure in the cavity and the sound pressure in free space.

10.19. Automatic BH Curve Tracer.[125,126]—The magnetic behavior of magnetic tapes and other small samples of magnetic materials cannot be measured by the conventional ballistic galvanometer method. A system
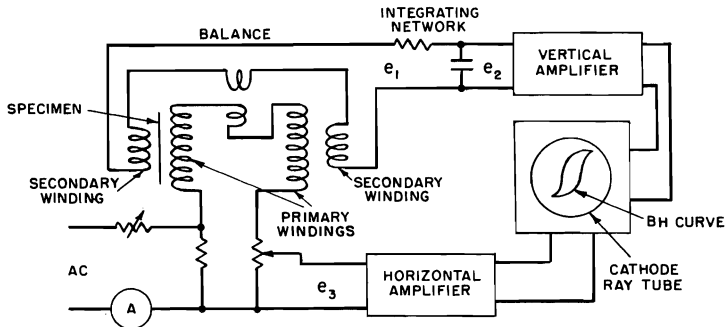
FIG. 10.58. Schematic arrangement of the apparatus for obtaining the B–H characteristic of small samples of magnetic materials.

for the BH characteristics of small magnetic samples is depicted in Fig. 10.58. Two large solenoids are used with a small pickup coil inside of each one. The primary windings are excited by 60-cycle alternating current. The outputs of the two secondaries are connected in opposition and fed to an integrating network. The output of the integrating network is connected to an amplifier, the output of which is connected to the vertical deflection system of a cathode-ray oscilloscope. The 60-cycle voltage is fed to the horizontal deflection system of the cathode-ray oscilloscope. In the absence of a magnetic specimen in one of the solenoids, there should be no vertical

[125] Crittenden, Smith, and Olsen, *Rev. Scien. Inst.*, Vol. 17, No. 10, p. 372, 1946.
[126] Arndt, J. P., Jr., Brush Strokes, March, 1951.

indication on the cathode-ray oscilloscope. If there is a deflection, it can be balanced out by means of the balance coil system. Now if the sample of magnetic material is inserted in one of the secondary coils, the output voltage e_1 will be proportional to the change in flux, that is, $\frac{d\phi}{dt}$. The BH curve will be depicted on the cathode-ray oscilloscope.

10.20. Electronic Measurement of Roughness.[127]—The type of surface produced by finish machining process is of great importance in all industry. The attainment of a particular surface finish may be essential
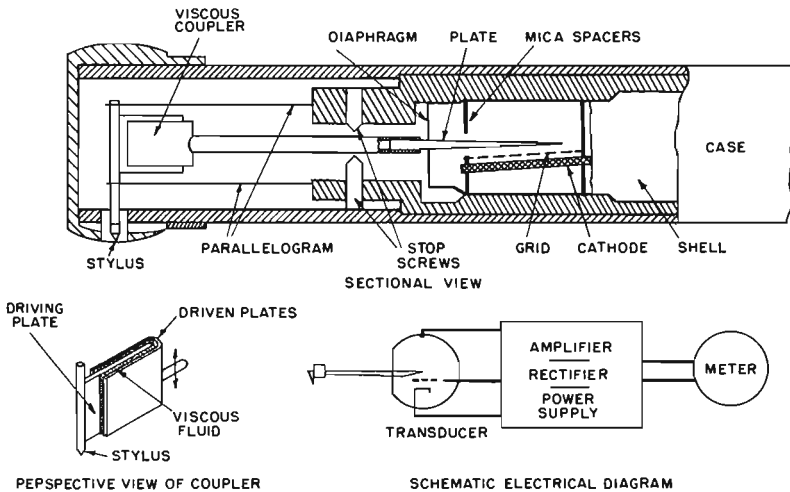
FIG. 10.59. Schematic arrangement of the apparatus for measuring the roughness of a surface, showing a sectional view of the mechano-electronic transducer, the viscous coupler, and the electrical diagram.

to the satisfactory performance of many machine elements. This means that some means must be provided for the measurement of the finish of a surface. An electronic system for measuring the roughness of a surface has been developed and termed "Surfindicator." The surfindicator shown in Fig. 10.59 consists of a diamond stylus which is used as a feeler to follow the surface contours, and mechano-electronic transducer which converts the motion into an electrical signal and a viscous coupler which prevents excessive motion from damaging the transducer. The indicating portion of the surfindicator consists of an amplifier and a vacuum tube voltmeter. The surfindicator reads the arithmetic average deviation of the surface. This parameter is the average of the vertical distances between the mean line and the profile. The viscous coupler and the stylus spring support make it possible to measure flat, concave, and convex surfaces. A set of standard blocks are used to calibrate the instrument.

[127] Hagen and Lindberg, *Gen. Motors Eng. Jour.*, Vol. 1, No. 7, p. 18, 1954.

10.21. Vibration Measurements.[128],[129]—The measurement of the vibration of a machine, the elements of a machine, the supports for a machine, the floor, walls or ceiling of a room, and other vibrating elements plays an important role in the reduction and control of vibration and the attendant noise. Vibration pickups have been described in Sec. 9.4. The output of the pickup can be analyzed by means of the analyzers described in this chapter. By the use of suitable electrical circuits acceleration, velocity or displacement may be measured. See Sec. 9.4. The particular quantity that is measured will depend upon the use to which the measurements are made. For example, for studying the radiation of sound by vibrating parts a measure of the velocity is the most useful because it can be translated into the radiation of sound in conjunction with the acoustical radiation resistance.

[128] American Standards Association, "Shock and Vibration Terminology," Z24.1a–1955.

[129] American Standards Association, "Analyzers Used in Analysis of Sounds and Vibrations," Z24.15–1955.

11

ARCHITECTURAL ACOUSTICS AND THE COLLECTION AND DISPERSION OF SOUND

11.1 Introduction.—The advent of sound reproducing systems has changed the problems involving architectural acoustics. Before the introduction of sound reproducing systems the major concern was the optimum reverberation time and the proper geometrical configuration for the best artistic effects in music and the maximum intelligibility of speech. By means of sound reproducing systems speech can be rendered intelligible where before it was either too weak to be heard above the general noise level or too reverberant. Furthermore, these instruments have opened a field for all manner of artistic effects never before possible.

The theaters which suffer most from insufficient loudness are, of course, the large enclosed theater and the open-air theater. Sound reproducing systems have opened new vistas in musical renditions both by reproduction and reinforcement. In certain instances the volume range of an orchestra is inadequate for full artistic appeal or to utilize the full capabilities of the hearing range. In these cases, means are required for augmenting the intensity of the original sound. The systems for accomplishing this objective are termed sound reinforcing systems.

The acoustic problems involving the reproduction of sound motion pictures are quite unlike those of stage presentations. The acoustics of radio broadcasting differ from those of the stage and sound motion pictures in that the action cannot be seen. Therefore, sound carries the entire load of the transmission of intelligence. Television acoustics are the most complex of all because they involve a part of stage, sound motion pictures, and radio techniques, as well as entirely new problems. It is quite evident that reproduced sound offers greater possibilities for obtaining the proper artistic effects by the use of the following expedients: incidental sound, a wide volume range, the control of the reverberation or room characteristics, and various sound effects.

For large outdoor gatherings such as state occasions and athletic events in large stadiums and parks, sound reproducing systems are employed to amplify the speaker's voice.

In department stores, hotels, hospitals, schools, and factories sound reproducing systems are employed to transmit sound from a central point

to several independent rooms or stations. The systems for accomplishing this objective have been termed general announce or call systems.

It is the purpose of this chapter to outline the applied phases of architectural acoustics and the applications of the collection and dispersion of sound.

11.2. Dispersion of Sound.[1,2,3,4,5,6,7,8,9,10,11]—A. *Sound Absorption and Reverberation.*—When a source of sound is started in a room the energy does not build up instantly due to the finite velocity of a sound wave. Each pencil of sound sent out by the source is reflected may times from the absorbing walls of the room before it is ultimately dissipated. A steady state condition obtains when the energy absorbed by the walls equals the energy delivered by the sound.source. In the same way, when the source is stopped, some time is required before the energy in the room is completely absorbed. The reverberation time has been arbitrarily defined by Sabine as the time required for the sound to decrease to one millionth of its original intensity after stopping the source.

The equation[12] for the decay of the sound in a room is

$$E = E_0 \epsilon^{-cAt/4V} \qquad\qquad 11.1$$

where E = sound energy density, after a time t seconds, after stopping the source, in ergs per cubic foot,

A = total number of absorption units, in sabins (see definition below),

$E_0 = 4P_0/cA$,

P_0 = rate at which sound is generated by the source, in ergs per second,

t = time, in seconds,

c = velocity of sound, in feet per second, and

V = volume of the room, in cubic feet.

[1] Sabine, W. C., "Collected Papers in Acoustics," Harvard Univ. Press, Cambridge, Mass.

[2] Watson, F. R., "Acoustics of Buildings," John Wiley and Sons, New York, N.Y., 1923.

[3] Begenal and Wood, "Planning for Good Acoustics," Methuen, 1931.

[4] Knudsen, V. O., "Architectural Acoustics," John Wiley and Sons, New York, N.Y., 1932.

[5] Sabine, P. E., "Acoustics and Architecture," McGraw-Hill Book Co., New York, N.Y., 1932.

[6] Olson and Massa, "Applied Acoustics," 2nd Ed., P. Blakiston's Son and Co., Philadelphia, Pa., 1939.

[7] Rettinger, "Applied Architectural Acoustics," Chemical Publishing Company, Brooklyn, N.Y., 1947.

[8] Knudsen and Harris, "Acoustical Designing in Architecture," John Wiley and Sons, New York, N.Y., 1950.

[9] Beranek, "Acoustics," McGraw-Hill Book Company, New York, N.Y., 1954.

[10] Olson, "Musical Engineering," McGraw-Hill Book Company, New York, N.Y., 1952.

[11] Rettinger, "Practical Acoustics," Chemical Publishing Company, New York, N.Y., 1955.

[12] Franklin, W. S., *Phys. Rev.*, Vol. 16, p. 372, 1903.

The *acoustic absorptivity* (or *absorption coefficient*) of a surface is the ratio of the flow of sound energy into the surface on the side of incidence to the incident rate of flow. The sabin is a unit of equivalent absorption and is equal to the equivalent absorption of 1 square foot of a surface of unit absorptivity, that is, of 1 square foot of surface which absorbs all incident sound energy.

From equation 11.1 the time required for the sound in a room to decay to one millionth of its original intensity is

$$T = .050 \frac{V}{A} \qquad\qquad 11.2$$

where T = time, in seconds,

$\quad V$ = volume, in cubic feet, and

$\quad A$ = total absorption, in sabins.

Later work[13] has shown that equation 11.2 is unsatisfactory for large rooms or rooms with very large absorption. The equation developed by Eyring is

$$T = \frac{.05V}{-S \log_\epsilon (1 - a_{av})} \qquad\qquad 11.3$$

where V = volume, in cubic feet,

$\quad S$ = total area, in square feet, and

$\quad a_{av}$ = average absorption per square foot, in sabins.

A tabulation of sound absorption coefficients for various building materials and objects is shown in Table 11.1. The coefficients in this table were obtained upon small samples in chambers having long reverberation times. In general, these measurements do not agree with those obtained under actual conditions in practice. That is, field measurements yield smaller values than laboratory measurements. However, the values of Table 11.1 show the relative absorption coefficients of the various materials. For a complete résumé of this subject, see the Anniversary issue of the *Journal of the Acoustical Society of America*, Vol. 11, No. 1, Part 1, July, 1939.

There are a number of methods available for measuring the decay of sound in a room. Sabine and others have used an organ pipe and stop watch and have determined by ear the time required for the sound to decay to one millionth of its original intensity. At least two dozen instrumental methods have been developed for the measuring of the reverberation time of a room. At the present time, high-speed level indicators and recorders appear to be the most suitable means for obtaining the reverberation time of a room (see Sec. 10.3A2).

[13] Eyring, C. F., *Jour. Acous. Soc. Amer.*, Vol. 1, No. 2, p. 217, 1930.

TABLE 11.1. ABSORPTION COEFFICIENTS OF VARIOUS ACOUSTICAL MATERIALS, BUILDING MATERIALS AND OBJECTS

Material	Thick-ness (in.)	Mount-ing	Frequency						Author
			128	256	512	1024	2048	4096	
			Coefficient						
Corkoustic	1¼	2	.11	.34	.67	.47	.57	.53	A.M.A.
Cushionstone	⅞	2	.17	.58	.70	.90	.76	.71	,,
Sanacoustic Pad, with Metal Facing	1⅛	3	.25	.56	.99	.99	.91	.82	,,
Fibretex	1	2	.14	.28	.81	.94	.83	.80	,,
Absorbatone	1	2	.15	.28	.82	.99	.87	.98	,,
Acoustex 60R	1	2	.14	.28	.81	.94	.83	.80	,,
Fiberglass Tile	1	2	.22	.46	.97	.90	.68	.52	,,
Acoustone F.	15/16	1	.16	.31	.87	.92	.83	.87	,,
Acousti-Celotex C-4	1¼	2	.28	.56	.98	.78	.59	.49	,,
Acousteel Pad with Metal Facing	1¼	3	.25	.52	.99	.99	.81	.60	,,
Transite Acoustical Panels	1 3/16	2	.32	.58	.72	.85	.76	.67	,,
Q-T Ductliner	1	4	.29	.41	.78	.89	.88	.78	,,
Hushkote Plaster	½	2	.13	.24	.45	.71	.56	.49	B.S.
Sabinite Plaster	½	2	.26	.16	.32	.70	.73	.72	A.M.A.
Draperies Hung Straight, in Contact with Wall, Cotton Fabric, 10 oz. per sq. yd.			.04	.05	.11	.18	.30	.44	P.S.
The Same, Velour, 18 oz. per sq. yd.			.05	.12	.35	.45	.40	.44	,,
The Same as Above, Hung 4″ from Wall			.09	.33	.45	.52	.50	.44	,,
Felt, All Hair, Contact with Wall	1		.13	.41	.56	.69	.65	.49	,,
Balsam Wool, Paper Backing and Cloth Covering	1		.14	.33	.50	.71	.70	.60	,,
Rock Wool	1		.35	.49	.63	.80	.83		V.K.
Carpet, on Concrete	0.4		.09	.08	.21	.26	.27	.37	B.R.
Carpet, on ⅛″ Felt on Concrete	0.4		.11	.14	.37	.43	.27	.27	B.R.
Cork Board	1			.08	.30	.31	.28		F.W.
Firetex, on 2″ × 4″ − 16″ O.C.	¼		.22	.21	.28	.31	.44	.55	V.K.
Masonite, on 2″ × 4″ − 16″ O.C.	½		.18	.25	.32	.35	.33	.31	V.K.
Concrete, Unpainted			.010	.012	.016	.019	.023	.035	V.K.
Wood Sheeting, Pine	0.8		.10	.11	.10	.08	.08	.11	W.S.

Abbreviations in the above table are as follows: A.M.A., American Materials Association; W.S., Wallace Sabine; P.S., P. E. Sabine; F. W., F. R. Watson; V.K., V. O. Knudsen; B.R., Building Research Station, England; B.S., U.S. Bureau of Standards. Mountings in the above table are as follows:

1. Cemented to plaster board.
2. Nailed to 1″ × 2″ furring 12″ O.C.
3. Attached to metal supports applied to 1″ × 2″ wood furring.
4. Laid on 24 ga. sheet iron, nailed to 1″ × 2″ wood furring 24″ O.C.

TABLE 11.1. (*Continued*)

Material	Thick-ness (in.)	Mount-ing	Frequency						Author
			128	256	512	1024	2048	4096	
			Coefficient						
Brick Wall, Un-painted.			.024	.025	.031	.042	.049	.070	W.S.
Brick Wall, Painted .			.012	.013	.017	.020	.023	.025	W.S.
Concrete Porous Block, Set in 1:3 Cement, Sand, Mortar.	2		.15	.21	.43	.37	.39	.51	B.R.
Plaster, Lime on Wood Lath on Wood Studs, Rough Finish .	½		.039	.056	.061	.089	.054	.070	P.S.
Plaster, Gypsum on Wood Lath on Wood Studs, Rough Finish .	½		.023	.039	.039	.052	.037	.035	P.S.
Ozite	¾		.09	.19	.28	.51	.56	.47	P.S.

Individual Object	Absorption Units in sq. ft. (Sabins)						
Audience, per Person, Man with Coat .	2.3	3.2	4.8	6.2	7.6	7.0	B.S.
Auditorium, Chairs, Solid Seat and Back .	.15	.22	.25	.28	.50		P.S.
Auditorium Chairs, Upholstered .		3.1	3.0	3.2	3.4		F.W.

Abbreviations in the above table are as follows: A.M.A., American Materials Association; W.S., Wallace Sabine; P.S., P. E. Sabine; F.W., F. R. Watson; V.K., V. O. Knudsen; B.R., Building Research Station, England; B.S., U.S. Bureau of Standards. Mountings in the above table are as follows:
1. Cemented to plaster board.
2. Nailed to 1″ × 2″ furring 12″ O.C.
3. Attached to metal supports applied to 1″ × 2″ wood furring.
4. Laid on 24 ga. sheet iron, nailed to 1″ × 2″ wood furring 24″ O.C.

B. *Mechanism of Sound Absorption by Acoustical Materials.*[14,15,16,17,18,19,20,21,22]—The mechanism of sound absorption may be illustrated by means of the acoustical impedance concept. This phase of sound absorption of

[14] Rettinger, M., *Jour. Acous. Soc. Amer.*, Vol. 6, No. 3, p. 188, 1935.
[15] Gemant, A., *Jour. App. Phys.*, Vol. 12, No. 10, p. 725, 1941.
[16] Morse, Bolt, and Brown, *Jour. Acous. Soc. Amer.*, Vol. 12, No. 2, p. 217, 1940.
[17] Brown and Bolt, *Jour Acous. Soc. Amer.*, Vol. 13, No. 4, p. 337, 1942.
[18] Beranek, L., *Jour. Acous. Soc. Amer.*, Vol. 13, No. 3, p. 248, 1942.
[19] Morse and Bolt, *Rev. Mod. Phys.*, Vol. 16, No. 2, p. 69, 1944.
[20] Rettinger, "Applied Architectural Acoustics," Chemical Publishing Company, Brooklyn, N.Y., 1947.
[21] Knudsen and Harris, "Acoustical Designing in Architecture," John Wiley and Sons, New York, N.Y., 1950.
[22] Beranek, "Acoustics," McGraw-Hill Book Company, New York, N.Y., 1954.

acoustical materials has been considered by a number of investigators. Expressions have been worked out for the normal acoustical impedance per unit area of the acoustical material. It is the purpose of this section to illustrate the concept of acoustical impedance as applied to acoustical materials. The considerations will be confined to the frequency range in which the thickness of the material is small compared to the wavelength. It will be assumed that the back of the material is placed in contact with a rigid wall as shown in Fig. 11.1. The acoustical impedance, in acoustical ohms per unit area, is given by

$$z_{A1} = \frac{r_{A1D}d}{3} + \frac{j\omega dm\rho}{3} - \frac{j\rho c^2}{\omega Pd} \qquad 11.4$$

where r_{A1D} = d-c acoustical resistance of the material per unit cube, in acoustical ohms,

     d = thickness of the material, in centimeters,

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

     m = ratio of the effective density of the air in the pores to its density in the open,

     P = porosity, the ratio of the volume of air in the pores to the total volume,

     c = velocity of sound, in centimeters per second,

     ω = $2\pi f$, and

     f = frequency, in cycles per second.

In Fig. 11.1, the acoustical resistance, in acoustical ohms, per unit area is

$$r_{A1} = \frac{r_{A1D}d}{3} \qquad 11.5$$

The inertance per unit area is

$$M_1 = \frac{dm\rho}{3} \qquad 11.6$$

The acoustical capacitance per unit area is

$$C_{A1} = \frac{Pd}{\rho c^2} \qquad 11.7$$

Methods for measuring the d-c acoustical resistance or flow resistance and porosity of acoustical materials are given in Secs. 10.9 and 10.10. The value of the a-c acoustical resistance and density ratio m will depend upon the motion of the porous material itself if it is a yielding structure. For materials with a rigid structure the dynamical acoustical resistance equals the d-c acoustical resistance. The value of m is a little more obscure. In general, the effective density of the gas particles is greater than the actual density.

The measured and computed acoustical resistance and reactance for Permacoustic as a function of frequency are shown in Fig. 11.2. The computed values were obtained from d-c acoustical resistance measurements

and porosity measurements. For this material the d-c acoustical resistance was found to be 220 acoustical ohms per unit cube and the porosity was determined as .85. There does not appear to be any satisfactory
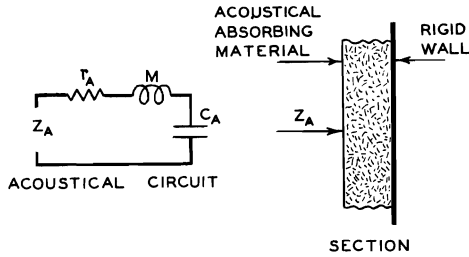
FIG. 11.1. Sectional view of sound absorbing material and the acoustical circuit. In the acoustical circuit: z_A = the acoustical impedance of the material per unit area. r_A, M, and C_A = the acoustical resistance, inertance and acoustical capacitance of the material.

method for determining m save by assuming a reasonable value. In this case m was assumed to be 1.5. At the low frequencies the computed acoustical resistance is somewhat larger than the measured value. This means that the d-c acoustical resistance is somewhat greater than the a-c
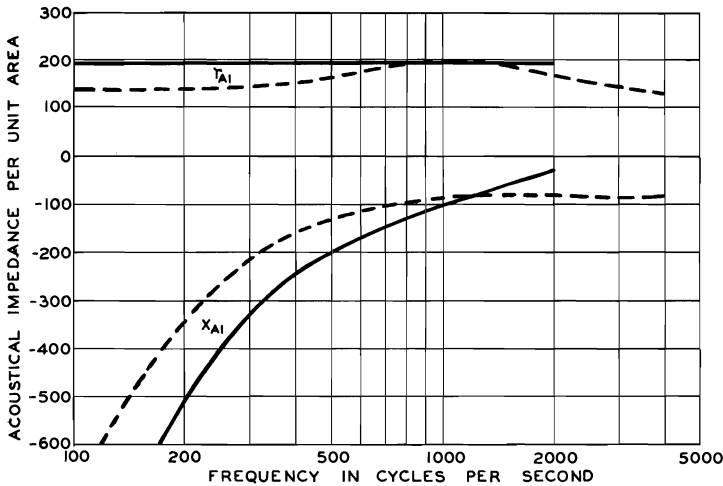
FIG. 11.2. The acoustical resistance, x_{A1}, and acoustical reactance, r_{A1}, characteristics of Permacoustic 1 inch in thickness and backed by a rigid wall. Dashed lines: experimentally measured characteristics. Solid lines: computed characteristics. The ordinates represent the acoustical impedance per square centimeter.

acoustical resistance. The computed acoustical reactance is larger than the measured value.

In the case of a material like Permacoustic, the a-c and d-c acoustical resistances are practically the same. This would be the case if sound energy enters the material by air penetration and the absorption is due to viscosity (see Sec. 5.2). If the sound energy enters the material by compressional vibration and is absorbed by internal damping in addition to viscous damping in the pores the d-c acoustical resistance will be much greater than the a-c acoustical resistance. In some cases the d-c acoustical resistance may be as much as a hundred times the a-c acoustical resistance.

The absorption coefficient may be obtained from the acoustical impedance by means of the following equation,

$$\alpha = 1 - \left|\frac{z_{A1} - \rho c}{z_{A1} + \rho c}\right|^2 \qquad\qquad 11.8$$

where z_{A1} = acoustical impedance of the material, in acoustical ohms per square centimeter,

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

c = velocity of sound, in centimeters per second.

Equation 11.8 may be written

$$\alpha = \frac{4 r_{A1} \rho c}{(r_{A1} + \rho c)^2 + x_{A1}{}^2} \qquad\qquad 11.9$$

where r_{A1} = acoustical resistance of the material, in acoustical ohms per square centimeter, and

x_{A1} = acoustical reactance of the material, in acoustical ohms per square centimeter.

The absorption coefficient frequency characteristic computed from the data of Fig. 11.2 and equation 11.9 and the absorption coefficient frequency characteristic as obtained from reverberation chamber measurements are shown in Fig. 11.3. The absorption coefficient obtained in the reverberation chamber is larger than the coefficient computed from acoustical impedance measurements. In general, the absorption coefficient obtained in reverberation chambers is larger[23] than the values obtained under actual conditions.

C. *Functional Sound Absorbers.*[24,25]—Conventional sound absorbing materials are designed to serve a twofold function—namely, as a building material and as a sound absorber. Because of this compromise the sound absorbing efficiency is low. There are certain applications where the principal problem is to absorb sound. There are some rooms where conventional

[23] Morris, Nixon and Parkinson, *Jour. Acous. Soc. Amer.*, Vol. 9, No. 3, p. 234, 1938.
[24] Olson, H. F., *RCA Review*, Vol. 7, No. 4, p. 503, 1946.
[25] Functional sound absorbers are commercially available in various shapes, materials, and constructions.

materials cannot be applied to the ceiling and walls. For these applications the logical solution is the use of a functional sound absorber of relatively high efficiency. The amount of energy absorbed depends upon the sound pressure of the source and the acoustical impedance of the medium and the sound absorber. The acoustical impedance of the medium and the sound absorber are controlled by the design of the sound absorber. It is the
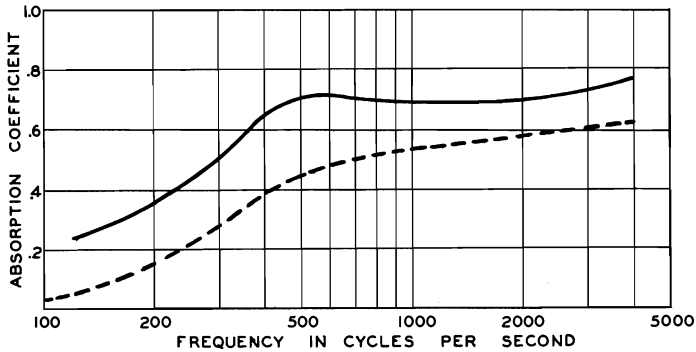
FIG. 11.3. The absorption coefficient characteristics of Permacoustic 1 inch in thickness and backed by a rigid wall. Solid line: reverberation chamber measurements. Dashed line: computed from acoustical impedance characteristic of Fig. 11.2.

purpose of this section to describe a functional sound absorber of high efficiency.

Conventional acoustical absorbing materials are employed as a wall covering on the boundaries of the room. The absorbing mechanism may be depicted by an acoustical network with lumped constants. A consideration of the acoustical circuit of Fig. 11.4A shows that the maximum absorption occurs when the acoustical impedance, z_A, of the material is an acoustical resistance equal to the characteristic acoustical resistance, r_{AG}, of air. Under these conditions the absorption of sound is 100 per cent. In the case of the absorbing wall, the maximum efficiency that can be obtained is 100 per cent, because the ratio of the area of the wavefront to the area of the wall is unity. In most practical cases, 100 per cent absorption is not attainable because a material with high absorption is not suitable as a wall material and the average absorption is usually about 50 per cent. To increase the absorption beyond 100 per cent requires a reduction in the value of the generator acoustical impedance. This can be accomplished by the use of diffraction as shown in Fig. 11.4B. For this condition the value of the source acoustical impedance, z_{AG}, can be made very small. A consideration of the acoustical circuit shows that an appropriate value for the absorber acoustical impedance, z_A, will yield an absorption coefficient which is more than unity. The above lumped constant theory applies in the frequency

region in which the dimensions of the absorber are small compared to the wavelength.

The functional sound absorber is made in the form of a thin shell of acoustical absorbing material. The magnitude of the acoustical resistance of the shell is selected to yield the highest absorption of sound.

The absorption coefficient frequency characteristic of a typical functional sound absorber, obtained from reverberation chamber measurements, is shown in Fig. 11.4C. It will be seen that the absorption per unit area is about two times that of conventional materials. The shell-type functional
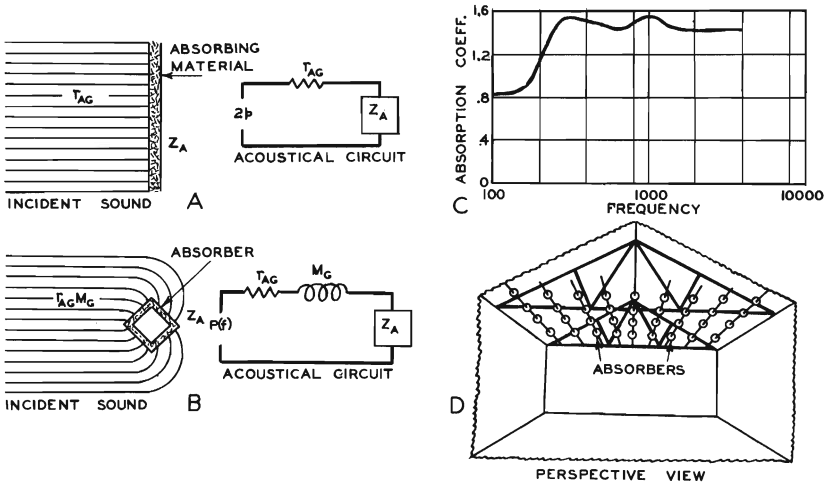
FIG. 11.4. A. Conventional sound absorbing material. In the acoustical circuit, r_{AG} = the characteristic acoustical impedance of air for a plane wave. z_A = the acoustical impedance of the sound absorbing material. p = the sound pressure in free space. B. Functional sound absorber. In the acoustical circuit, r_{AG} and M_G = the acoustical resistance and inertance of the acoustical generator. z_A = the acoustical impedance of the functional sound absorber. $p(f)$ = the actuating sound pressure. C. The absorption coefficient frequency characteristic of the functional sound absorber. D. Perspective view of an installation of the functional sound absorber.

sound absorber is very economical in the use of material, because the sound absorption, in sabins per pound, is about twelve times that of conventional absorbing materials.

The functional sound absorber is very easy to install because it is merely suspended on wires. A typical installation is shown in Fig. 10.4D. This is a truss type roof where the installation of conventional materials would be very costly. In the case of skylights, conventional materials, installed as a false ceiling, of course, impair the lighting. On the other hand, the functional sound absorber will not impair skylighting. It is particularly useful in industrial applications where the principal objective is to absorb sound and appearance is not a factor.

D. *Resonator Sound Absorber*.[26,27,28]—The resonator sound absorber consists of a simple resonant acoustical system. One type consists of a Helmholtz resonator with some absorbing material located in the cavity of the resonator, or with a cloth over the mouth to provide additional acoustical resistance. The acoustical performance is practically the same for all types of simple resonators. Therefore, a consideration of the performance of the Helmholtz type will indicate the characteristics of resonant sound absorbers.

A sectional view and acoustical network of a Helmholtz resonator are shown in Fig. 11.5. The performance of the system can be determined from the acoustical network and the constants of the system. In one application, the problem is to reduce the sound pressure over a small volume. Under these conditions, the acoustical resistance r_{A2} should be made as small as possible. A typical measured sound-pressure frequency characteristic at the mouth of the resonator is shown in Fig. 11.6. The characteristic shows that there is a reduction in sound pressure in the frequency range above 72 cycles and an increase in sound pressure in the frequency range below 72 cycles. Since there is an increase of pressure over a certain portion of the frequency range, it is obvious that the simple resonator is not suitable for a free-field, spot-type sound reducer.

When the resonator is used as a sound absorber, the action is somewhat different. In order to obtain the maximum sound absorption, the acoustical resistance r_{A1} should be equal to the acoustical resistance, r_{A2}. This can be accomplished by selecting the proper value of the acoustical resistance r_{A2} provided by the cloth over the opening of the resonator. If a bank of resonators is used, so that the ultimate acoustical resistance r_{A1} is obtained, the maximum efficiency of sound absorption will be obtained. A typical sound-absorption frequency characteristic of a Helmholtz resonator is shown in Fig. 11.7. An examination shows that a high value of sound absorption is obtained over only a very narrow frequency range. At least twenty resonators would be required to cover the frequency range from

SECTIONAL VIEW

ACOUSTICAL CIRCUIT
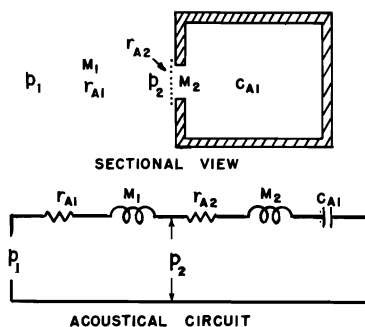
FIG. 11.5. Sectional view and acoustical network of a Helmholtz resonator sound absorber. In the acoustical network, p = the sound pressure in free space. M_1 and r_{A1} = the inertance and acoustical resistance of the air load. M_2 = the inertance of the aperture. C_{A1} = the acoustical capacitance of the volume. r_{A2} = the acoustical resistance of the cloth over the opening of the resonator. p_2 = the sound pressure at the mouth.

[26] Harris and Malloy, *Jour. Acous. Soc. Amer.*, Vol. 24, No. 1, p. 1, 1952. This paper contains references to publications of other investigators on the subject of resonator sound absorbers.

[27] Ingard, Uno, *Jour. Acous. Soc. Amer.*, Vol. 25, No. 6, p. 1037, 1953.

[28] Olson and May, *Jour. Acous. Soc. Amer.*, Vol. 24, No. 6, p. 1130, 1953.
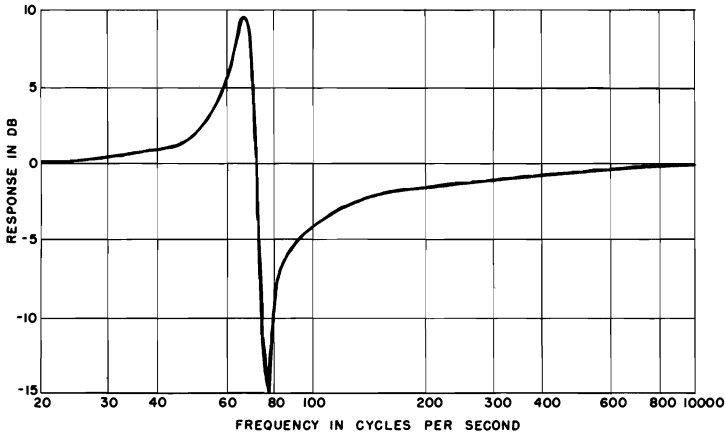
FIG. 11.6.   Sound pressure reduction frequency characteristic of a Helm-
holtz resonator having a volume of 2 cubic feet.

30 to 200 cycles with tolerably good absorption.   Since the average cubical
content of each resonator is two cubic feet, the total cubical content would
be 40 cubic feet.   Since a bank of resonators is required for each frequency,
the entire assembly becomes an arrangement of tremendous bulk.
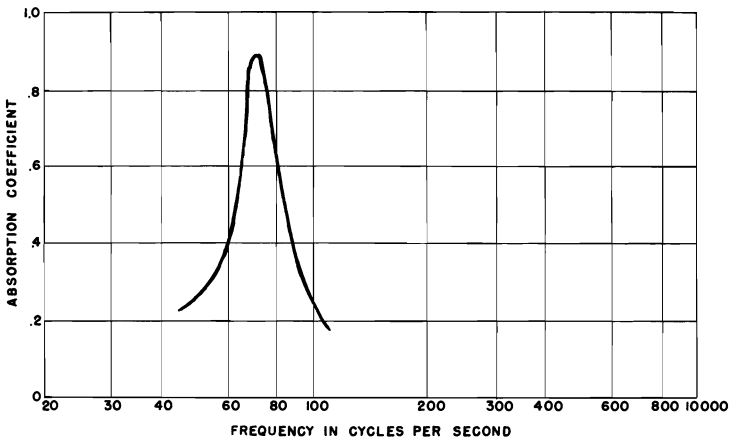
FIG. 11.7.   Sound absorption frequency characteristic of a Helmholtz
resonator having a volume of 2 cubic feet.

Resonators with inertance in the form of a flat diaphragm with acoustical
resistance damping and coupled on the back side by an acoustical capacitance
in the form of a volume has also been employed as a resonator-type sound
absorber.

E. *Electronic Sound Absorber.*[29]—The conventional sound absorber is a passive type of sound absorber in which the mechanism of sound absorption is obtained by the direct conversion from acoustical to heat energy. On the other hand, the electronic sound absorber is an active type of sound absorber in which the mechanism of sound absorption is obtained by means of an electronic system. The electronic sound absorber has been described in Sec. 9.18B.

Conventional sound absorbing materials with high efficiency in the low-frequency range are extremely bulky. Therefore, conventional sound absorbing systems are unsuitable where compact high-efficiency sound absorbing systems are required. When the electronic sound absorber is commercialized it will provide a compact system of high-absorbing efficiency in the low-frequency range.

The electronic sound absorber may be used as a diffraction sound absorber as shown in Fig. 11.8. In order to obtain a high acoustical resistance load, the absorbers should be mounted in the corner at the intersection of the three surfaces, as shown in Fig. 11.8. As contrasted to the resonant sound absorber with a narrow frequency range of sound absorption, the electronic sound absorber exhibits high efficiency over several octaves in the low-frequency portion of the audio-frequency range.

Fig. 11.8. Three electronic sound absorbers located in the corner of a room.

F. *Articulation and Reverberation Time.*—The articulation[30] (see Sec. 10.16) of unamplified speech in auditoriums of various sizes as a function of the reverberation time is shown in Fig. 11.9. The average power of unamplified speech is much less than that required for distinct hearing. A greater reverberation time increases the intensity of sound at the auditor. However, increased reverberation decreases the intelligibility of speech. These two factors oppose each other with the result that there is an optimum reverberation for each auditorium which yields maximum articulation as shown in Fig. 11.9. The obvious solution is the use of sound reinforcing equipment. The articulation for a weak, average, and loud talker without amplification as compared to amplified speech is shown in Fig. 11.9. By proper selection and placement of the loudspeakers the articulation characteristic may be made considerably higher.

G. *Sound Motion Picture Reproducing Systems.*[31]—The resultant sound

[29] Olson and May, *Jour. Acous. Soc. Amer.*, Vol. 25, No. 6, p. 1130, 1953.
[30] Knudsen, V. O., *Jour. Acous. Soc. Amer.*, Vol. 9, No. 3, p. 175, 1938.
[31] Olson and Massa, *Jour. Soc. Mot. Pict. Eng.*, Vol. 23, No. 2, p. 63, 1934.

energy density at the absorption of the auditor in a theater depends upon the response and the directional characteristics of the loudspeaker and upon the reverberation characteristics of the theater. From the standpoint of the auditor, it may be said that there are two sources of sound energy—namely, the direct sound, which travels directly from the loudspeaker to the auditor; and the generally reflected sound, which is reflected from the boundaries before reaching the auditor.

In a theater, free from acoustical difficulties, the energy density of the generally reflected sound is practically the same for all parts of the theater.
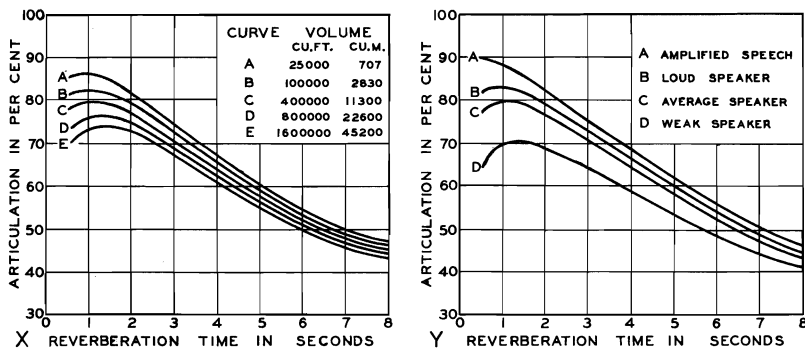
FIG. 11.9.   X. The articulation of a speaker in auditoriums of various volumes. Y. The articulation of amplified speech, loud, average, and a weak speaker in an auditorium of 400,000 cubic feet.   (After Knudsen.)

Therefore, the solution of the problem of achieving uniform sound energy density is to employ reproducers that will project the same direct sound energy to all parts of the theater. The example which follows will illustrate how this may be accomplished by employing a directional loudspeaker.

An elevation view of a reproducer in a theater is shown in Fig. 11.10. The two extreme points to be supplied are indicated as A and E. If the loudspeaker were nondirectional, the ratio of the direct sound energy densities at the two points would be inversely proportional to the ratio of the squares of the distances from the reproducer. In this particular case, the difference in level would be 13 db. Obviously, such a large variation in sound intensity precludes the possibility of satisfactory reproduction over the entire area to be supplied. Therefore, a compensating means must be provided to counteract the variation of intensity with the distance from the reproducer. The directional loudspeaker furnishes a solution of the problem.

The directional characteristics of the loudspeaker are shown in Fig. 11.10. In this particular case, the difference of level for a point 40° from the axis, as compared with the level at a point on the axis, is 13 db. The loudspeaker is adjusted until the axis of the characteristic passes through the point E. Then the position of the loudspeaker is adjusted until the angle θ is 40°. The distribution over the distance under consideration is shown in Fig. 11.10.

Summarizing, the variation of the sound pressure with the angle between the axis and the line joining the observation point and the reproducer has been employed to compensate for the decrease of the sound energy with the distance.

From the response frequency characteristic of the loudspeaker the pressure at any distance r centimeters on the axis may be obtained from the following equation.

$$p = p_0 \frac{x_0}{r} \qquad 11.10$$

where p_0 = pressure, in dynes per square centimeter, obtained at a distance x_0, in centimeters.

To obtain the pressure for a point not on the axis, the above equation must be multiplied by a factor obtained from the directional characteristic
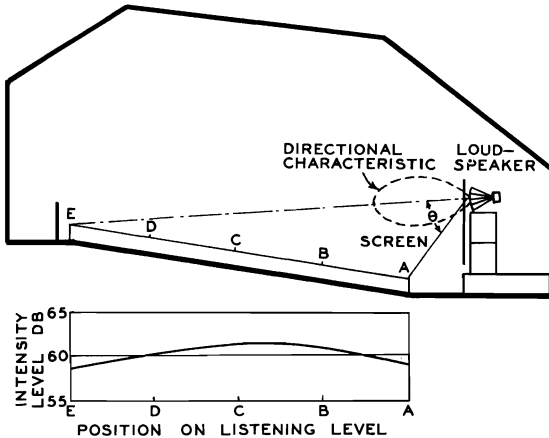
FIG. 11.10. Arrangement of the loudspeaker for a sound motion-picture reproducing system in a theater. The graph shows the intensity level at points along the listening level.

at this frequency. The direct radiation from the loudspeaker can then be obtained for any point in the space.

The energy density, ergs per cubic centimeter, due to direct radiation from the loudspeaker is

$$E_D = \frac{p_0^2 x_0^2 R_\theta^2}{r^2 \rho c^2} \qquad 11.11$$

where R_θ = ratio of the sound pressure at angle θ to $\theta = 0$,
ρ = density of air, in grams per cubic centimeter, and
c = velocity of sound, in centimeters per second.

To analyze the distribution of the direct sound over the area, the plan view of the theater and the directional characteristics of the reproducer in the horizontal plane must be considered. The angle subtended at the loud-speaker by the area to be covered will determine the effective dispersion angle of the reproducer.

The sound energy density due to the generally reflected sound is a function of the absorption characteristics of the theater and the power output of the reproducer. The sound energy density, ergs per cubic centimeter, due to the generally reflected sound is given by

$$E_R = \frac{4P}{caS} [1 - \epsilon^{(cS[\log_\epsilon (1-a)]t)/4V}] (1 - a) \qquad 11.12$$

where a = the average absorption per unit area, absorption coefficient,
  S = the area of the absorbing materials, in square centimeters,
  V = the volume of the room, in cubic centimeters,
  t = time, in seconds,
  c = the velocity of sound, in centimeters per second, and
  P = the power output of the loudspeaker, in ergs per second.

The total sound energy density at any point in the theater will be the sum of the direct and the generally reflected sound, and may be expressed by

$$E_T = E_D + E_R \qquad 11.13$$

A method has been outlined above, employing directional loudspeakers for obtaining a uniform energy distribution of the direct sound. The energy density of reflected sound, as shown by equation 11.12, is inde-pendent of the observation point. As a consequence, by employing direc-tional loudspeakers, the total sound energy density will be the same in all parts of the theater. Furthermore, the *effective reverberation* of the reproduced sound (the ratio of generally reflected to direct sound) is the same for all parts of the theater.

The distribution of a reproducing system in a theater is usually checked by means of a response measuring system. The plan and elevation view of a typical theater are shown in Fig. 11.11. The response characteristics for positions A, B, C, and D on the orchestra level and E and F on the balcony level are shown in the respective graphs. These characteristics show that it is possible to obtain uniform response in all parts of the theater by following the procedures outlined in the preceding discussions.

In sound motion picture reproduction, the loudspeakers are usually placed behind a perforated screen upon which the picture is projected. Fig. 11.11. Sound is transmitted through the screen by actual vibration of the screen and by the perforations. In general the transmission by vibration, is negligible. The perforations usually consist of small circular holes about a millimeter in diameter. These holes form an inertance and acoustical resistance (see Secs. 5.3 and 5.15). The acoustical resistance of the holes introduces attenuation which is usually small. The acoustical reactance

due to the inertance increases with frequency, and therefore the attenuation increases with frequency. The response frequency characteristic of the screen shows more or less constant attenuation in the low- and mid-frequency ranges due to the acoustical resistance of the holes. However, the attenuation in the high-frequency range increases with frequency due to the acoustical reactance of the holes. The inertance increases with the thickness of the screen and decreases as the ratio of the open to closed area of the screen increases. For example, for 3 db attenuation at 10,000 cycles the hole area is usually 15 to 20 per cent of the screen area. If the hole
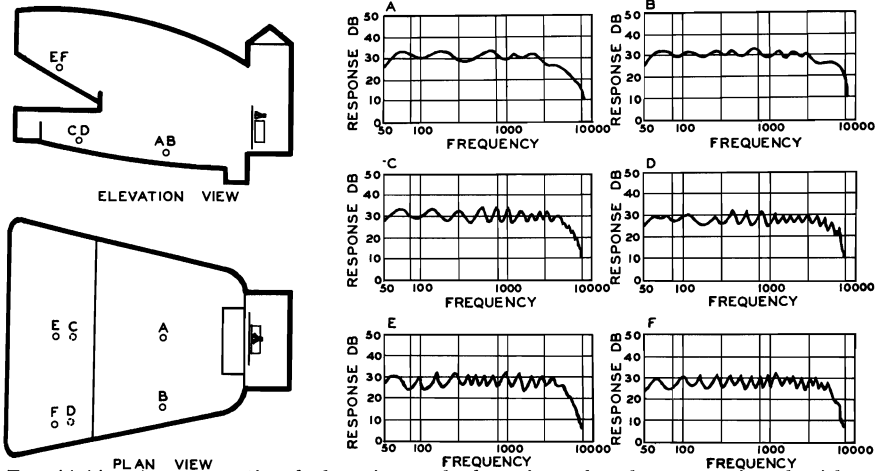
FIG. 11.11. A cross-sectional elevation and plan view of a theater equipped with a loudspeaker for the reproduction of sound motion pictures. The graphs show the response frequency characteristics in various parts of the theater.

area is 7 to 15 per cent the attenuation is about 6 db at 10,000 cycles and about 3 db at 5000 cycles. These examples show that the screen is an important problem in wide frequency range sound reproduction.

The sound motion-picture systems in the preceding considerations in this section and depicted in Figs. 10.10 and 10.11 are of the single-channel type. In the past three years wide screen systems with stereophonic sound[32] have been introduced on a wide scale. Stereophonic sound motion-picture reproducing systems are described in Secs. 9.9 and 13.7. The sound is reproduced from four magnetic tracks on the positive motion-picture film. Three of the channels supply the three loudspeaker stations located behind the screen. The other channel supplies loudspeakers distributed around the theater. The loudspeaker arrangement is depicted in Fig. 11.12. Each stage loudspeaker station should supply a uniform distribution of sound over the entire theater. The principles outlined for Figs. 11.10 and 11.11 are used to obtain a uniform distribution of sound.

[32] Stereophonic Sound, *Jour. Soc. Mot. Pic. Tel. Engs.*, Vol. 61, No. 3, Part II, p. 353, 1953. This issue contains several papers on stereophonic sound in motion pictures.

ELEVATION VIEW   PLAN VIEW
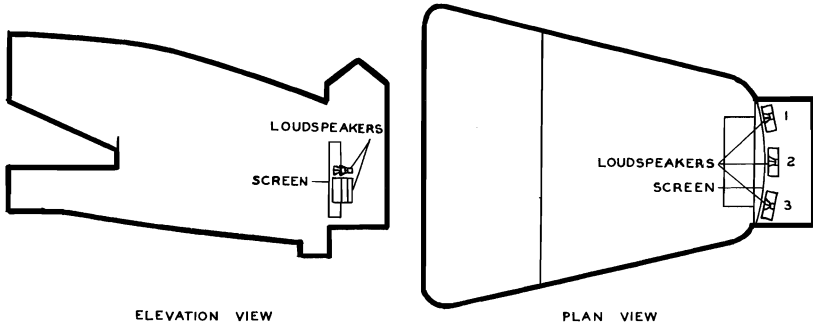
FIG. 11.12. A cross-sectional and plan view of a theater equipped with three loud-speakers behind the screen for stereophonic sound reproduction of sound motion pictures. Loudspeakers for special effects are located on the sides of the theater.

H. *Sound Reinforcing Systems.*[33]—A large theater equipped with a sound reinforcing system is shown in Fig. 11.13. Microphones are concealed in the footlight trough for collecting the sound on the stage and others are placed in appropriate positions in the orchestra. The loudspeakers are located above the stage in the proscenium arch. The volume control and microphone mixing system is usually located in a box or booth in the balcony.

In this system there are two sources of direct sound—namely, the original sound and the augmented sound from the loudspeakers. Usually the intensity of the original sound issuing from the stage will be quite satisfactory on the orchestra floor near the stage and, as a consequence, it is not necessary to augment the sound in this portion of the theater. As the distance from the stage increases, the original sound intensity decreases. To make up for this loss the sound energy from the loudspeakers must progressively increase toward the rear of the theater. It is the purpose of this section to describe and analyze a sound reinforcing system.

Consider the system shown in Fig. 11.13. If the distance between the source of the original sound and the point of observation is r centimeters, the sound energy density, in ergs per cubic centimeter, due to the direct sound is

$$E_{D1} = \frac{P_{D1}}{4\pi r^2 c} \qquad\qquad 11.14$$

where P_{D1} = power output of the sound source, in ergs per second, and
  c = velocity of sound, in centimeters per second.

The sound intensity may be obtained from the sound energy density by multiplying it by the velocity of sound. The reference intensity for intensity level comparisons is 10^{-9} ergs per second or 10^{-16} watts per square centimeter. The intensity level of a sound is the number of decibels above the reference level.

[33] Olson, H. F., *RCA Review*, Vol. 1, No. 1, p. 44, 1936.

The intensity level on the orchestra floor resulting from a sound source as, for example, a speaker or singer on the stage, is given by the curve O of Fig. 11.13. It will be seen that the intensity level of the direct sound in the rear of the house is inadequate for good hearing. The arrangement and characteristics of the sound reinforcing system should be chosen so that the resultant intensity level, due to the direct sound from the com-
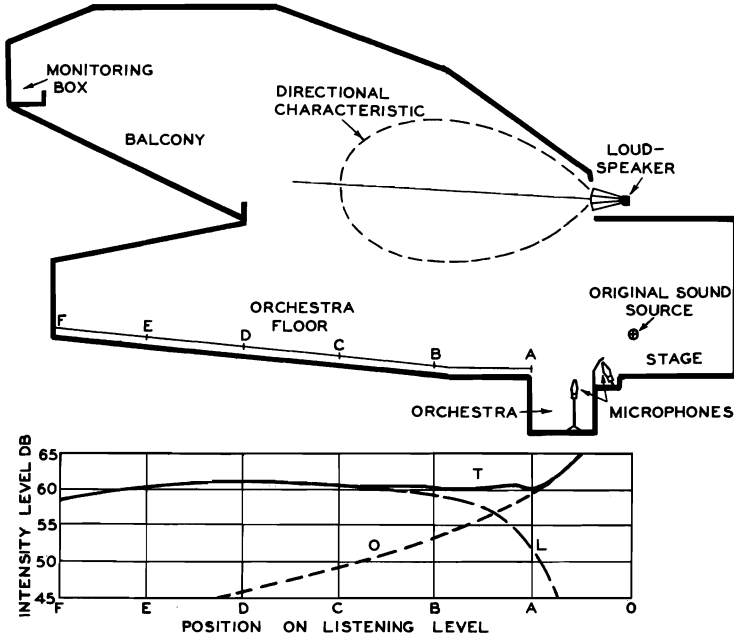
FIG. 11.13. Arrangement of the components of a sound reinforcing system in a theater. The graph shows the intensity level due to direct sound at the points indicated on the orchestra floor. Curve O is the intensity due to the original source of sound. Curve L is the intensity level due to the loudspeaker. Curve T is the resultant intensity level.

bination of the original source and the loudspeakers, is the same for all parts of the audience.

The energy density, in ergs per cubic centimeter, at a distance r centimeters due to direct radiation from the loudspeaker, from equation 11.11, is

$$E_{D2} = \frac{p_0{}^2 x_0{}^2 R_\theta{}^2}{r^2 \rho c^2} \qquad 11.15$$

where p_0 = pressure, in dynes per square centimeter, obtained at a distance x_0 centimeters,

R_θ = ratio of the pressure at the angle θ to $\theta = 0$,

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

c = velocity of sound, in centimeters per second.

The problem is to select a loudspeaker with suitable directional characteristics (see Sec. 11.2E), and then to adjust the power output and orientation so that the sum $E_{D1} + E_{D2}$ of equations 11.14 and 11.15 is a constant for all parts of the listening area of the theater. The intensity level on the orchestra floor, in Fig. 11.13, due to the direct sound from a loudspeaker having directional characteristics as shown, is given by the curve L. The intensity level due to the combination of the original sound and augmented sound from the loudspeaker is shown by curve T of Fig. 11.8. The resultant intensity is quite uniform over the orchestra floor. A similar analysis will show that the intensity level in the balcony is also relatively uniform. Further consideration of the characteristic of Fig. 11.13 shows that the total intensity level characteristic remains uniform when the output of the loudspeakers, that is, the gain in augmented sound, is varied over wide limits.

The energy density, in ergs per cubic centimeter, in the theater due to generally reflected sound is

$$E_R = \frac{4(P_{D1} + P_{D2})}{caS} \left[1 - \epsilon^{(cS[\log_\epsilon (1-a)] \, t)/4V}\right] (1 - a) \qquad 11.16$$

where a = average absorption per unit area, absorption coefficient,
$\quad\quad S$ = area of absorbing materials, in square centimeters,
$\quad\quad V$ = volume of the theater, in cubic centimeters,
$\quad\quad t$ = time, in seconds,
$\quad\quad c$ = velocity of sound, in centimeters per second,
$\quad P_{D1}$ = power output of the original sound, in ergs per second, and
$\quad P_{D2}$ = power output of the loudspeaker, in ergs per second.

The aid obtained from reflected sound in a directional sound reinforcing system is relatively small, ranging from 2 to 6 db.

The preceding considerations have been concerned with a sound reinforcing system for a large theater or auditorium. In the system of Fig. 11.13 a single loudspeaker station is employed. This is sometimes termed a high-level sound system. This is in contrast to a sound system employing a large number of loudspeakers distributed over the area to be covered. The latter system is termed a low-level sound system. Examples of these two types of sound reinforcing systems[34] are depicted in Fig. 11.14. In Fig. 11.14A the loudspeakers are located in a cluster above the stage. In Fig. 11.14B the loudspeakers are distributed around the walls of the room. Either system may be used with equally good results. The single loudspeaker station located above the original sound source produces somewhat more realistic results since the original and augmented originate in the same general direction. In the system of distributed loudspeakers it is somewhat easier to obtain good results because the feedback difficulties are reduced. Furthermore, the problem of uniform distribution of the sound

[34] Olson, "Musical Engineering," McGraw-Hill Book Company, New York, N.Y., 1952.

is simpler. The latter arrangement is almost universally used for school-rooms and churches.

In a sound reinforcing system of the type shown in Fig. 11.14B for most locations in the room, the sound will not appear to be from the source on the stage, but from the loudspeaker nearest to the listener. This is due to the delay effect described in Sec. 12.18. This disconcerting condition can
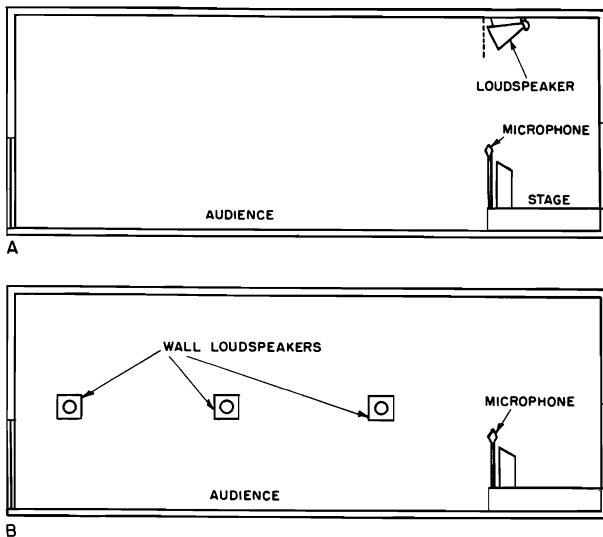
FIG. 11.14. Arrangements for a sound-reinforcing system in a large room, as for example, a schoolroom, a church, an auditorium, etc. A. A horn loudspeaker cluster located above the stage. B. A large number of direct-radiator loudspeakers distributed around the walls of the room.

be overcome by employing a delay system[35] as shown in Fig. 11.15. The time delay between loudspeaker 1 and loudspeaker 2 or 3 is made the same or greater than the time required for sound to travel in air from loudspeaker 1 to loudspeaker 2 or 3. The total time delay between loudspeaker 1 and loudspeakers 4 and 5 is made the same or greater than the time required for sound to travel in air from loudspeaker 1 to loudspeakers 4 and 5, etc. If a system of the type shown in Fig. 11.15 is used, the sound will always appear to emanate from the stage. Furthermore, the intelligibility will be far superior to the conventional system of Fig. 11.14B without delay.

The microphones for collecting the sounds are usually concealed in the footlight trough.[36] By employing directional loudspeakers, as outlined above, the sound level at the microphones due to the loudspeakers is low and thereby reduces the possibility of oscillations due to acoustic feedback or regeneration in the reproducing system. In large theaters, having an

[35] Haas, H., *Akustica*, Vol. 1, No. 2, p. 49, 1951.
[36] Olson, H. F., *RCA Review*, Vol. 1, No. 1, p. 49, 1936.

expansive stage, the pickup distance will be very large. Consequently, the sound which reaches the microphones from the original source will be small and will require considerable amplification which increases the tendency for feedback. In cases where difficulties are experienced, due to acoustic feedback, a further reduction in coupling can be obtained by employing directional microphones. Furthermore, the stage collecting system
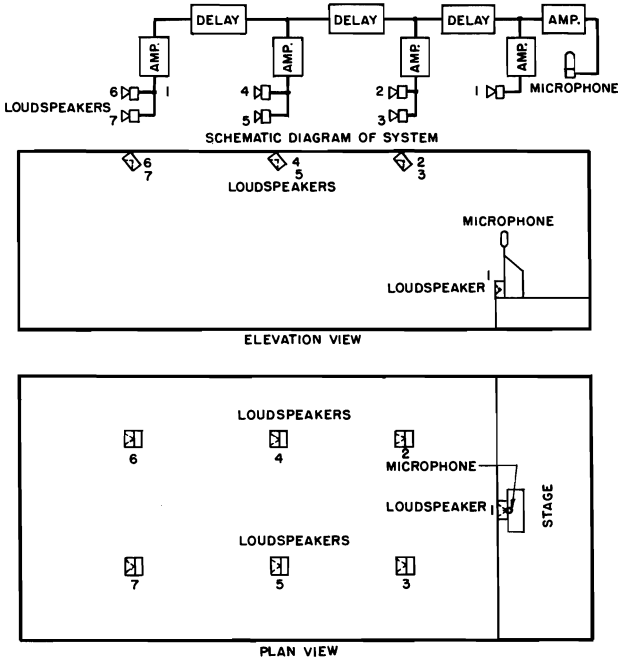
FIG. 11.15. Elevation and plan views of a sound reinforcing system with progressive delay between the microphone and the loudspeakers. The delay time is approximately the transit time of sound propagation in air from the microphone to the loudspeaker.

should not be responsive to sound originating in the orchestra or audience. In case the microphones are located in the footlights, the shielding effects of the apron, together with a velocity microphone, Fig. 11.16, are in general sufficient to accomplish this objective. Where it is impossible to shield the microphones in this manner the unidirectional microphone has been found to be very useful, as will be seen from a consideration of the directional characteristics of this microphone shown in Fig. 11.16.

In order to "cover" the action from any part of the stage several microphones are employed, usually spaced at intervals of 10 feet. The output

of each stage microphone and orchestra microphone is connected to a separate volume control on the mixer panel. This mixer and volume control system is located in the monitoring box. By means of this system the operator follows the action by selecting the microphone nearest the action on the stage. The operator also controls the ratio of the volume of the stage sound to that received from the orchestra when there is an orchestral accompaniment as well as the over-all intensity of the augmented sound. The monitoring box is usually located in the rear balcony, the position which is the most
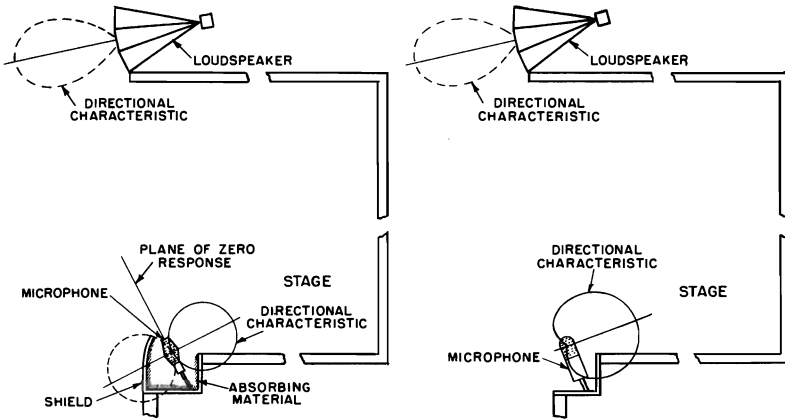
FIG. 11.16. Arrangements depicting the use of directional loudspeakers and microphones for reducing feedback between the loudspeaker and the microphone. The arrangement on the left employs a velocity microphone. A shield is used to reduce sound pickup from the orchestra and audience. The arrangement on the right employs a unidirectional microphone. The directional characteristics of this microphone are particularly adapted for collecting sounds on the stage and discriminating against sounds coming from the orchestra and audience.

susceptible to the augmented sound and one which also furnishes a good view of the action.

I. *Theater Acoustics.*[37]—Many theaters have major acoustical defects which cause echoes and objectionable concentrations due to focusing of the reflected sound. These reflections may be more important than the reverberation time.

When a sound wave strikes a wall of a theater, a part is reflected, a part absorbed, and a part transmitted. The reflection, for surfaces large compared to the wavelength, is analogous to specular reflection. The reflected sound, in a poorly designed theater, produces highly concentrated zones of reflected energy. For proper sound reflection control in an auditorium the acoustical treatment and shape of the walls and ceiling must be such as to thoroughly diffuse the reflected sound. The reflected sound energy received in any auditorium location should not come from one particular reflecting

[37] Standardization Committee, *Jour. Soc. Mot. Pic. Eng.*, Vol. 36, No. 3, p. 267, 1941.

area but should be contributed by numerous reflecting surfaces. The sound energy from any reflection should be small compared to the total reflected sound energy at any point in the auditorium. This also provides a more uniform decay of the reverberant sound.

Two of the most common defects in a theater attributable to poor shape design are echoes and sound concentrations. These, as well as other defects,
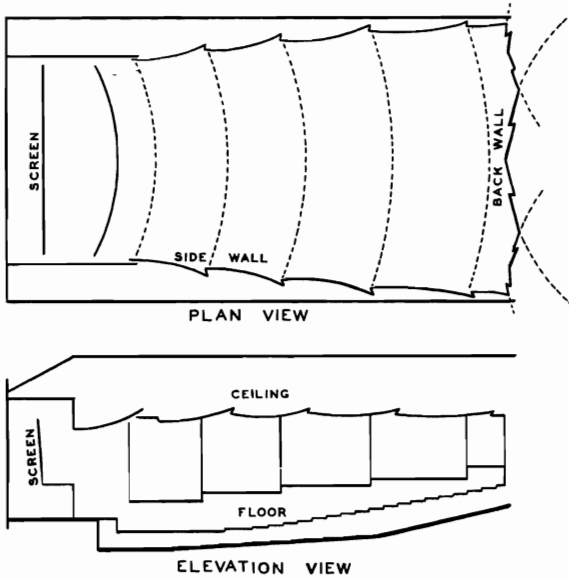
FIG. 11.17. Plan and elevation of a theater designed for good acoustics.

can be avoided and optimum results obtained by observing the following general rules.

(1) The cubical content should be kept to a minimum consistent with the number of seats required.

(2) The auditorium width should be from 50 to 70 per cent of the length, and the ceiling height not more than 40 per cent of the length.

(3) Nonparallel surfaces should be used.

(4) Convex, rather than concave, walls and ceiling sections should be provided. The wall and ceiling surfaces should also otherwise be broken up so as to diffuse the sound thoroughly.

(5) The average absorption per square foot on the floor and ceiling should not be appreciably different from the average absorption per square foot on the side walls.

(6) Well-upholstered seats and ozite-lined carpet in the aisles should be provided.

(7) The backstage volume should be so shaped and so acoustically treated that resonant reinforcements of sound will not be reflected into the auditorium to distort the sound quality.

The design of Fig. 11.17 is one method of applying the above principles to obtain the desired conditions. The fully convex rear wall and convex sections on the side walls and ceiling are ideal design features. However, a design including three convex surfaces on the rear wall as shown by the solid lines will also give excellent results.

J. *Reverberation Time of a Theater for the Reproduction of Sound.*—The optimum reverberation time of theaters for the reproduction or the reinforcement of sound as a function of the volume of the auditorium, for a
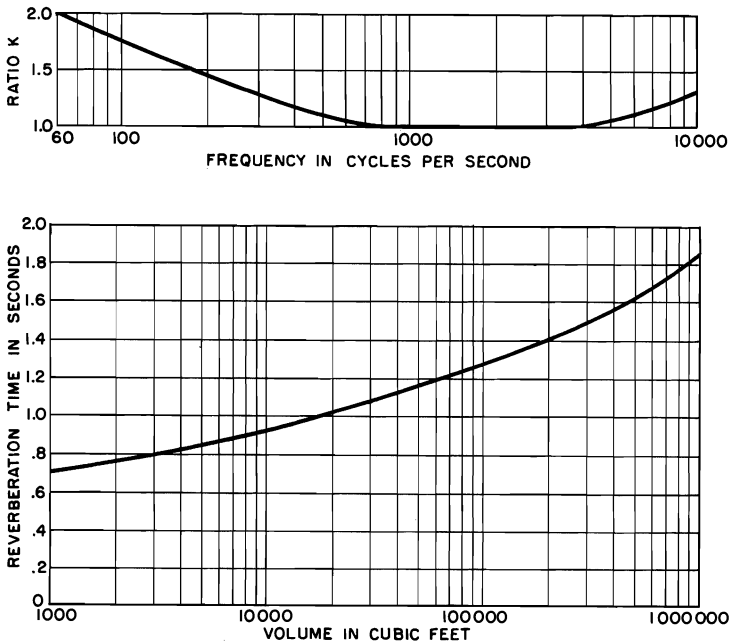
FIG. 11.18. Lower graph shows the optimum reverberation time for a theater as a function of the volume for 1000 cycles. Upper graph shows the relation between the reverberation time and the frequency, that is, the reverberation time at other frequencies is obtained by multiplying by K.

frequency of 1000 cycles, is shown in the lower graph of Fig. 11.18. The reverberation time for other frequencies can be obtained by multiplying by the factor K, obtained from the upper graph of Fig. 11.18. The reverberation time increases at the lower and higher frequencies so that the aural rate of decay of pure tones will be approximately the same for all frequencies (see Secs. 12.6 and 12.7 and Figs. 12.11 and 12.13).

K. *Power Requirements for Reproducing Systems.*[38]—The power require-ment is an important factor in the motion picture and sound reinforcing systems. The minimum intensity which these systems should be capable of producing is 80 db. 0 db = .0002 dyne per square centimeter. The graph of Fig. 11.19 shows the acoustical power required, as a function of the volume, in auditoriums to produce sound levels of 70, 80, 90, and 100 decibels. In large auditoriums where the orchestra is also reinforced the
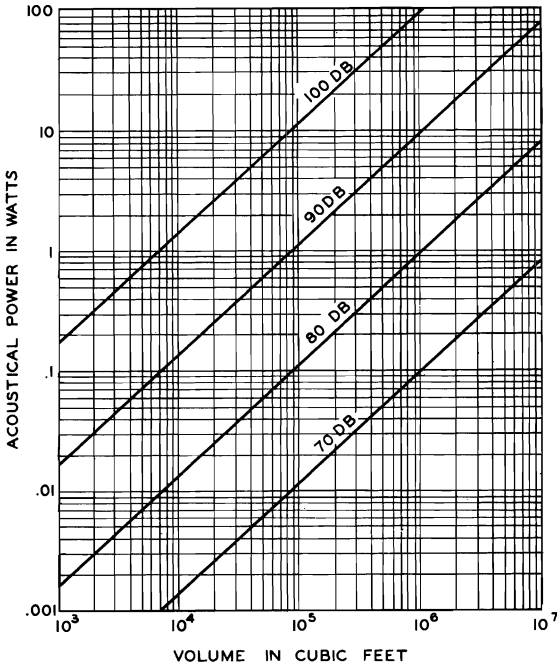
FIG. 11.19. The acoustical power required to produce sound intensity levels of 70, 80, 90, and 100 decibels in a room, or auditorium as a function of the volume of the room or auditorium.

power available should be greater. For example, to render full artistic appeal, the system should be capable of reproducing a level up to 100 db. This means a power of 100 times the power for 80 db. Systems for produc-ing this sound level without distortion usually require special amplifiers and loudspeakers.

L. *Noise at Different Locations.*—The ease with which speech may be heard and understood depends upon the noise conditions as well as upon the other characteristics of a sound reproducing system. The full artistic effects of musical reproduction can only be obtained with a wide volume

[38] Olson, H. F., *RCA Review*, Vol. 1, No. 1, p. 49, 1936.

range.   This volume range, of course, depends upon the noise level at the
listening point.   The tolerable level of the noises generated in any repro-
ducing system depends upon the noise level at the reproducing point.

The noise level[39] of residences, business offices, and factories is shown in
Fig. 11.20.   The reference level is 10^{-16} watts per square centimeter.   It
will be seen that there is a wide variation in the noise from one location to
another.   For example, 5 per cent of the residences have a noise level of
33 db while another 5 per cent have a noise level of 52 db.

The average noise level in empty theaters[40] is 25 db.   With an audience
the average noise level is 42 db.   It may rise to 48 db and go down to as
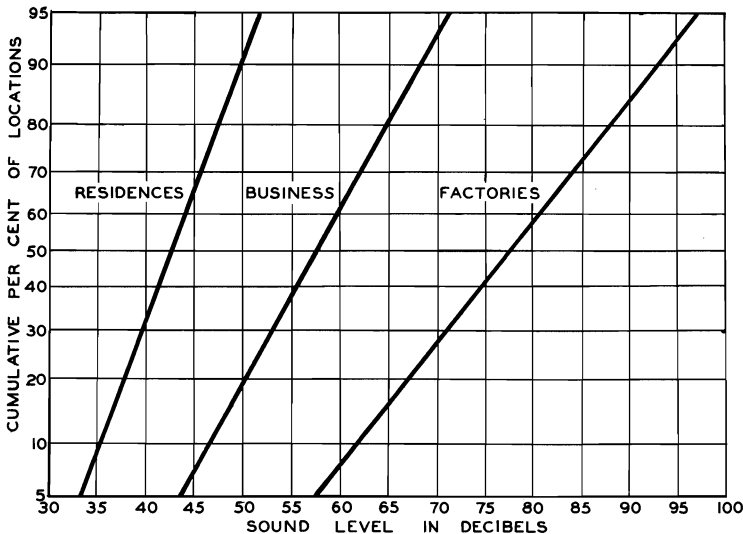
FIG. 11.20.   Room noise in residence, business, and factory locations.
(After Seacord.)

low as 32 db during a quiet dramatic passage.   The average dialogue peak
level of reproduced speech in sound motion-picture theaters is 65 db and
the minimum dialogue level is 48 db.

The noise level in various locations is shown in Table 11.2.   All data
were obtained with a noise meter employing the characteristics of Fig.
10.49.

M. *Public Address Systems.*[41]—The term public address system ordinarily
refers to a sound reproducing apparatus for use in addressing large assemb-
lages.   There are innumerable specific applications of sound reproducing
apparatus for this purpose.   The problems in all these situations are

---

[39] Seacord, D. F., *Jour. Acous. Soc. Amer.*, Vol. 12, No. 1, p. 183, 1940.
[40] Muller, W. A., *Jour. Soc. Mot. Pic. Eng.*, Vol. 35, No. 1, p. 49, 1940.
[41] Olson, H. F., *RCA Review*, Vol. 1, No. 1, p. 49, 1936.

TABLE 11.2. NOISE LEVELS FOR VARIOUS SOURCES AND LOCATIONS

Source or Description of Noise		Noise Level in Decibels
Threshold of Pain		130
Hammer Blows on Steel Plate .	. 2 ft.	114
Riveter	35 ft.	97
Factory		78
Busy Street Traffic		68
Large Office		65
Ordinary Conversation . .	. 3 ft.	65
Large Store		63
Factory Office		63
Medium Store		62
Restaurant		60
Residential Street		58
Medium Office		58
Garage		55
Small Store		52
Theatre		42
Hotel		42
Apartment		42
House, Large City		40
House, Country		30
Average Whisper . . .	. 4 ft.	20
Quiet Whisper. . . .	. 5 ft.	10
Rustle of Leaves in Gentle Breeze . .		10
Threshold of Hearing		0

practically the same. It is the purpose of this section to consider some typical examples of the use of public address systems.

Regardless of the size of the athletic field or baseball park, a public address system is useful for informing those in the stands of what is happening on the field. In general, the chief requirements are as follows: uniform distribution of sound intensity in all parts of the stand, adequate power to override any anticipated noise level of the maximum crowd, and facilities available for microphones at predetermined points.

A large stadium equipped with a public address system is illustrated by the left portion of Fig. 11.21. Due to the size and configuration of the audience area it is practically impossible to obtain satisfactory sound level and coverage with a single loudspeaker. Consequently, the loudspeakers are placed at intervals near the boundary of the field sufficiently close together so that uniform response is obtained in the horizontal plane. The elevation view of Fig. 11.21 shows how uniform sound distribution is obtained in the vertical plane by means of the directional characteristics.

A baseball field equipped with a public address system is illustrated by the right portion of Fig. 11.21. As contrasted to the stadium, here a single loudspeaker station is used to supply the entire audience area. The distance between the loudspeakers and the auditors is very large. Therefore, the vertical coverage angle is very small, which means that practically any system will have a distribution angle sufficiently broad to supply the required vertical spread. However, for conservation of power the vertical spread

of the loudspeaker should correspond to the vertical angle subtended by the audience at the loudspeaker. Since the distance of those nearest the loudspeaker to those farthest removed (that is, considering the vertical angle only) is very nearly the same, the sound intensity from the loudspeaker will be practically the same for all parts of the audience through any vertical plane and the use of compensation by means of the directional characteristics for change in distance in the vertical plane to obtain uniform response is not necessary. In the horizontal plane the spread of the loudspeaker should correspond to the angle subtended by the stands at the loudspeaker. Since
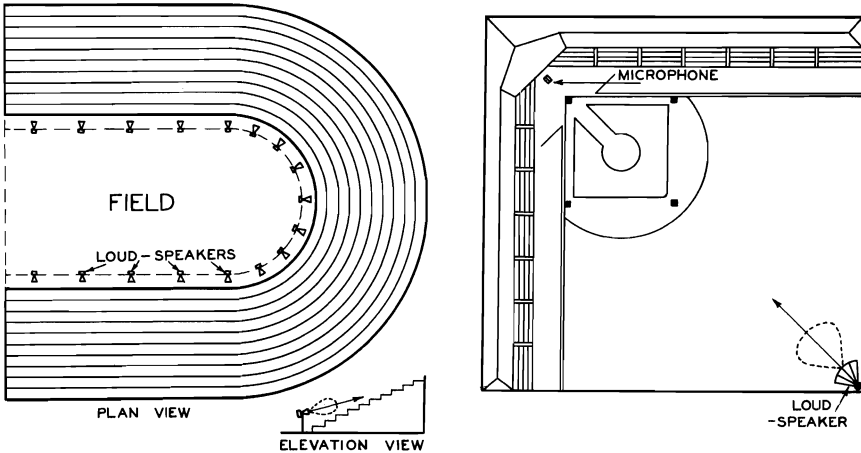
FIG. 11.21. Two arrangements of sound systems for addressing assemblages in large grandstands. For the stadium on the left a large number of loudspeakers are used, each loudspeaker covering a small portion of the total area. For the ball park on the right, a single loudspeaker cluster is used to supply the entire grandstand.

the center line is farthest removed, a directional characteristic of the shape shown is necessary for obtaining the same sound level in all parts of the grandstand. To eliminate any difficulties due to feedback, a velocity microphone is used and oriented so that the plane of zero reception passes through the loudspeaker system.

The sound level required for public address work of the type considered above will be determined by the noise level of the maximum crowd. In general, it is not practical to employ a system with sufficient power to override the sound level during cheering, applause, etc. However, the power should be sufficient to override the general noise during relatively quiet intervals. The noise level may be determined by means of a noise meter. The power available should be sufficient to produce a minimum sound level of 80 db or, for very noisy conditions, 20 to 30 db above the noise during the relatively quiet intervals. In the two examples cited above, and, in fact, for all outdoor public address work, the only consideration is direct sound. The problem is to select amplifiers and loudspeakers

with characteristics which will deliver the required sound level over the distances and areas considered. The steps in the selection of a system may be as follows: First, the directional characteristics should be determined, as outlined in the preceding discussion, so that uniform response is obtained over the audience area. Second, either a single or a group of loudspeakers having the desired directional characteristics should be selected. Third, the response characteristic of the system on the axis at a specified input and distance should be available to show the amplifier power required to supply the desired intensity level. Fourth, the power handling capacity of the loudspeakers and amplifiers should be adequate to supply the required intensity level.

Two types of sound reinforcing installations for an outdoor theater are shown in Fig. 11.22. The system on the left employs a single loudspeaker
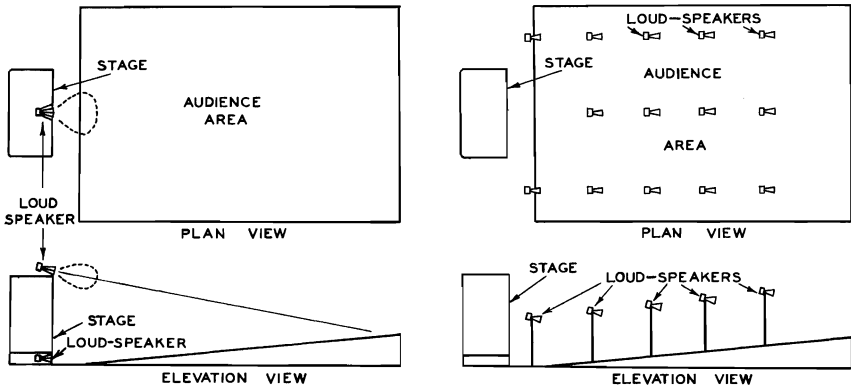
FIG. 11.22. Two arrangements of sound reinforcing systems for an outdoor theater. The arrangement on the left employs a single loudspeaker located above the stage and having suitable directional characteristics to produce a uniform intensity level over the audience area. An alternative arrangement employs a loudspeaker located below the stage floor. The arrangement on the right employs a number of loudspeakers, each covering a small portion of the audience.

station located either above or below the stage as shown in Fig. 11.22. If the stage is quite low the logical position for the loudspeakers is at the top of the stage. The same procedure for obtaining uniform sound coverage and adequate intensity level of the sound from the loudspeakers as used in the preceding considerations is applicable in this case. If the stage is very high the separation between the loudspeakers and the action on the stage will be very large. As a consequence, the wide difference in the direction of the direct and reinforced sound will be particularly disconcerting to listeners in the front portion of the seating area. Under these conditions, it may be desirable to locate the loudspeaker under the stage as shown in Fig. 11.22. The system depicted on the right employs a large number of loudspeakers, each one supplying a small portion of the audience.

The directional characteristics of the loudspeakers should be selected so that each individual area is adequately supplied. Cognizance must be taken of the energy supplied from adjacent loudspeakers.

There are certain advantages in each system. In the case of the single loudspeaker system, better illusion is obtained because the augmented sound appears to come from the stage. On the other hand, the intensity level outside the audience area in a backward direction falls off very slowly. At a distance equal to the length of the audience area the level is only 6 db lower than that existing in the audience area. In certain locations the sound levels produced by such systems will cause considerable annoyance to those located in the vicinity of the theater. By dividing the theater area into small plots, each supplied by a loudspeaker, and by directing the loudspeakers downward, the sound intensity level outside the audience area will be considerably lower than in the case of the single loudspeaker station and usually eliminates any annoyance difficulties. The short sound projection distance is another advantage of the multiple loudspeaker system.

The above typical examples of outdoor public address and sound reinforcing systems illustrate the principal factors involved in this field of sound reproduction.

N. *Sound Motion-Picture "Drive-In" Theater.*—A "drive-in" motion-picture theater is an outdoor system in which the audience attends the show while seated in an automobile. A type of sound motion-picture "drive-in" theater system is shown in Fig. 11.23. Sound motion-picture
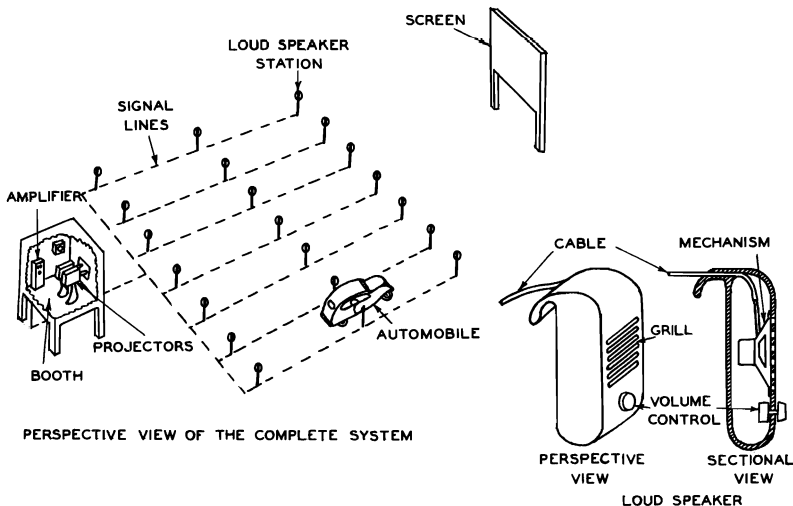
Fig. 11.23. A perspective view showing the elements of a "drive in" sound motion-picture theater. Perspective and sectional views of the loudspeaker used in the "drive in" theater are shown in the lower right. (After Olson, "Musical Engineering," McGraw-Hill Book Company, New York, N.Y., 1952.)

reproducing systems are described in Sec. 13.7.   The photoelectric cell and optical system are housed in the picture projector of Fig. 11.23. The picture projector amplifier and master volume controls are located in the projection booth.   In the drive-in theater, the customers are seated in their automobiles to view the picture and hear the sound.   Individual loudspeakers are provided to supply the sound to each automobile.   Perspective and sectional views of the loudspeaker are shown in Fig. 11.23. In use, the loudspeaker is hung inside the automobile.   The volume control incorporated with the loudspeaker makes it possible for the listener to adjust the level of the reproduced sound.   When not in use the loudspeakers are hung on posts provided for the purpose.   Signal lines connect the individual loudspeakers to the amplifier in the booth.

O. *Orchestra and Stage Shell.*—When orchestra and stage productions are conducted in outdoor theaters it is desirable to provide a shell to augment

PERSPECTIVE VIEW   ELEVATION VIEW
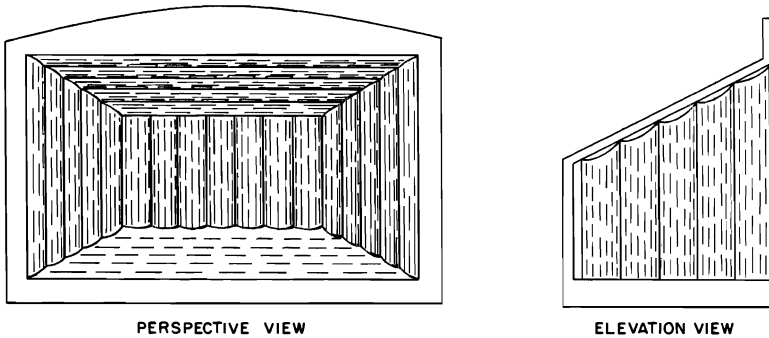
FIG. 11.24.   Perspective and elevation views of an orchestra shell with polycylindrical surfaces.

and direct the sound to the audience, to surround the orchestra with reflecting surfaces and to protect the performers and instruments against wind, dew, and other undesirable atmospherics.   Most of the outdoor orchestra shells have been of the concave type which produce intense and sharp concentrations of reflected sound in both the shell and audience area. These acoustical effects are particularly undesirable when the sound is picked up by microphones on the stage for sound reinforcing and broadcasting.   Under these conditions the intensifications and discriminations make it appear that the orchestra is unbalanced with relation to the various instruments.   Furthermore, it is impossible for the conductor to obtain a true balance because these undesirable acoustical effects also exist at the conductor's platform on the stage.   The undesirable acoustical effects can be overcome by means of an orchestra shell in which the boundaries are polycylindrical surfaces as shown in Fig. 11.24.   These surfaces reflect the sound in a diffuse manner and thereby obviate concentrations of sound energy on the stage and in the audience area.   The acoustics of this type

of orchestra shell make it possible for the conductor to obtain an improved balance of the orchestra.   This type of shell produces a uniform distribution of sound in the audience area.   The polycylindrical shell provides good acoustics for microphone pickup for sound reinforcing or broadcasting.

P. *General Announce and Paging Systems.*[42]—General announce systems are useful in factories, warehouses, railroad stations, airport terminals, etc. A typical installation is depicted on the left portion of Fig. 11.25.   For this type of work intelligibility is more important than quality.   The deleterious effect of reverberation upon articulation can be reduced, and a better control of sound distribution can be obtained, by reducing the low-frequency
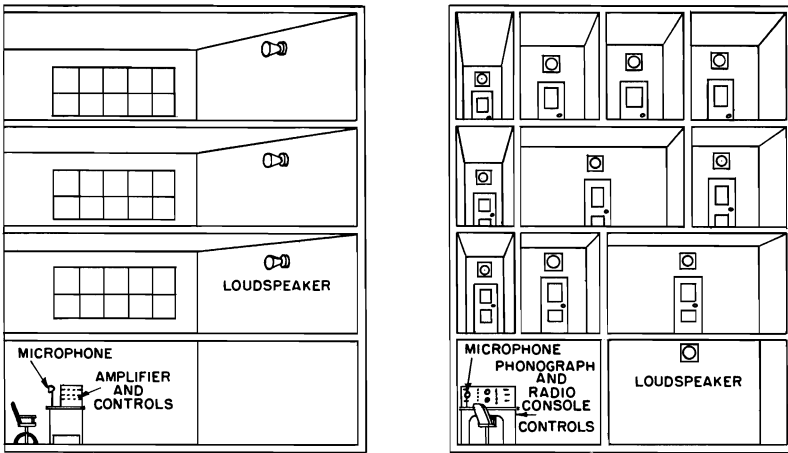
FIG. 11.25.   Two uses of call, general announce, and sound distributing systems. On the left, a high efficiency horn loudspeaker is used to obtain a high sound level over a large floor area as in a factory or warehouse.   On the right, small, direct-radiator loudspeakers are used to supply the small rooms at a relatively low level as in paging, announcing, and centralized radio installations used in hospitals, hotels, or schools.

response of the system.   Furthermore, the cost of the amplifiers and loud-speakers is also reduced by limiting the frequency range.   To find the power required, the sound intensity level under actual operating conditions should be determined.   The system should be designed to produce an intensity level 20 to 40 db above the general noise level.   Under no conditions should the system be designed to deliver an intensity level of less than 80 db.   The loudspeakers should be selected and arranged following an analysis similar to that outlined in the preceding sections, so that uniform sound distribution and adequate intensity levels are obtained.

For certain types of general announce, paging, and sound distributing installations used in schools, hospitals, department stores, hotels, etc., the intensity level required is relatively low and the volume of the average room is usually small.   For most installations of this type, save in noisy

[42] Olson, H. F., *RCA Review*, Vol. 1, No. 1, p. 49, 1936.

locations, an intensity level of 70 db is more than adequate. Higher intensity levels tend to produce annoyance in adjacent rooms. From a consideration of the data of Fig. 11.19, it will be seen that the power requirements for the loudspeakers will be small. To blend with the furnishings of the room, it is desirable to mount the loudspeakers flush with the wall surface. Therefore, for these applications, a direct radiator loudspeaker of the permanent magnet dynamic type is most suitable. In this connection it should be mentioned that these loudspeakers have a very low efficiency, being of the order of 1 per cent as compared to 25 per cent to 50 per cent for the horn loudspeakers.

For large rooms requiring large acoustical outputs it is more economical to use a high-efficiency loudspeaker and effect a corresponding reduction in the power amplifier requirements. On the other hand, for an installation of the type depicted on the right side of Fig. 11.25 and requiring a large number of units, it is more economical to use relatively inefficient low-cost loudspeakers and correspondingly larger amplifiers.

Q. *Intercommunicating Systems.*—Intercommunicating systems are loudspeaking telephones for use in communicating between two rooms. The
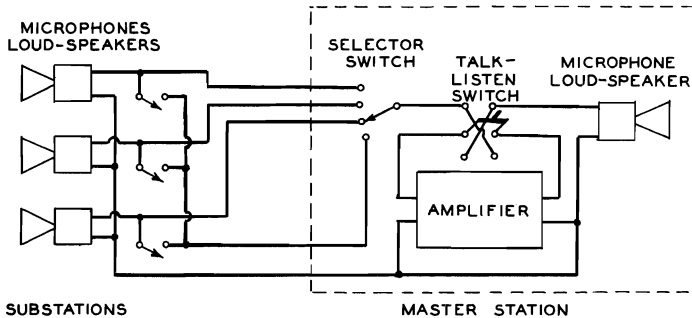
Fig. 11.26. A schematic arrangement of the elements of a simple intercommunicating system.

more elaborate systems are similar to the general announce system described in Sec. 11.2P with the addition of microphone positions in more than one room. The simplest system consists of two units for use between two stations. The master unit contains an amplifier, microphone, loudspeaker, and a talk-listen switch. The remote unit consists of a microphone, loudspeaker, and talk-listen switch. In the simplest systems the loudspeaker with suitable electrical compensation is also used as a microphone as shown in Fig. 11.26. Additional stations and appropriate switching systems may be added for communicating between a number of rooms. The voice currents are carried in two ways: in one by direct wire, and in the other by using a high-frequency carrier on the power mains. The latter system does not require wiring but has the disadvantage that, in large buildings having several separate systems, cross-talk may occur.

R. *Radio Receiver Operating in a Living Room.*—The radio receiver and phonograph represent by far the largest number of complete reproducing systems. For this reason, the performance of a radio receiver in a room is an extremely important problem. Equations 11.10, 11.11, and 11.12 for the direct and reflected sound are applicable to a radio receiver in a room. In the case of a theater it is possible to adjust the loudspeakers so that the direct sound is the same in all parts of the auditorium. It is not practical to arrange the loudspeakers in a radio receiver so that there will be no variation of the direct sound with distance. In view of the rather small distances and relatively small volume of the room this is not very important. It is important, however, that the directional characteristic be independent of the frequency and sufficiently broad to send direct sound into all listening areas.

The response frequency characteristics of a good radio receiver taken at various listening positions in a typical living room are shown in Fig.
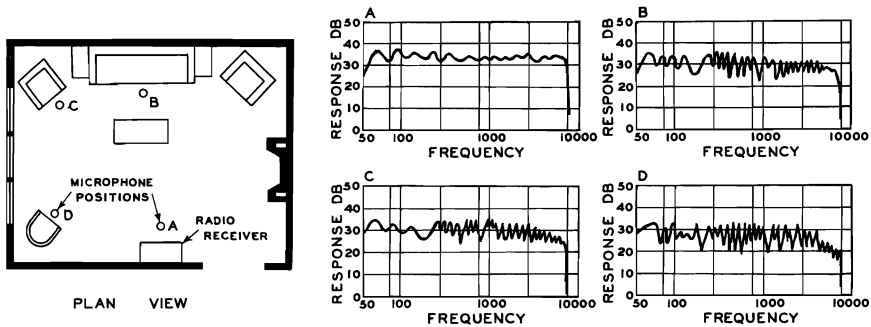
FIG. 11.27. A plan view of a living room with a radio receiver. The graphs show the response frequency characteristics for various positions in the room.

11.27. Graph *A* shows the response frequency characteristic very close to the receiver and, therefore, indicates the direct sound output. The directional characteristics of this receiver were uniform over an angle of 120°. The sharp variations in response frequency characteristics taken in other parts of the room are due to the reflected sound. The direct sound energy density and the reflected sound energy density are approximately equal at a distance of 5 feet from the receiver for the average living room and average reproducer. It is interesting to note that the response frequency characteristics taken in various positions in the room have the same shape as that taken very close to the receiver. The reverberation time characteristic of this room was quite uniform with respect to frequency; therefore, the reflected sound does not vary appreciably with frequency since the output of the receiver is independent of the frequency (see equation 11.12).

The response frequency characteristics upon the ears, Fig. 12.11, to be considered in Sec. 12.6, show that corresponding to the intensity of a 1000 cycle note there is an intensity at another frequency that will sound as

loud. These characteristics show that if the sound is reproduced at a lower level than that of the original sound it will appear to be deficient in low-frequency response. In general, the reproduction level in the home is much lower than the level of the original reproduction. In order to compensate for the low-frequency deficiency, the volume control in most radio receivers and phonographs is designed so that the low-frequency response is accentuated in an inverse ratio to the relative sensitivity of the ear in going from the original level to the lower level of reproduction. This type of volume control is termed an acoustically compensated volume control.[43]

S. *Radio Receiver Operating in an Automobile.*—The primary loudspeaker in an automobile is usually located in one of four locations, as shown in Fig. 11.28, namely, LS1 in the front of the instrument enclosure, LS2 in the top of the instrument enclosure, LS3 in the fire wall or dash, and LS4 in the header above the windshield. The header location gives somewhat better distribution of the high-frequency response in the back seat than the three
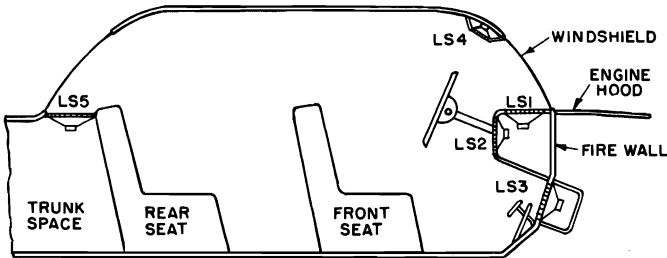
FIG. 11.28.   Loudspeaker locations in an automobile.

other locations. However, the low-frequency response of a loudspeaker mounted in the header is usually attenuated due to the small volume behind the loudspeaker, see Sec. 6.10. The dash or fire wall location gives fair distribution of high-frequency response in the front seat, but very poor distribution in the rear seat. The low-frequency response in this position can be made very good by employing a large loudspeaker case or by venting the back of the case into the engine compartment. Sometimes a combination of a low-frequency dash loudspeaker and a high-frequency header or instrument panel loudspeaker is employed. At the present time the favored position for the loudspeaker appears to be in the instrument panel because in this location the radio receiver, loudspeaker, and controls may be combined into a single compact unit. The distribution of sound is excellent in the front seat and good in the back seat. The stiffness presented to the back of the cone is small because the entire radio receiver case volume is used to enclose the back of the loudspeaker. In this manner the response can be maintained in the low-frequency range. In order to improve the reproduction of sound in the rear seat, a secondary loudspeaker is used in location LS5 in back of the rear seat.

[43] Wolff and Cornell, *Electronics*, Vol. 6, No. 2, p. 50, 1933.

The conditions under which an automobile radio receiver operates differ widely from those of a loudspeaker in the living room. It will be seen that the automobile is a small enclosure with short distances between the loudspeaker and the listener. This is fortunate because wind, road rumble, and engine noises mask the reproduced sound. The power output of the receiver should be sufficient to override these noises and give intelligible reproduction of speech and pleasing reproduction of music. In view of the fact that the sound level delivered to the loudspeaker is quite high under the noise conditions, it is important that the response frequency characteristic be smooth and free of peaks, otherwise the reproduced sound will be disagreeable due to the high level of the peaks relative to the general level. It is also important that the nonlinear distortion be kept at a low value because spurious components in the reproduction of sound are more apparent at the higher sound levels.

T. *Absorption of Sound in Passing Through Air.*—The absorption[44] of a plane progressive sound wave in passing through air may be several times
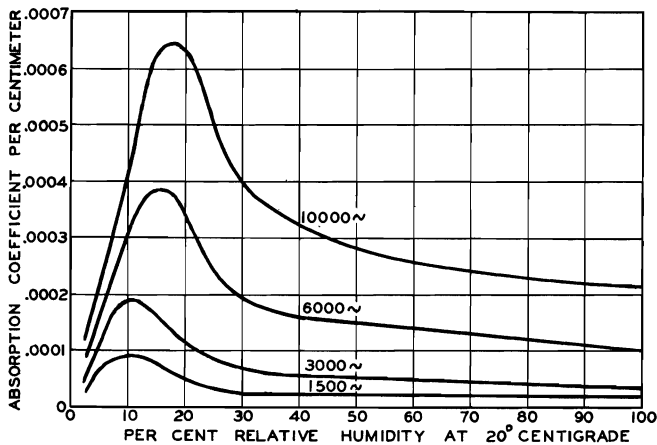
FIG. 11.29. Curves showing the absorption of a plane sound wave in passing through air, at 20° Centigrade, for different frequencies, as a function of the relative humidity. The intensity after a plane wave has traveled a distance x centimeters is $I_0 \epsilon^{-mx}$, where I_0 is the intensity at $x = 0$ and m is the coefficient given by the above graph. (After Knudsen.)

that predicted by the classical theory. The anomalous absorption is primarily dependent upon the humidity, although it is also affected by impurities such as H_2O, H_2, H_2S, and NH_3. This, of course, means that there may be considerable frequency discrimination of the reproduced sound in large theaters where the sound travels a long distance. In addition, the reverberation time will be reduced at the higher frequencies. The coefficient

[44] Knudsen, V. D., *Jour. Acous. Soc. Amer.*, Vol. 6, No. 4, p. 199, 1935.

per centimeter for 1500, 3000, 6000, and 10,000 cycles as a function of the humidity is shown in Fig. 11.29.

U. *Sound Transmission Through Partitions.*[45,46,47,48,49,50,51,52,53]—The problem of sound transmission through partitions and walls is complicated because of the many factors involved. The problem of the mass-controlled single-wall partition is very simple. The sound insulation of this type of partition is proportional to the mass and frequency. For the usual building materials and walls of ordinary dimensions supported at the edges, the problem is that of the clamped rectangular plate with distributed resistance throughout the plate and lumped damping at the edges. Obviously, the performance of this system depends upon the size, the ratio of the two linear dimensions, the weight of the material, the damping in the material, and the edge supports. This type of problem is not amenable to an analytical solution.

The transmittivity of a partition is defined as the ratio of the intensity in the sound transmitted by the partition to the intensity in the sound incident upon the partition. The transmission loss, in decibels, introduced by the partition is given by

$$\text{T.L.} = 10 \log_{10} \frac{I_i}{I_t} = 10 \log_{10} \frac{1}{\tau} \qquad \qquad 11.17$$

where I_i = intensity of the incident sound,
   I_t = intensity of the transmitted sound, and
   τ = transmittivity or transmission coefficient.

The coefficient of transmission τ is a quantity which pertains alone to the partition and is independent of the acoustic properties of the rooms which it separates.

The reduction factor is the ratio of the sound energy density in the room containing the sound source to the sound energy in the adjoining receiving room. The reduction factor, in decibels, is given by

$$\text{R.F.} = \text{T.L.} + 10 \log_{10} \frac{A}{S} \qquad \qquad 11.18$$

where A = total absorption in the receiving room, in sabins, and
   S = area of the partition, in square feet.

[45] Rayleigh, "Theory of Sound," Macmillan Co., London, 1926.
[46] Eckhardt and Chrisler, Bureau of Standards, Paper No. 526.
[47] Knudsen, "Architectural Acoustics," John Wiley and Sons, New York, N.Y., 1932.
[48] Sabine, "Acoustics and Architecture," McGraw-Hill Book Co., New York, N.Y., 1932.
[49] Watson, "Acoustics of Buildings," John Wiley and Sons, New York, N.Y., 1923.
[50] Morrical, K. C., *Jour. Acous. Soc. Amer.*, Vol. XI, No. 2, p. 211, 1939.
[51] Rettinger, "Applied Architectural Acoustics," Chemical Publishing Company, Brooklyn, N.Y., 1947.
[52] Knudsen and Harris, "Acoustical Designing in Architecture," John Wiley and Sons, New York, N.Y., 1950.
[53] Beranek, "Acoustics," McGraw-Hill Book Company, New York, N.Y., 1954.

TABLE 11.3.   NOISE REDUCTION THROUGH VARIOUS STRUCTURES

Material	Weight in lbs. per sq. ft.	Transmission Loss in DB — Frequency					Av. T.L. in DB	Average τ	Author
		128	256	512	1024	2048			
Aluminum, .025"	.35		18	13	18	23	16	.025	B.S.
Iron, .03" galvanized	1.2		25	20	29	35	25	.0032	B.S.
Lead, ⅛"	8.2		31	27	37	44	32	.00063	B.S.
Plywood, ¼"	.73		21	21	25	26	21	.008	B.S.
Celotex, Standard ¼"	.30		14	15	18	24	15	.032	B.S.
Celotex, Standard ½"	.66		22	17	23	27	20	.01	B.S.
Hair Felt, 1"	.75	4.9	4.6	6.0	7.1	6.7	6	.32	P.E.S.
Hair Felt, 4"		7.5	12.5	15	19	19	14	.04	P.E.S.
Wood Studs, 2" × 4" Plywood ¼"	2.2	23	24	25	29	27	26	.0025	
Wood Studs, Plaster-board ½"	5.4	32	34	35	38	37	35	.00032	
Wood Studs, 2" × 4" Wood Lath, Lime Plaster	18	27	29	38	47	43	43	.00005	P.E.S.
Wood Studs, 2" × 4" Metal Lath, Plaster ¾"	21	36	38	40	45	41	40	.0001	
Tile, 2" Gypsum	20	25	34	44	51	63	48	.000016	P.E.S.
Cinder Block 4" Unfinished	32	12	18	22	23	30	22	.0063	
Cinder Block 4" Plastered Both Sides	37	36	38	40	44	42	40	.0001	
Wood Studs, 2" × 4", Staggered, Metal Lath, Plaster, ¾", 8" Spacing	21	44	46	48	52	51	48	.000016	
Tile Clay 6" × 12" × 12" Plastered both sides	37		41	35	45	52	40	.0001	B.S.
Brick, 8" Plastered both sides	87		50	48	55	63	50	.00001	B.S.
Brick, 16" Plastered both sides	174	50	52	55	58	60	55	.0000032	
Double Cinder Block, Each 8", Air Space 4", Outside Plastered	132	55	56	58	67	60	60	.000001	
Door, Light 4 Panel		13	16	20	23	22	22	.0063	P.E.S.
Door, Oak		15	18	23	26	25	25	.0032	P.E.S.
Door, Steel ¼"		25	27	31	36	31	35	.00032	P.E.S.
Window Glass, Plate ¼"	3.5		33	31	33	35	30	.001	B.S.
Window Glass, Small Panes 3/16"		19	20	24	31	28	29	.0013	P.E.S.
Double Window Glass ¼", Air Space 1"	7.0	42	43	44	48	45	45	.000032	
Finish and Rough Flooring, Plaster Ceiling		34	35	37	39	39	37	.0002	
Concrete Slab 4"	48	38	39	42	46	48	42	.000063	
Concrete Slab 4" Suspended Plaster Ceiling	51	46	50	55	55	58	52	.0000063	

The abbreviations in the above table are as follows: B.S., Bureau of Standards; P.E.S., P. E. Sabine.

Equation 11.18 shows that the reduction is due to both the loss introduced by the partition and the absorption in the receiving room.

The choice of a partition for insulating a room against sound involves a number of considerations. Some of the factors are the frequency distribution and intensity level of the components of the objectionable sound, the transmission frequency characteristics of the partition, the ambient noise or sound level in the receiving room which will mask the objectionable sound, and the response frequency characteristics of the ear.

Measurements have been made by various investigators upon the transmission by single partitions. The results of these measurements are shown in Table 11.3.

The mass-controlled partition with air between the partition elements is a low-pass filter in which the mass of the wall is the series element and the volume between the partitions is the shunt element. The partitions in this case are mounted in edge supports which allow freedom of motion without cracks which would pass air-borne sound.

11.3. Collection of Sound.—A. *Sound Collecting System.*[54]—When a source of sound is caused to act in a room, the first sound that strikes a collecting system placed in the room is the sound that comes directly from the source without reflection from the boundaries. Following that comes sound that has been reflected once, twice, and so on, meaning that the energy density of the sound increases with the time, as the number of reflections increase. Ultimately, the absorption of energy by the boundaries equals the output of the source and the energy density at the collecting system no longer increases; this is called the steady state condition. Therefore, at a given point in a room there are two distinct sources of sound, namely: first, the direct and, second, the generally reflected sound. For rooms that do not exhibit abnormal acoustical characteristics it may be assumed that the ratio of the reflected to the direct sound represents the *effective reverberation* of the collected sound.

Consider a sound collecting system, Fig. 11.30, the efficiency of reception of which may be characterized as a function of the direction with respect to some reference axis of the system. (The nondirectional collecting system is a special case of the directional system in which the efficiency of reception is the same in all directions.) The output of the microphone may be expressed as

$$e = \mathscr{Q}pf_1(\psi) \qquad\qquad 11.19$$

where e = voltage output of the microphone, in volts,

   p = sound pressure, in dynes per square centimeter,

   $\mathscr{Q}$ = sensitivity constant of the microphone, and

   ψ = angle between incident pencil of sound and the reference axis of the microphone.

[54] Olson, H. F., *Proc. Inst. Rad. Eng.*, Vol. 21, No. 5, p. 655, 1933.

If the distance, in centimeters, between the source of the sound and the collecting system is D, the energy density at the microphone due to the direct sound is

$$E_D = \frac{E_0}{D^2 4\pi c} \qquad 11.20$$

where E_0 = power output of the sound source, in ergs per second, and
c = velocity of sound, in centimeters.

To simplify the discussion, assume that the effective response angle of the microphone is the solid angle Ω steradians. The direction and phase of the reflected sound are assumed to be random. Therefore, the reflected sounds available for actuating the directional microphone are the pencils of sound within the angle Ω. The energy response of the directional microphone to generally reflected sound will be $\Omega/4\pi$, that of a nondirectional microphone. The generally reflected sound to which the directional microphone is responsive is, therefore, given by

$$E_R = \frac{4E_0\Omega}{caS4\pi} \left[1 - \epsilon^{(cS\,[\log_\epsilon\,(1-a)]\,t)/4V} \right] (1 - a) \qquad 11.12$$

where a = absorption per unit area, absorption coefficient,
S = area of absorbing material, in square centimeters,
V = volume of room, in cubic centimeters, and
Ω = solid angle of reception, in steradians,
t = time, in seconds.

The ratio of the generally reflected sound to the direct sound is a measure of the effective reverberation of the collected sound and is given by

$$\frac{E_R}{E_D} = \frac{4D^2\Omega \left[1 - \epsilon^{(cS\,[\log_\epsilon\,(1-a)]\,t)/4V} \right] (1 - a)}{aS} \qquad 11.22$$

If the sound continues until the conditions are steady, equation 11.22 becomes

$$\frac{E_R}{E_D} = \frac{4D^2}{aS} \Omega(1 - a) \qquad 11.23$$

From equations 11.22 and 11.23, it will be seen that the received reverberation can be reduced by decreasing the distance D, by increasing the absorption aS, or by decreasing Ω.

For a given room employing a directional microphone, the receiving distance can be increased $\sqrt{4\pi/\Omega}$ times that in the nondirectional system with the same collected reverberation in both cases.

The absorption characteristic of a studio is shown in Fig. 11.30A. The direct sound picked up by a nondirectional microphone and two directional microphones is the same because the distance between the sound source and the microphones is assumed to be the same for all three cases (Figs. 11.30B, 11.30C, and 11.30D). The generally reflected sound picked

up by a nondirectional microphone is shown in Fig. 11.30B. The generally reflected sound picked up by a velocity or unidirectional microphone in which $\Omega = 4\pi/3$ is shown in Fig. 11.30C. The generally reflected sound picked up by an ultradirectional microphone in which $\Omega = \pi/10$ is shown
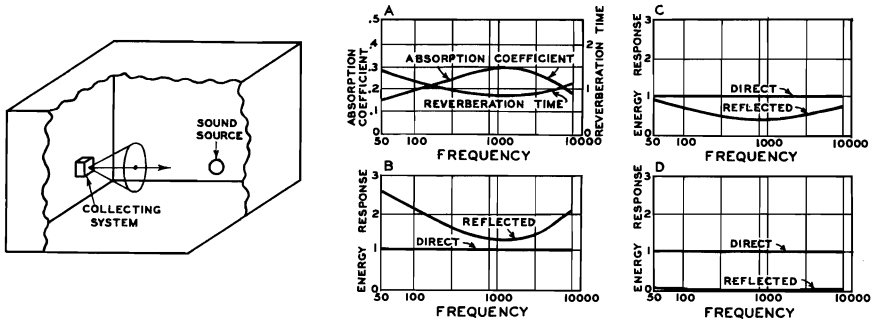
FIG. 11.30. Sound collecting system in a studio. Graph A shows the reverberation time and the absorption coefficient of the boundaries of a typical studio. Graphs B, C, and D show the energy response for the direct and reflected sounds for various microphones as follows: B. Nondirectional microphone. C. Bidirectional velocity or unidirectional microphone. D. Ultradirectional microphone.

in Fig. 11.30D. The effectiveness of a directional sound collecting system in overcoming reverberation and undesirable sounds is graphically depicted in Fig. 11.30.

Directional microphones, in addition to discriminating against noise and generally reflected sounds, have been found to be extremely useful in arranging actors in dialogue and for adjusting the relative loudness of the instruments of an orchestra.

A plan view[55] of a velocity microphone and a number of sound sources is shown in Fig. 11.31A. Suppose that sources 2 and 5 represent two actors who are carrying on a dialogue. In view of the fact that this microphone receives with the same efficiency in two directions, it is possible to have the actors face each other, which is an advantage from a dramatic standpoint. Suppose that the sources of sound 1, 2, 3, and 5 represent the instruments of an orchestra. All the sources are located at the same distance. This means that 1 will produce 0.7 times the voltage output produced by 2 for the same loudness. In the same way 3 will be 0.5 of 2. Source 4 is considered as objectionable and is placed in the zero reception zone. With this microphone the relative loudness of these sources can be adjusted by the angular position relative to the microphone axis as well as the distance. Obviously, this is a great advantage in balancing the instruments of an orchestra. In the case of a nondirectional microphone, the relative loudness can only be adjusted by the distance.

[55] Olson, H. F., *Proc. Inst. Rad. Eng.*, Vol. 21, No. 5, p. 655, 1933.

The same procedure[56] for balancing the instruments of an orchestra may be used in connection with a unidirectional microphone (Fig. 11.31B.) The unidirectional microphone is particularly useful when all the instruments are grouped in front of the microphone and the objectionable sounds originate behind the microphone.

The velocity microphone is used in recording music and other sound in sound motion pictures. In some instances noises produced by the camera
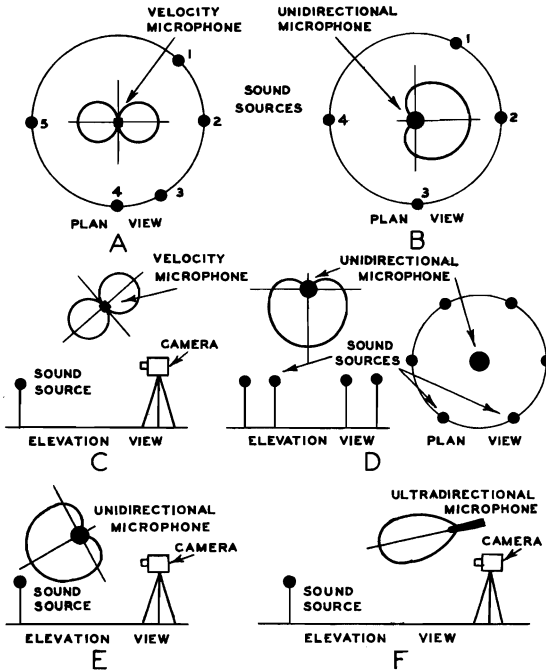
FIG. 11.31. Examples illustrating the use of directional microphones.

and devices are objectionable and must be reduced. The directional characteristics of the velocity microphone are useful in overcoming objectionable noises.[57] It is possible to orient the microphone so that the objectionable noise lies in the plane of zero response of the microphone as shown in Fig. 11.31C.

In certain types of recording[58] it is desirable to place the microphone at the center of action directed downward and collect sounds over an angle of 360° with respect to the microphone. (Fig. 11.31D illustrates the use of a unidirectional microphone for this application.)

[56] Olson, H. F., *Jour. Soc. Mot. Pic. Eng.*, Vol. 27, No. 3, p. 284, 1936.
[57] Olson, H. F., *Jour. Soc. Mot. Pic. Eng.*, Vol. 16, No. 6, p. 695, 1931.
[58] Olson, H. F., *Jour. Soc. Mot. Pic. Eng.*, Vol. 27, No. 3, p. 284, 1936.

The unidirectional microphone is almost universally employed for speech pickup in sound motion pictures and television (Fig. 11.31E). The broad coverage in the forward direction makes it possible to follow the action. The high discrimination against pickup of sounds originating in the rear is useful in eliminating noises from the camera and lights.

The narrow directional pattern of the ultradirectional microphone[59] provides a high order of discrimination against reverberation and other undesirable sounds (see Sec. 8.6D). With this directional pattern a very large sound pickup distance may be employed. The ultradirectional microphone has been used in tests in sound motion pictures and television (Fig. 11.31F). These tests have shown that a sharp directional pattern is useful. However, there are some practical problems to be overcome before the ultradirectional microphone becomes a useful tool in sound pickup.

Other examples of the use of directional microphones are shown in Fig. 11.16.

B. *Broadcasting Studios.*[60,61,62,63,64,65,66,67]—In the early days of broadcasting it was customary to make the reverberation time of the studios as low as possible. This imposed quite a strain upon the orchestra and singers to keep in tune. The almost universal use of directional microphones during the past few years has eliminated the necessity of extremely dead studios. As a result, the quality and artistic effects of the collected sound are materially enhanced.

The studios in a large broadcasting station should be graduated in size and in corresponding acoustical condition to accommodate anticipated loading to the best advantage. The control booths should be provided with soundproof windows located so that the studio engineer has an unobstructed view of the studio.

The studios should be insulated against all types of extraneous noises. Cinder composition has been found to give very good insulation. Resilient mounting of the walls, floor, and ceiling reduces mechanical transmission.

Air-borne noises carried in the air conditioning ducts may be suppressed by lining the ducts with felt, rock wool, etc., to obtain suitable attenuation. Mechanical transmission of sound by the ducts may be reduced by isolating the sections of the duct.

The reflected sound in a studio produces standing wave systems. These standing wave systems exhibit variations in sound pressure from point to point in a room. It is desirable to reduce this variation to as small a value as possible. This can, of course, be done by making the walls very

[59] Olson, H. F., *Proc. Inst. Rad. Eng.*, Vol. 27, No. 7, p. 438, 1939.
[60] Morris and Nixon, *Jour. Acous. Soc. Amer.*, Vol. 8, No. 2, p. 81, 1936.
[61] Chinn and Bradley, *Proc. IRE*, Vol. 27, No. 7, p. 421, 1939.
[62] Potwin and Maxfield, *Jour. Acous. Soc. Amer.*, Vol. 11, No. 1, Part 1, p. 48, 1939.
[63] Nixon, G. M., *RCA Review*, Vol. 6, No. 3, p. 259, 1942.
[64] Volkmann, J. E., *Jour. Acous. Soc. Amer.*, Vol. 13, No. 3, p. 234, 1942.
[65] Boner, C. P., *Jour. Acous. Soc. Amer.*, Vol. 13, No. 3, p. 244, 1942.
[66] Content and Green, *Proc. Inst. Rad. Eng.*, Vol. 32, No. 2, p. 72, 1944.
[67] Nygren, A., *FM and Television*, Vol. 6, No. 5, p. 25, 1946.

absorbing which leads to the undesirable characteristic of a very low reverberation time. The problem of obtaining a better sound pressure distribution can be accomplished by the use of wall surfaces which diffuse, distribute, and disperse the sound reflected from the walls. Four typical wall treatments for obtaining a diffuse and uniform sound pressure distribution characteristic are shown in Fig. 11.32. In Fig. 11.32A, the absorbing material is distributed in discrete spots[68] on the wall surface. This distribution of material breaks up the reflected wave front and thereby produces a diffuse sound field. In Fig. 11.32B, C, and D, polycylindrical,[69] serrated[70] and spherical surfaces[71] are employed to produce a diffuse sound field. These surfaces have been used for walls and ceilings in
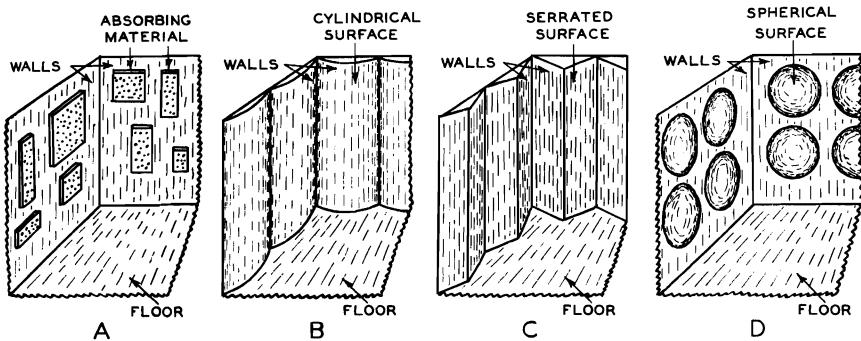
FIG. 11.32. Wall structures for diffusing the reflected sound. A. Absorbing material located in spots. B. Cylindrical surfaces. C. Serrated surfaces. D. Spherical surfaces.

broadcast studios. The polycylindrical and spherical surfaces increase the wave-front of the reflected sound. The convex surfaces also reduce the interference effect between direct and reflected sounds. The treatments shown in Fig. 11.32 are also applied to the ceiling. The use of these expedients produces a more uniform sound-decay curve and reduces echoes and flutters.

Broadcast studios may be divided into two general classes; in the first class the entire studio is used exclusively by the performers and in the second class the studio is used for both the performers and the audience.

The first class of studio is used for all manner of programs and groups. Under these conditions, it has been found that a studio with uniform acoustics is more useful than the live and dead end type. Uniform acoustics throughout the studio are obtained by a uniform distribution of the absorbing material. The various types of wall construction shown in

---

[68] Potwin and Maxfield, *Jour. Acous. Soc. Amer.*, Vol. 11, No. 1, Part 1, p. 48, 1939.
[69] Volkmann, J. E., *Jour. Acous. Soc. Amer.*, Vol. 13, No. 3, p. 234, 1942.
[70] Nixon, G. M., *RCA Review*, Vol. 6, No. 3, p. 259, 1942.
[71] Nygren, A., *FM and Television*, Vol. 6, No. 5, p. 25, 1946.

Fig. 11.32 are used to break up discrete reflections and thereby obtain a uniform distribution of reflected sound energy in the studio. In general, studios of this type are rectangular parallelopipeds.

Various expedients, in addition to the wall structures of Fig. 11.32, are used to break up flutters and echoes. In some designs the walls are inclined to eliminate parallelism between opposite walls. In other designs the ceiling and/or walls are broken up into two or more nonplanar surfaces.

The ratio of the dimensions[72] of the studio is important in distributing the characteristic resonant frequencies uniformly over the frequency range. The graph of Fig. 11.33 shows the ratio of the dimensions for small, medium,
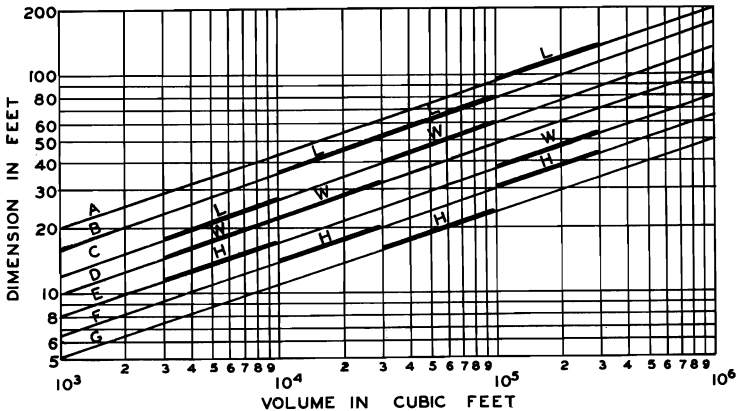
FIG. 11.33. Preferred studio dimensions. In the graph, H = height, W = width, L = length. Small rooms, $H:W:L = 1:1.25:1.6 = E:D:C$. Average shape rooms, $H:W:L = 1:1.6:2.5 = F:D:B$. Low ceiling rooms, $H:W:L = 1:2.5:3.2 = G:C:B$. Long rooms, $H:W:L = 1:1.25:3.2 = F:E:A$. (After Volkmann.)

and large studios. The most desirable ratio of the dimensions would be in the ratio of the cube root of 2. This separates the dimensions by one-third octave. This ratio is possible for small studios but is not practical for large studios, in that the ceiling height becomes too great. The dimensions of the small rooms are given by the lines C, D, and E of Fig. 11.33. For medium studios the ratio of the dimensions is near the cube root of 4. The dimensions of medium studios are given by the lines B, D, and F of Fig. 11.33. This is approximately the ratio 2 : 3 : 5 which has been frequently used in the design of broadcast studios. For very large studios the dimensions are given by the lines B, C, and G of Fig. 11.33.

In the second class of studio, termed the auditorium type, the performers occupy one end of the room and the audience the other end. A plan and sectional view of an auditorium type broadcast studio is shown in Fig. 11.34. The wall and ceiling surfaces of the stage are arranged to provide

[72] Volkmann, J. E., *Jour. Acous. Soc. Amer.*, Vol. 13, No. 3, p. 234, 1942.

sound diffusion so that the reflected sounds are properly mixed and the tonal quality of the performer or performing group is enhanced. The stage ceiling is broken in a saw-tooth fashion to provide a sound diffusing condition and to conceal the border lights and spotlights from the eyes of the audience. The rear wall of the stage is constructed of a series of plaster polycylindrical surfaces to provide diffuse reflection of the sound from this boundary. The side walls of the stage are provided with spherical surfaces for diffusing the reflected sound. The acoustical treatment of the stage,
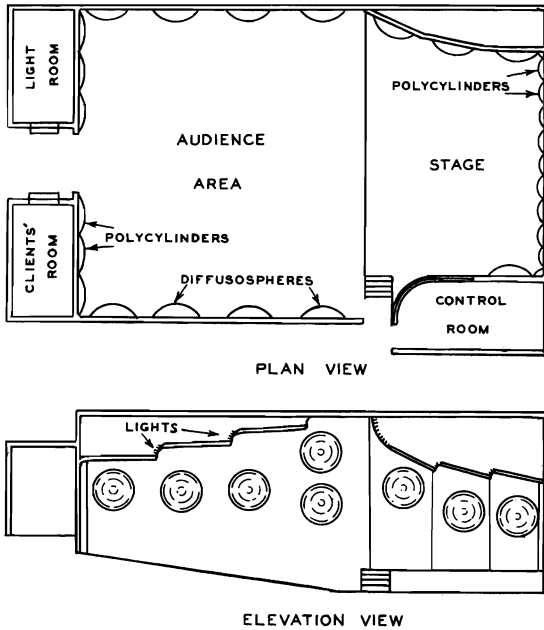
Fig. 11.34. Plan and elevation views of an auditorium-type broadcasting studio. (After Nixon and Nygren.)

save for the diffusospheres on the side walls and polycylinders at the rear of the stage, is rockwool, 2 inches in thickness, covered with perforated sheet metal or perforated sheet asbestos. The diffusospheres throughout the studio and polycylinders at the rear of the stage are made of plaster and backed by rockwool. The ceiling and the side walls in the rear two thirds of the auditorium section are untreated. The walls and ceiling in the front of the auditorium section are treated with 2 inches of rockwool covered by perforated asbestos. Heavy upholstered chairs in the audience area provide substantially the same acoustical conditions with and without an audience present in the studio. The control room is located so that the occupants have an unobstructed view of the stage and studio seating section. The clients' room is located so that the sponsors may watch and listen to

the progress of the program. Lighting booths are also provided in the rear for lighting the stage.

C. *Sound Pickup Arrangement for a Radio Broadcast.*—A broadcast studio with a setup for the sound pickup for a sound broadcast is shown in Fig. 11.35. The monitor room is located next to the studio. A large window between the monitor room and the studio makes it possible for the monitor man to view the action in the studio. Separate microphones are used for the orchestra, actors, and announcer. A transcription turntable supplies
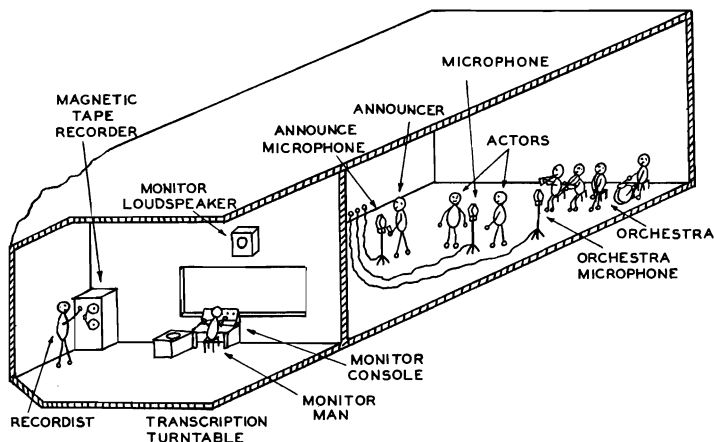
FIG. 11.35. Perspective view of a radio broadcasting studio and monitor room showing the arrangement of the elements used in radio broadcasting studio.

recorded program material. The program may also be recorded on a magnetic tape recorder.

A common type of radio broadcast takes place in a studio of the type shown in Fig. 11.34. In general, shows are played to an audience because the studio is designed for an audience. The arrangements of the announcer, actors, orchestra, and sound effects are shown in Fig. 11.36. The announcer, actors, and orchestra are located on the stage. The sound effects may be located on or off the stage. The monitoring engineer controls and mixes the pickup by the different microphones.

D. *Scoring and Recording Studios.*[73,74,75,76,77,78]—Scoring and recording studios are used for recording the music in sound motion pictures and phonograph records. In recent years, considerable effort has been expended

[73] Rettinger, M., *Proc. IRE*, Vol. 28, No. 7, p. 296, 1940.

[74] Volkmann, J. E., *Jour. Acous. Soc. Amer.*, Vol. 13, No. 3, p. 234, 1942.

[75] Rettinger, M., *Jour. Soc. Mot. Pic. Eng.*, Vol. 39, No. 3, p. 186, 1942.

[76] Livadary and Rettinger, *Jour. Soc. Mot. Pic. Eng.*, Vol. 42, No. 6, p. 361, 1944.

[77] Slyfield, C. O., *Jour. Soc. Mot. Pic. Eng.*, Vol. 42, No. 6, p. 367, 1944.

[78] Ryder, L. L., *Jour. Soc. Mot. Pic. Eng.*, Vol. 42, No. 6, p. 369, 1944.

in the improvement of the acoustics of scoring and recording studios. To obtain good acoustics, particularly for large musical aggregations, the studio should be large. The studio should be designed so that the reflected sound is thoroughly diffused. The studio should be well soundproofed. A scoring and recording stage satisfying these requirements is shown in Fig. 11.37. The maximum dimensions for the height, width, and length are respectively 30, 50, and 75 feet. A shell is provided for the orchestra at the live end of the studio. The voluminous part of the studio is made sound absorbent to simulate an imaginary audience. The reflecting portion of the convex surfaces constituting the side walls of the stage are made of ¼-inch plywood.
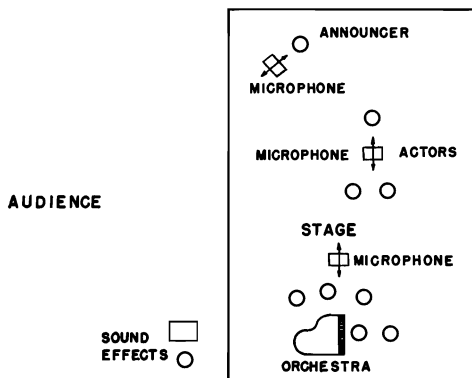
FIG. 11.36. Arrangement of the announcer, actors, and sound effects for a sound broadcast show in the studio of Fig. 11.33. (After Olson, "Musical Engineering," McGraw-Hill Book Company, New York, N.Y., 1952.)

One fourth of the convex surface is made absorbent as shown in Fig. 11.37. The ceiling construction is similar to the wall surface save that one fourth of the convex surface is equipped with ventilating grills instead of absorbing material. Wood polycylindrical surfaces comprise the rear wall of the shell. The treatment on the side walls and rear wall consists of rockwool packed between 2 by 4 inch vertical studs. Wood strips, 1 by 2 inches, were applied to vertical studs graduated in spacing from 27 inches near the wainscoting to 12 inches near the ceiling. Fiberboard ½ inch thick and plywood ⅜ inch thick were applied to the studs between the stripping to produce a series of horizontal rockwool, fiberboard and plywood panels. The construction of the ceiling is similar to the walls save that the plywood panels are omitted and the fiberboard panels made narrower due to the reflective floor parallel to it. Since the wood strips were thicker than the plywood and fiberboard panels, a sheet of muslin is stretched over the walls and ceiling to form a monolithic surface broken only by a narrow decorative molding fastened to the furring strips. The wall construction provides a uniform absorption

and eliminates concentrations of the reflected sound. The live shell with convex surfaces provides an ideal environment for the orchestra as well as providing a means for eliminating sound concentrations and for directing the flow of sound toward the absorbing part of the studio.

PLAN VIEW
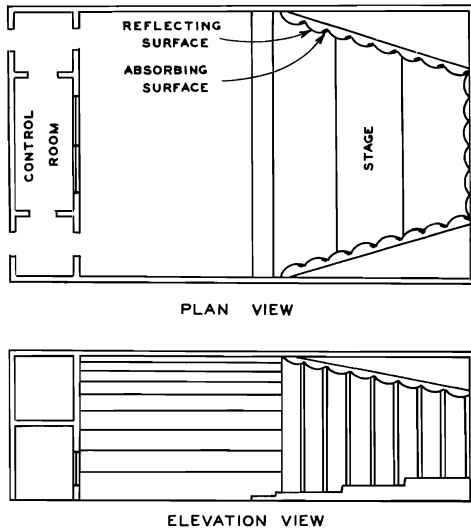
ELEVATION VIEW

FIG. 11.37.   Plan and elevation views of a scoring and recording studio.   (After Rettinger.)

E. *Sound Pickup Arrangements for Orchestra.*[79]—The sound pickup arrangement for broacasting or recording a symphony orchestra is shown in Fig. 11.38.   The orchestra is usually located on the stage of a broadcasting studio or recording studio, as shown in Figs. 11.34, 11.35, and 11.37.   In the case of a broadcast, there is usually an audience.   A single-velocity microphone located 15 to 25 feet in front of the orchestra at a height of 15 feet is usually used for the pickup of a symphony orchestra.   For average acoustical conditions, this gives very good instrument balance and reverberation characteristics.   Definition is determined by the ratio of direct to reflected sound.   If the microphone is placed closer to the orchestra, the definition is increased.   As the microphone is moved away from the orchestra, the definition decreases but the blending of the various instruments is improved owing to a decrease in the ratio of direct to reflected sound.   The microphone distance should be adjusted to give the optimum balance between definition and blending.   The balance between the various instruments is determined by the distance from the microphone, the directivity pattern of the microphone, and the directivity patterns of the musical instruments.   For certain conditions it may be desirable to use separate microphones for the pickup of solo instruments.   A separate velocity microphone is used

[79] Gurin, H., *Audio Engineering*, Vol. 32, No. 2, p. 9, 1948.

for the soloist, oriented so as to discriminate against sound from the orchestra.

The sound pickup arrangement for a small concert orchestra is shown in Fig. 11.39. As in the case of the symphony orchestra, a single velocity microphone located 10 to 15 feet in front of the orchestra at a height of 10 feet is used.

For the pickup of a dance band or orchestra, for either broadcasting or recording, a large number of microphones is used, as shown in Fig. 11.40. Each microphone covers one instrument or group of instruments. In the case of popular music, a high order of definition is desirable. Because of the
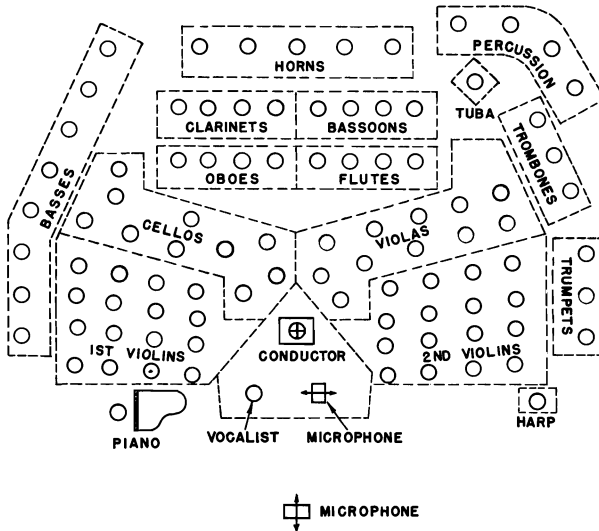
FIG. 11.38. Arrangement of the instruments and microphones for recording or broadcasting a symphony orchestra. (After Gurin.)

fast tempo, the ratio of direct to reflected sound must be kept large or the music will be blurred. In order to accomplish this objective, it is necessary to use a very small distance between the microphone and the instruments. This in turn requires a number of microphones, as shown in Fig. 11.40, in which each microphone covers a single instrument or group of instruments. In this arrangement, the balance of the orchestra is controlled by the monitoring engineer. Sometimes dance bands are picked up for broadcasting from very noisy surroundings. This is another reason for the use of a large number of microphones placed close to the instruments because a larger ratio of desired sound to noise is obtained by this type of pickup.

The preceding considerations of the pickup of sound have been concerned with single-channel sound reproduction.

Within the past three years stereophonic sound reproduction has been developed and commercialized in sound motion pictures on a wide scale.
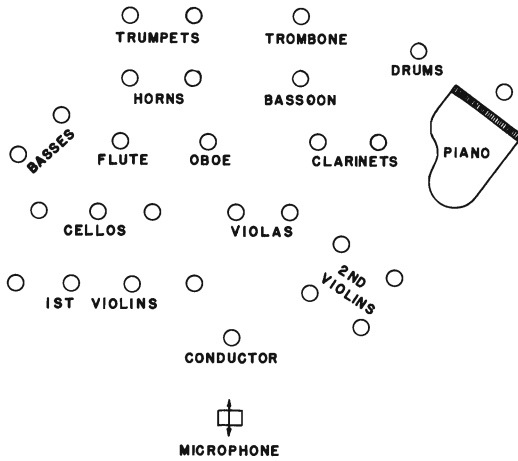
FIG. 11.39. Arrangement of the instruments and microphones for recording or broadcasting a small concert orchestra. (After Gurin.)

See Secs. 9.9 and 13.7B. The system has been of the three-channel type. Stereophonic sound recordings for reproduction in the home in the form of both disk and magnetic tape have been developed and commercialized during the past three years. See Sec. 13.5B. The system has been of the
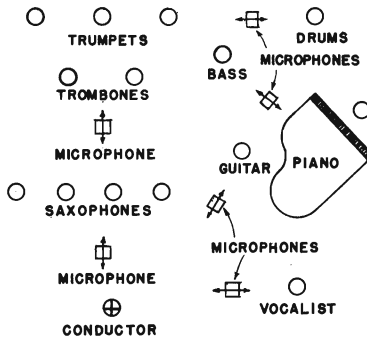
FIG. 11.40. Arrangements of the instruments and microphones for recording or broadcasting a dance or popular orchestra. (After Gurin.)

two-channel type. The microphone arrangement for stereophonic sound in the form of an orchestra or other large area action is shown in Fig. 11.41. The main idea is to confine the sound pickup in each channel to specific areas of the action. For example, in the two-channel system in Fig. 11.41A, microphone M_1 picks up sound over the left half of the area of action and

microphone M_2 picks up sound over the right half of action. In reproduction the loudspeaker corresponding to microphone M_1 will reproduce the left portion of the area of action and the loudspeaker corresponding to Microphone M_2 will reproduce the right portion of the orchestra. Directional microphones must be employed in order to accomplish this division of sound pickup. Unidirectional microphones are shown in Fig. 11.41. Bidirectional microphones may also be used. If nondirectional microphones are used, there will be considerable overlapping between the two channels because the only division will be due to distance. There is also some overlapping in the case of the directional microphones in mid-portion of the area, but this is necessary and desirable. A three-channel stereophonic pickup is shown in Fig. 11.41B. The ideas are the same as in the case of the two-
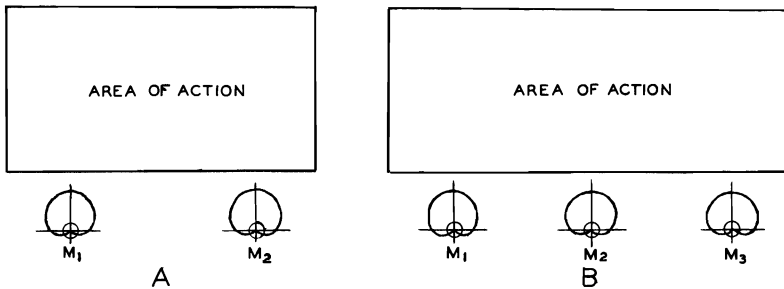
FIG. 11.41. Arrangement of the microphones with respect to area of action for two- and three-channel stereophonic sound pickup.

channel system. Each microphone covers a portion of the action. The use of three channels gives a smoother distribution. For this reason in some of the recent motion-picture systems of the road-show type, five channels are used.

F. *Vocal Studios.*[80,81]—In sound motion pictures when scoring an orchestra and one or more vocalists, it has been the practice to record the orchestra on one film channel and the vocalists on the second or separate film channel. This permits great latitude in musical balance when the two sound tracks are dubbed together. Frequency discrimination or accentuation of various portions of the frequency ranges in either or both the vocal and orchestra recording may be made without any relation between the two. Compression may be carried out in either or both channels. Synthetic reverberation may be added in either or both channels. It is quite evident that the use of two separate channels permits a wide range of artistic effects which would be impossible if a single original record were made.

The vocal studio should be located adjacent to the orchestra studio. A window between the two studios should be placed so that the vocalists or

[80] Mounce, Portman, and Rettinger, *Jour. Soc. Mot. Pic. Eng.*, Vol. 42, No. 6, p. 375, 1944.

[81] Ryder, L. L., *Jour. Soc. Mot. Pic. Eng.*, Vol. 42, No. 6, p. 379, 1944.

vocal group can see the conductor. The vocalists hear the orchestra by means of telephone receivers which reproduce the orchestra. In general, the number in the vocal studio will not exceed thirty.

The acoustics of the vocal room should be similar to that of a small standard broadcast studio (see Sec. 11.3B). One of the most important considerations in the design of a vocal studio is the sound isolation between the vocal studio and the orchestra. The sound level of the orchestra in the vocal studio must be sufficiently low so that it will not be recorded in the output of the microphones in the vocal studio. The type of wall and window construction for the vocal studio to obtain the desired value of sound isolation can be determined as outlined in Sec. 11.2R.

G. *Reverberation Time of Broadcasting, Recording, and Scoring Studios.*[82] —The optimum reverberation time of broadcasting, recording, and scoring
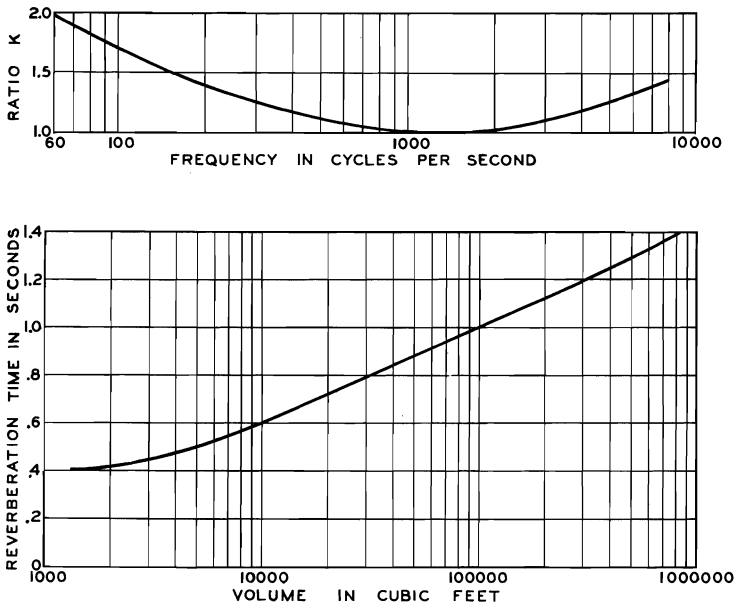
FIG. 11.42. Lower graph shows the reverberation time for a recording or broadcasting studio as a function of the volume for 1000 cycles. Upper graph shows the relation between the reverberation time and the frequency, that is, the reverberation time at other frequencies is obtained by multiplying by K. (After Morris and Nixon.)

studios as a function of the volume of the studio, for a frequency of 1000 cycles, is shown in the lower graph of Fig. 11.42. The reverberation for other frequencies can be obtained by multiplying by the factor K, obtained from the upper graph of Fig. 11.42. The reverberation time is greater at the lower and higher frequencies so that the aural rate of decay of pure tones

[82] Morris and Nixon, *Jour. Acous. Soc. Amer.*, Vol. 8, No. 2, p. 81, 1936.

will be approximately the same for all frequencies (see Secs. 12.6 and 12.7 and Figs. 12.11 and 12.13).

H. *Sound Stages for Motion Pictures and Television.*[83,84,85,86,87]—A sound stage is a large acoustically treated room used to house a stage setting in sound motion picture recording or television broadcasting. The sound stage is equipped with catwalks, power outlets, air conditioning, and other facilities required for the production of sound motion pictures or television. Most sound stages are equipped with an adjacent recording and monitoring room. In the case of large stages portable sound booths are used on the stage.

The technique of the pickup of sound in motion pictures and television differs from that of radio broadcasting and phonograph recording in that the microphone must be kept out of the picture. In the case of radio broadcasting and phonograph recording the microphone can be placed in a position which yields the best sound pickup. For the broadcasting of speech, the distance from the speaker to the microphone can be made very small so that the received reverberation is negligible (see Sec. 11.3A). However, in sound motion-picture recording and television sound pickup the microphone must be kept out of the picture. This means that the pickup distance will be quite large. Under these conditions, the received reverberation can be kept low by making the reverberation time of the sound stage as low as possible. When the reverberation time of the stage is low, the setting determines the acoustics of the sound picked up by the microphone. In the case of sets consisting of small rooms the acoustics of the set masks the acoustics of the sound stage. In the early days of sound motion-picture recording it was customary to make the sets of acoustical materials having good transmission at the low frequencies and high absorption at the high frequencies. In this way it was possible to keep the reverberation time of the set very low. With the advent of directional microphones it has been possible to use conventional materials for the construction of sets.

Typical over-all dimensions for a large studio are as follows: height, 45 feet; width, 100 feet; and length, 140 feet. A reverberation time of about one-half second is possible for stages with a volume of about 500,000 cubic feet. In the case of smaller stages a lower reverberation time may be obtained. It is usually standard practice to erect several sets on a single stage. This procedure may render some of the absorbing material of the stage ineffective and thereby increase the reverberation time. These

[83] Ringel, A. S., *Jour. Soc. Mot. Pic. Eng.*, Vol. 15, No. 3, p. 352, 1930.

[84] Rettinger, "Applied Architectural Acoustics," Chemical Publishing Company, Brooklyn, N.Y., 1947.

[85] Knudsen and Harris, "Acoustical Designing in Architecture," John Wiley and Sons, New York, N.Y., 1950.

[86] Olson, "Musical Engineering," McGraw-Hill Book Company, New York, N.Y., 1952.

[87] Bloomberg, Pond, and Rettinger, *Jour. Soc. Mot. Pic. Tel. Eng.*, Vol. 63, No. 1, p. 19, 1954.

undesirable effects may be overcome by the use of heavy sound absorbing curtains which shield the different sets from each other.

The floors of the sound stage should be rigid and massive to prevent transmission of sound along the floor due to impacts, as, for example, in the case of large dancing groups.   An improvement in the case of the floor can be effected by dividing the floor into sections and isolating each section mechanically.

The sound stage should be sound proof and isolated against vibrations coming through the ground and from adjacent rooms and buildings.   For average conditions a relatively light, double-wall construction may be used.   Under these conditions, the outer wall consists of 1-inch fiberboard sheathing nailed to the vertical studs.   On the outer face of this fiberboard a layer of building paper and stucco wire netting is applied.   Stucco 1 inch thick is applied to the wire netting.   The inner wall consists of vertical 2 by 4 inch studs spaced from the outer wall by at least 2 inches.   A layer of $\frac{1}{2}$-inch plasterboard is applied to the outside face of the studs.   The space between the 2 by 4 inch studs is filled with rockwool battens 4 inches thick.   In the case of very noisy locations massive double-wall construction will be required as, for example, a concrete outer wall.   The only noise in which the roof and ceiling are involved is that of airplanes.   With the ever-increasing number of airplanes, particular consideration must be given to the roof and ceiling.   In the past, 4-inch rockwool battens have been applied directly to the underside of the roof.   This, in general, does not provide adequate shielding and a ceiling of fiberboard and rockwool separated from the roof is required.

I. *Sound Pickup Arrangements for Sound Motion Pictures and Television.*— A sound stage with a setup for sound motion-picture recording is shown in Fig. 11.43.   In this case the initial monitoring is done at a console located on the sound stage.   This permits the monitor man to have an excellent view of the action.   Most sound stages are equipped with a sound recording room located next to the sound stage.   This room may also be used as a monitoring room.   In the case of very large studios, portable monitoring and recording booths are located on the stage.

A complete sound motion-picture recording system is depicted in Figs. 13.21 and 13.22 and described in Sec. 13.7A.   The microphones and recorders used in recording sound motion pictures are described in Chapters VIII and IX.

The sound stages described in this section are used for the most part for recording dialogue.   The recording of music for motion pictures is carried out in scoring stages, see Sec. 11.3D.   In the recording of dialogue for motion pictures the microphone is suspended on a boom.   The boom operator maintains a minimum distance between the microphone and action and at the same time keeps the microphone out of the picture.   If the action covers a large area, more than one microphone and boom may be used.

In the recording of three-channel stereophonic dialogue, two techniques are in general use as follows: In one procedure three microphones are mounted

on a boom as shown in Fig. 13.27. The outputs of the three microphones are amplified and recorded simultaneously on three magnetic tracks. This is standard stereophonic sound pickup. The distance between the microphones is adjusted to correspond to the area of action. In this procedure very large pickup distances are used. As a consequence, in many cases presence in the reproduced speech is impaired. In the other procedure a single microphone is used as in the case of single-channel systems. Stereophonic sound reproduction is accomplished by rerecording the single channel to three channels. In this rerecording procedure the sound corresponding to the action is recorded on the proper sound track or tracks. Under these
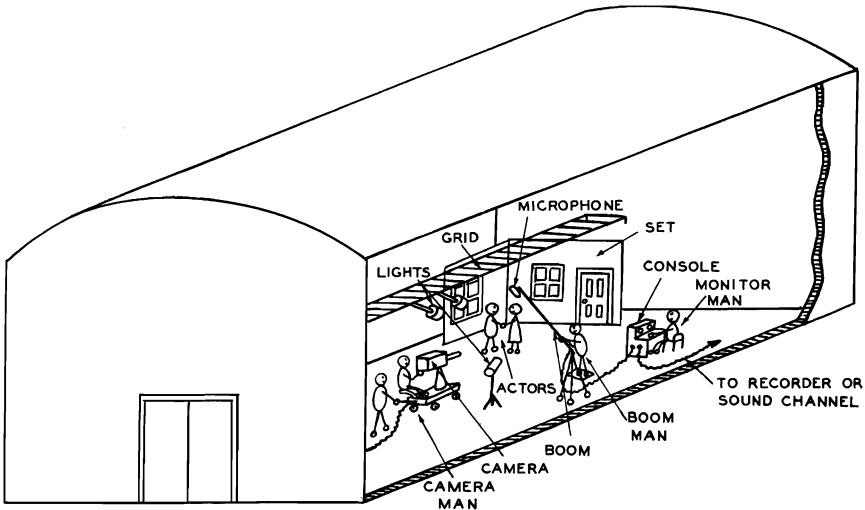
FIG. 11.43. Perspective view of a sound stage showing the arrangement of the elements used in recording a sound motion picture.

conditions the sound in reproduction in the theater will correspond to the action of the picture. The advantage of the single pickup procedure is the high order of presence inherent in single-channel pickup in which a minimum microphone-to-action distance is employed.

A sound stage with a setup for a television broadcast pickup is shown in Fig. 11.44. In this case the monitoring room is located next to the stage. A large window between the monitoring room and the stage makes it possible for the audio and video monitor men to view the action on the stage. The actors, orchestra, announcer, and sound effects men are all located on the stage. The stage cameras pick up the action of the actors. The title camera picks up the titles. Separate microphones are used for the different sounds at the appropriate time. The video men monitor and select the appropriate camera. For the main action, a boom type microphone is used employing the same technique as in the case of a sound motion-picture recording.

The technique of the pickup of sound in motion pictures and television differs from that of radio broadcasting and phonograph recording in that the microphone must be kept out of the picture. This is done by suspending the microphone from a boom so that it can be raised and lowered by the boom man (Figs. 11.43 and 11.44). The boom is also equipped with a suitable manually operated swivel arrangement so that the boom man can keep the directional microphone pointed at the action. Since the microphone must be kept out of the picture, the pickup distance is usually quite large. In the case of radio broadcasting and phonograph recording, the microphone can be placed in a position which yields the best sound pickup.
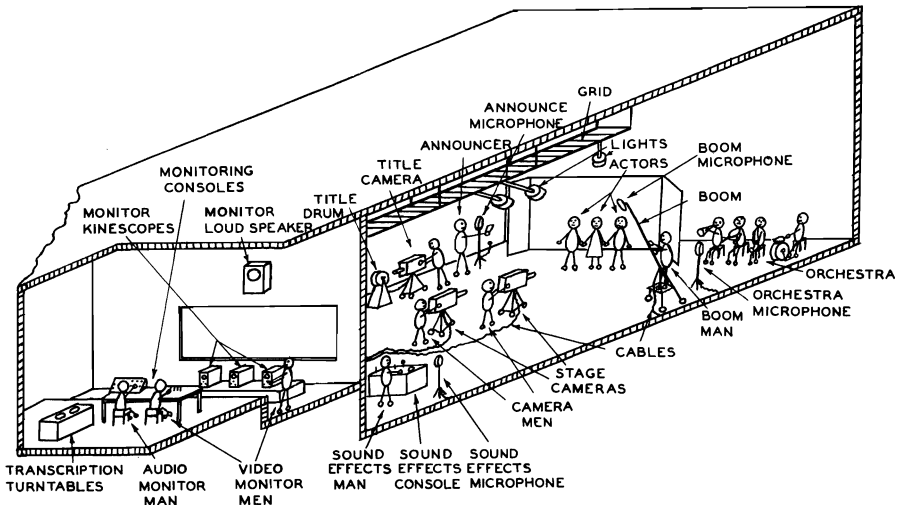
FIG. 11.44.   Perspective view of a sound stage and monitor room showing the arrangement of the elements used in television broadcasting.

For the broadcasting of speech, the distance from the speaker to the microphone can be made very small so that the received reverberation is negligible. However, for sound stages with large pickup distances, the received reverberation is kept low by making the reverberation time of the sound stage as low as possible and by the use of directional microphones. When the reverberation time of the stage is low, the setting determines the acoustics of the sound pickup by the microphone. In the case of sets consisting of small rooms, the acoustics of the set mask the acoustics of the sound stage. In the early days of sound motion-picture recording it was customary to make the sets of acoustical materials having good transmission at the low frequencies and high absorption at the high frequencies. In this way it was possible to keep the reverberation time of the set very low. With the advent of directional microphones it has been possible to use conventional materials for the construction of sets.

In the television show described above and depicted in Fig. 11.44, there

is no audience.   In another type of television show, the action takes place
before an audience.   For this type of show, conventional theaters have been
used.   In some of these theaters the stage has been increased in size.   The
broadcast studio of the type shown in Fig. 11.45 may be used for a television
show played to an audience.   In the special television studio designed for
a show played to an audience, an amphitheater is combined with a sound
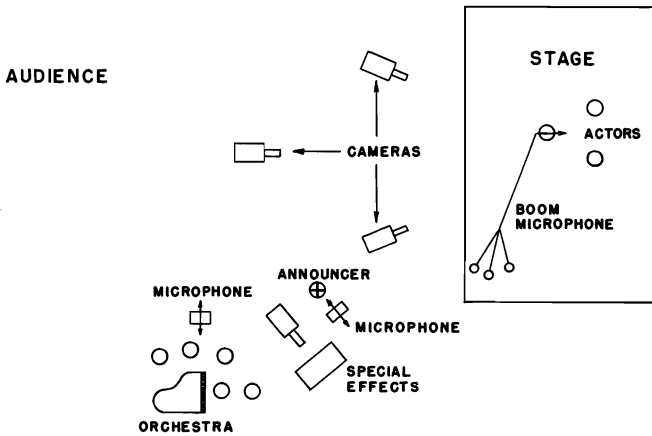
FIG. 11.45.   Arrangement of the announcer, actors, special effects,
orchestra, cameras, and microphones for a television show.

stage of Fig. 11.44.   A plan view of a typical arrangement used when a
television show is played to an audience is depicted in Fig. 11.45.   Both
picture and sound are picked up from the stage whereas only the sound is
picked up from the orchestra.   Titles and special effects are picked up on a
special camera.   The announcer uses a separate microphone.   Velocity
microphones are used for the orchestra and announcer sound pickup.   A
unidirectional is used as a boom to pick the sound of the action on the
stage.   Standard boom microphone techniques are employed for this sound
pickup.

12

SPEECH, MUSIC, AND HEARING

12.1. Introduction.—The major portion of this book has been concerned with the theory, design, and testing of acoustical apparatus for the reproduction of sound. The ultimate significant destination of all reproduced sound is the human ear. The physiological and psychological effects of the reproduced sound are the most important factors in any sound reproducing system. Sounds heard may be classified as speech, music, or noise. An enormous amount of valuable data relating to speech and hearing have been collected. This information is extremely useful in the development and design of sound reproducing equipment. It is beyond the scope of this book to present all the pertinent material of physiological and psychological acoustics. For information beyond that given in this chapter the reader may consult the references. It is the purpose of this chapter to show the principal characteristics of speech, music, and hearing and the relation between these characteristics and the objective characteristics discussed in the preceding chapters.

12.2. Hearing Mechanism.[1,2,3,4]—The hearing mechanism, shown in Fig. 12.1, may be divided into three parts: the outer ear, the middle ear, and the inner ear. The outer ear consists of the external ear or pinna and the ear canal which is terminated in the ear drum or tympanic membrane. Behind the ear drum is the middle ear, a small cavity in which three small bones—the hammer, the anvil, and stirrup—form the elements of a system for transmitting vibrations from the ear drum to an aperture, termed the oval window of the inner ear. The cavity in the middle ear is filled with air by means of a pressure equalizing tube, termed the Eustachian tube, leading to the nasal pharynx. The casing of the inner ear or cochlea is a bony structure of a spiral form (two and three quarter turns). The cochlea is divided into three parts by the basilar membrane and Riessner's membrane. These three parallel canals are wound into the spiral. On the one

---

[1] Fletcher, "Speech and Hearing," D. Van Nostrand Company, Princeton, N.J., 1929.

[2] Steinberg, "Acoustics"; Pender and McIlwain, "Electrical Engineers Handbook," *Communication and Electronics*, John Wiley and Sons, New York, N.Y., 1936.

[3] Fletcher, "Speech and Hearing in Communication," D. Van Nostrand Company, Princeton, N.J., 1955.

[4] Stevens and Davis, "Hearing," John Wiley and Sons, New York, N.Y., 1938.

side of the basilar membrane is the organ of Corti, which contains the nerve terminals in the form of small hairs extending into the canal of the cochlea. These nerve endings are stimulated by the vibrations in the cochlea.

A schematic cross-sectional view of the ear and the acoustical network of the vibrating system is shown in Fig. 12.2. When a sound wave impinges upon the ear, it enters the ear canal and causes the eardrum to vibrate. The vibration of the eardrum is transmitted to the inner ear or cochlea by the three bones of the middle ear. The cochlea may be considered to be made up of distributed constants as shown in Fig. 12.2. The meters indicate the volume currents $\dot{X}_1$, $\dot{X}_2$ ... $\dot{X}_K$ in the branches 1, 2 ... K.
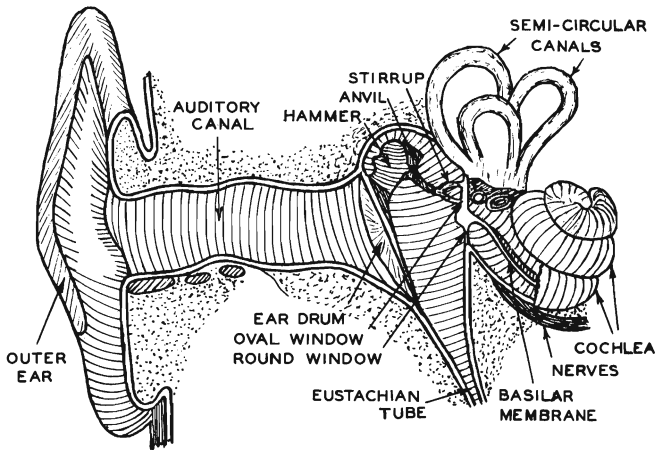
FIG. 12.1. Sectional and perspective views of the hearing mechanism.

These volume currents in turn actuate the nerves. High-frequency sounds excite the portion of the cochlea nearest the oval window as shown in Fig. 12.2. Low frequencies are associated with the extreme end removed from the oval window. In other words, the cochlea is a frequency discriminating system in which a certain vibration frequency is associated with a certain definite section of the cochlea. The auditory nerves which terminate all along the cochlea are stimulated by the vibrations. The activated nerve sends a pulse to the brain which in turn is translated into a definite pitch. The frequency depends upon the nerve which is actuated.

The acoustical impedance looking into the ear canal is of interest in the design of artificial ears for testing insert type telephone receivers. The components of the acoustical impedance looking into the ear canal are shown in Fig. 12.3. The dimensions of the average ear canal are as follows: length, 2.2. centimeters, cross-sectional area, .45 square centimeter and volume, 1 cubic centimeter. The acoustical reactance characteristic

of Fig. 12.3 shows that the effective volume is somewhat greater than the actual volume which is due to the resilient walls and eardrum.

SCHEMATIC SECTIONAL VIEW

ACOUSTICAL NETWORK
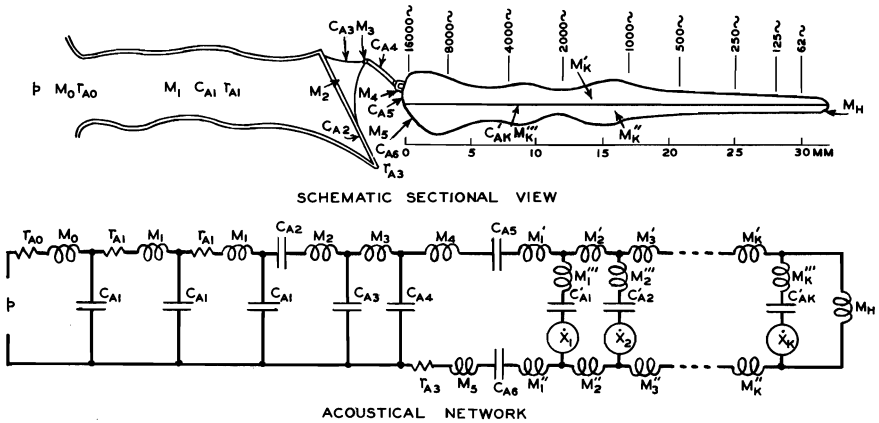
FIG. 12.2. Schematic sectional view and acoustical network of the hearing mechanism. In the acoustical network: p = the actuating sound pressure. M_0 and r_{A0} = the inertance and acoustical resistance of the air load upon the opening to the ear canal. M_1, C_{A1}, and r_{A1} = the distributed inertance, acoustical capacitance, and acoustical resistance of the ear canal. M_2 = the inertance of the eardrum and hammer. C_{A2} = the acoustical capacitance of the eardrum and tensor tympani. C_{A3} = the acoustical capacitance of the hammer handle. M_3 = the effective inertance of the ossicles and hammer-anvil joint. C_{A4} = the acoustical capacitance of the hammer-anvil joint and anvil arm. M_4 = the inertance of the stirrup and oval window. C_{A5} = the acoustical capacitance of the oval window and the tensor stapedius. M_1', M_2', $M_3' \ldots M_K'$ represent inertances of the liquid in the scala tympani. M_1'', M_2'', $M_3'' \ldots M_K''$ represent inertances of the liquid in the scala vestibula. M_1''', $M_2''' \ldots M_K'''$, and C_{A1}', C_{A2}', and C_{AK}' represent the inertances and acoustical capacitances of the basilar membrane which separates the upper from the lower liquid. M_H = the inertance of the liquid in the heliotrema. The nerve terminals are represented by the volume current meters X_1, $X_2 \ldots X_K$. C_{A6}, M_5, and r_{A3} = the acoustical capacitance, inertance, and acoustical resistance of the round window. (After Steinberg.)

12.3. Voice Mechanism.[5,6,7]—The voice mechanism, shown in Fig. 12.4, consists of three parts: the lungs and associated muscles for maintaining a flow of air, the larynx for converting the steady air flow into a periodic modulation, and the vocal cavities of the pharynx, mouth, and nose which vary the harmonic content of the output of the larynx. The vocal cords do not receive excitation at the frequency of vibration. The source of power is the steady air stream.

The voice mechanism is analogous to the vacuum tube oscillator in that it converts a direct current flow into a pulsating flow. The elements of a

---

[5] Fletcher, "Speech and Hearing," D. Van Nostrand Company, Princeton, N.J., 1929.

[6] Wegel, R. L., *Jour. Acous. Soc. Amer.*, Vol. 1, No. 3, Part 2, p. 1, 1930.

[7] Fletcher, "Speech and Hearing in Communication," D. Van Nostrand Company, Princeton, N.J., 1955.

simplified larynx are shown in Fig. 12.5. The electronic analogy of this vibrating system is also shown in Fig. 12.5. The electronic system may be replaced by a generator having an internal pressure p_G and an internal acoustical resistance r_{AG}. M represents the mutual coupling between branch 1 and branch 2 of Fig. 12.5. The acoustical circuit under these conditions is depicted in Fig. 12.5. The frequency of the vibration is
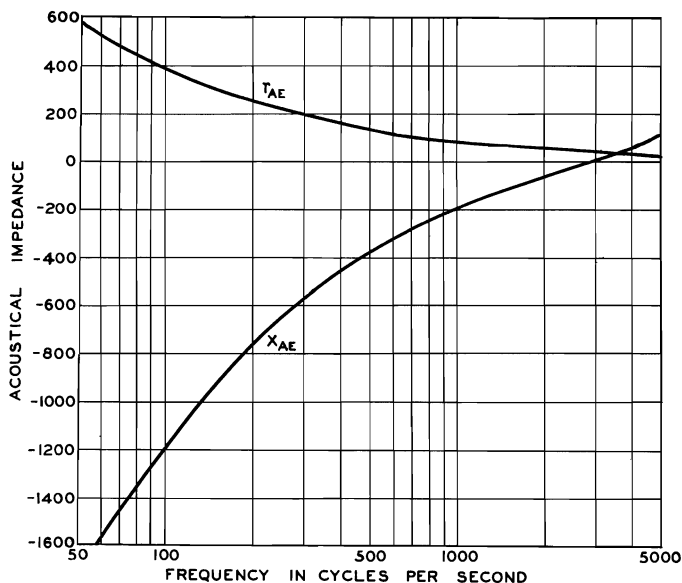
FIG. 12.3. The acoustical impedance components of acoustical resistance r_{AE} and acoustical reactance x_{AE} characteristics at the input to the ear canal. (After Steinberg.)

governed by all the elements of the vibrating system, that is, the acoustical capacitance, C_{A1}, of the vocal cords incurred by tension, the inertance, M_1, and acoustical resistance, r_{A1}, of the vocal cords, the inertance, M_2, and acoustical resistance, r_{A2}, of the aperture and the load acoustical impedance, z_{AV}, due to the vocal cavities. A schematic view and the acoustical network of the vocal cavities can be seen in Fig. 12.6. This shows that the nature of the input acoustical impedance, z_{AV}, to the acoustical cavities is extremely complex. The inertances M_1, M_2, and M_3 and the acoustical capacitances C_{A1} and C_{A2} can be varied by changing the sizes of the apertures and the volumes of cavities.

The oscillation of the vocal cords is of the relaxation type rather than the conventional sinusoidal variation. This is borne out[8] by the rapid starting and stopping in the case of some sounds. The oscillator shown in Fig. 12.5 will produce waves of the relaxation type providing the circuit

[8] Drew and Kellogg, *Jour. Acous. Soc. Amer.*, Vol. 12, No. 1, p. 95, 1940.

constants and the nonlinear elements are suitable. The wave shape of a relaxation oscillator corresponds to the general wave shape of the output of the vocal cords[9] shown in Fig. 12.7. The output of the vocal chords was measured with a pressure microphone in the pharynx with the mouth and nose cavities damped. The microphone was terminated in a long damped pipe one inch in diameter. Referring to Fig. 12.7, it will be noted that the bottom of the wave corresponding to the closing of the vocal chords is flattened. This means that the vocal chords are closed or practically
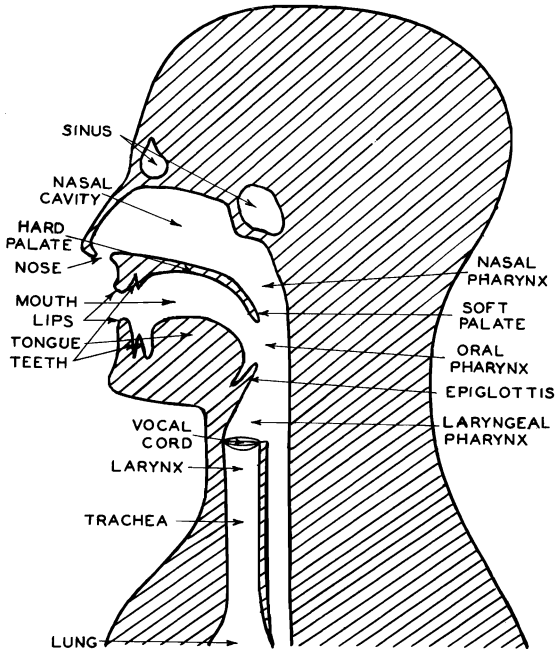
FIG. 12.4. Sectional view of the head showing the voice mechanism.

closed during a portion of the cycle. Nevertheless, the general shape of the wave is that of a saw tooth. A saw-tooth wave contains the fundamental and all the harmonics. Therefore, the generator, p_G, should produce the fundamental frequency and all the harmonics of the fundamental frequency. When the vocal cords are set into vibration as outlined above, the output of the larynx consists of a steady stream with superimposed impulses (Fig. 12.7). This pulsating air stream passes through the air cavities of the head. The harmonic content of the output is modified due to the discrimination introduced by the acoustical network of Fig. 12.6. The effect of the vocal cavities is illustrated in Fig. 12.7, which shows the wave shape

9 Olson, H. F., Unpublished Report.

of the sound output of the mouth and nose corresponding to the wave shape of the output of the vocal cords. When the shape of the vocal cavities is altered the acoustical elements of the acoustical network of Fig. 12.6 are altered which in turn alters the output harmonic content. These changes together with a change in the fundamental frequency of the vocal cords
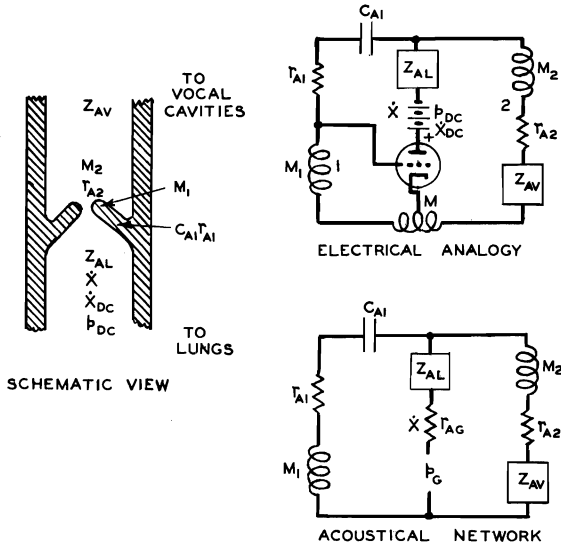
FIG. 12.5. Sectional view of a vibrating system approximating the larynx. In the electrical analogy and acoustical network: p and X = the d-c air pressure and d-c volume current supplied by the lungs. M_2 and r_{A2} = the inertance and acoustical resistance of the aperture formed by the vocal cords. M_1, C_{A1}, and r_{A1} = the inertance, acoustical capacitance, and acoustical resistance of the vocal cords. z_{AV} = the input acoustical impedance to the vocal cavities. r_{AG} = the acoustical resistance of the generator. p_G = the pressure of the equivalent a-c generator. z_{AL} = the acoustical impedance of the trachea and lungs. M = the mutual coupling between branch 1 and branch 2.

make it possible to produce an infinite number of different sounds. The tongue plays the major role in altering the shape of the vocal cavities. The shape of the vocal cavities for four vowel sounds is shown in Fig. 12.8. It will be seen that the mouth opening, tongue, and epiglottis are the principal elements which are altered in these examples. Of course, the fundamental frequency of the vocal cords is also different in the four examples. The true vowels and diphthongs are produced by the above outlined resonance method. The so-called unvoiced constants, as, for example "S," are produced by air from the lungs passing over the sharp edges and through the narrow passages in various parts of the mouth and nose. The
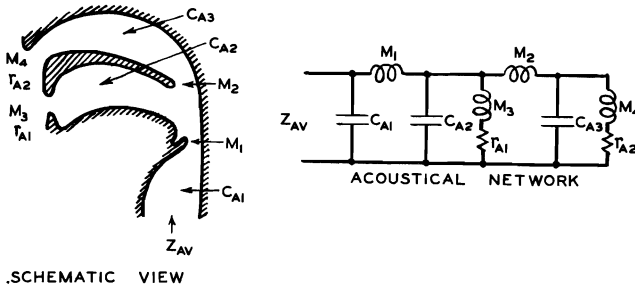
.SCHEMATIC VIEW

FIG. 12.6. Schematic sectional view and the acoustical network of the vocal cavities. In the acoustical circuit: z_{AV} = the input acoustical impedance to the vocal cavities. C_{A1} = the acoustical capacitance of the laryngeal pharynx. M_1 = the inertance of the narrow passage determined by the epiglottis. C_{M2} = the acoustical capacitance of the mouth cavity. M_2 = the inertance of the passage connecting the mouth and nasal cavities. C_{A3} = acoustical capacitance of the nasal cavity. M_3, M_4, r_{A1}, r_{A2} = the inertances and acoustical resistances of the mouth and nose openings and the air load upon these openings.

vocal cords are not used in the production of these sounds. The voice constants are produced by a combination of the two systems.

The voice mechanism then consists of a number of acoustical elements which can be varied by the person at will to produce a wide variation of tones differing in frequency, quality, loudness, duration, growth, and decay.
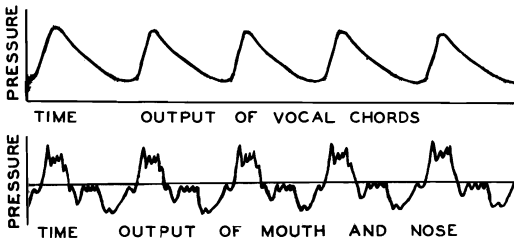
FIG. 12.7. Wave shapes of the output of the vocal cords and the mouth and nose for the vowel sound ĕ.

12.4. Artificial Voice Mechanisms. A. *Artificial Larynx.*[10]—A surgical operation known as tracheotomy leaves no connection between the lungs and mouth. It is performed in an emergency to prevent the patient from dying by suffocation incurred by a swelling of the throat due to an injury or infection. Following the operation the process of breathing is accomplished by drawing air in and out through a small opening in the neck. Since the larynx is bypassed the individual can make no vocal sounds. Under these conditions the individual can learn to talk by means of an

[10] Riesz, R. R., *Jour. Acous. Soc. Amer.*, Vol. 1, No. 2, p. 273, 1930.
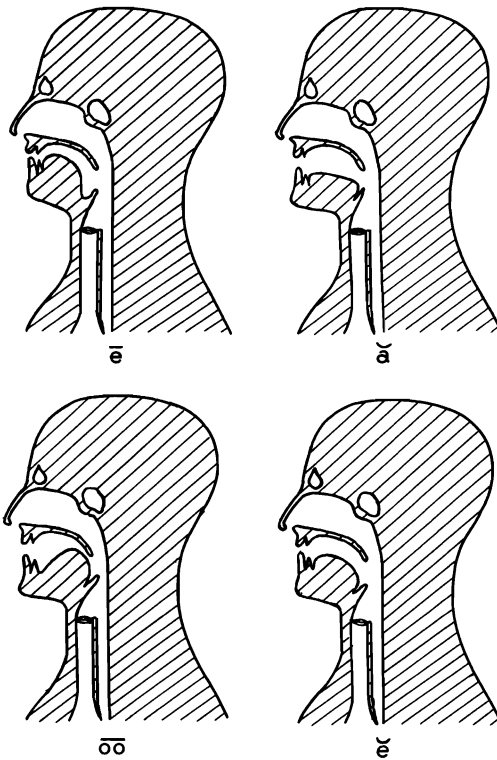
FIG. 12.8.   Sectional views of the head showing the
configuration of the mouth cavity for different vowel
sounds.

artificial larynx.   The artificial larynx, depicted in Fig. 12.9, consists of
a reed actuated by the air from the opening in front of the throat, through
which breathing takes place.   The complex sound output of the reed is
conducted to the mouth cavity by means of a small tube.   The quality
of the sound emitted by the reed is altered by the resonance of the cavities
of the head (see Sec. 12.3 and Fig. 12.6).   The artificial larynx is another
illustration of the major part which the mouth and lip shapes play in the
differentiation of speech sounds.

B.   *Voder*.[11]—The vocal tone, as outlined in Sec. 12.3, consists of a buzzer-
like tone.   The breath tone is a hisslike noise.   By suitable selection of
the frequency, intensity, quality, duration, growth, and decay of a tone it
should be possible to imitate any vocal sound.   The addition of a hisslike
noise will simulate the consonants.   A system of this type has been built
consisting of a series of electron tubes for performing the various functions.

[11] *Bell Labs. Record*, Vol. 17, No. 6, p. 170, 1939.

The frequency, quality, intensity, duration, growth, and decay of the various complex tones are changed by altering the various parameters in the system. The parameters are changed by means of keys forming a small keyboard similar to that of a musical instrument. The output of the electronic system is coupled to a loudspeaker which converts the electrical variations into the corresponding sound vibrations. This instrument is termed a voder. A skilled operator can "play" the voder and produce the sounds of speech. In this way an operator can "talk" by means of the voder.
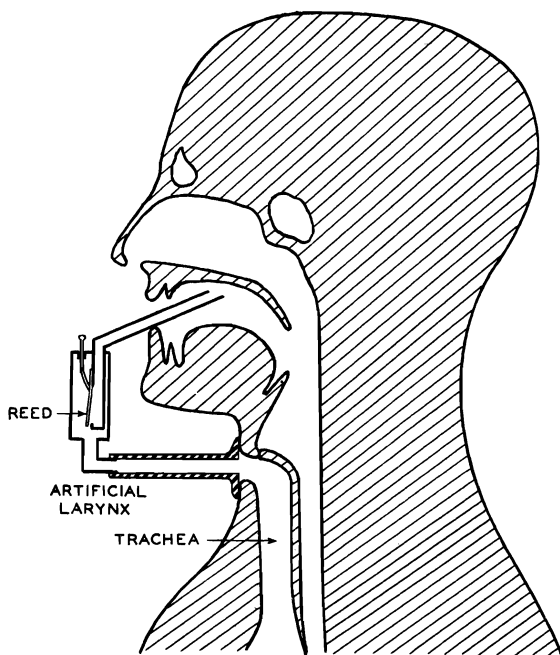
FIG. 12.9. The elements of an artificial larynx.

C. *Vocoder*.[12]—The vocoder employs a system similar to the voder save that the voice is used to actuate the system instead of the mechanical keys. Speech is picked up by means of a microphone, amplified and fed to pitch and spectrum analyzers. Control of the frequency, intensity, quality, growth, and decay is provided by the talker's speech. The original voice frequencies are analyzed and used to control these quantities. Remade speech of good intelligibility is produced by this means. The currents used in the controls contain only low syllabic frequencies of the order of 10 cycles per second as contrasted to the frequency range of 100 to 3000 cycles in the remade speech. The system described above has been termed

12 Dudley, H., *Jour. Acous. Soc. Amer.*, Vol. 11, No. 2, p. 169, 1939.

"vocoder" because it operates on the principle of coding the voice and then reconstructing the voice in accordance with the code.

D. *Speech Synthesizers.*—Various speech synthesizers have been developed. These systems are based upon the remaking of speech by the use of the fundamental physical characteristics of speech.

Speech can be synthesized by feeding the proper low-frequency currents to actuate the reproducing system of the vocoder.

Speech can also be synthesized by means of the electronic music synthesizer by breaking down the sounds of speech into the characteristics of frequency, intensity, growth, duration, decay, portamento, timbre, and vibrato. See Sec. 12.36.

Speech has been synthesized[13] by the use of the building blocks of speech termed modules. Speech sounds or modules are stored on a magnetic drum. These sounds are reproduced by means of magnetic heads in contact with the drum. To synthesize speech, the modules are reproduced at the proper instant and fed to a tape recorder. The tape containing the synthesized speech can be reproduced to produce the synthesized speech.

Speech has been synthesized[14] by employing an electrical analog of the vocal tract and suitable electronic generators. The electrical analog of the vocal tract contains electrical elements which correspond to the acoustical elements of the vocal tract. The electrical generator, corresponding to the vocal cords, produces pulses consisting of a series of saw-tooth type waves. The fundamental frequency of the waves can be varied. This generator produces the vowel sounds. The consonant sounds are generated by an electronic random noise generator. The system is operated by a switching system which actuates the generators, controls the frequency of the vowel generator, and varies the electrical elements of the analog of the vocal tract. Speech sounds can be produced by supplying the appropriate information to the switches.

Speech has been synthesized[15] by painting the frequency on a cellulose acetate base. These patterns are in fact copies of visible speech patterns. A multitone wheel, light source, and photoelectric cell are used to reproduce positive copies of the hand painted original. When light shines through a certain translucent area of the positive film, it passes through a certain area of the tone wheel. When the resultant interrupted light falls on the photocell, the electrical output corresponds to the light interruptions. The frequency and amplitude of the electrical output corresponds to the particular translucent area of the positive film. In this way, the formants of speech sounds can be produced and hence complete words and sentences can be synthesized.

Speech has been synthesized[16] by means of four tuned circuits, representing

[13] Harris, Cyril M., *Jour. Acous. Soc. Amer.*, Vol. 25, No. 5, p. 970, 1953.

[14] Stevens, Kasowski, and Fant, *Jour. Acous. Soc. Amer.*, Vol. 25, No. 4, p. 734, 1954.

[15] Cooper, Delattre, Liberman, Borst, and Gerstman, *Jour. Acous. Soc. Amer.*, Vol. 24, No. 6, p. 597, 1952.

[16] Stevens and Bastide, *Jour. Acous. Soc. Amer.*, Vol. 27, No. 1, p. 207, 1955.

the resonances of the vocal tract, and actuated by saw tooth and noise generators. Coded information from a punched paper tape controls the resonant frequency of the tuned circuits, the frequency of the saw-tooth generators, and the amplitude of the noise generator. Sounds, words, and sentences have been synthesized by means of this system.

12.5. Visible Speech.[17]—Visible speech is an electronic method of changing spoken words into visible patterns that someone may learn to read. A schematic arrangement of the apparatus for depicting speech in visible patterns is shown in Fig. 12.10. Speech is picked up by the microphone and converted into the corresponding electrical variations. These variations are amplified and limited in amplitude so that the amplitude range is
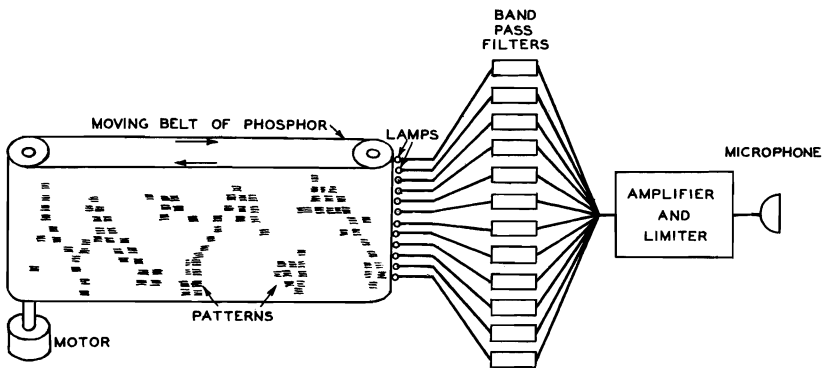
FIG. 12.10.   The elements of a system for producing visible speech.   (After Potter.)

confined within relatively narrow limits. The output of the amplifier is coupled to twelve band pass filters. Each filter covers a frequency band of 300 cycles. The entire frequency range covers the band from 150 to 3750 cycles. The output of each filter is coupled to a lamp. When a lamp is illuminated it produces a trace on the moving belt of phosphor. With this apparatus, a complex sound wave is divided into twelve discrete frequency bands. The portions of the frequency range, with intensity sufficient to produce illumination on the phosphor screen, will leave a trace. A complex wave or a series of complex waves will leave patterns on the moving belt of phosphor. Each vowel and consonant sound produces a unique and distinguishable pattern. Under these conditions, speech picked up by the microphone may be read from the moving belt of phosphor. Music or any other sound may also be picked up by the microphone and portrayed on the screen. Some of the uses of the visible speech apparatus are as follows: visual hearing for the deaf, teaching of the deaf to speak, speech correction, aid in the study of vocal music, etc.

[17] Potter, R. K., *Bell Labs. Record*, Vol. 24, No. 1, p. 7, 1946.   Also a group of papers on the subject of Visible Speech by a number of authors in *Jour. Acous. Soc. Amer.*, Vol. 18, No. 1, pp. 1 to 89 inc., 1946.

12.6. Response Frequency Characteristics of Ears.—The loudness of a pure tone depends upon the frequency and intensity (see Sec. 12.7 for the definition of loudness). This relation is revealed in the Fletcher-Munson[18] equal loudness level curves shown in Fig. 12.11. The 1000-cycle tone is the reference tone in these determinations. The loudness level of other tones is the intensity level of the equally loud 1000-cycle tone. These characteristics show that the ear is most sensitive in the region between 3000 and 4000 cycles. The sensitivity of the ear decreases above and below this frequency. The response frequency characteristics of ears are useful
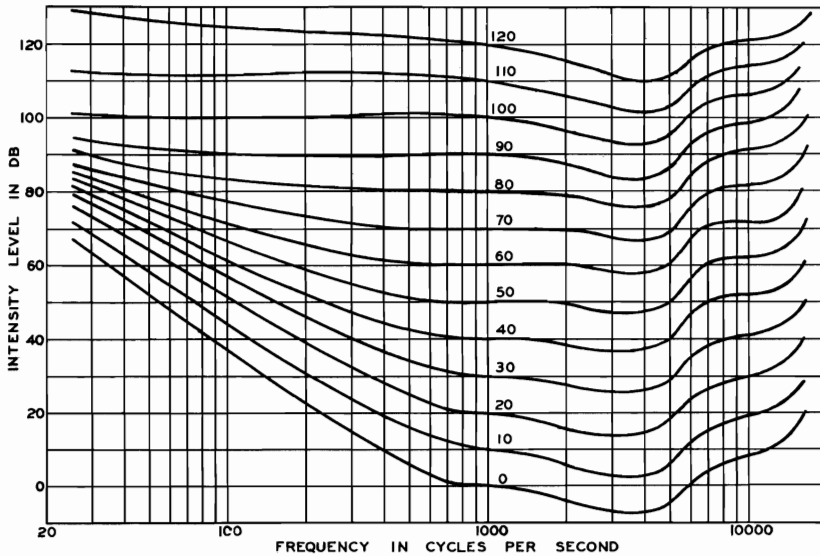
FIG. 12.11. Contour lines of equal loudness for normal ears. Numbers on curves indicate loudness level in phons. 0 db = 16^{-16} watts per square centimeter. 0 db = 0.000204 dyne per square centimeter. (After Fletcher and Munson.)

and of fundamental importance in the design of reproducing systems. For example, the threshold of hearing at 60 cycles is 48 db higher than that at 1000 cycles.

In general, sound is reproduced at a level lower than that of the original sound. To compensate for the difference in frequency balance, due to the lower reproduction level, an acoustically compensated volume control is used to increase the relative low-frequency response as the level is reduced (see Sec. 11.2R).

These characteristics must be considered in the measurement of noise. The response frequency characteristic of the noise meter is adjusted to correspond to the ear characteristics (see Sec. 10.13).

[18] Fletcher and Munson, *Jour. Acous. Soc. Amer.*, Vol. 5, No. 2, p. 82, 1933.

12.7. Loudness.—Loudness of a sound is the magnitude of the auditory sensation produced by the sound. The units on the scale of loudness should agree with common experience in the estimates made upon sensation magnitude. A true loudness scale must be constructed so that when the units are doubled the sensation will be doubled and when the scale is trebled the sensation will be trebled, etc. Units on the scale are called loudness units, abbreviated LU. The loudness level of the reference tone, expressed in phons, is the intensity level of the reference tone (1000 cycles) in decibels. The loudness level of any other sound is determined by adjusting the
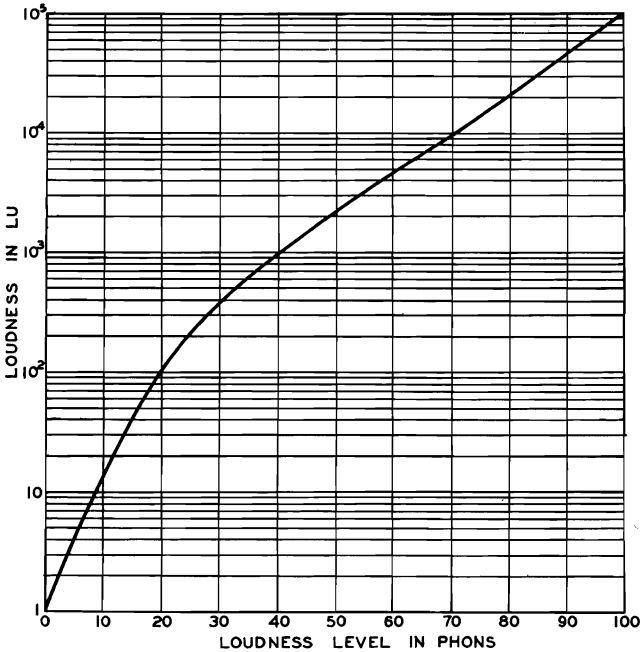
FIG. 12.12. Loudness vs. loudness level. (After Fletcher.)

reference tone until it sounds equally loud. The loudness level, in phons, of a sound is numerically equal to the intensity level, in decibels, of the 1000-cycle pure tone which is judged by the listeners to be equivalent in loudness. The phon is the unit of loudness level as specified in the preceding sentence. A scale[19] showing the relation between loudness level, in phons, and the loudness, in loudness units, is shown in Fig. 12.12. Recently a new unit of loudness, termed the sone, has been adopted. It is defined as follows: a pure tone of 1000 cycles per second, 40 db above the normal threshold of hearing, produces a loudness of one sone. One sone is equivalent to 1000 loudness units.

[19] Fletcher H., *Jour. Acous. Soc. Amer.*, Vol. 9, No. 4, p. 275, 1938.

The loudness[20,21] of pure tones of various frequencies is shown in Fig. 12.13.   For tones between 800 and 2000 cycles the loudness is the same for the same pressure.   The difference is small up to 8000 cycles.   For higher frequencies than this the loudness decreases as the frequency increases.
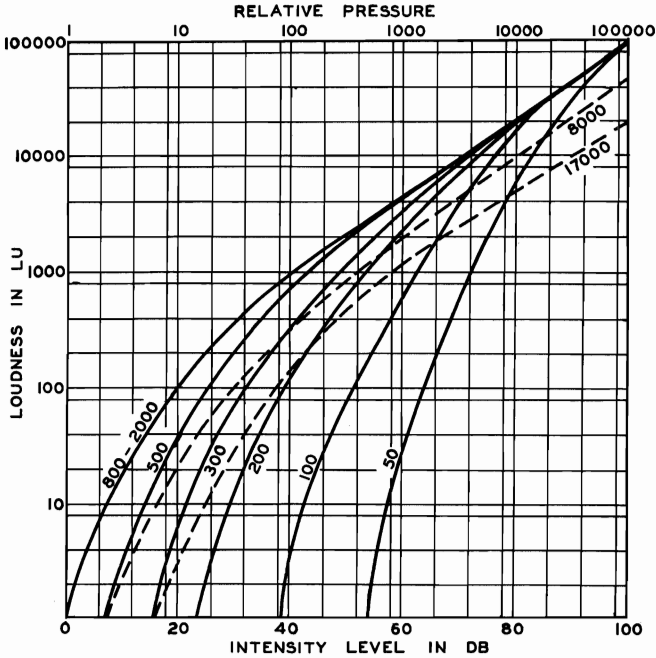
FIG. 12.13.   The relation between the intensity level and the loudness of pure tones of the frequencies indicated.   0 db = 0.000204 dyne per square centimeter.   (After Fletcher.)

Further, it will be seen that for a 50-cycle tone the intensity required to reach the threshold of hearing is 250,000 times that required for a reference 1000-cycle tone.

12.8. Pitch.—Frequency of a sound wave is the number of cycles per second executed by the particles of the medium in which a sound is being propagated.   Pitch is that subjective quality of sound which determines its position in a musical scale.   Pitch may be measured as the frequency of a pure tone having a specified sound pressure which seems to the average ear to occupy the same position in a musical scale.   Thus it will be seen that there is definite distinction between frequency and pitch.   For example, a tone of a fixed frequency of a few hundred cycles decreases in pitch as the

---

[20] Munson and Gardner, *Jour. Acous. Soc. Amer.*, Vol. 22, No. 2, p. 177, 1950.
[21] Beranek, Marsholl, Cudworth, and Peterson, *Jour. Acous. Soc. Amer.*, Vol. 23, No. 3, p. 261, 1951.

intensity is increased. The change[22] in pitch with loudness is shown in Fig. 12.14.

The mel is a unit of pitch. By definition, a simple tone of frequency 1000 cycles per second, 40 decibels above a listener's threshold, produces a
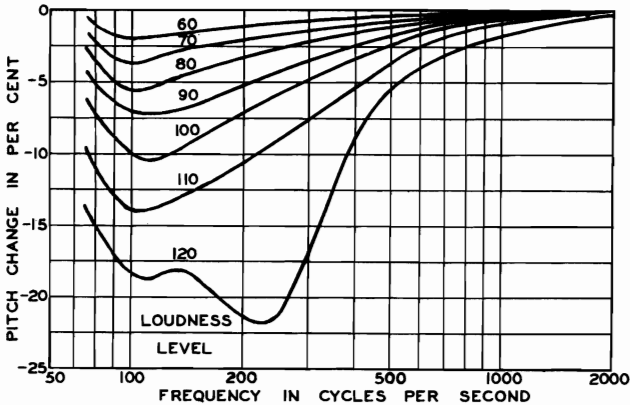
FIG. 12.14. Contours of constant loudness level. Curves show the amount by which the pitch of a pure tone of any frequency is shifted as the tone is raised in loudness level from 40 to the level of the contour. For example, a 100-cycle tone will be changed 10 per cent downward in pitch if raised from a loudness level 40 to a loudness level 100, but a 500-cycle tone will be changed only 2 per cent for the same loudness level increase. (After Snow.)

pitch of 1000 mels. The pitch of any sound that is judged by the listener to be n times that of a 1-mel tone is n mels.

12.9. Masking.[23,24]—The reduction of the ability of a listener to hear one sound in the presence of other sounds is known as masking. In testing the masking properties of a sound, pure tones are generally used as the masked sound. The number of decibels that the threshold level of a pure tone is shifted, due to the presence of noise, is called the masking in decibels at the frequency corresponding to that of the pure tone.

The masking effect of a pure tone, a narrow band of thermal noise and a wide band of thermal noise is shown in Fig. 12.15. The figures on each of the curves show the intensity level of the masking tone or noise. The ordinates, in each of the charts, show the decibels above the threshold that the various frequencies must be raised in order to be just heard in the presence of the particular masking tone or noise. For example, referring to the 4000-cycle tone having an intensity of 93 db, it is only necessary to raise a 2000 cycle tone 20 db to be heard. On the other hand, a 10,000-cycle tone must be raised 66 db to be heard.

[22] Snow, W. B., *Jour. Acous. Soc. Amer.*, Vol. 8, No. 1, p. 14, 1936.

[23] Wegel and Lane, *Phys. Rev.*, Vol. 23, No. 2, p. 266, 1924.

[24] Fletcher and Munson, *Jour. Acous. Soc. Amer.*, Vol. 9, No. 1, p. 1, 1937.
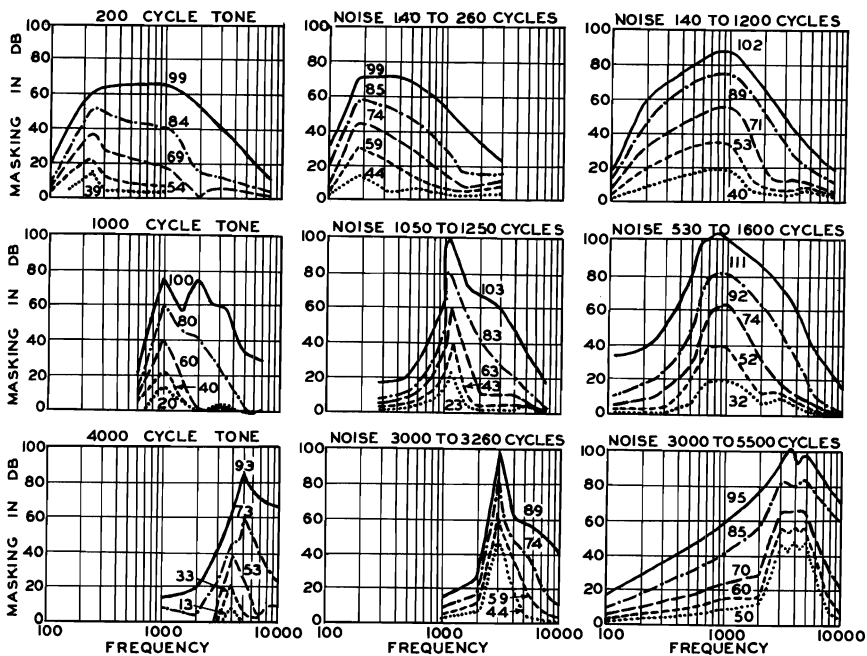
FIG. 12.15. Masking audiograms for single frequency tones, narrow bands of thermal noise, and wide bands of thermal noise. The curves are labeled in db above the threshold. (After Fletcher and Munson.)

12.10. Nonlinearity of the Ear.[25]—When a pure tone of a suitable intensity is impressed upon the ear a series of harmonics or overtones of the original frequency, are heard. Furthermore, when two loud tones are sounded together, a group of tones is heard consisting of the sums and differences of the two primary tones and their harmonics. These phenomena show that the ear is a nonlinear system.

The levels above threshold of the fundamental at which the various harmonics first become detectable,[26,27] are shown in Fig. 12.16. The subjective effects of the harmonics generated in the ear are more pronounced at the lower frequencies. Furthermore, the harmonics appear at a lower level at the lower frequencies.

12.11. Effect of Phase Relations Among the Harmonics.[28,29,30]— The phase of a harmonic affects the threshold of perceptible distortion as well

[25] Stevens and Davis, "Hearing," John Wiley and Sons, New York, N.Y., 1939.

[26] Wegel and Lane, *Phys. Rev.*, Vol. 23, No. 2, p. 266, 1924.

[27] Fletcher, "Speech and Hearing," D. Van Nostrand Company, Princeton, N.J., 1929.

[28] Chapin and Firestone, *Jour. Acous. Soc. Amer.*, Vol. 5, No. 3, p. 173, 1934.

[29] Lewis and Larsen, *Nat. Acad. Sci.*, Vol. 23, p. 415, 1937.

[30] Stevens and Davis, "Hearing," John Wiley and Sons, New York, N.Y., 1938.
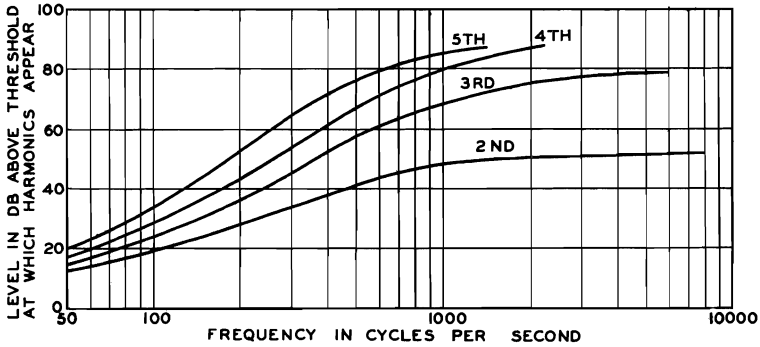
FIG. 12.16. The level above threshold at which harmonics are generated in the ear at the various frequencies. (After Wegel and Lane.)

as the quality of a complex sound. This statement contradicts the so-called Ohm's Auditory Law: that the ear tends to analyze the compounds of a complex sound regardless of the phase relations. There is a definite phase relation which will produce the greatest loudness and another which will produce the least loudness. For example, a harmonic in the actuating sound may reinforce or cancel an aural harmonic.

12.12. Modulation (Vibrato).[31,32]—Amplitude, phase, or frequency modulation refers to a change in the amplitude, phase, or frequency of a sound wave. Suppose that an oscillator is connected to a loudspeaker, Fig. 10.8. If the volume control is varied, the result is amplitude modulation. If the frequency control is varied, the result is frequency modulation. If the position of the loudspeaker is varied with respect to the observation point, the result is phase modulation.

The vibrato is used as an artistic embellishment by singers. It is an example of frequency modulation. The average rate of the vibrato is 7 cycles per second.

When two tones of nearly the same frequency are sounded together, they produce beats at a rate equal to the frequency difference between them. In the case of very slow beats the intensity seems to rise and fall continuously. Faster beats appear as intermittent impulses.

12.13. Minimum Perceptible Differences.—The minimum perceptible difference in frequency[33] is of interest in any type of sound reproduction where a change or fluxation in the frequency may occur, as, for example, phonographs and sound motion pictures (see Sec. 10.6). The minimum perceptible change in frequency as a function of the sensation level is shown in Fig. 12.17. It will be seen that the ear is most sensitive to frequency changes at the higher frequencies.

[31] Seashore, C. E., "Psychology of Music," p. 33, McGraw-Hill Book Company, New York, N.Y., 1938.
[32] Stevens and Davis, "Hearing," John Wiley and Sons, New York, N.Y., 1938.
[33] Shower and Biddulph, *Jour. Acous. Soc. Amer.*, Vol. 3, No. 2, Part I, p. 275, 1931.
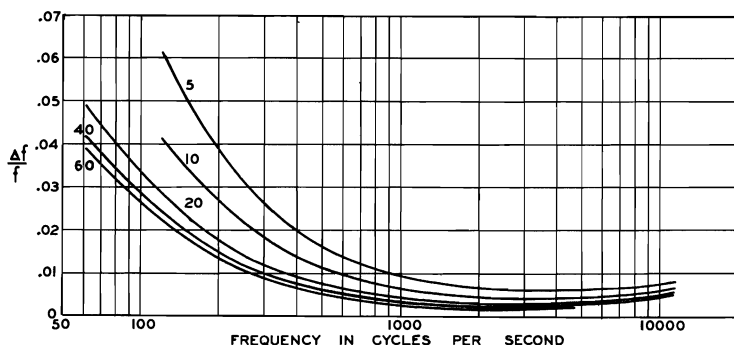
FIG. 12.17. The variation of $\Delta f/f$ with frequency for various sensation levels. Δf is the change in frequency. Sensation level is level above threshold. (After Shower and Biddulph.)
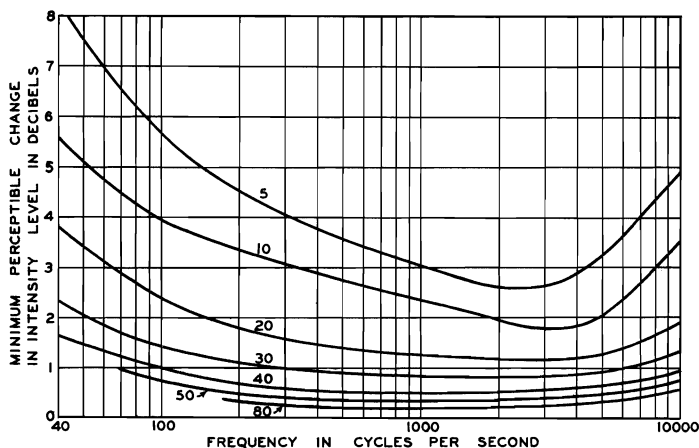
FIG. 12.18. The minimum perceptible change in intensity level of pure tones as a function of the frequency for various sensation levels. Sensation level is level above threshold. (After Fletcher, "Speech and Hearing," D. Van Nostrand Company.)

The minimum time required for a pure tone to excite the ear in order to be sensed as a pure tone is of interest in sound reproduction from the standpoint of the transient response and hangover. It appears that the time is independent of the frequency and is about one twentieth of a second.

The minimum perceptible change in intensity[34] which the ear can detect is of interest in certain types of sound reproduction where the level may change as, for example, fluxations in the voltage gain of an amplifier. The minimum perceptible change in intensity level of pure tones as a function

[34] Fletcher, "Speech and Hearing," D. Van Nostrand Company, Princeton, N.J., 1929.

of the frequency for various sensation levels is shown in Fig. 12.18. These characteristics show that the ear is most sensitive to intensity level changes at the higher sensation levels.

12.14. Timbre (Tone quality).[35,36,37]—In general, it is said that the three characteristics which describe a tone are loudness, pitch, and timbre or quality. These quantities are not sufficient to describe a tone. Three more are required, as follows: vibrato, duration, and growth and decay. Loudness, pitch, vibrato, duration, and growth and decay are defined in other sections. It is the purpose of this section to describe timbre.

Timbre is that characteristic of a tone which depends upon its harmonic structure as modified by the other physical factors that describe a tone. The harmonic structure of a tone is expressed in the number, intensity, distribution, and phase relations of its components. Timbre, then, may be said to be the instantaneous cross section of the tone. It ranges from a pure tone through an infinite number of variations in complexity up to a pitchless sound such as thermal noise. Work has been carried out on the subjective measurements of timbre. However, the subject of timbre is more complex than that of loudness and pitch, because it is an interrelated function of the intensity, pitch, duration, growth, and decay.

12.15. Duration.—The duration of a note in music is indicated by the kind of a note, as, for example, a whole, a half, a quarter, or an eighth note. It is quite evident that the duration of a tone influences the aspect as perceived by the ear. A fine musical ear may detect a difference in the length of two tones as small as 0.01 second. The duration is one of the important means that the artist has for the interpretation of music.

12.16. Growth and Decay.[38]—In the case of a certain instrument the instantaneous cross section of the tone may be exactly similar to that of another instrument but to the ear the sound appears entirely different. The difference is due to the growth and decay of tone. In the case of an organ pipe, time is required for the tone to build up and die down. In the piano the build up time is very fast and decay time is very long when the key is depressed. It is quite evident that growth and decay play an important part in the nature of musical tones.

There is a special case of growth and decay, namely, portamento. Portamento is the passage from a tone of one frequency to a tone of a different frequency in one continuous glide through all the intervening frequencies.

12.17. Auditory Localization.[39,40,41]—The human hearing mechanism

[35] Seashore, "Psychology of Music," McGraw-Hill Book Company, New York, N.Y., 1938.

[36] Fletcher, H., *Amer. Jour. Phys.*, Vol. 14, No. 4, p. 215, 1946.

[37] Olson, "Musical Engineering," McGraw-Hill Book Company, New York, N.Y., 1952.

[38] Olson, "Musical Engineering," McGraw-Hill Book Company, New York, N.Y., 1952.

[39] Stewart, G. W., *Phys. Rev.*, Vol. 15, No. 5, p. 425, 1920.

[40] Stevens and Davis, "Hearing," John Wiley and Sons, New York, N.Y., 1938.

[41] Steinberg and Snow, *Bell Syst. Tech. Jour.*, Vol. 13, No. 2, p. 245, 1934.

can localize sounds with great accuracy. This property is due to two effects, namely, the difference in intensity and the difference in phase between the sound at the two ears. The difference in phase between the sounds at the two ears is due to the difference in time arrival at the two ears. The difference in intensity at the two ears is due to diffraction. The pressure at the two ears may be obtained from Fig. 1.5 by assuming the head to be a rigid sphere. This assumption has been verified experimentally.

The binaural phase effect is confined principally to the lower frequencies, namely, below 1000 cycles. The binaural phase effect has been utilized for the location of airplanes, submarines, etc.

If attachments can be made to the ears which will virtually separate them further (for example, spread the microphones on the dummy of Fig. 13.7), then a small rotation of the apparatus will mean a larger difference in phase at the receivers than at the unaided ear. Of course, there may be ambiguity as to whether the sound comes from in front or behind, but this does not vitiate the method.

The difference in intensity at the two ears due to diffraction is very small, below 1000 cycles. However, at the higher frequencies the difference in intensity may be 20 to 30 db.

The human hearing mechanism is also a directional collecting system. Using the system of Fig. 13.7, the reverberation in a room appears to be normal. However, if only one microphone is used, the apparent reverberation will be increased. This indicates that the human hearing mechanism is very directional and discriminates against reverberation and other undesirable sounds. For this reason it has been found necessary to use either very small pickup distances or directional collecting systems in the monaural collection of sound.

12.18. Delay Effect.[42]—It has been established that if there are several separated sources of sound, identical in content and amplitude but displaced with respect to time, that the sound will appear to come from the source which leads the other in time. The experiment which illustrates this phenomena is shown in Fig. 12.19. The same signal is reproduced from loudspeakers 1 and 2. The signal from loudspeaker 1 can be delayed by means of the delay system. For each value of delay, the ratio of the voltage input to the two loudspeakers is varied until it is impossible to distinguish which loudspeaker appears to be the source. The results of this test are shown by the graph of Fig. 12.19. This experiment shows that there can be considerable unbalance before the sound ceases to appear to come from undelayed source. With the same intensity of sound emanating from both loudspeakers, the source always appears to be the undelayed loudspeaker. The delay phenomena plays an important role in stereophonic systems.

12.19. Hearing Acuity in the United States Population.[43]—Hearing tests at New York and San Francisco World's Fairs in 1939 were conducted

[42] Haas, H., *Akustika*, Vol. 1, No. 2, p. 49, 1951.
[43] Steinberg, Montgomery, and Gardner, *Jour. Acous. Soc. Amer.*, Vol. 12, No. 2, p. 291, 1940.
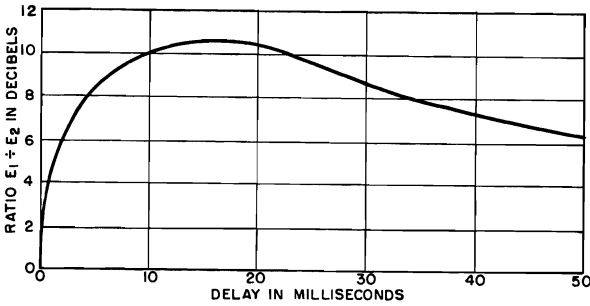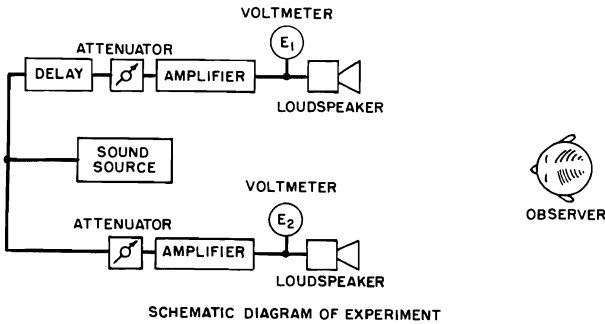
FIG. 12.19.   Arrangement of the apparatus for demonstrating
the delay effect.   The graph shows the ratio of the voltages
E_1 and E_2 as a function of the delay for the conditions when
it is impossible to distinguish which loudspeaker is the source.
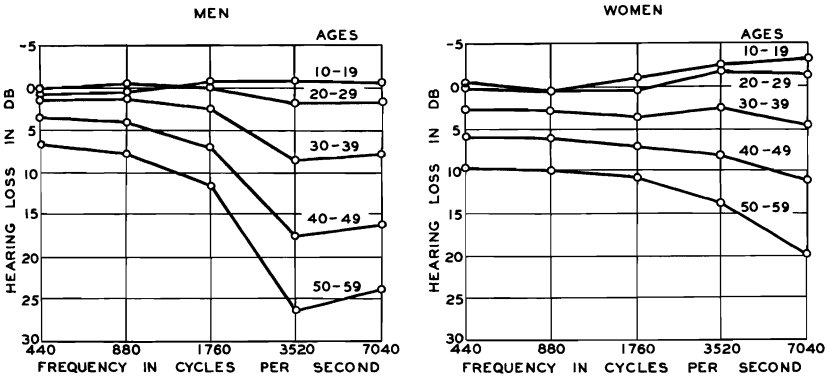
FIG. 12.20.   Hearing loss frequency characteristic of men and women for different age
groups.   (After Steinberg, Montgomery, and Gardner.)

as a part of the Bell System Exhibits. About one-half million records were obtained. The tests were divided into five age groups, 10–19, 20–29, 30–39, 40–49, and 50–59 years. Many cross checks were made with laboratory tests to insure accuracy as, for example, the effect of background noise, calibration of the equipment, estimation of age, etc.

The results of the tests for men and women are shown in Fig. 12.20. It will be seen that the hearing acuity falls off with age, particularly in the high-frequency ranges.

The composite results of these tests have been depicted in another manner in Fig. 12.21. The upper curve is the threshold of feeling level. The lower
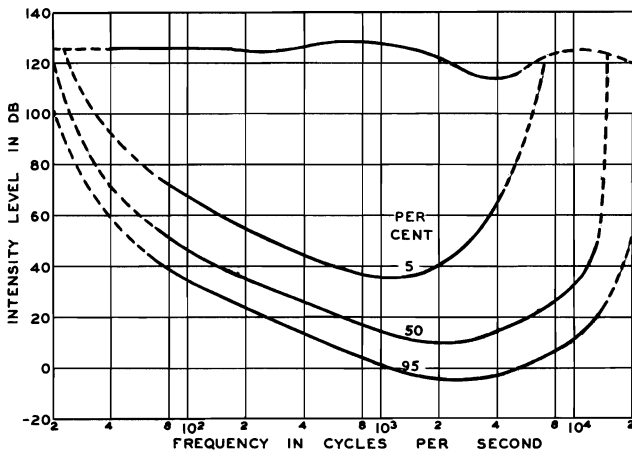
Fig. 12.21. Contours of hearing loss. (After Steinberg, Montgomery, and Gardner.)

curve of Fig. 12.11 is the threshold of hearing level for standard normal hearing. The lower curve labeled 95 indicates that 95 out of 100 persons in a typical group cannot hear pure tones whose frequency and intensity level lie below this curve. The curve labeled 50 indicates that 50 out of 100 persons cannot hear these tones until they exceed the intensity level indicated by this curve. The curve labeled 5 indicates that 5 out of 100 cannot hear these tones until they exceed the intensity levels indicated by this curve.

12.20. The Frequency and Volume Ranges of Speech and Music.— The frequency range of the average normal ear is from 20 to 20,000 cycles. The frequency range of most reproducing channels such as the radio, the telephone, the phonograph, and the sound motion picture is considerably less than that of the hearing range of the ear. It is interesting to note the effect of the frequency range upon the intelligibility of speech and the quality of music.

The frequency[44] and volume ranges of speech and orchestral music are shown in Fig. 12.22. The reproduction of speech with perfect fidelity requires a frequency range of from 100 to 8000 cycles and a volume range of 40 db. The reproduction of orchestral music with perfect fidelity requires a frequency range of from 40 to 14,000 cycles and a volume range of 70 db.
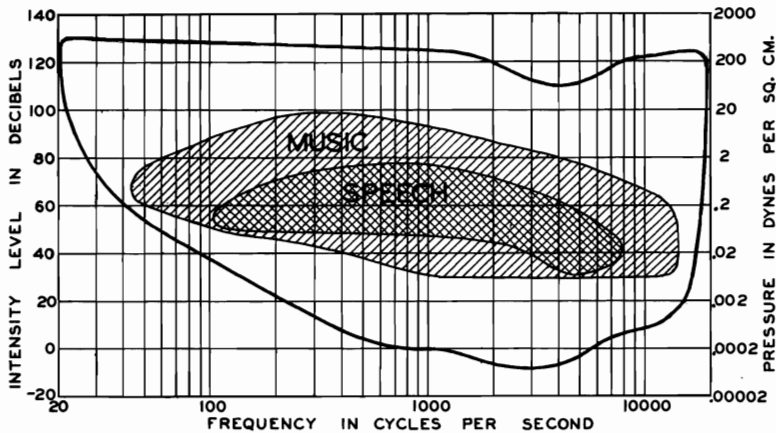
FIG. 12.22. Frequency and volume ranges of speech and music. The solid lines depict the boundaries of normal hearing, that is, the upper and lower limits of intensity and frequency. (From *Bell Laboratories Record*, June, 1934.)

The frequency ranges[45] required for the reproduction of speech, musical instruments, and noises without any noticeable frequency discrimination of distortion are shown in Fig. 12.23.

12.21. The Effect of Frequency Discrimination, Amplitude, Frequency Shift, Reverberation, Nonlinear Distortion and Noise upon the Articulation of Reproduced Speech.[46,47,48]—The recognition of transmitted speech is an important aspect of a sound transmission system.

The sound transmission system may be the air between the mouth and the ear in the open air or in a room, or it may be a sound reproducing system such as a telephone, phonograph, radio, sound motion picture, or television. The method for measuring recognition is to have the speaker read aloud speech sounds, syllables, or words to a listener who writes down what he thinks he hears. A comparison of the sounds, syllables, or words recorded by the listener with those uttered by the speaker shows the fraction that is interpreted correctly. This fraction is termed articulation. See Sec. 10.17.

[44] *Bell Labs. Record*, Vol. 12, No. 6, p. 314, 1934.

[45] Snow, W. B., *Jour. Acous. Soc. Amer.*, Vol. 3, No. 1, Part 1, p. 155, 1931.

[46] Fletcher and Galt, *Jour. Acous. Soc. Amer.*, Vol. 22, No. 2, p. 89, 1950.

[47] Hirsh, Reynolds, and Joseph, *Jour. Acous. Soc. Amer.*, Vol. 26, No. 4, p. 530, 1954.

[48] Fletcher, "Speech and Hearing in Communication," D. Van Nostrand Company, Princeton, N.J., 1955.
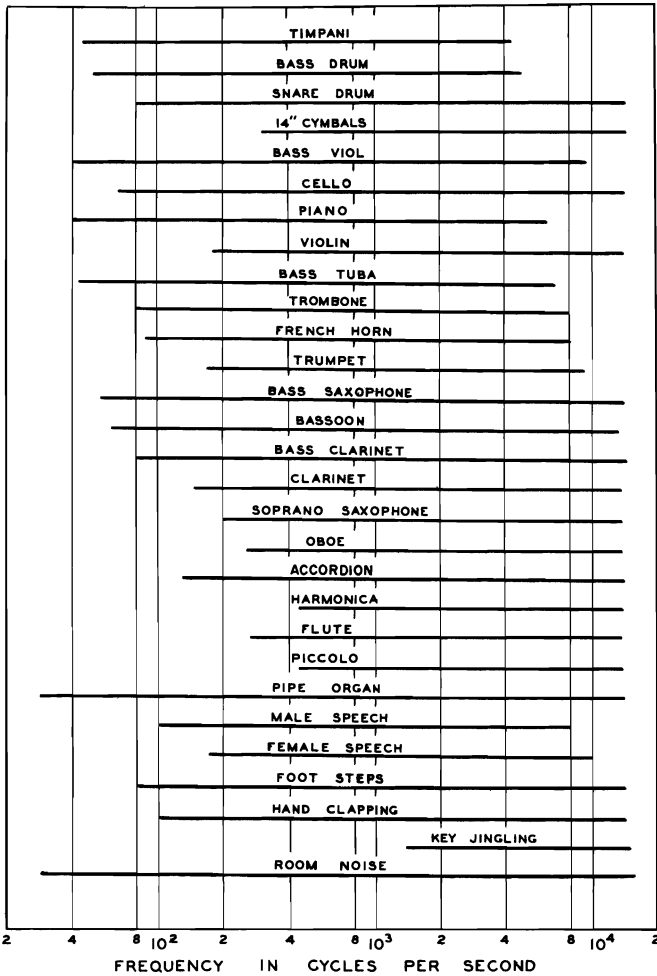
FIG. 12.23. The frequency ranges required for the reproduction of speech, musical instruments, and noises without any noticeable distortion. (After Snow.)

The effect of reducing the high- and low-frequency range upon speech articulation is shown in Fig. 12.24. These data were obtained at a talking level into the system of 70 decibels. The reproduction level was also 70 decibels.

The syllable articulation in a sound reproducing system equipped with low-pass filters with cutoff frequencies at 750, 1000, 1500, 1950, 3700, and 7000 cycles is shown in Fig. 12.25, as a function of the gain in the system. The talking level is an over-all measure of the acoustic level and is defined

as to be the long time average at a distance of one meter directly in front of the speaker's lips in a free field.   The talking level in Fig. 12.25 is 69 decibels above threshold reference level.   The reproduction level of 0 gain corresponds to the talking level of 69 decibels.   It will be seen that the
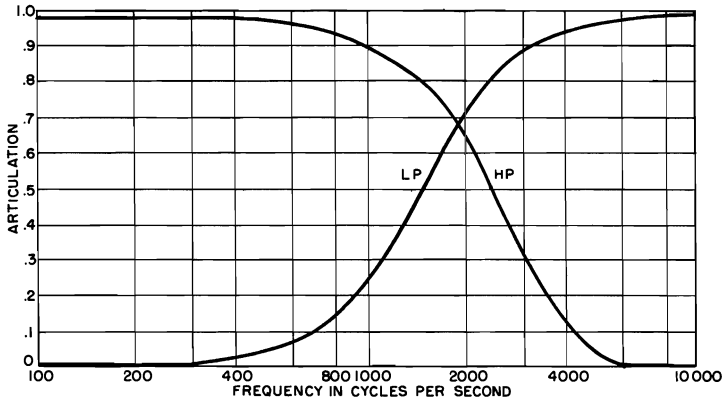
FIG. 12.24.   The effect of the frequency range upon the syllable articulation of speech.   HP High-pass filter—all frequencies below the frequency given by the abscissa removed.   LP Low-pass filter—all frequencies above the frequency given by the abscissa removed.   (After French and Steinberg.)

maximum syllable articulation occurs for a gain of about 15 decibels.   This is a level of 84 decibels above threshold reference level.   The data of Fig. 12.25 are in general agreement with the low-pass characteristic of Fig. 12.24.
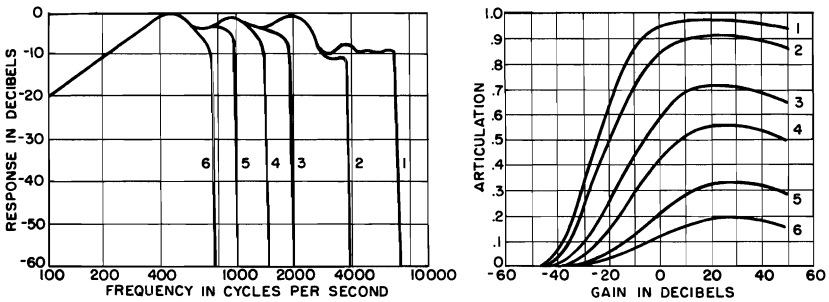
FIG. 12.25.   The syllable articulation as a function of the reproduced sound level for various low-pass transmission characteristics.   (After Fletcher, "Speech and Hearing in Communication," D. Van Nostrand Company.)

The syllable articulation in a sound reproducing system equipped with high-pass filters with cutoff frequencies at 750, 1500, 1900, and 2850 cycles as a function of the gain in the system is shown in Fig. 12.26.   The talking level is 69 decibels above threshold reference level.   The reproduction level

of 0 gain corresponds to the talking level of 69 decibels. It will be seen
that the maximum syllable articulation occurs for a gain of about 20 deci-
bels. This is a level of 89 decibels above the threshold level. The data of
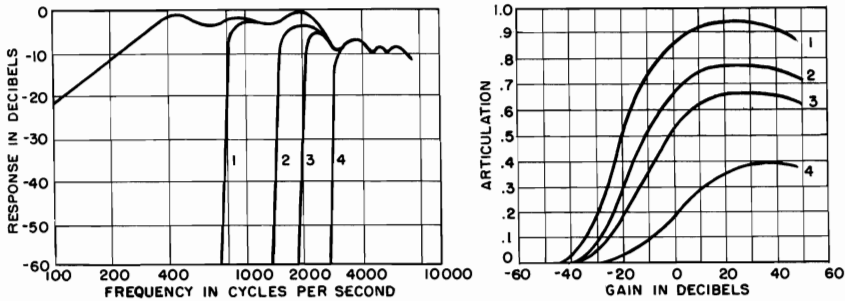
FIG. 12.26. The syllable articulation as a function of the reproduced sound level for
various high-pass transmission characteristics. (After Fletcher, "Speech and Hearing
in Communication," D. Van Nostrand Company.)

Fig. 12.26 are in general agreement with the high-pass characteristic of
Fig. 12.24.

The syllable articulation in a sound reproducing system equipped with a
resonant type of network, which exhibits a response frequency characteristic
with a single major peak, as a function of the gain in the system is shown in
Fig. 12.27. There are five response frequency characteristics with five
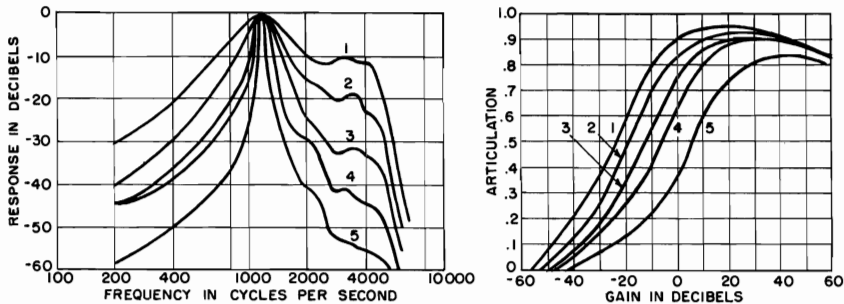
FIG. 12.27. The syllable articulation as a function of the reproduced sound for limited
frequency range transmission characteristics. (After Fletcher, "Speech and Hearing in
Communication," D. Van Nostrand Company.)

different values of damping in the resonant system. The talking level of
Fig. 12.27 is 69 decibels above threshold. The reproduction level of 0 gain
corresponds to the talking level of 69 decibels. It will be seen that a high
order of articulation can be obtained with a response frequency characteristic
of very limited frequency range.

From the foregoing, the indications are that for maximum articulation
there should be an optimum sound level for each band in the frequency range.

For the optimum level of 69 decibels above the threshold level, the data of Fig. 12.28 show the level in decibels of each critical frequency band. It is to be noticed that response level falls off in the low- and high-frequency ranges.

The data of Figs. 12.24, 12.25, and 12.26 indicate that some frequency regions are more important than others in contributing to articulation or
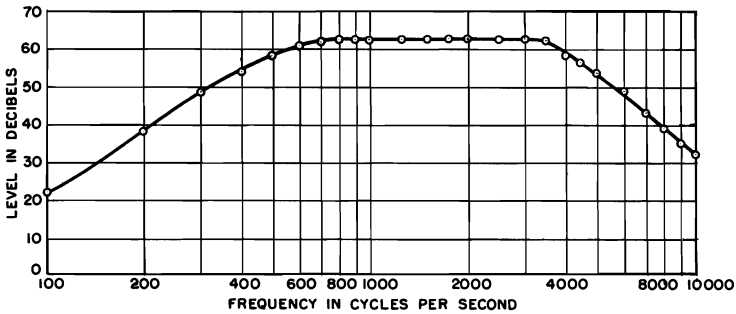
FIG. 12.28. Level of each critical frequency band of undistorted speech above threshold, when speech is reproduced at the optimum level for interpretation. (After Fletcher, "Speech and Hearing in Communication," D. Van Nostrand Company.)

intelligibility of speech. The importance function $D(f)$, that is, the importance for articulation or intelligibility per cycle, is shown in Fig. 12.29. The importance for articulation or intelligibility is the integral $\int_0^f D(f)\,df$ which is depicted in Fig. 12.29.
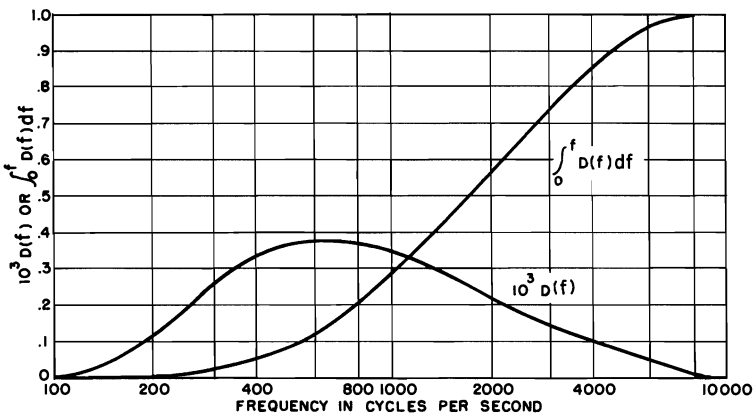
FIG. 12.29. The importance function per cycle, $D(f)$ and the importance integral $\int_0^f D(f)\,df$ of each frequency region for articulation or intelligibility. (After Fletcher, "Speech and Hearing in Communication," D. Van Nostrand Company.)

The preceding considerations of articulation have been concerned with the effects of a nonuniform response frequency characteristic and the level of the reproduced sound.   There are other types of distortion which influence the articulation and intelligibility.   One interesting experiment is that of multiplying all frequencies by the same number.   This experiment can be performed by recording a phonograph record at one turntable speed and reproducing it at a different turntable speed.   The syllable articulation as a function of the ratio of the speed of rotation of a phonograph record in reproducing to the speed of rotation in recording is shown in Fig. 12.30.
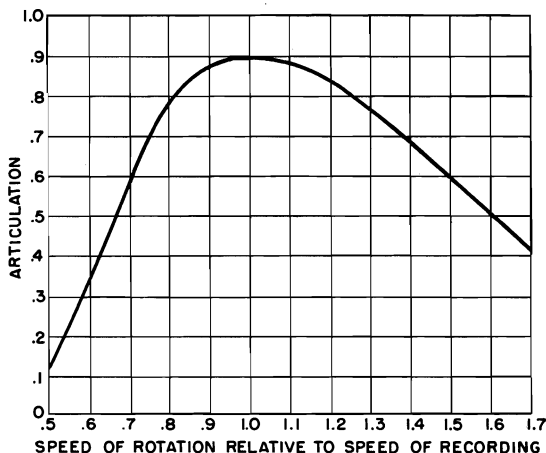
FIG. 12.30.   The syllable articulation of a reproduced disk phonograph record as a function of the ratio of the rotational speed in reproduction to the rotational speed in recording.   (After Fletcher, "Speech and Hearing in Communication," D. Van Nostrand Company.)

The interesting fact is that there can be a large ratio of frequency shift before the articulation is impaired.

The reverberation of a room influences the articulation and intelligibility of speech.   The effect of reverberation upon the articulation is described in Sec. 11.2F and depicted in Fig. 11.9.

Nonlinear distortion is produced when there is a deviation from linearity in the input-output characteristic of the reproducing system.   See Secs. 12.26 and 12.27.   One of the most common examples of nonlinear distortion is the overloading of the amplifier in a sound reproducing system.   The effect[49] of nonlinear distortion produced by the overloading of an amplifier in a sound reproducing system upon articulation is depicted in Fig. 12.31. The syllable articulation characteristic labeled 1 in Fig. 12.31 was obtained on a wide frequency range, high quality feedback amplifier with no over-loading.   The talking level was 70 decibels above threshold reference level.

[49] Pollack, Irwin, *Jour. Acous. Soc. Amer.*, Vol. 24, No. 5, p. 538, 1952.

The reproduction level of 0 decibels corresponds to the talking level of 70 decibels. The syllable articulation characteristic labeled 2 was obtained on the same amplifier but equipped with input and output attenuators so that overload began at 0 decibels. It will be seen that the syllable articulation is highly impaired by the overloaded amplifier. The principal reason being that, in the case of a feedback amplifier, overload produces tremendous nonlinear distortion.

Noise is another factor that would be expected to impair the articulation and the intelligibility of speech. The syllable articulation labeled 1 in Fig. 12.32 was obtained on a wide frequency range, high quality system in
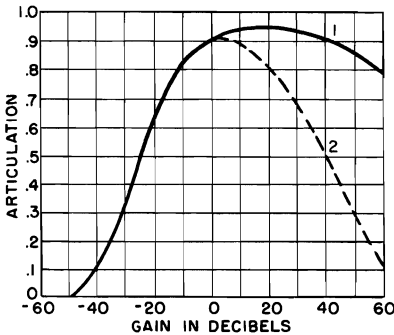
FIG. 12.31. The syllable articulation of a wide frequency range, high-quality sound reproducing system for two conditions of operation as a function of the reproduced sound level for two conditions of operation. 1. No nonlinear distortion. 2. With nonlinear distortion starting at 0 decibels gain.
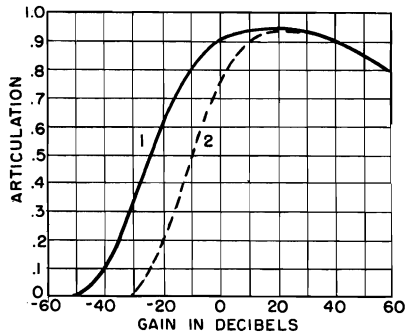
FIG. 12.32. The syllable articulation of a wide range, high-quality sound reproducing system as a function of the reproduced sound level for two conditions of operation. 1. No noise. 2. With residence room noise of 43 decibels.

the absence of noise. The talking level is 70 decibels above threshold level. The reproduction level of 0 gain corresponds to the talking level of 70 decibels. The measured noise level was very near the threshold of hearing, when room noise having the spectrum distribution shown in Fig. 12.38 for a residence was introduced. This corresponds to a total noise level of 43 decibels. It will be seen that the articulation is reduced by a large factor at the lower levels of reproduction. However, as would be expected, at the higher levels of reproduction the articulation is not reduced.

Another form of distortion which may be applied to speech is clipping.[50] It is interesting to note that when the level of clipped speech is increased the articulation approaches that of unclipped speech.

From the foregoing data it will be seen that a reasonably good articulation and intelligibility of speech is obtained under conditions of severe interference and distortion, as for example, a narrow frequency band, in the

[50] Martin, D. W., *Jour. Acous. Soc. Amer.*, Vol. 22, No. 5, p. 614, 1950.

presence of noise and large nonlinear distortion. However, the quality of the reproduced speech is very much impaired by these distortions. On the other hand the intelligibility of a limited frequency range may be actually superior to a wider frequency range due to the introduction of additional noises and distortions in a wider frequency 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.

12.22. The Effect of Frequency Discrimination upon the Quality of Reproduced Music.—The effect of the frequency range[51] upon the quality of reproduction of orchestral music is shown in Fig. 12.33. It will be seen that the frequency range required for no appreciable loss in quality is
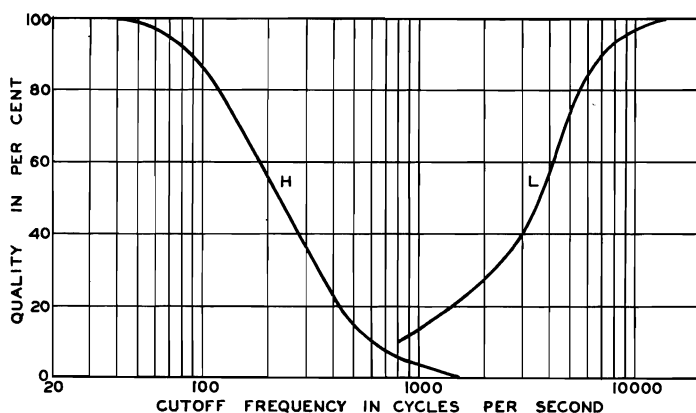
FIG. 12.33. The effect of the frequency range upon the quality of orchestra music. H. High-pass filter—all frequencies below the frequency given by the abscissa removed. L. Low-pass filter—all frequencies above the frequency given by the abscissa removed. (After Snow.)

from 40 to 14,000 cycles. A good radio transmitter and receiver in the broadcast band will cover a frequency range of from 40 to 8000 cycles and a volume range of 50 to 60 db. These frequency and volume ranges will cover the speech area of Fig. 12.22, but will not cover the orchestral music area. For the latter case some form of volume compression must be used.

12.23. Absolute Amplitudes and Spectra of Speech, Musical Instruments and Orchestras.[52,53,54,55]—The average and peak outputs of speech and musical instruments are of importance in the design of all types of

[51] Snow, W. B., *Jour. Acous. Soc. Amer.*, Vol. 3, No. 1, Part 1, p. 155, 1931.
[52] Sivian, Dunn, and White, *Jour. Acous. Soc. Amer.*, Vol. 2, No. 3, p. 330, 1931.
[53] Sivian, L. J., *Jour. Acous. Soc. Amer.*, Vol. 1, No. 2, Part 2, p. 1, 1930.
[54] Hathaway, J. L., *Electronics*, Vol. 12, No. 11, p. 29, 1939.
[55] Drew and Kellogg, *Jour. Acous. Soc. Amer.*, Vol. 12, No. 1, p. 95, 1940.
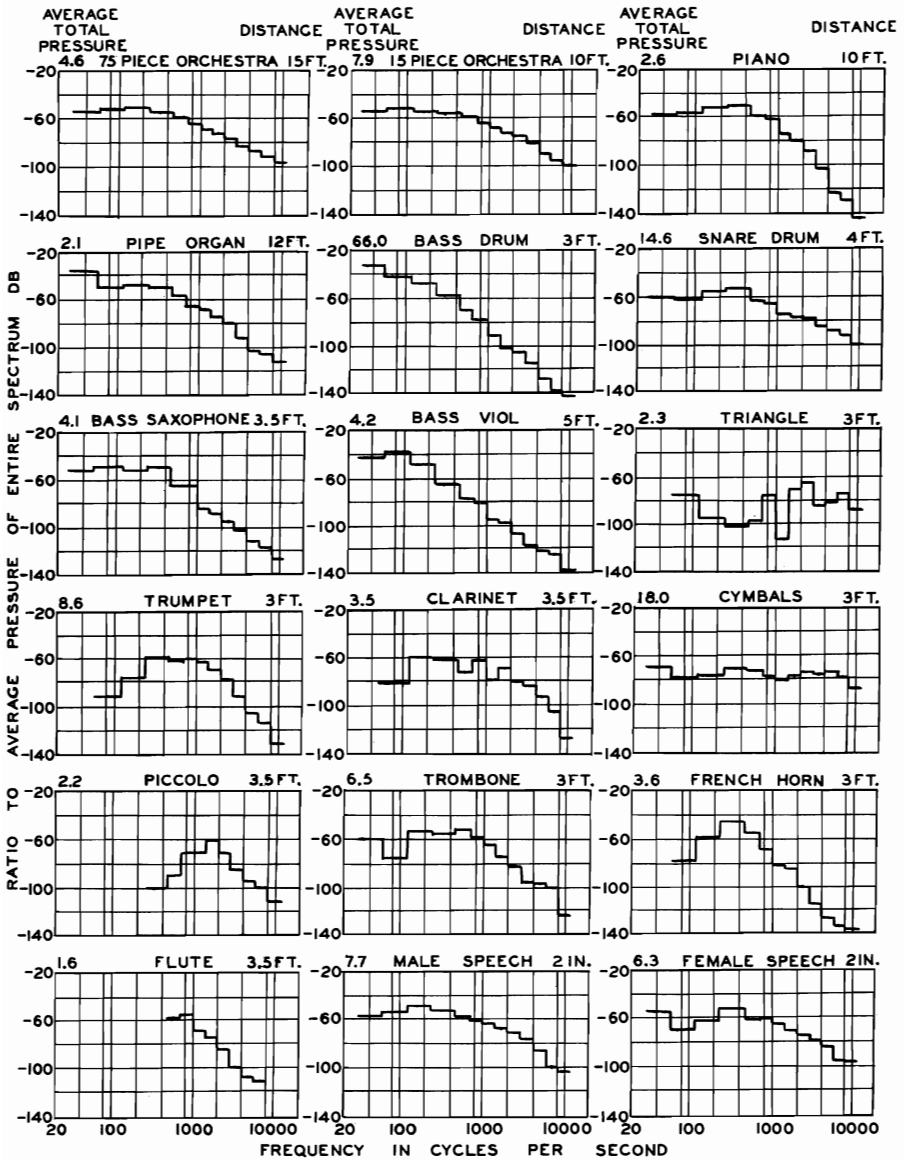
FIG. 12.34. Ratio of the average pressure per cycle to the average total pressure of the entire spectrum for speech, various musical instruments, and orchestras. The distance and average total pressure, in dynes per square centimeter, are shown above each graph. (After Sivian, Dunn, and White.)
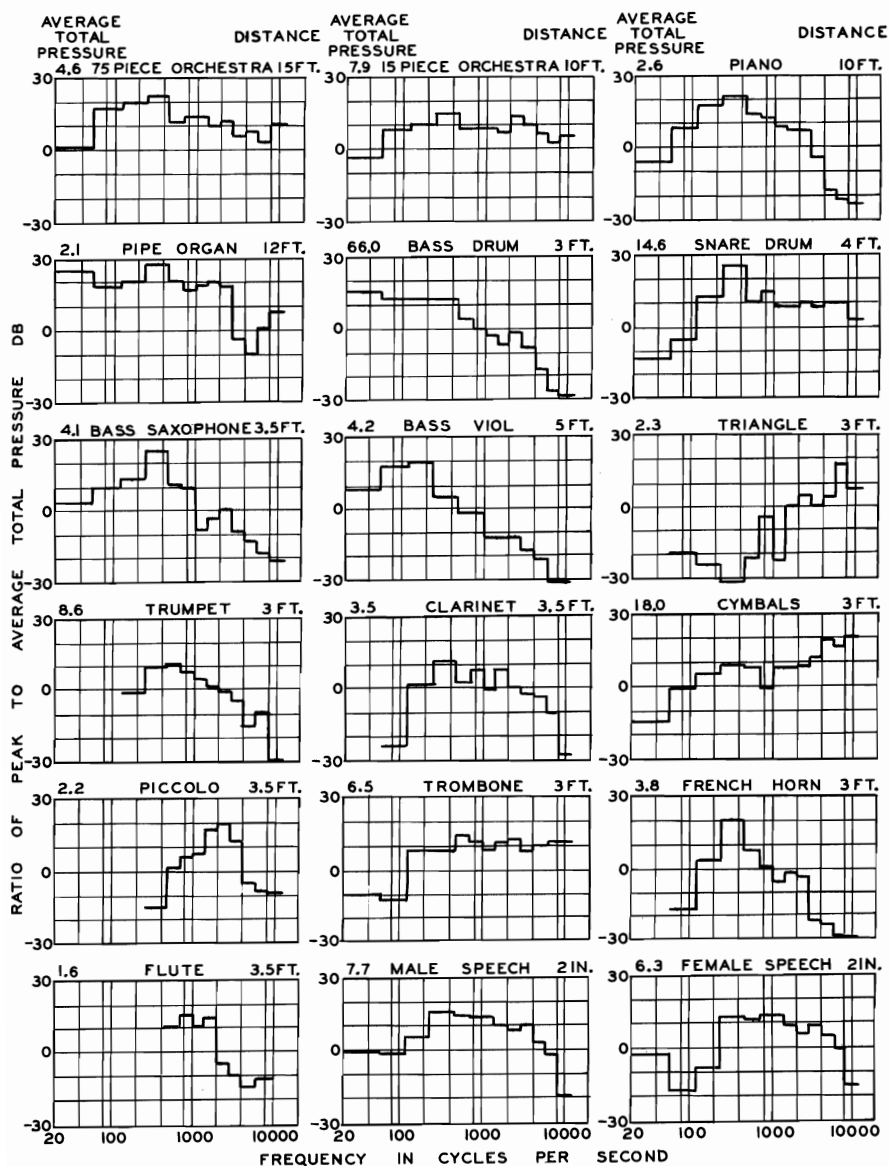
FIG. 12.35. Ratio of the peak pressure to the average total pressure of the entire spectrum for speech, various muscial instruments, and orchestras. The distance and average total pressure, in dynes per square centimeter, are shown above each graph. (After Sivian, Dunn, and White.)

reproducing equipment. For example, the average power output involves such factors as the heating of the voice coil, the heating of audio-power transformers, etc., while the peak power output fixes the overload point of the system.

The ratio of the average sound pressure per cycle to the average total pressure of the entire spectrum for speech, various musical instruments,
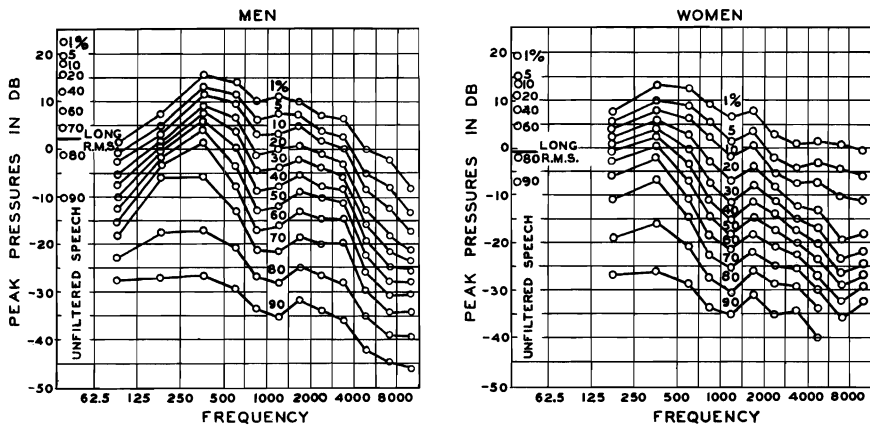
FIG. 12.36. Peak pressures in one-eighth-second intervals of conversational speech, at 30 centimeters from the mouth; composite from the voices of 6 men and 5 women. Measurements were made in the bands indicated by divisions of the frequency scale, and the percentages are those of intervals having peak pressures greater than the indicated ordinates. At the left in the graphs, peak measurements on speech as a whole are given. The rms pressure over a long-time interval is also given. 0 db = 1 dyne per square centimeter. (After Dunn and White.)
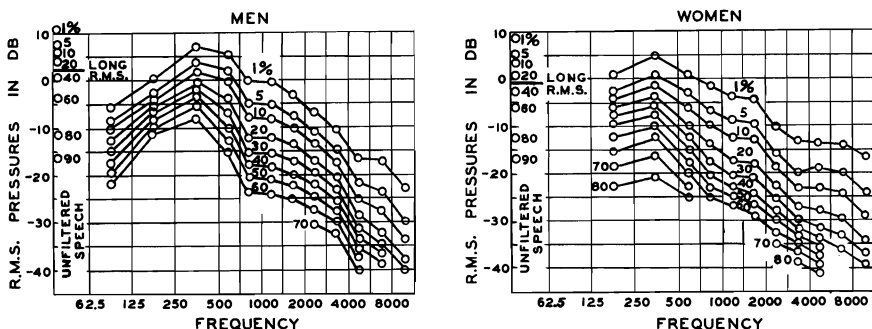
FIG. 12.37. Rms pressures in one-eighth intervals of conversational speech, at 30 centimeters from the mouth; composite of 6 men and 5 women. Measurements were made in the bands indicated by the divisions of the frequency scale, and the percentages are those of intervals having peak pressures greater than the indicated ordinates. At the left in the graphs, rms measurements on speech as a whole are given. The rms pressure over a long-time interval is also given. 0 db = 1 dyne per square centimeter. (After Dunn and White.)

and orchestras is shown in Fig. 12.34. The ratio of the peak pressure to the average pressure of the entire spectrum for speech, various musical instruments, and orchestras is shown in Fig. 12.35. The peak or total power output can be computed from the pressure and the distances following the procedure is outlined in Sec. 10.3D.

The peak sound pressures[56] in speech for various frequency bands at a distance of 30 centimeters are shown in Fig. 12.36. The percentages on the graphs are those of intervals having peak pressures greater than the indicated ordinates. For example, the characteristic labeled 20 per cent means that 20 per cent of the peaks in the particular frequency interval exceeded the ordinate value of the characteristic. At the left of the graphs, peak measurements of the entire frequency band are given.

The rms sound pressures in speech for various frequency bands at a distance of 30 centimeters are shown in Fig. 12.37.

12.24. Noise in Reproducing Systems.—Static, hiss, scratch, surface noise, and hum are terms which have been introduced to describe various kinds of noises produced by reproducing systems. A few of the types of noises in reproducing systems will be discussed in this section.

In radio broadcasting systems random noise is produced by thermal agitation[57] in conductors, and the small shot effect[58] and fluctuating noise in vacuum tubes are atmospherics and man-made interference. The energy produced by thermal noise and the small shot effect is proportional to the width of the frequency band. Of course, the effects of atmospherics and man-made static can be reduced by increased power. Accentuating the high-frequency response in transmission and introducing complementary equalization in the receiver will reduce the noise, since the noise is proportional to the band width. For the same reason, frequency modulation also reduces noise in radio reproducing systems.

Hum[59] is another source of noise in radio, phonograph, and sound motion picture reproducing systems because practically all of these systems are operated from the alternating current mains. Hum is due to inadequate filtering in the high-voltage supply and inductive and capacitive coupling between the power source and some part of the audio system.

Surface noise or record scratch in phonograph records is due to dirt or foreign particles in the groove and to a granular characteristic of the record material. The record noise generated in a pickup which produces constant output for constant velocity is proportional to the frequency for a narrow frequency band of constant width. It is for this reason that scratch is such a troublesome problem at the high frequencies in record reproduction.

Surface noise in film motion picture reproduction is caused by the modulation of the light falling on the photocell, by dirt, scratches, and the natural grain of the film. In the case of film the noise is proportional to the

[56] Dunn and White, *Jour. Acous. Soc. Amer.*, Vol. 11, No. 3, p. 278, 1940.
[57] Johnson, J. B., *Phys. Rev.*, Vol. 32, No. 1, p. 97, 1928.
[58] Schottky, W., *Ann. d. Phys.*, Vol. 57, p. 541, 1918.
[59] Standards on Radio Receivers, Institute of Radio Engineers, 1938.

frequency band width. Noise reduction systems increase the ratio of signal to noise in film reproduction by reducing the light to the least possible for the instantaneous modulation.

Superimposed on the average atmospheric pressure are fluctuations caused by thermal velocities of air molecules.[60] This noise places a lower limit upon the air as a transmitting medium. The ear exhibits the greatest sensitivity between 1000 and 6000 cycles. The rms sound pressure due to thermal noise in this frequency range is about .00005 dyne per square centimeter. The maximum threshold sensitivity of the ear from Fig. 12.11 is .00008 dyne per square centimeter. In very sensitive ears the threshold at 1000 cycles may be .0002 to .0001 dyne per square centimeter—that is, of the same order of magnitude as thermal noise. For exceptionally good ears a further increase in physiological sensitivity would be useless in the presence of thermal noise. It is interesting to note in passing that the thermal noise of the air molecules acting upon the diaphragm of a very sensitive microphone may be of the same order as the thermal noise generated in the electrical system (see Sec. 8.15).

The noises referred to above occur in the reproducing system. The room noise in the studio, home, office, factory, and theater also influences the reproduction of sound. Room noise and the reproduction of sound will be considered in the next section.

12.25. Room Noise and the Reproduction of Sound.[61]—The hearing curves of Figs. 12.11 and 12.21 set the limits for an ideal transmission system with the listener in a quiet place. The ideal of no noise is seldom realized by listeners. Therefore, the lower limit is determined by the ambient room noise. It is the purpose of this section to show the effect of room noise upon the reproduction of sound.

The average noise spectrum[62] may be obtained from the measurements of room noise. In general, the noise spectrum is the same for all types of rooms. From Fig. 11.20, the total noise level for an average living room is 43 decibels. The spectrum for average room noise having a total level of 43 decibels is shown in Fig. 12.38. The ordinates, depicting the spectrum level, are given by

$$B = 10 \log \frac{I}{WI_0} \qquad\qquad 12.1$$

where I = sound intensity in a frequency band width W, and

I_0 = zero reference level of 10^{-16} watts per square centimeter.

In the case of thermal noise or wide-band random noise it is possible to obtain the masking[63] from the spectrum level. The masking contours for wide-band sounds are shown in Fig. 12.39. These masking contours have been drawn after a careful consideration of all data on the subject of masking.

[60] Sivian and White, *Jour. Acous. Soc. Amer.*, Vol. 4, No. 4, p. 288, 1933.
[61] Fletcher, H., *Proc. IRE*, Vol. 30, No. 6, p. 266, 1942.
[62] Hoth, D. F., *Jour. Acous. Soc. Amer.*, Vol. 12, No. 4, p. 499, 1941.
[63] Fletcher and Munson, *Jour. Acous. Soc. Amer.*, Vol. 9, No. 1, p. 1, 1937.

M is the difference between the threshold of hearing in the presence of noise and a quiet place. B is the spectrum level defined by equation 12.1. The masking curve can be deduced from the intensity level per cycle curve, termed the spectrum level curve, Fig. 12.38, the threshold curve, Fig. 12.11,
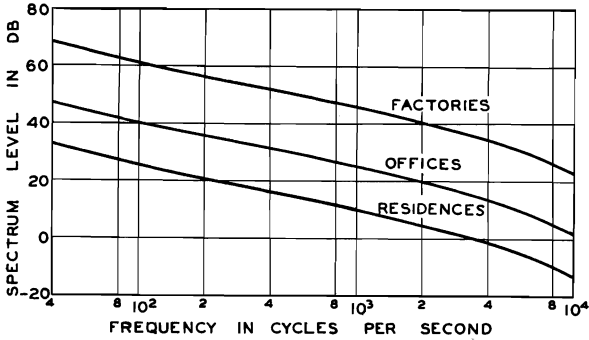
FIG. 12.38. Average noise spectrum for residences, offices, and factories. 0 db = 0.000204 dyne per square centimeter. (After Fletcher and Noth.)
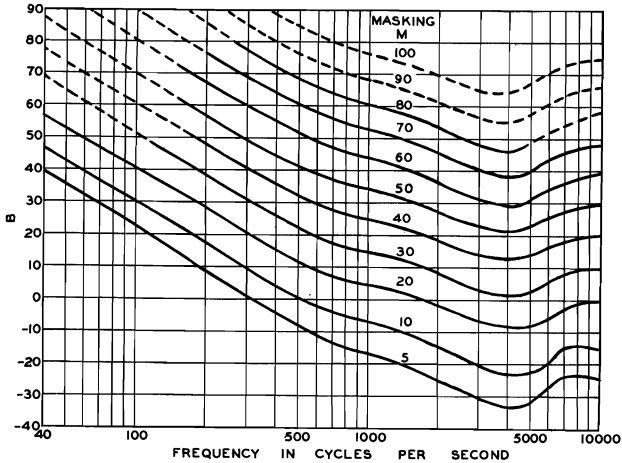
FIG. 12.39. Masking contours for wide-band noise. M is the difference between the threshold of hearing in the presence of noise and a quiet place. B is the spectrum level. (After Fletcher and Munson.)

and the masking contours of Fig. 12.39. For example, for a spectrum level of 9 db at 1000 cycles, the masking level is 25 db. The masking curve for average room noise obtained by these means is shown in Fig. 12.40. The curves for an average business office and an average factory are also shown in Fig. 12.40.

From the masking curve of Fig. 12.40 it is possible to determine the permissible hum and noise level in a radio receiver or phonograph used in the home. For example, the 60-cycle hum must be kept below a 57-db level. The 120-cycle component must be kept below a 41-db level.
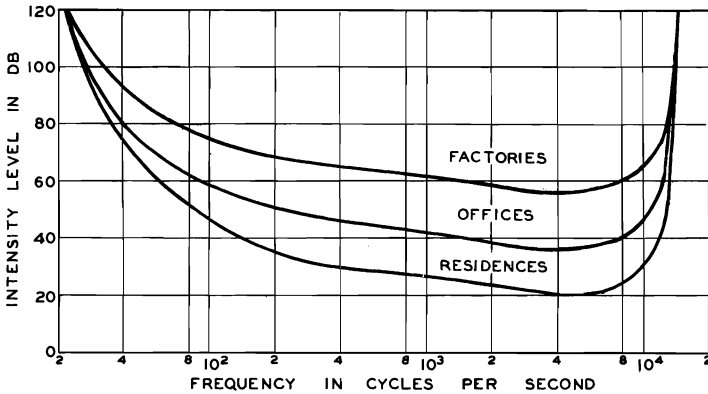
FIG. 12.40. Hearing limits for pure tones. The above characteristics are for a typical listener in typical residence, office, and factory noise. (After Fletcher.)

The masking curve also shows that the volume range of the reproduced sound is reduced by the ambient noise. For example, in the case of factory noise the volume range is quite limited. Masking by the ambient noise is one of the limitations for high-quality reproduction in noisy locations.

12.26. Combination Tones and Nonlinear Transducers.—In most of the discussions in this book the elastic restoring force of the elements of a vibrating system have been considered to be proportional to the first power of the displacement. If a second power term is included the element is asymmetrical, the restoring force being different in magnitude for positive and negative displacements. According to most investigators the structure of the ear is of such an asymmetrical character.

Assume that the displacement of the nonlinear element may be expressed as follows

$$y = ap + bp^2 \qquad\qquad 12.2$$

where p = the actuating force,

a = a constant, and

b = a constant.

Suppose two harmonic forces $p_1 = p_0 \cos \omega_1 t$ and $p_2 = p_0 \cos \omega_2 t$ are impressed upon the system. The total force on the nonlinear element is

$$p = p_1 + p_2 = p_0 \cos \omega_1 t + p_0 \cos \omega_2 t \qquad\qquad 12.3$$

Substituting equation 12.3 in 12.2 the resulting equation may be put in the form

$$y = ap_0 \cos \omega_1 t + ap_0 \cos \omega_2 t + \tfrac{1}{2}bp_0^2 \cos 2\omega_1 t + \tfrac{1}{2}bp_0^2 \cos 2\omega_2 t$$
$$+ bp_0^2 \cos (\omega_1 + \omega_2)t + bp_0^2 \cos (\omega_1 - \omega_2)t + bp_0^2 \qquad 12.4$$

If the element under consideration is the ear, it will be seen that six different frequencies will be heard as follows: the first primary frequency $\omega_1/2\pi$, the second primary frequency $\omega_2/2\pi$, the second harmonic of the first primary frequency ω_1/π, the second harmonic of the second primary frequency ω_2/π, the summation frequency $(\omega_1 + \omega_2)/2\pi$, the difference frequency $(\omega_1 - \omega_2)/2\pi$. The last term bp^2 represents a steady force and produces no sound.

Combination or sum and difference tones may be produced in any nonlinear system as, for example, an intense sound in the air, in the throat of a horn, by overloaded vacuum tube amplifiers, by diaphragms, and by suspension systems.

12.27. Effect of Nonlinear Distortion upon the Quality of Reproduced Speech and Music.—In an ideal reproducing system the elements are invariant with respect to the time. However, in practical systems the elements are nonlinear. These elements introduce nonlinear distortion.
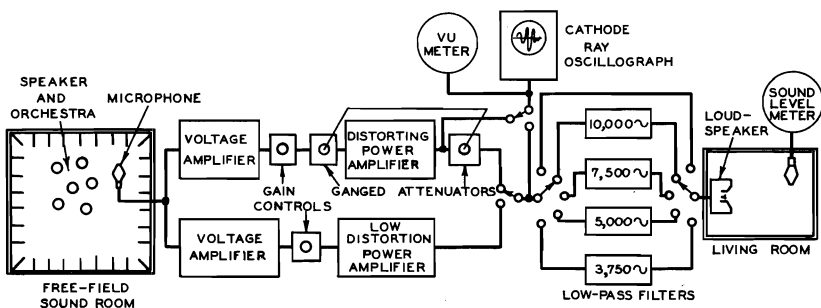
FIG. 12.41. Schematic arrangement of the apparatus for a subjective determination of the relation between nonlinear distortion and frequency range in reproduced sound.

Some idea of the effect of nonlinear distortion can be obtained from a study of the masking curves. From these data it will be seen that the high order harmonics are noticeable at much lower levels than the lower order harmonics. Furthermore, as the high-frequency range is increased the effect of the harmonics is more noticeable. In the complex waves of speech and music, sum and difference tones are also an important phase of the problem of nonlinear distortion.

The effect of various types of nonlinear distortion[64] on the reproduction of speech and music has been determined through the system shown in Fig. 12.41. The over-all response frequency characteristic of the microphone,

[64] Olson, H. F., Unpublished Report.

amplifier, and loudspeaker was uniform to within 2 db from 45 to 15,000 cycles (Fig. 12.34). Low-pass filters were provided with cutoff frequencies at 3750, 5000, 7500, and 10,000 cycles. The nonlinear distortion in the over-all reference channel was very low. In addition, the distortion components were principally second and third harmonics. The pickup studio for these tests was the free-field sound room described in Sec. 10.3A4.
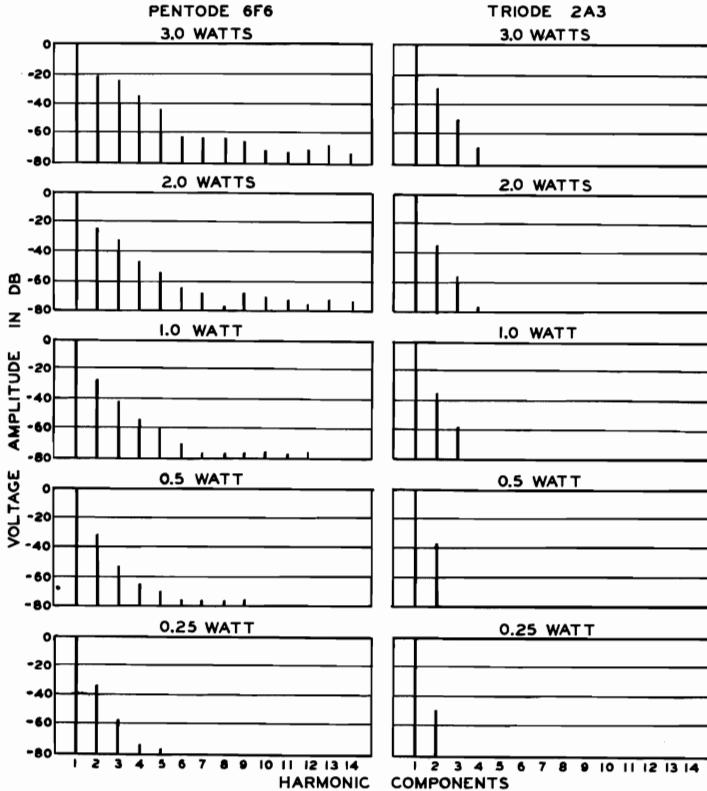
FIG. 12.42. Distortion characteristics of single-ended pentode and triode amplifiers.

The sound was reproduced in a room with acoustics similar to a typical living room. The noise level at the pickup point was about 0 db in the absence of any performers. The noise level in the listening room was about 25 db.

Two types of distorting amplifiers were used—namely, a single-ended triode and a single-ended pentode. The distortion components for these two power output systems for various power levels are shown in Fig. 12.42.

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 to 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
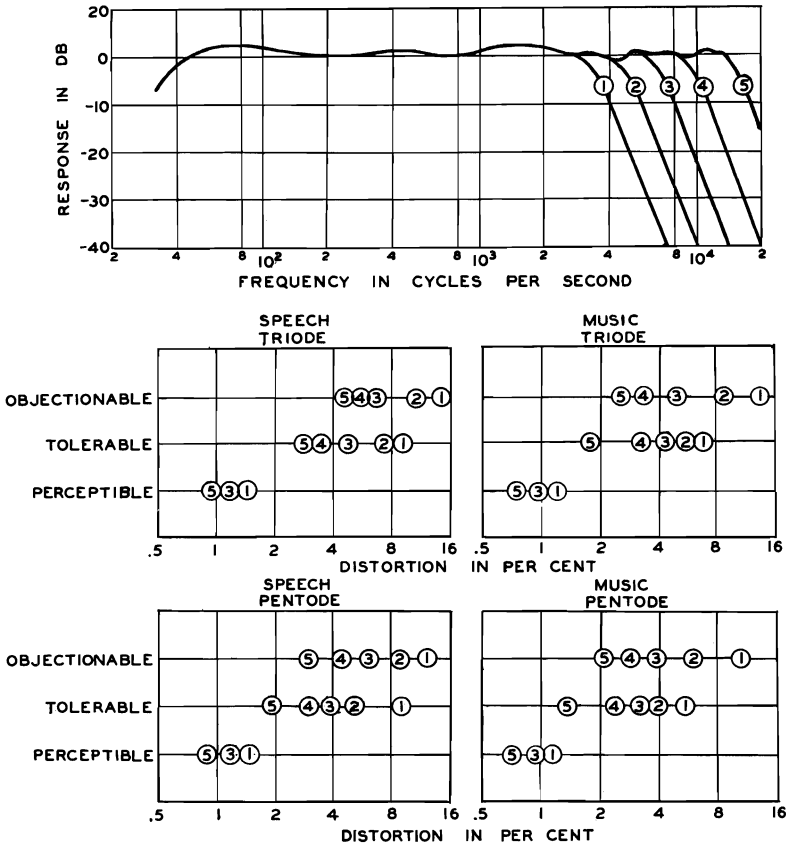
FIG. 12.43. Experimental results of subjective tests of reproduced speech and music depicting objectionable, tolerable, and perceptible nonlinear distortion for various high-frequency cutoffs. The numbers in the distortion data points correspond to the numbers which label the response frequency characteristics. The distortion, in per cent, is the ratio of the total rms of the second, third, fourth, etc., components of distortion to the rms of the fundamental, multiplied by one hundred.

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 reproduction.

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. 12.43. 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.

12.28. Frequency Ranges of Sound Reproducing Systems.—The frequency ranges of the most common sound reproducing systems are shown in Fig. 12.44. 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. 12.44.

The frequency range of the sound powered telephone is quite restricted. The efficiency is a function of the frequency range and decreases as the range is increased. The articulation is a function of the frequency range and intensity. The particular frequency range is a compromise between the various factors which yields the maximum articulation.

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 ranges of table model and console model radio and television receivers refer to commercial models of radio and television receivers sold during the last decade. A relatively small number of high-quality radio and television receivers have been built with frequency ranges varying from the commercial models up to the high-quality radio and television receiver. Most broadcast radio and television transmitters cover the entire audio range with good fidelity.

The frequency ranges of phonographs refer to commercial models sold during the last decade. The frequency range of the phonograph record is the average of commercial phonograph records pressed at the present time. The frequency range of transcription and high-quality phonographs is that of the systems used in broadcast stations and high-quality phonograph and sound systems.

Magnetic tape recorders and reproducers have been commercialized during the past decade. The small portable tape recorder exhibits a limited frequency range as compared to the wide range tape reproducing system.

The frequency range of a sound motion picture system refers to the average of existing reproducing systems in use in theaters at the present time. 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. 12.44.

The average frequency range of commercial sound systems used in public address, announce, paging, reinforcing, intercommunicating, and other applications of sound systems, is shown in Fig. 12.44. A small number of
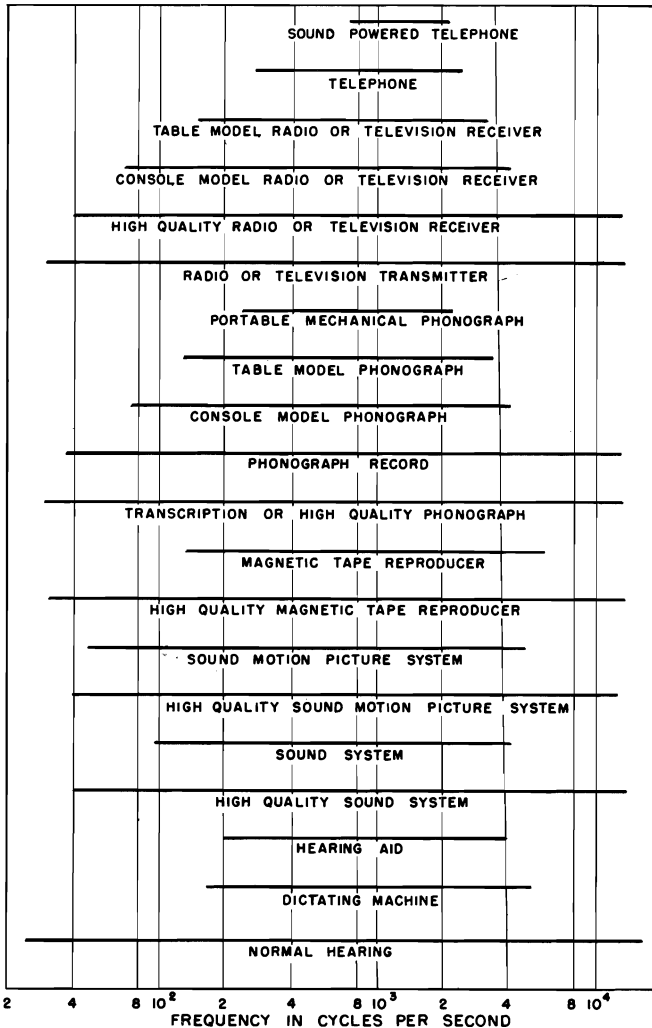
FIG. 12.44.   The frequency ranges of sound reproducing systems.

high-quality sound systems are also in use in sound reinforcing systems and other applications.

The frequency range of the hearing aid shown in Fig. 12.44 represents the average response of high-quality vacuum tube hearing aids in use today. The low-frequency range may be somewhat greater but in general this added range cannot be used due to rumble and other low-frequency noises.

An examination of Fig. 12.44 shows that the general run of commercial receivers, phonographs, television receivers, sound motion-picture systems, magnetic tape reproducers, and sound systems sold during the past decade

exhibit response frequency ranges between 100 and 5000 cycles. On the other hand, the frequency range of radio and television transmitters, sound motion-picture records, phonograph records, and magnetic tape is practically that of the human ear. In other words, the potential frequency ranges of radio and television receivers, phonographs, sound motion-picture and magnetic tape reproducers are not limited by the source material or the system. In this connection it is interesting to note that the sale of wide frequency range high-quality radio and television receivers, phonographs, sound motion-picture, and magnetic tape reproducers has increased at a high rate during the past five years. In the case of some of these systems great strides have been made to higher quality. There can be no doubt but that there is a definite trend toward higher quality in reproduced sound. One of the principal reasons for this state of affairs is due to the reduction in nonlinear distortion.

12.29. Frequency Range Preference for Reproduced Speech and Music.[65]—In the preceding sections data have been presented depicting the response frequency characteristics of the human ear, the manner in which these characteristics vary with age, the loudness range, the effect of masking sounds, the effect of nonlinear distortion, the effect of the frequency range upon the articulation of speech and upon the quality of music, the frequency ranges of speech and music, and the frequency ranges of reproducing systems. A study has also been made of the frequency range preference of a representative cross section of broadcast listeners.

As contrasted to 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. 12.45 and were designated as wide, medium, and narrow frequency ranges.

The results of the tests are shown in Fig. 12.45. 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.

It is interesting to note that the frequency range preference of a representative cross section of broadcast listeners is essentially the same as the frequency range of commercial radio receivers, phonographs, sound motion picture systems, and sound systems.

[65] Chinn and Eisenberg, *Proc. Inst. Rad. Eng.*, Vol. 33, No. 9, p. 571, 1945.
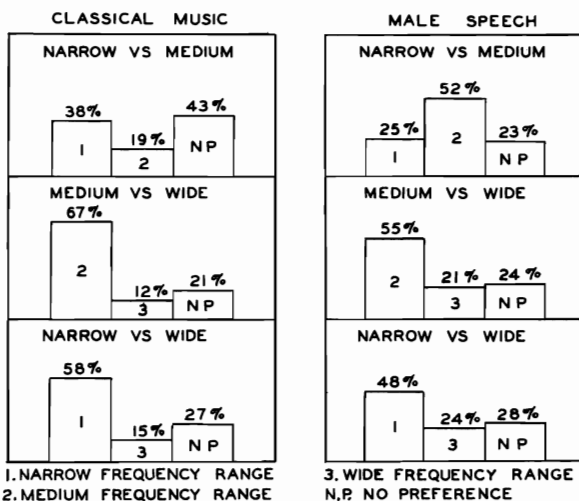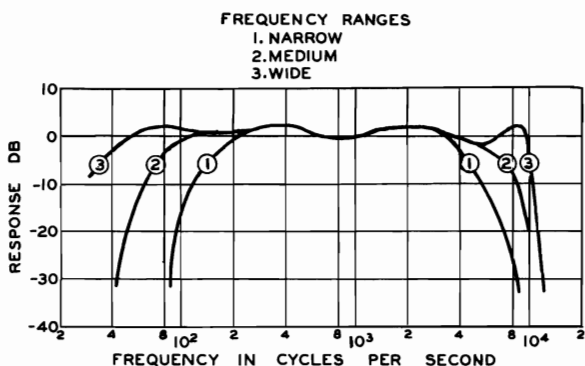
FIG. 12.45. The frequency range preferences of a cross-section of listeners for classical music and male speech. The graph depicts the response frequency characteristics for the following: 1 = narrow frequency range. 2 = medium frequency range. 3 = wide frequency range. The block diagrams depict the preference. NP means no preference. When the narrow range was compared to 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 very markedly for the narrower bands for both classical music and male speech. (After Chinn and Eisenberg.)

12.30. Frequency Range Preference for Live Speech and Music.[66]—

The frequency range preference for reproduced speech and music was considered in the preceding section. These tests indicate that listeners prefer

[66] Olson, H. F., *Jour. Acous. Soc. Amer.*, Vol. 19, No. 4, Part 1, p. 549, 1947.

a restricted frequency range in monaural reproduced speech and music.
There are three possible reasons for the results of these tests, as follows:
A. The average listener, after years of listening to the radio and the phono-
graph, has become conditioned to a restricted frequency range and feels
that this is the natural state of affairs. *B.* Musical instruments are not
properly designed and would be more pleasing and acceptable if the produc-
tion of fundamentals and overtones in the high-frequency range were
suppressed. *C.* 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.
       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.

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 arrange-
ment of the test is shown in Fig. 12.46.   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. 12.46.   The response frequency
characteristic of the acoustical filter shown in Fig. 12.46 approximates
commercial good radio or phonograph reproduction in the high-frequency
range.   The acoustical filter is composed of ten units with each unit pivoted
at the top and bottom.   The ten 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°.   The acoustical filters are shown in
the full frequency-range position in Fig. 12.46.   A sheer cloth curtain
which transmits sound with no appreciable attenuation over the frequency
range up to 10,000 cycles and less than 2-db attenuation from 10,000 cycles
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 5000 cycles low-pass transmission—is shown on an A-B 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 the room was about 70 db. 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 indicated a preference for the 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. 12.29.
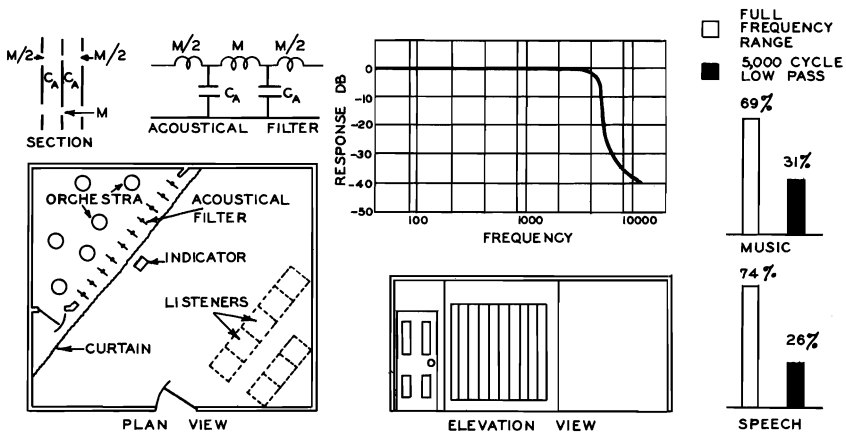
Fig. 12.46. Plan and elevation views of the schematic arrangement of the apparatus for direct frequency preference testing of speech and music. A sectional view, acoustical network and response frequency characteristic of the acoustical filter used in the tests and the results are also depicted.

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. 12.27, 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.

12.31. Frequency Range Preference for Stereophonic Reproduced Speech and Music.[67]—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. 12.30. These tests were all acoustic. The sound was not reproduced. Therefore, there were no electroacoustic transducers

[67] Olson, Preston, Woodward, May, Morgan, and Bleazey, Unpublished Report.

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 as to whether similar results can be obtained with reproduced sound.　The only reason that the same results could not be
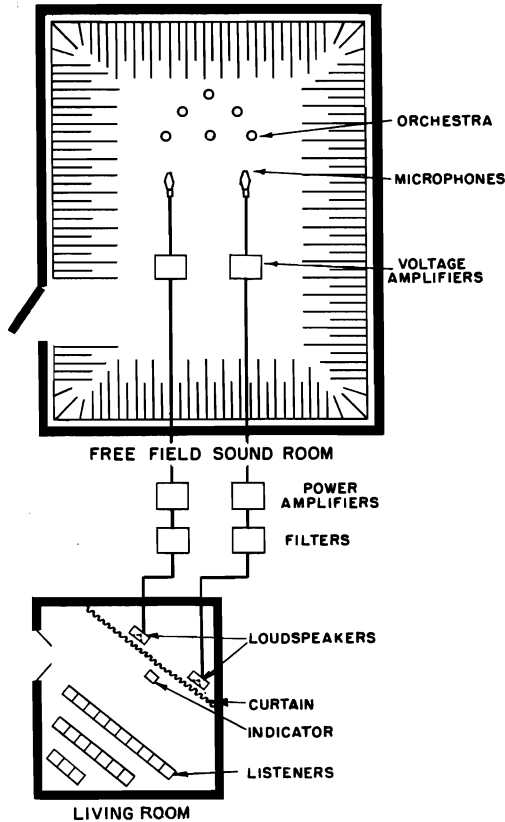
FIG. 12.47.　Plan view of the schematic arrangement of the apparatus for frequency range preference for stereophonic reproduced speech and music.

obtained with reproduced 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. 12.30.

Following the all-acoustic frequency range preference tests, it appeared logical to follow these tests with frequency range preference tests employing 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. 12.47. The idea of the test was 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 free-field sound room was 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. See Sec. 10.3A3. The level
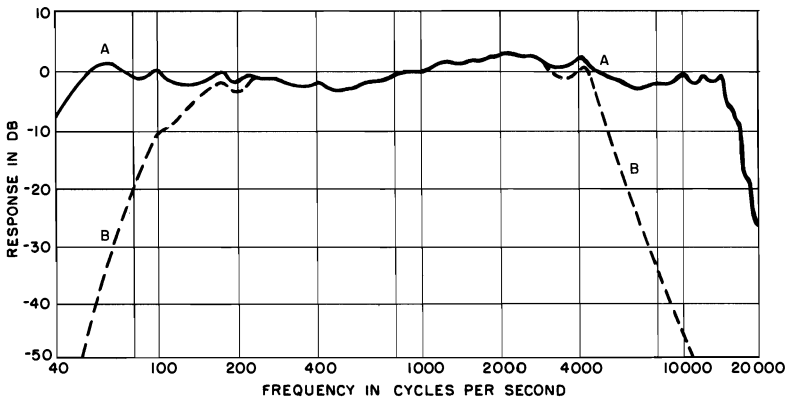
FIG. 12.48. Over-all response frequency characteristic of the two conditions used in the frequency range preference for stereophonic-reproduced speech and music. A. Wide frequency range. B. Restricted frequency range.

of the reflected sound at the microphones for steady state sound conditions is about 50 db below the direct sound. Therefore, it is impossible to detect any acoustics of the studio in the reproduced sound.

The listening room is the same as that used in the all-acoustic test. The listening room was designed to be the acoustical equivalent of an average living room.

The reproducing system used in these tests employed two channels. Each channel consisted of a RCA 44BX Velocity Microphone, a RCA OP-6 Amplifier, a laboratory developed triode-type push-pull power amplifier, laboratory developed high- and low-pass electrical filters, and a 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. 12.48. In the restricted range condition there is attenuation in both the low- and high-frequency ranges. The product of the low and high cutoff frequencies is 500,000 (cycles)[2]. This value has been recommended by many investigators.[68] 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 db.

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 overall 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. 12.27, 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 db to 80 db 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.

12.32. Comparison of Live and Reproduced Symphony Orchestra.[69] —A demonstration was conducted on July 29, 1947, at the Berkshire Festival in Tanglewood, Massachusetts, in which the full Boston Symphony Orchestra was compared with a phonograph record of the orchestra. The demonstration was made before an overflow audience in the Music Shed at Tanglewood, Massachusetts.

Plan and elevation views of the Music Shed are shown in Fig. 12.49. The main audience area with a ceiling height of 40 feet contains over 6000 seats. The colonnade, twenty-seven feet in width, along the outside edge,

[68] Hanson, Rackey, and Nixon, "Down to Earth on High Fidelity," Eng. Dept. N.B.C., New York, N.Y., March 27, 1944.
[69] Olson, Preston, Riefsteck, Reiskind, and Pulley, Unpublished Report.

with a ceiling height of twenty feet will accommodate an additional 3000 persons. From a consideration of the dimensions and the seating capacity, it will be seen that the Music Shed is a very large structure.

The reverberation time of the audience area, with all the seats occupied, ranges from 1 to 1.5 seconds in the mid-frequency range. The relatively
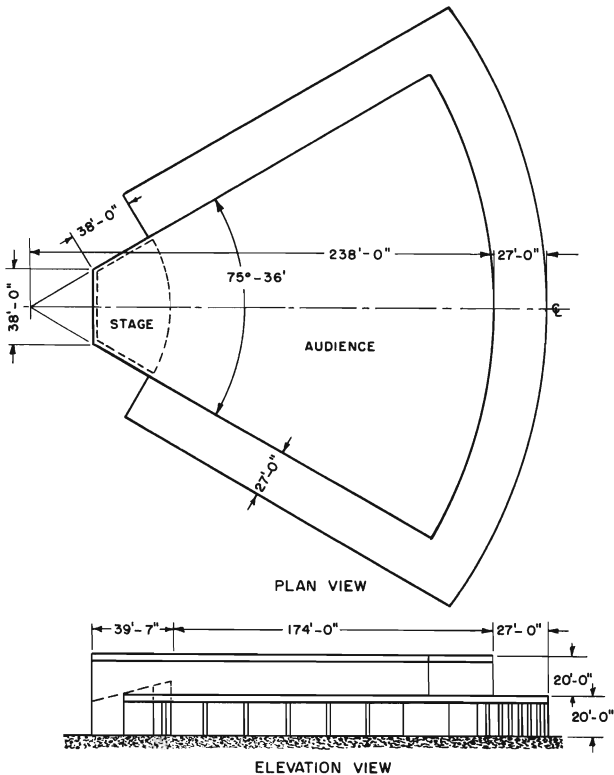
FIG. 12.49. Plan and elevation views of the Music Shed at Tanglewood, Mass.

low reverberation time for a structure of this size is due to the openings to free space on all sides which represent 100 per cent absorption. However, the reverberation time of the stage and shell is much greater. Therefore, the general impression of perspective of the orchestra is lost save for the first few rows of seats.

Plan and elevation views of the stage showing the location of the microphones used in recording and the loudspeakers used in reproducing are shown in Fig. 12.50.

A schematic diagram of the recording system is shown in Fig. 12.50.

Three RCA Type 44BX Velocity Microphones were used.   The gain in each of the three microphone channels was the same.

A schematic diagram of the sound reproducing system is shown in Fig. 12.51.   Tests indicated that 12 RCA Type LC1A Loudspeakers driven by 3-40 watt amplifiers would reproduce the peak level of the orchestra.   In reproducing the record, twelve Duo Cone Loudspeakers were placed at the front of the stage as shown in Fig. 12.50.   In this way, the acoustics of the reproduced sound matched the acoustics of the orchestra because the loud-speakers carried on, as it were, from where the microphones picked up the sound.   Furthermore, due to the relatively reverberant stage and relatively dead listening area, it was not necessary to use perspectively reproduced
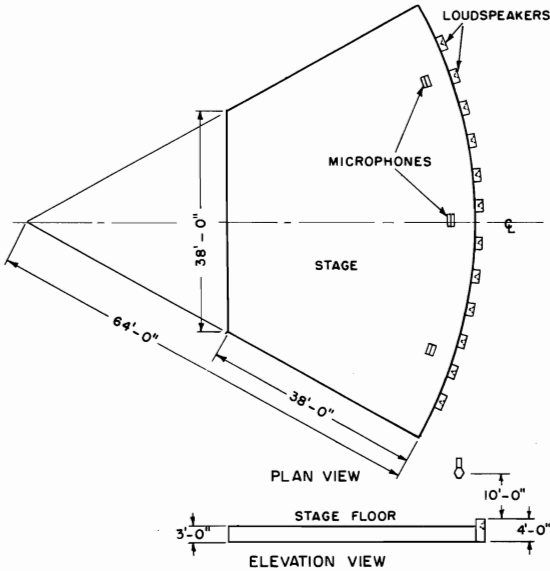
FIG. 12.50.   Plan and elevation views of the stage of the Music Shed showing the locations of the microphones and loudspeakers.

sound to maintain the illusion that the reproduced sound was identical with the original sound.

The peak sound level attained by the full orchestra at the box seats, which are located about 85 feet from the stage, was 95 db.   This value agrees with measurements made on many other orchestras in many other locations. It was found that this level could be matched with 120 watts input to the twelve loudspeakers in reproducing the sound.   The noise level with the full audience and when the audience was quiet was about 40 db.

The over-all response frequency characteristic depicting the ratio of the sound pressure output from the loudspeaker in free space to the sound pressure at the microphone in free space is shown in Fig. 12.52.

The last four minutes of Beethoven's overture to "Egmont" was recorded by the Boston Symphony Orchestra with Serge Koussevitsky conducting. In the demonstration the Boston Symphony Orchestra, with Serge Koussevitsky conducting, played the preceding portion of the selection and the
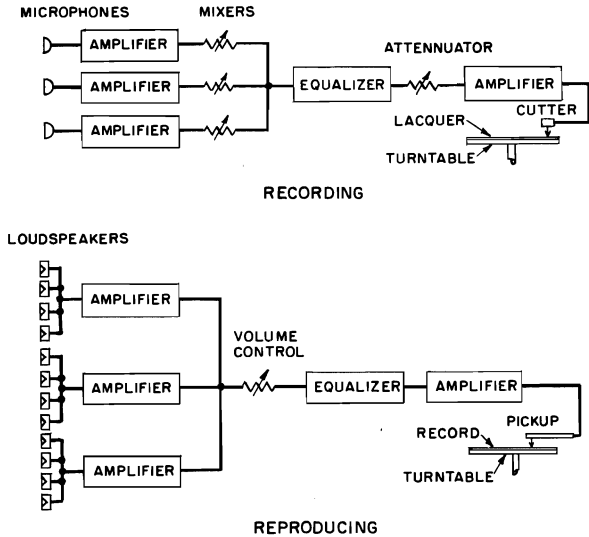
FIG. 12.51. Schematic diagrams of the recording and reproducing systems used in the reproduction of a symphony orchestra.
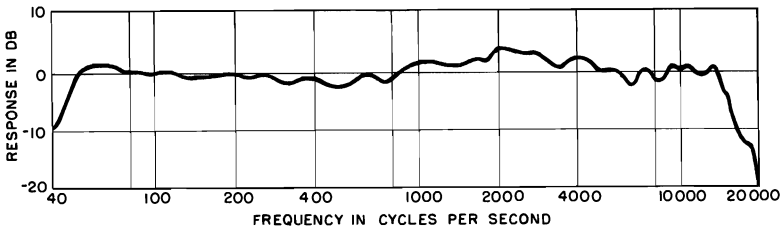
FIG. 12.52. Over-all response frequency characteristic of the recording and reproducing systems used in the reproduction of a symphony orchestra.

switch was made from original sound to reproduced sound. Many rehearsals were carried out to insure continuity of the music as well as the amplitude during the switch from live to reproduced music.

Comments from both music critics and layman were that tone color, dynamic range, and general fidelity of the reproduced sound matched the

orchestra so closely as to be practically indistinguishable from the original sound.

A few excerpts from the comments made by critics on the demonstration of the comparison of the reproduction of a record of the Boston Symphony Orchestra with the live orchestra are as follows:

"I have heard this instrument (system) and find it unprecedented in the fidelity of its performance."—Serge Koussevitsky.

"Under the severe test of this juxtaposition (pitting the live orchestra against the reproduced sound) one could still at times note a difference in tone quality .... but the impression was often one of identity or remarkable close approximation."—Francis D. Perkins, N.Y. Herald Tribune.

"Research in electronics has now made possible hearing life-like reproduction of actual performances ... The advance over other recorded per-
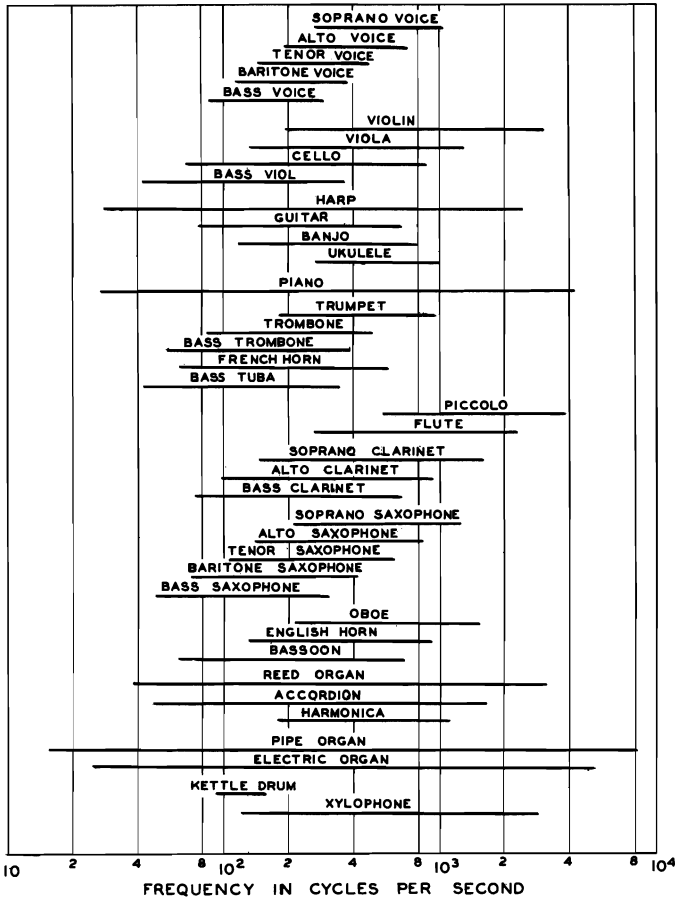
FIG. 12.53.   Frequency ranges of the fundamental frequencies of voices and various musical instruments.

formances . . . is little short of revolutionary in fidelity of reproduction."—
Miles Kastendieck, Journal American.

". . . in an almost imperceptible switch. It was scarcely apparent where
the live orchestra left off and the canned music began, so alike in volume
and tone quality was the music which came from the machine. . . . It is the
closest I have yet heard mechanical reproduction come to the real thing."—
Robert Hague, N. Y. PM.

"For a moment, even the listeners who anticipated the daring gesture
were not quite sure which was performing. . . . But it (the reproduced sound)
carried on with a veracity of tone color and dynamic range that a verdict in
its favor could not be denied."—Irving Kolodin, N. Y. Sun.

**12.33. Fundamental Frequency Ranges of Voices and Musical
Instruments.**—Musical instruments and the voice produce fundamental
frequencies and overtones of fundamental frequencies. The overtone
structure is one of the characteristics which distinguish various instruments
and voices. If musical instruments produced the fundamental without
overtones, each instrument would produce a pure sine wave and would
therefore be the same as the output of an oscillator and loudspeaker combina-
tion. The fundamental frequency is the lowest frequency component in the
complex sound wave. When a musician speaks of the "range" of a voice
or musical instrument, he means the frequency range of the fundamental
frequency. The fundamental frequency ranges of voices and various musical
instruments are shown in Fig. 12.53. There may be some variation from
these frequency ranges among various instruments and voices but in general
the ranges are typical. Comparing the frequency ranges of the fundamentals
of Fig. 12.53 with the entire frequency spectrum of musical instruments of
Fig. 12.23, it will be seen that the overtones of the instruments extend the
frequency ranges by a factor of two or more octaves.

12.34. Musical Scale.[70]—An octave is the interval between any two
tones whose frequency is 2 : 1.

A tone is a sound giving a definite sensation of pitch.

A scale is a series of tones ascending or descending in frequency by definite
intervals suitable for musical purposes.

A cent is the interval between any two tones whose frequency ratio is
the twelve-hundredth root of two.

For the practical production of music a so-called equally tempered scale
has been found to be most satisfactory. The equally tempered scale is a
division of the octave into twelve equal intervals called equally tempered
half tones.

The standard pitch for America is based on the frequency 440 cycles per
second for the tone A on the pianoforte keyboard. The frequencies of a
piano having an equally tempered scale are shown in Fig. 12.54.

The relative frequencies of the natural or diatonic or just scale and tem-
pered scale are shown below.

[70] Olson, "Musical Engineering," McGraw-Hill Book Company, New York, N.Y.,
1952.

	C	D	E	F	G	A	B	C
Natural Scale . . .	1.000	1.125	1.250	1.333	1.500	1.667	1.875	2.000
Tempered Scale. . .	1.000	1.122	1.260	1.325	1.498	1.682	1.887	2.000

Singers and players of instruments whose pitch can be varied by breath or touch prefer the natural scale to the equally tempered scale because it seems to be more artistic. Of course, the difference between the two scales is small. However, in the case of chords it is said that the difference is noticeable.

Pianos and other fixed-pitch instruments are tuned to the equally tempered scale so that music may be played in various keys without retuning.
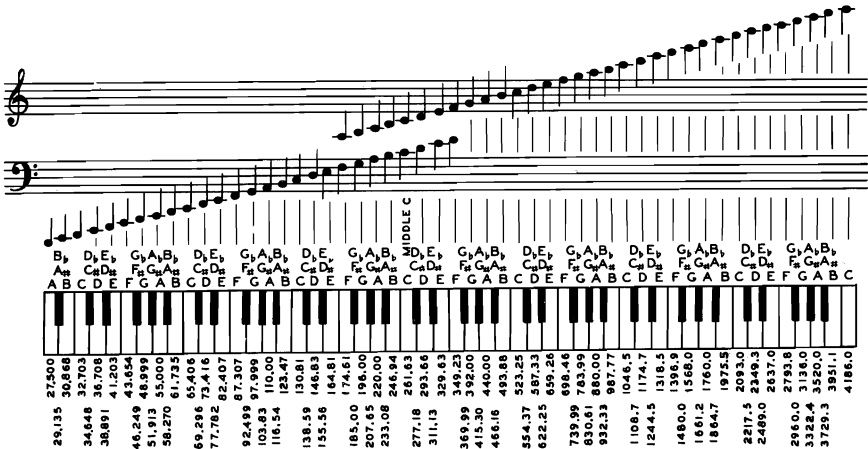
FIG. 12.54. The piano keyboard and the frequencies of the equally tempered scale as used in music according to the American Standard Pitch.

12.35. Electrical Musical Instruments.[71,72,73,74]—The vacuum tube oscillator and amplifier have opened an entirely new field for the production of sound of practically any frequency, quality, or amplitude. Many musical instruments employing various types of vibrating systems and associated vacuum tube oscillators and amplifiers have been developed.

[71] Miessner, B. F., "Electronic Music and Instruments," Proc. IRE, Vol. 24, No. 11, p. 1427, 1936.

[72] Young, Robert W., "Twenty-five Years of Musical Acoustics," Jour. Acous Soc. Amer., Vol. 26, No. 6, p. 955, 1954.

[73] Olson, "Musical Engineering," McGraw-Hill Book Company, New York, N.Y., 1952.

[74] Dorf, "Electronic Musical Instruments," Radio Magazines, Inc., Mineola, N.Y., 1954.

The simplest system for the amplification of string instruments like violins, guitars, banjos, pianos, etc., consists of a vibration pickup attached to the body or sounding board, an amplifier, and a loudspeaker.

Electric pianos[75] have been developed in which the vibrations of the strings are converted into the corresponding electrical variations. In one system, the variation in capacity between the string and an insulated plate is used in a manner similar to the condenser microphone. In another, the string acts as an armature in an electromagnetic system. The outputs of the pickup systems are amplified and reproduced by means of loudspeakers. Electric pianos have also been developed and commercialized in which the vibrations of reeds are converted into electrical variations by an electrostatic system similar to the condenser microphone. The reeds are set into vibration by being struck by hammers in a manner similar to that of conventional pianos.

One type of electric organ[76] consists of a number of small alternators (one for each note), a keying and mixing system for adjusting the quality, and amplifier, and a loudspeaker. Another electric organ[76a] employs wind driven reeds. The vibrations of the reeds are converted into the corresponding electrical variations, amplified and reconverted into sound by means of loudspeakers.

In an all electronic organ the frequencies in the highest octave of the equally tempered scale are generated by means of twelve vacuum tube oscillators. The frequencies in the lower octaves are obtained by electronic dividing. This procedure insures that all the overtones are harmonics of the fundamental. The overtone structure of a tone is obtained by suitable mixing of the fundamental and harmonics.

Electric carillons[77] consisting of tuned coiled vibrators, magnetoelectric translators, amplifiers, and reproducers possess qualities which are quite similar to the conventional carillons.

12.36. Electronic Music Synthesizer.[78]—Music is the art of producing pleasing, expressive, or intelligible combinations of tones. The sounds of original music are produced by the human voice or by an instrument actuated by a musician. Most music is recorded and translated into sounds by means of a symbolic notation on paper. The ultimate objective destination of all musical sounds is the human ear. Thus the production of music consists of the following processes: The symbolic notation upon paper by the composer, the translation of the symbolic notation into musical sounds by the musician, employing either his own voice or a musical instrument or both, and the actuation of the human hearing mechanism by the musical sounds.

The medium of transmission from the musician and musical instrument to the listener is sound waves. These sound waves carry the musical tones.

[75] Meissner, B. F., *Proc. IRE*, Vol. 24, No. 11, p. 1427, 1936.
[76] Hammond, B. F., U.S. Patent, 1,956,350.
[76a] Hoschke, U.S. Patent 2,015,014.
[77] Curtiss, A. N., U.S. Patent, 2,026,342.
[78] Olson and Belar, *Jour. Acous. Soc. Amer.*, Vol. 27, No. 3, p. 595, 1955.

The properties of a musical tone are frequency (pitch), intensity (loudness), growth, duration, decay, portamento, timbre, and vibrato. These characteristics of a tone are depicted in Fig. 12.55. Descriptions of the properties of a tone have been given in preceding sections in this chapter.

Once a sound or a tone has been described by means of the characteristics shown in Fig. 12.55, it is possible to generate or produce this tone by electronic
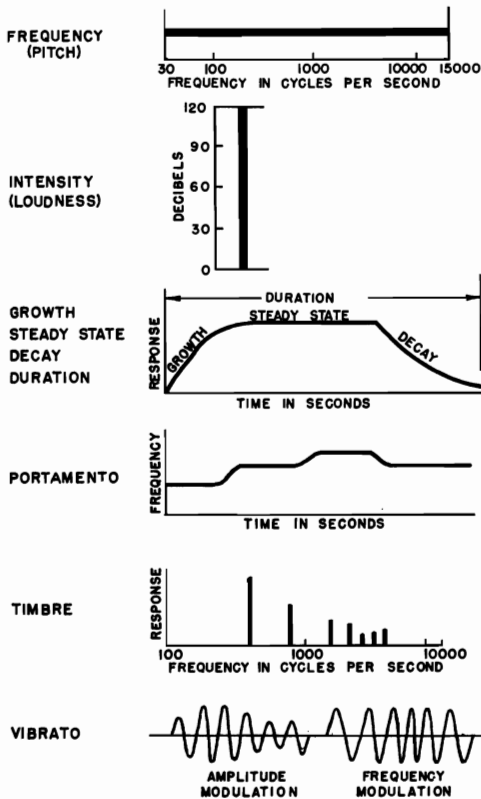
FIG. 12.55.   Characteristics of a tone.

means.   Thus it will be seen that it is possible to generate any tone produced by a voice or a musical instrument by employing an electronic system.   In addition, it is possible to produce musical tones which cannot be produced by the voice or conventional instruments.   In other words, the process of translating the musical notation on paper into the corresponding musical sounds can be accomplished by an electronic system.   Furthermore, the electronic system can reproduce or create any sound or combinations of sounds, which have or have not been produced, that may have any possible musical significance.   The electronic system for the production of musical sounds has been termed electronic music synthesizer.

One of the uses for an electronic music synthesizer is for the production of phonograph records.   Of course, the phonograph records thus produced can be played in the conventional manner.   The use of a synthesizer for the production of musical sounds opens an entirely new field for the production of recorded music.   For example, there is the possibility of entirely new tone complexes and combinations which cannot be achieved in conventional instruments.   Furthermore, in the case of conventional instruments, the musician is limited to the use of ten fingers on two hands and two feet to perform the different operations.   This limitation does not exist in the synthesizer.   Conventional instruments produce various noises such as the rushing of wind in wind instruments, bow scratch in the viol family, various clatters and rattles in plucked and struck string instruments, and mechanism rattle in any instrument in which keys, valves, levers, and shafts are used. These undesirable noises do not exist in the electronic music synthesizer. With the advent of the electronic method for the production of musical tones, new musical compositions can be written which take advantage of the superior characteristics of the electronic synthesizer.

From the preceding discussions and Fig. 12.55, it is evident that, in order to synthesize any musical tone whatsoever, the synthesizer must provide the following facilities: Means for producing a tone with any fundamental frequency within the audio frequency range: Means for producing a tone with any overtone structure.   Means for producing a tone of any growth, duration, or decay characteristic.   Means for changing the overtone structure at any time.   Means for introducing a vibrato.   Means for changing the intensity of the tone.   Means for providing a portamento or glide from a tone of one frequency to a tone of a different frequency.

A schematic block diagram of an electronic music synthesizer with means for producing all the characteristics of a musical tone outlined above is shown in Fig. 12.56.   The coded paper record controls all the functions of the electronic music synthesizer.   The information is recorded and stored in the paper record in the form of perforations.   When the paper record is run through the machine, brushes slide over the paper record.   An electrical circuit is closed when a brush passes over a perforation.   The brushes which pass over the paper record actuate electrical circuits in the relay trees.   The use of a relay tree makes it possible to record the information on the paper record in the binary code system.   There is a separate relay tree for each of the characteristics of a tone.   In this way any of the characteristics of a tone can be obtained or changed at any instant.   Thus it will be seen that any tone whatsoever can be produced by providing the proper information in the coded paper record.   The output of the synthesizer is recorded on a disk record.   The coded paper record and the phonograph recorder are driven in synchronism by an interconnecting cable drive.   In the complete electronic music synthesizer two complete channels, as shown in Fig. 12.56, are used and operated from the single coded paper record.   This makes it possible for the coded paper record to set up one channel while the other channel is in operation and producing a tone.   Furthermore, one channel
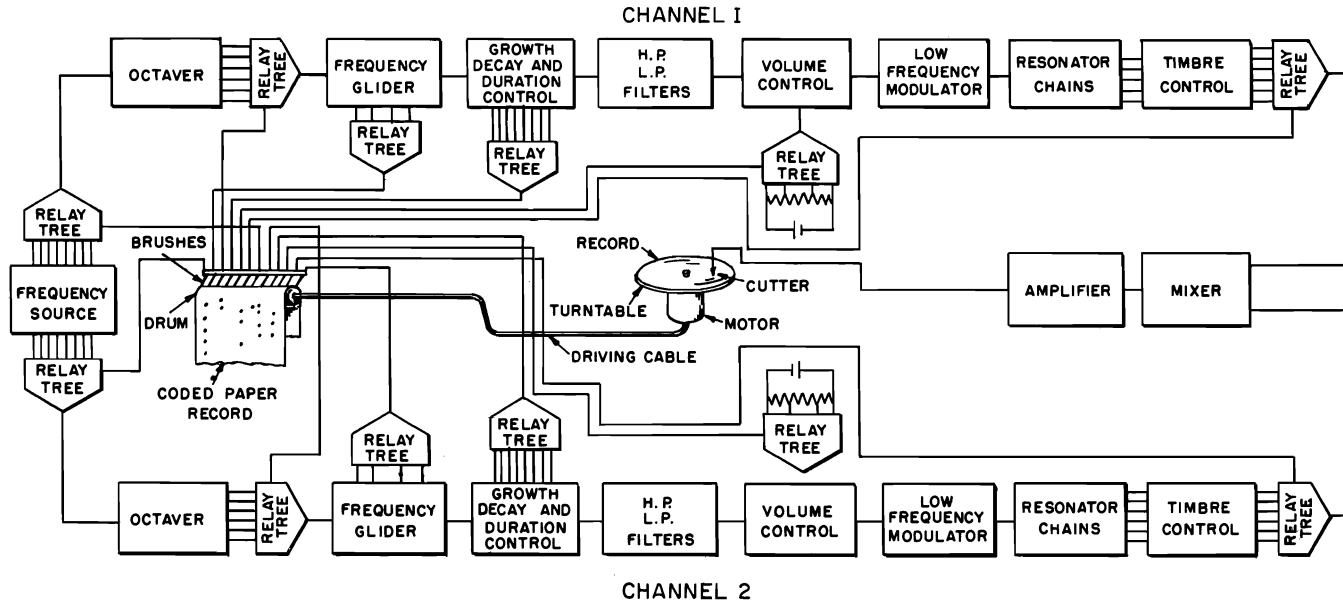
FIG. 12.56.   Schematic diagram of the electronic music synthesizer.

can start playing a tone before the other channel stops playing a tone. Everything is duplicated in the second channel except the twelve tuning fork oscillators which supply the tones in one octave.

A schematic perspective view of the complete electronic music synthesizer is shown in Fig. 12.57. The seven racks contain all the electronic equipment for performing the different functions. The information contained in the paper record in the form of perforations is recorded by means of a puncher actuated by keys forming a keyboard similar to that of a typewriter.

From the fundamental knowledge and the characteristics and performance of the synthesizer, as outlined above, it is not difficult to prove from
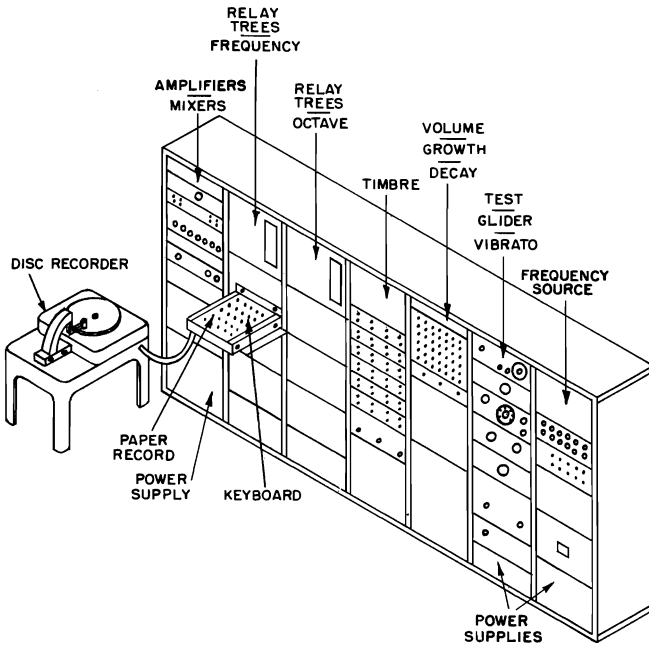
FIG. 12.57.   Perspective view of an electronic music synthesizer.

theory what can be done with the synthesizer.   However, to make a practical demonstration, it is necessary to train an operator or synthesist.   In this connection, the music synthesizer is an entirely new genus of apparatus. Were it merely an improvement of existing species, immediate commercial applications would be evident.   One of the uses foreseen for the music synthesizer is to make music for sale in the form of phonograph records. To make an artistic record, or a hit, novelty alone or technical excellence is not sufficient.   There are also other ingredients many of which are intangible. However, the synthesizer will facilitate the production of such a hit.   For example, the synthesizer can produce any kind of sound that can be imagined. Then if a person can image a hit, then the synthesizer will facilitate the

production of the hit.   The hit producing attributes are not amenable to analysis in the acoustical laboratory, so this factor must be left out.   To demonstrate the possibilities and performance of the synthesizer, an experiment was conducted as follows:

Two different piano selections, namely, "Polonaise in A Flat Op. 53" (Chopin) and "Clair de Lune" (Debussy) and a violin and piano selection, "Old Refrain" (Kreisler), all played by famous artists and reproduced from commercial disk records were compared with synthesized versions of the same selections.   The piano selections were recorded by Iturbi, Rubinstein, and Horowitz, and the violin in "Old Refrain" was recorded by Kreisler. The synthesizer versions, completed on August 5, 1953, and the identical passages from the commercial phonograph records were recorded on magnetic tape and intermixed and played to various people.   They were asked to tell which was synthesized and which was not.   Interpreting the results by standard statistical methods, it can be said with 70 per cent certainty that only one out of four persons can tell which is which.

It should be noted that this work was done with only the help of conventional equipment to aid in the analysis which is not as complete as demanded for synthesis; thus much of the work was done by cut and try.   This points to an important feature of the synthesizer.   Manual dexterity is not required. A synthesis once learned can be added to the fund of knowledge without further practice to be able to perform it.   The ability of the synthesist is, therefore always increasing.

If a composer has in mind what he wants to achieve, the effects can be obtained by means of the electronic music synthesizer, regardless of whether he can play a musical instrument or not.   The composer or musician can produce the sound of any existing musical instrument as well as other sounds, regardless of whether they have ever existed.   The results which the composer and musician wishes to achieve can be obtained and demonstrated as the music is being composed and played.   Once a particular result has been obtained, it can be retained forever.   Thus it will be seen that the electronic music provides a powerful tool for the composer or musician because he can reproduce or create any sound or combination of sounds which have or have not been produced, that may have any musical significance.

The above experiment demonstrates the potential capabilities of the synthesizer in the ability to copy existing selections.   For an acoustical engineer to copy the performance of famous artists in the matter of a few weeks shows the tremendous possibilities of the synthesizer.   For example, it would be impossible for even a genius with no previous experience in playing a musical instrument to imitate all of these artists on these two instruments with the order of fidelity that was achieved.   Thus, it seems quite obvious that a trained or professional musician could produce great musical renditions by means of the electronic music synthesizer.

The above experiment illustrates another use for the synthesizer, in addition to those which have been listed, namely, the rejuvenation of old

recordings where the master is in poor condition. That is, a new record without distortion and noise can be made by the synthesizer.

One use envisioned for the music synthesizer is the production of music for sale in the form of phonograph records. In order to further demonstrate the potentialities of the music synthesizer, complete musical selections were synthesized. A partial list of these selections indicating the style of music and the date on which the synthesis was completed was as follows:

"Blue Skies" (Berlin) April 1, 1952.	In the style of a dance band.
"Nola" (Arndt) May 28, 1952	In the style of the piano.
Stephen Foster Medley, December 12, 1952	"Oh Susanna," "De Camptown Races," "My Old Kentucky Home," "Old Black Joe," "Old Folks at Home," and "Hard Times Come Again No Mo." In the style of bowed, plucked, and struck instruments; air, mechanical, and lip reed instruments; and percussion instruments.
"Holy Night" (Adams) December 15, 1952.	In the style of the organ.
"Fugue No. 2 from Well Tempered Clavichord" (Bach) July 15, 1953.	In the style of ancient struck and plucked strings in several variations.
"Hungarian Dance No. 1" (Brahms) September 3, 1953	In the gypsy style without copying any particular instrument, but varying colors adapted for easy synthesis.
"Sweet and Low" (Tennyson-Barnby) January 15, 1954	Voice and instrumental accompaniment to show that voice can be synthesized.
"Spoken Voice" March 26, 1954	A few spoken sentences were synthesized to show the versatility of the synthesizer.

The results obtained with the electronic music synthesizer as exemplified by the musical selections outlined above demonstrate that excellent musical performance can be produced by means of this new system; in fact, the performance of the musical synthesizer speaks for itself.

A phonograph record, RCA Victor LM-1922, entitled "The Sounds and Music of the RCA Electronic Music Synthesizer" illustrates the characteristics of a tone and music produced by the synthesizer.

12.37. Phonetic Typewriter.[79, 80]—The printing press, as developed about 400 years ago, made it possible to disseminate information in the form of the printed page. The advent of sound reproduction in various forms made it possible to reproduce sound at the same point or some other point, either at the same time or some subsequent time. Another logical step in the field of communication is the conversion of speech sounds into the corresponding printed words on a printed page. This process involves the use of speech sounds to actuate machinery in accordance with the information carried by the speech. There are many applications for machines that will

[79] Phonetic typewriter is a term used to designate a voice-operated machine which types or prints on paper the words spoken into the microphone input to the machine.

[80] Olson and Belar, *Jour. Acous. Soc. Amer.*, Vol. 28, No. 6, p. 1072, 1956.

make it possible for speech to perform certain operations. One of the important and useful machines in this category is a speech typewriter which prints or types on paper the words which are spoken into the machine.

The important factors involved in the development of a phonetic typewriter are as follows: The particular form in which the words are typed. The means for analyzing the sounds of speech. The process for the identification of the analyzed sounds. The means for encoding, storing and decoding of the sounds for the operation of the actuating mechanism. The mechanism for actuating the typewriter. A solution of these problems constitutes the development of a phonetic typewriter. Thus, it will be seen that the development of a phonetic typewriter involves many interrelated problems in diverse fields.

The particular form in which the words are typed is an important aspect of a phonetic typewriter because it involves two interrelated considerations, namely, the complexity of the machine and the fidelity of the typed material. The form in which the words are typed may be divided into many categories, as for example, phonetic, syllables or words. The phonetic system requires a smaller memory in the machine, but results in an output which requires a knowledge of phonetic symbols similar to the form used in stenography in order to interpret the output. In the other systems, the form may range from recognizable syllables to complete words typed by employing conventional letter symbols. Considering both intelligibility, spelling accuracy and the memory system, the syllable type of presentation shows the greatest promise. In view of this, the phonetic typewriter described in this section was directed toward the objective of employing and evolving a syllable system.

The model of the phonetic typewriter which has been developed consists of the following elements: a microphone, an electronic compressor, analyzers, a visual display, encoders, memories, decoders, a typewriter actuating mechanism and a conventional typewriter. A schematic diagram of an elementary system which converts the sounds of speech into the corresponding words typed on a page of paper is shown in Fig. 12.58.

The speech sounds are picked up by the microphone and converted into the corresponding electrical variations. The output of the microphone is amplified and fed to the compressor.

In the course of normal speech, as used in dictation, the amplitude level may vary over wide limits. Since the analysis is a function of the amplitude level, some means must be provided to maintain a relatively constant output level over a wide range of input levels. This can be accomplished by the use of an electronic volume compressor. An electronic volume compressor is a system which reduces the amplification of a vacuum tube amplifier when the signal being amplified is relatively large and increases the amplification when the signal is relatively small. The output of the compressor is amplified and fed to the analyzing system.

A system for analyzing and separating the sounds of speech into discrete categories constitutes one of the basic elements of any machine for the

control of typing by the sounds of speech. The analyzer shown in Fig. 12.58 is an amplitude, frequency and time system. The analyzer follows the compressor amplifier. The input to the analyzer passes through a high-pass filter to an amplifier. The output of the amplifier is fed to the inputs of the amplifiers in channels 1 to 9. The channel amplifiers 1 to 8 are
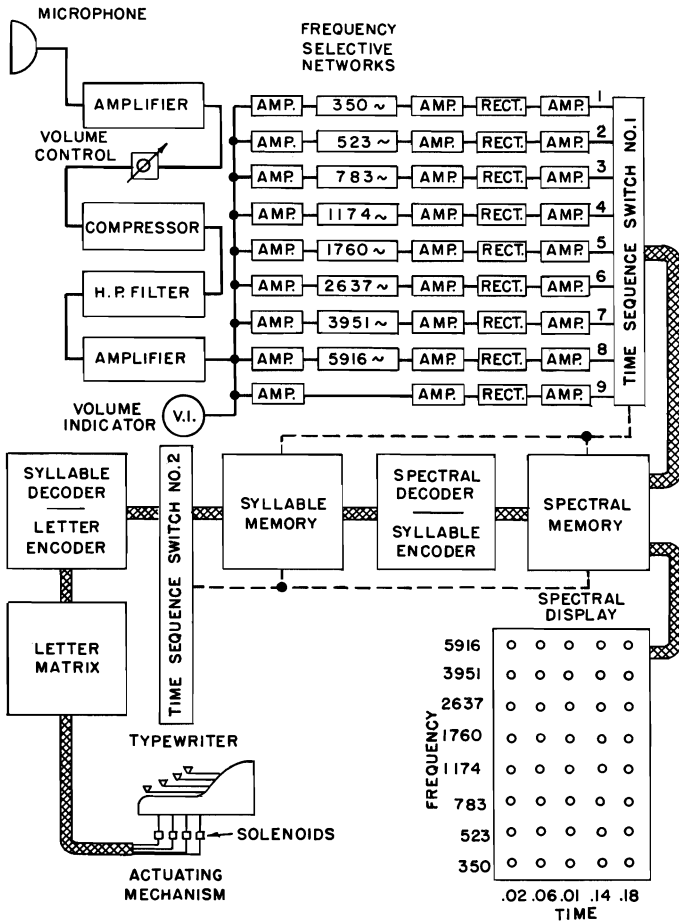
FIG. 12.58. Schematic diagram of the phonetic typewriter.

coupled to frequency selective networks. The output of each frequency selective network is followed by another stage of amplification, a rectifier and a direct current amplifier. The outputs of the direct current amplifiers are used to actuate the relays in the spectral memory system. Channels 1 to 8 inclusive are similar, except for different constants in the frequency selective networks. Each of the frequency selective networks covers a narrow frequency band. There are no frequency selective circuits in

channel 9 and the response frequency characteristic is practically flat. Channel 9 is used to activate the system.

The outputs of channels 1 to 9 inclusive are fed to the input of the rotary sequence switch designated as time sequence switch No. 1. Time is divided into five intervals. The output of channel 9 actuates the stepper in the rotary sequence switch through a relay which sets in motion the ten wiper arms of the switch. The tenth circuit of the switch is used for sequence of operation functions. The wipers make contact with the terminals 1 to 40 inclusive in the time interval 0 to .20 seconds. The wipers stay in contact with terminals 1 to 8 for the time interval 0 to .04 seconds, terminals 9 to 16 for the time interval .04 to .08 seconds, terminals 17 to 24 for the time interval .08 to .12 seconds, terminals 25 to 32 for the time interval .12 to .16 seconds, and terminals 33 to 40 for the time interval .16 to .20 seconds.

The spectral display shown in Fig. 12.58 is connected to the relays of the spectral memory and thereby indicates the relays which have been actuated by the sound. The visible spectral display is useful for analyzing and establishing the code for the system.

The output of the relays of the spectral memory are connected to the spectral decoder and syllable encoder in the form of terminal boards. The output of the syllable encoder is connected to the input of the syllable memory.

The syllable memory consists of seven relays. Each relay corresponds to a syllable, which means that the memory can store seven syllables. In another experiment, a memory of ten syllables was used. The output of the syllable encoder is fed to the syllable memory. When the syllable corresponding to this code is spoken into the microphone, the corresponding relay in the syllable memory will close and stay closed until it is released following the typing of the syllable. The output of the syllable memory is fed to the syllable decoder.

The syllable decoder, letter encoder and letter matrix consist of a rotary selector switch, designated as time sequence switch No. 2, and a 26 × 40 matrix. The output of the syllable memory is connected to the input of the time sequence switch No. 2. This element of the sequence switch constitutes the syllable encoder. The sequence switch can be connected to the matrix so that up to five letters will be typed for one sweep of the wipers. This element of the sequence switch and matrix constitutes the letter encoder. The output of the matrix is connected to the 26 solenoids of the electrical actuating mechanism of the typewriter.

The elements of the phonetic typewriter have been described above. The operation of the phonetic typewriter will now be described. To illustrate the operation, it will be assumed that the vowel sound I is spoken into the microphone. The output of the microphone is amplified by the preamplifier and the compressor amplifier and fed to the frequency selective amplifiers, designated as channels 1 to 8 inclusive and the wide band amplifier designated as channel 9. These outputs are fed to the time sequence switch No. 1. The output of channel 9 starts the time sequence switch No. 1 when the level of

the sound attains a certain specified level. The wipers of the sequence switch start and pass the terminals 1 to 40 in the sequence. The spectrum display for the vowel I will be as shown on the spectral display. That is, the relays corresponding to these lights will be actuated. The output of the spectral memory is connected for the code of the spectral display to the spectral decoder. The syllable is encoded and connected to the syllable memory. The switching circuit of relay 1 of the syllable memory is connected to channel 1 of the syllable decoder. When the switching circuit of relay 1 of the syllable memory closes, it activates the time sequence switch No. 1.

The elements of the matrix are connected so that the circuit corresponding to the letter I will be closed through the actuating mechanism which is
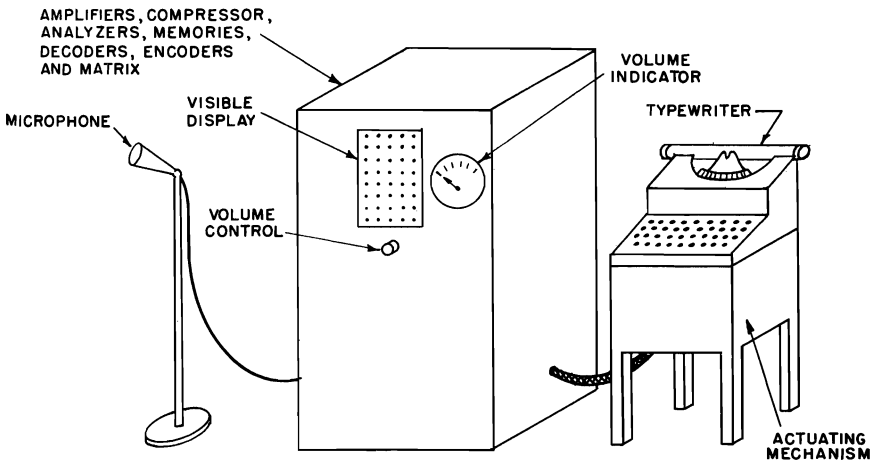
FIG. 12.59.   Perspective view of the phonetic typewriter.

connected to the key of the letter I of a conventional typewriter. A D.C. current is sent through the solenoid of the actuating mechanism which causes the letter I to be typed. Following this, the time sequence switch No. 2 returns to the start position, the relays of the spectral memory are all released, the relay of the syllable memory is released and the time sequence switch No. 1 returns to the start position and the system is ready for the next syllable.

The elementary development model of the phonetic typewriter is shown in Fig. 12.59. The cabinet houses all the equipment except the microphone and typewriter. The spectrum display, volume control and volume indicator are visible in the cabinet.

Two series of words were selected for use in the performance tests of the elementary version of the phonetic typewriter. For the seven syllable memory the following words were used: I - can - see - you - type - this - now. For the ten syllable memory the following words were used: are - see - a - I -

can - you - read - it - so - sir. In either of these series it is possible to arrange the words in various permutations to make up a large number of intelligible sentences. When the code is set up for a particular person, quite reliable performance can be obtained if care is used to insure clear enunciation of the words spoken into the microphone. For example, using several hundred series of words, the typing accuracy was such that 98 per cent of the words were typed correctly.

The simplified version of the phonetic typewriter which has been described will type up to ten syllables. This machine demonstrates and illustrates the principles involved in the electronic and mechanical translation of the sounds of the voice into the corresponding typed syllables.

13

COMPLETE SOUND REPRODUCING SYSTEMS

13.1. Introduction.[1]—The reproduction of sound is the process of picking up sound at one point, and reproducing it at the same point, or at some other point either at the same time, or at some subsequent time. The most common sound reproducing systems are the telephone, the phonograph, the radio, the sound motion pictures, and television.

The telephone is the oldest sound reproducing system. There are 55,000,000 telephones in this country. In other words, almost every family has a telephone instrument, which means that any person can talk to any other person in the matter of a very short interval of time.

The phonograph was the first sound reproducing system which made it possible for all the people of the world to hear statesmen, orators, actors, orchestras, and bands when previously only a relatively few could hear them at first hand. The phonograph is used in every country and clime. In this country there are 40,000,000 phonographs. The popularity of the phonograph is due to the fact that the individual can select any type of information or entertainment and reproduce it whenever he wants it.

The radio, like the phonograph, is a home-type instrument. There are 3200 broadcasting stations, 125,000,000 home-type receivers, and 35,000,000 automobile receivers in this country. Practically every family owns a radio receiver, which means that they can hear famous persons in the government, in the industries, in the schools, in the theater, and in all manner of musical aggregations.

The addition of sound to the motion picture made this type of expression complete. It was the first time that picture and sound were synchronized and reproduced at the same time. There are 18,000 motion-picture theaters in this country. The average weekly attendance is 90,000,000 persons.

Television is the latest system in which picture and sound are reproduced at the same time. Sound is, of course, important to television, because without it, it would be the same as the silent motion picture. There are 450 television transmitters and 36,000,000 television receivers in this country. In addition to the telephone, phonograph, radio, sound motion picture, and television, there are many other sound reproducing systems. These include,

---

[1] Olson, H. F., "Review of 25 Years of Sound Reproduction," *Jour. Acous. Soc. Amer.*, Vol. 26, No. 5, p. 637, 1954.

sound systems,[2] magnetic tape reproducers, hearing aids, and dictating machines.

The radio, phonograph, sound motion picture, and television have made it possible for all the people of the world to hear famous statesmen, artists, actors, and musical aggregations where only a relatively small number had been able to hear them first hand.   It is evident that the reproduction of sound has produced in a relatively short time a great change in the education and entertainment of this and other countries.   The impact of the telephone, phonograph, radio broadcasting, sound motion pictures, and television upon the dissemination of information, art, and culture has been tremendous.   The reproduction of sound in these fields has been as important to the advancement of knowledge as the printing press and the printed page.

It is the purpose of this chapter to describe complete sound reproducing systems.

13.2. Distortion and Noise Considerations.—The two factors which limit and change the reproduced sound in a sound reproducing system are noise and distortion.   Noise places a lower limit on the amplitude range of the signal.   Distortion changes the nature and form of the reproduced sound from that of the original sound.   It is the purpose of this section to present a generalized consideration of noise and distortion.

A complete sound reproduction system may be represented as shown in Fig. 13.1.   The first element in the chain is the information source which produces the original sound which is sent to the recorder or transmitter. The original sound contains the ambient noise which occurs in the local environment of the source of sound.   The original sound is recorded or transmitted by means of a recorder or transmitter.   The recorder or transmitter adds the inherent noise in these systems to the signal which is recorded or transmitted.   In the case of recording the signal there will be a storage medium and in the case of the transmitting system there will be a transmission medium.   Both of these mediums add noise inherent in these elements. The signal is reproduced at the ultimate destination by a reproducer or receiver.   This reproducer and receiver add noise.   The final link in the chain is the information destination.   In a sound reproduction system the ultimate useful destination of all reproduced sound is the human ear.

Referring to the system of Fig. 13.1, it will be seen that there are two important parameters involved in the transmission, namely, signal and noise.   The capacity C for the transmission of information of a sound reproducing system is given by

$$C = W \log \frac{P + N}{N} \qquad 13.1$$

where W = band width, in cycles,
P = power of the signal, in watts, and
N = power of the noise (white noise), in watts.

---

[2] Sound systems include: public address, sound reinforcing, electric megaphone, intercommunicating, announce, paging, and call systems.

With proper precautions or modifications, equation 13.1 may be applied to all systems for the reproduction of sound. Equation 13.1 outlines the fundamental philosophy relating to signal and noise in sound reproducing systems. The capacity for the transmission of information is proportional to the frequency band width and a logarithmic function of the signal and noise.

In the reproduction of sound, distortion is a term used to designate any departure in the form or nature of reproduced sound from that of the original
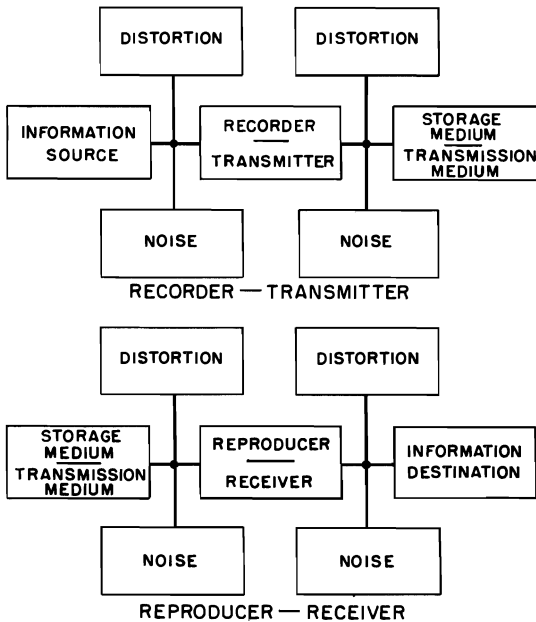
FIG. 13.1. A schematic diagram of the elements of a generalized communication system consisting of a transmitting or recording section and a receiving or reproducing section.

sound. Distortion is produced in some degree by all of the elements used in the reproduction of sound. The distortions in sound reproducing systems are depicted in a generalized fashion in Fig. 13.1. The first element in the chain in this system is the information source which produces the original sound that is sent to the recorder or transmitter. To the original sound is added the distortion introduced by the local environment of the sound source. The resultant sound is recorded or transmitted by means of a recorder or transmitter. The recorder or transmitter adds the distortion, which is produced in these systems, to the signal that is recorded or transmitted. In the case of recording the signal there will be a storage medium, and in the case of the transmitting system there will be a transmission medium. Both of these mediums add distortion inherent in these elements.

The signal is reproduced at the ultimate destination by a reproducer or receiver. The receiver or reproducer adds distortion. The final link in the system is the information destination. In a sound reproducing system the ultimate useful destination of reproduced sound is the human ear. The distortions in sound reproducing systems may range from imperceptible, perceptible, noticeable, tolerable to objectionable. Thus, it will be seen that, although distortions occur in all reproducing systems, the subjective aspects of the distortion determine the acceptability of the system from the standpoint of distortion. The distortions which occur in the elements of sound reproducing systems have been considered in detail in this book. Since distortion is a fundamental characteristic of all sound reproducing systems, it seems appropriate to outline and describe these distortions in this generalized consideration of sound reproducing systems.

The distortions which occur in a sound reproducing system are amplitude, spacial, nonlinear, phase, and transient. These distortions are depicted by the following characteristics, namely, response frequency, directional, nonlinear, phase, and transient.

The response frequency characteristic provides data on relationship between the output and input as a function of the frequency. The deviation from a constant ratio of the output to input as a function of frequency is a measure of the amplitude distortion or frequency discrimination.

The directional characteristic provides data on the response of an acoustical transmitting or receiving system as a function of the angle with respect to some axis of the system. The deviation from the desired directivity pattern is a measure of the spacial distortion.

The nonlinear distortion characteristic provides data on the ratio of the output to the input as a function of the input. The deviation from a constant relationship depicts the nonlinear distortion. The nonlinear distortion may also be described in terms of the generation of spurious components in the output which are not present in the input.

The phase frequency characteristic provides data on the phase angle between the input and output as a function of the frequency. The deviation from a constant phase angle is a measure of the phase distortion.

The transient response characteristic provides data on the response of a system to a sudden change in the input. The deviation in the envelope of the wave output to the envelope of wave input is a measure of the transient distortion.

The source, mechanism, and measurement of distortion in the different elements of a sound reproducing system from the source of sound to the ultimate useful destination of all reproduced sound, the human ear, have been discussed in detail in other parts of this book. Since the ear is the ultimate destination of reproduced sound, the considerations include the important subjective aspects of distortion.

13.3. Sound Systems.—Sound systems is a term used to designate all manner of sound reproducing systems as follows: public address, sound reinforcing, electric megaphone, intercommunicating, announce, paging,

and call systems.   In general, most of these systems involve microphones, amplifiers, and loudspeakers arranged in different ways to accomplish the desired results.   Public address systems are used to increase the coverage of the voice.   Sound reinforcing systems are used to augment the output of any sound source.   Electric megaphones are used as speech reinforcing
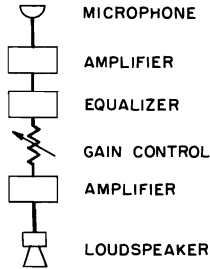
Fig. 13.2.   Schematic arrangement of the apparatus in a simple sound reproducing system.

systems.   Intercommunicating systems are used to communicate between different locations by means of loudspeaking systems.   Paging and call systems are used to cover many locations by means of loudspeaking systems. The equipment for the different applications involve electroacoustic transducers with widely differing characteristics.   In some systems, loudspeakers having an efficiency of over 50 per cent and capable of handling kilowatts
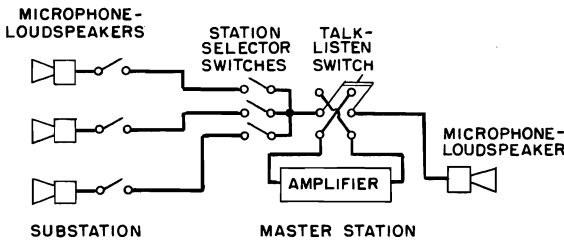
FIG. 13.3.   Schematic arrangement of the apparatus in an intercommunicating system.

input have been developed and commercialized.   Gradient microphones have been developed and commercialized that are capable of picking up intelligible speech in the presence of an ambient noise level of over 100 decibels.   Practically every school, church, theater, office, store, warehouse, and factory in the country employs some type of sound system.

A. *Monaural Sound System.*—The simplest sound reproducing system consists of a microphone, an amplifier, a gain control, and a loudspeaker, as shown in Fig. 13.2.   This arrangement is used in the simplest sound system.

Intercommunicating systems are extensions of the simple system of Fig. 13.2 and are used to communicate between a central location and one or more other stations. A schematic diagram of an intercommunicating system is shown in Fig. 13.3. The master unit contains an amplifier, microphone-loudspeaker, station selector, and talk-listen switch. The remote unit consists of a loudspeaker-microphone and a talk-listen switch. In the simplest systems, the small loudspeaker unit with suitable compensation is also used as the microphone, as shown in Fig. 13.2. Additional stations and appropriate switch systems may be added for communication between a number of rooms. A perspective view of the elements of an intercommunicating system is shown in Fig. 13.4.

MASTER STATION

SUBSTATIONS
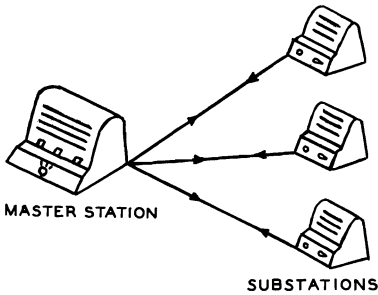
FIG. 13.4. Perspective view of the apparatus in an intercommunicating system.

The elements of an elaborate sound system, shown in Fig. 13.5, consists of one or more microphones, and voltage amplifiers, for sound pickup, record
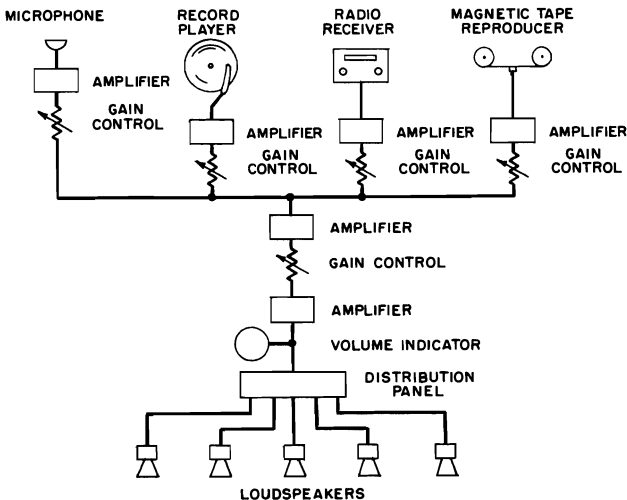
FIG. 13.5. Schematic arrangement of the apparatus in a sound reproducing system.

turntable for reproducing phonograph records, a radio receiver for providing radio program material, a magnetic tape recorder for recording and reproducing magnetic tape records, separate gain controls, amplifiers, over-all gain control, distribution panel, and loudspeakers. A perspective view of

the system of Fig. 13.5 is shown in Fig. 13.6.   The system depicted in Figs. 13.5 and 13.6 is used for sound reinforcement, public address, announce, paging and numerous other applications in theaters, churches, auditoriums,
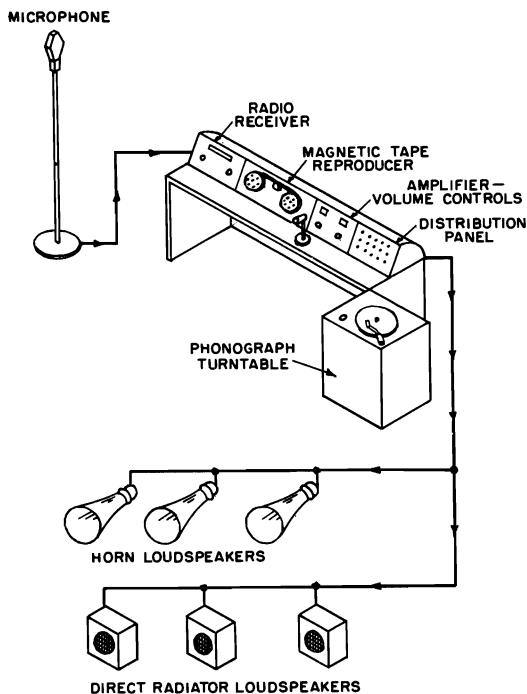
FIG. 13.6.   Perspective view of the apparatus in a sound reproducing system.

factories, offices, restaurants, outdoor theaters, mass meetings, athletic events, etc.

B. *Binaural Sound Reproducing System.*—An ideal binaural sound reproducing system[3] is shown schematically in Fig. 13.7 which indicates that the desired objective is obtained by effectively transferring the auditor to the point of scenic action through the intermediary of a double recording and reproducing channel.   Two microphones simulate the ears of a dummy, each receiving the component of the original sound that would normally be received were the dummy a human being.   Each component is reproduced through a separate audio channel, each channel terminating in a high-quality telephone receiver.   Each of the receivers is placed on the proper ear by the auditor and the sound produced in each of his ears will be identical to what would have been produced at the original set had he been there at the time.

[3] Olson and Massa, *Jour. Soc. Mot. Pic. Eng.*, Vol. 23, No. 2, p. 63, 1934.

MICROPHONES

DUMMY

AMPLIFIERS

GAIN CONTROLS

AMPLIFIERS

TELEPHONE
RECEIVERS

LISTENER
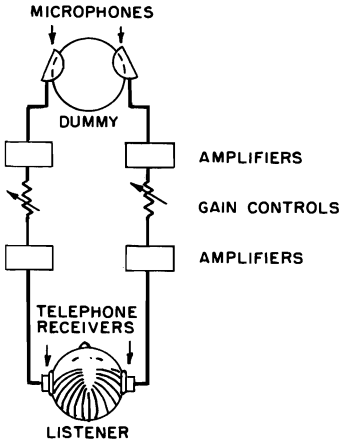
FIG. 13.7. Schematic arrange-
ment of the apparatus in a binaural
sound reproducing system.

The advantages of this system are quite obvious; the binaural effect is practically perfect, and the reverberation characteristic of the set (which should be designed to conform to the scene) is transferred unadulterated to the listener.

The binaural system may be used in subjective investigations of noise. As one example, a binaural tape recording system is used to record motor car noise while the machine is in motion. See Sec. 13.5C. Later the recording can be reproduced in the laboratory. The subjective effects of changes in noise due to alterations in the motor car can be evaluated by A—B comparisons. Such direct comparisons could not be made in the field because some of the changes in the motor car require several hours and it is impossible to carry the comparisons in mind for this period.

ORCHESTRA

CONDUCTOR

MICROPHONES   SET

AMPLIERS

GAIN CONTROLS

AMPLIFIERS

STAGE

LOUDSPEAKERS
THEATER
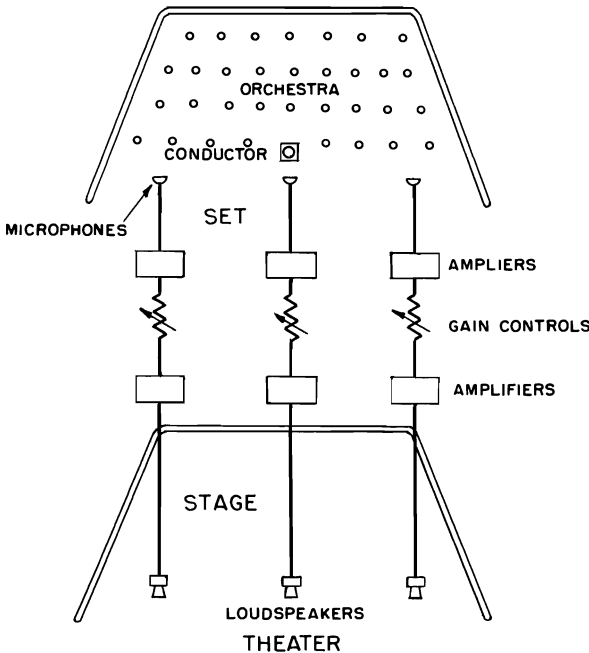
FIG. 13.8. Schematic arrangement of the apparatus in an auditory perspective sound reproducing system.

4 Apps, David C., *Jour. Acous. Soc. Amer.*, Vol. 24, No. 6, p. 660, 1952.

C. *Auditory Perspective System.*[5]—In the binaural reproducing system the ears of the auditor are effectively transferred to the original scene of action. A system for effectively transferring the original sources of sound from the studio to the theater stage is shown in Fig. 13.8. The sound is picked up by three microphones and amplified in separate channels, each channel feeding a separate loudspeaker. The three loudspeakers are arranged on the stage in the same positions as the microphones on the pickup stage. The system consisting of three channels is, of course, a compromise. To give true stereophonic reproduction an infinite number of channels must be used. However, excellent results have been obtained in sound motion-picture reproduction employing three channels in magnetic tape sound reproduction, and in phonograph record reproduction employing two channels. These systems will be described in greater detail in Secs. 13.5B, 13.6B, and 13.7B.
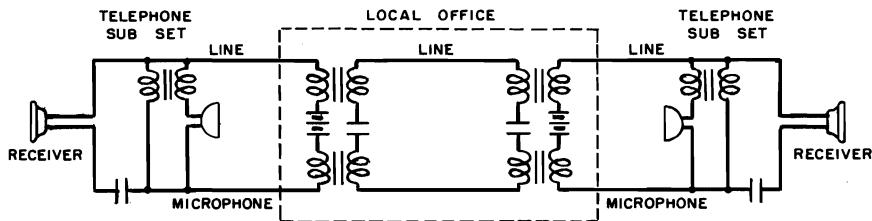
FIG. 13.9.   Schematic arrangement of the apparatus in a telephone system.

13.4. Telephone.—The telephone is a sound reproducing system consisting of a carbon microphone (sometimes termed a transmitter), a telephone receiver, and a battery. Carbon microphones used in telephony have been described in Sec. 8.2A. Telephone receivers used in telephony have been described in Sec. 9.2A. A schematic diagram of a telephone system is shown in Fig. 13.9. A modern subscriber dial telephone set is shown in Fig. 13.10. Each telephone station is connected by a line to the central office. The battery and ringing supplies and switching and signal equipment are located in the local or central office. The function of the central office is to connect any subscriber to any other subscriber. In large cities, there are many central offices, because it is not economical or practical for a central office to serve more than 10,000 subscriber stations. The local
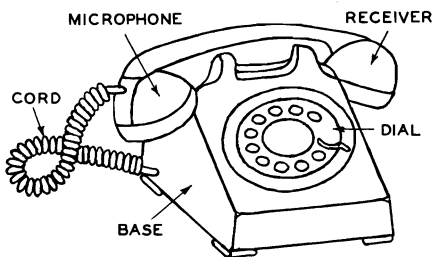
FIG. 13.10.   Perspective view of a subscriber dial telephone set.

[5] A complete discussion of the various factors involved in stereophonic sound reproduction is given in the *Journal of the Society of Motion Picture and Television Engineers,* Vol. 61, No. 3, Part II.

offices are interconnected by lines, as shown in Fig. 13.11. In local transmission the electrical output of the microphone fed to the input to the telephone receiver is sufficient for the telephone receiver to generate sound of ample loudness for intelligent transmission of speech. In long distance telephony, vacuum tube repeaters are used at regular intervals to restore the level of transmission to a normal value. The system shown in Fig. 13.11 depicts the electroacoustic elements of a telephone system. In addition, equipment must be supplied for the subscriber to signal the operator and to
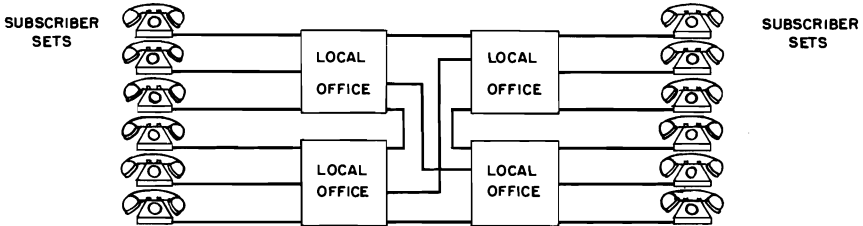
Fig. 13.11. Schematic arrangement of a telephone system with four offices and interconnecting lines.

permit the operator to send ringing current to the subscriber station. 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.

13.5. Magnetic Tape Sound Reproducing System.—The use of a moving magnetic wire as a means for reproducing audio signals was demonstrated over a half century ago. However, it is only within the last decade that theories, techniques, and materials have been developed which make it possible to reproduce audio signals with a satisfactory performance from the standpoint of frequency range, signal-to-noise, and distortion. Although satisfactory performance was obtained from wire, magnetic tape consisting of a plastic base coated with a thin layer of iron oxide is now almost universally used in magnetic sound reproducers.

A. *Monaural Magnetic Tape Sound Reproducing System.*—A complete monaural magnetic tape sound recording and reproducing system is shown in Fig. 13.12. The magnetic recorder system may be used to record in a studio, a room in the home, an office, a factory, etc., and from a signal input, as for example, a radio receiver. The magnetic recorder has been described in Sec. 9.7. Fig. 13.12 depicts all the elements of a magnetic tape sound recording and reproducing system. The output of the microphone or signal source is amplified and sent through an equalizer to compensate for the characteristics of the magnetic head and magnetic tape. The magnetic recording head, actuated by the amplifier, magnetizes the magnetic coating on the tape in a pattern which corresponds to the undulations in the original sound wave. That is, the magnetic flux pattern of the recorded tape consists of a series of magnetized sections. To overcome the nonlinearity of the

tape or wire, a high-frequency signal, termed a bias, is fed to the recording head together with the audio signal. The signal level fed to the recording head is governed by the volume control and is measured by the volume

HOME, STUDIO, OFFICE, ETC.
ACOUSTICS OF STUDIO
MICROPHONES
SIGNAL INPUT
AMPLIFIERS
MIXERS
AMPLIFIER
EQUALIZER
ATTENUATOR
AMPLIFIER
VOLUME INDICATOR
HIGH FREQ. OSCILLATOR
ERASING HEAD
RECORDING HEAD
MAGNETIC TAPE RECORD

MONITOR
ACOUSTICS OF
MONITORING ROOM
LOUDSPEAKER
AMPLIFIER
ATTENUATOR
EQUALIZER
VOLUME INDICATOR

MAGNETIC TAPE RECORDS ON REELS

HOME, OFFICE, STUDIO, ETC.
MAGNETIC TAPE RECORD
REPRODUCING HEAD
AMPLIFIER
EQUALIZER
VOLUME CONTROL
VOLUME INDICATOR
AMPLIFIER
LOUDSPEAKER
ACOUSTICS OF ROOM

FIG. 13.12. Schematic arrangement of the apparatus in a complete magnetic tape sound recording and reproducing system.

indicator. When the tape is reproduced, the magnetized sections move past the head and produce a magnetic field in the head which corresponds to the pattern on the tape. As a result, the electrical signal developed by the reproducing head corresponds to the electrical signal applied to the recording head when the signal was recorded on the tape. The output of the reproducing head is amplified, equalized, and fed to the loudspeaker.

The signal level fed to the loudspeaker is governed by the volume control and is measured by the volume indicator.

Perspective views of complete monaural magnetic tape sound reproducing systems are shown in Fig. 13.13. The studio or professional magnetic recorder and reproducer consists of microphones, monitoring console, monitoring loudspeaker, and rack or console-type magnetic recorder or reproducer. The system is used for professional studio recording on magnetic tape. Portable-type magnetic tape recorders are shown in Fig. 13.13.
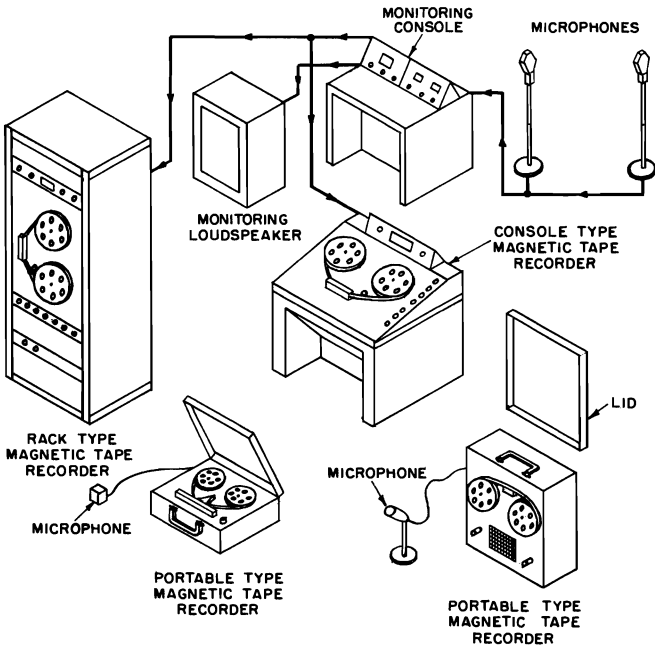
FIG. 13.13.   Perspective views of magnetic tape recorders.

The complete system for recording and reproducing by means of magnetic tape is housed in a single cabinet. The portable tape recorders are used for recording all types of sounds in homes, offices, outdoors, etc.

B. *Stereophonic Magnetic Tape Sound Reproducing System.*—A complete two-channel stereophonic magnetic tape sound reproducing system is shown in Fig. 13.14. Each channel consists of the elements of the system shown in Figs. 13.12 and 13.13. In recording an orchestra or any other sound source spread out or moving over an area, the sound is picked up by means of two microphones in channels A and B. The distance between the microphones and the distance from the microphones to the sources of sound are selected to provide the most realistic reproduction of sound when the

loudspeakers are separated by a distance of about ten feet in an average living room.

Prerecorded magnetic tapes both single and stereophonic have been commercialized and sold in the manner of disk phonograph records.
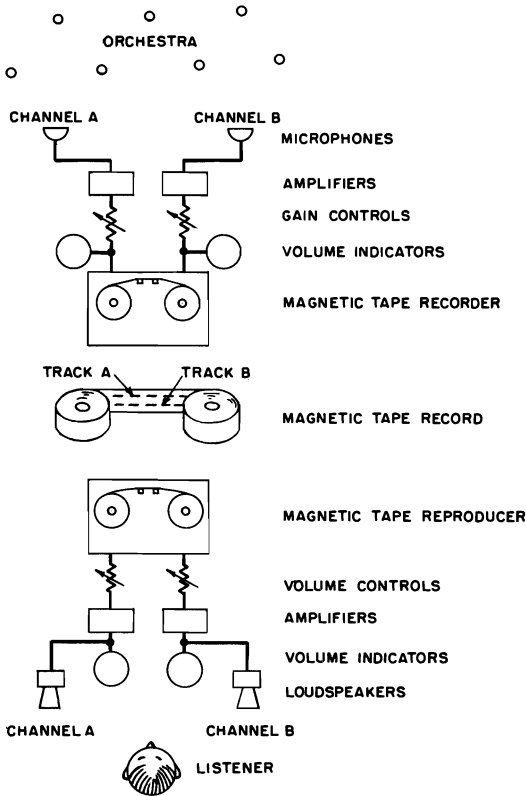
FIG. 13.14.   Schematic arrangement of the apparatus in a complete stereophonic magnetic tape sound reproducing system.

C. *Binaural Magnetic Tape Sound Reproducing System.*—The system shown in Fig. 13.14 may be employed as a binaural sound reproducing system by mounting the two microphones in a dummy head, as shown in Fig. 13.7, and substituting two telephone receivers for the loudspeaker.

Binaural sound reproducing systems have not been commercialized on a wide scale because a set of earphones must be worn throughout the rendition of the program.

The advantage of the system is that the binaural effect is practically perfect with the result that the listener is in effect acoustically transferred

to the point of sound pickup. For this reason the binaural sound reproducing is particularly useful for subjective investigations.

13.6. Disk Phonograph Reproducing System.—A. *Monaural Disk Phonograph Reproducing System.*—A complete disk recording system is shown in Fig. 13.15. The first element is the acoustics of the studio. The factors which influence the collection of sound have been discussed in Sec. 11.3. The output of each microphone is amplified and fed to a mixer. If more than one microphone is used, as for example, a soloist accompanying an orchestra, one microphone for the singer and one for the orchestra; the outputs of the two may be adjusted for the proper balance by means of the mixers. A compressor may be used to reduce a large amplitude range to that suitable for reproduction in the home. An equalizer provides the recording characteristic described in Sec. 9.3A4. The gain control controls
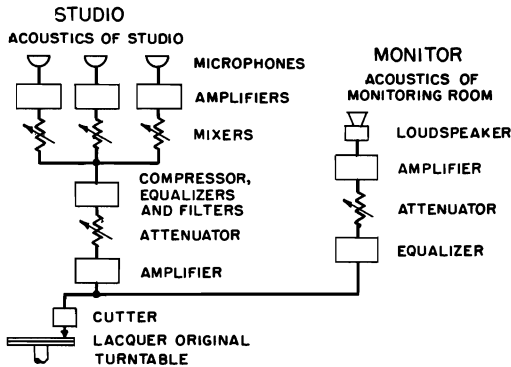
FIG. 13.15. Schematic arrangement of the apparatus in a complete disk sound recording system.

the over-all signal level. The cutter actuated by the amplifier cuts a wavy path in the revolving record corresponding to the undulations in the original sound wave striking the microphone. A minitoring system is used to control the recording operation.

During the past decade, in the recording of sound for phonograph disk records, the direct recording of the sound on the lacquer original has been replaced by first recording on magnetic tape by the system shown in Fig. 13.16. The magnetic record thus produced is then reproduced and the output used to record the lacquer original, as shown in Fig. 13.16. The recording and rerecording procedures are similar to that of the direct recording. There are many advantages in the use of magnetic tape as the master as follows: The tape can be edited and monitored to obtain the best lacquer original. Any number of similar lacquer originals can be obtained from the tape. The tape does not deteriorate with playing.

The processes in the record plant for the mass production of records is depicted in Fig. 13.17. The original lacquer disk termed the "original"

of Fig. 13.17A 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. 13.17B). The master is electroplated. This plating is separated from the master and reinforced by backing
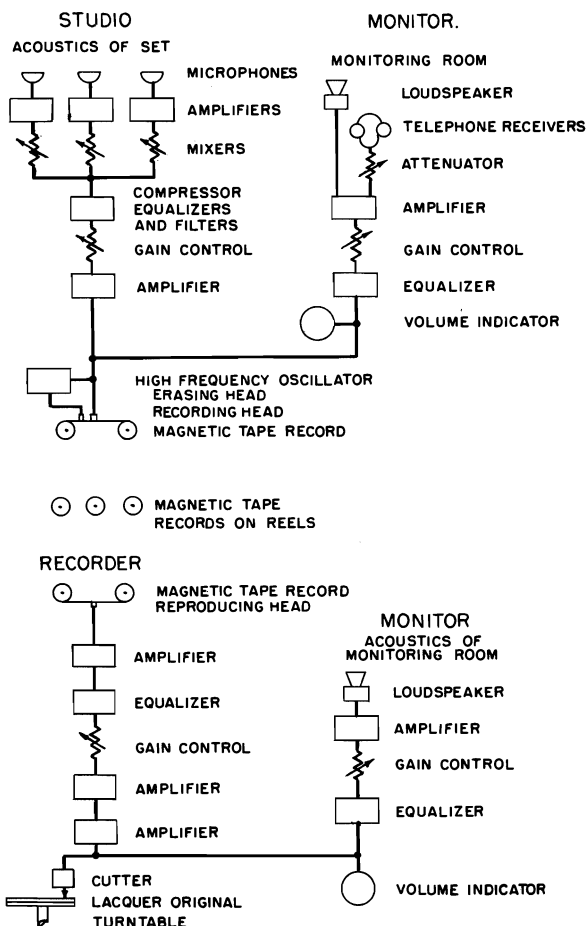
FIG. 13.16. Schematic arrangement of the apparatus in a complete magnetic tape sound recording and reproducing system and disk sound recording system.

with a solid metal plate. The assembly is termed the "mother" (Fig. 13.17C). 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. 13.17D). Several stampers may be made from each mother. One 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
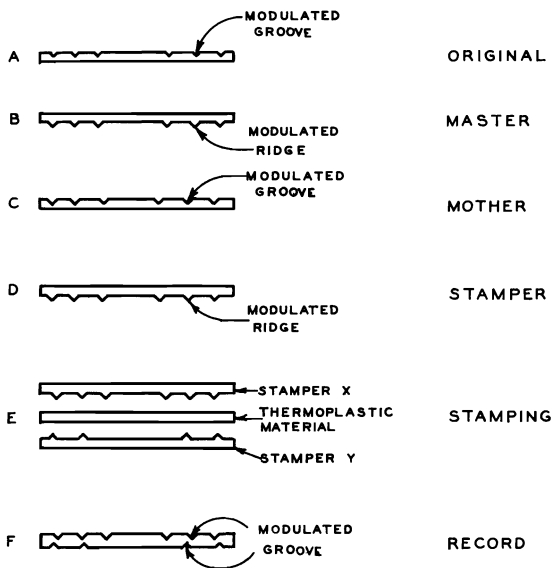
FIG. 13.17. The steps in the process for the mass production of disk phonograph records from the lacquer original.

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. 13.17E).

FIG. 13.18. Schematic arrangement of the apparatus in a complete disk sound reproducing system.

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 disk (Fig. 13.17F). 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 system for the reproduction of phonograph records is shown in Fig. 13.18. The record is turned at constant speed by the turntable. The stylus or needle of the pickup follows the wavy spiral groove in the record and generates a voltage corresponding to the undulations in the record
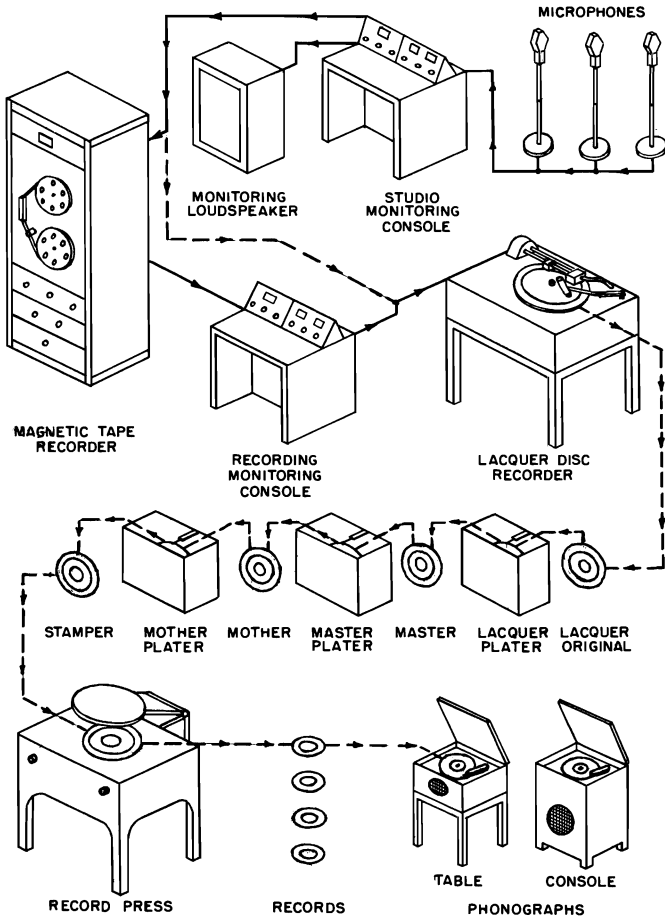
FIG. 13.19. Perspective view of a complete disk phonograph sound reproducing system.

(see Sec. 9.3B). The output of the pickup is amplified by an amplifier. This is followed by an equalizer which compensates for the equalization in the original recording. Filters or tone controls are provided for further equalization of the response according to the taste of the listener. A volume control controls the sound level of reproduction. The volume control is followed by an amplifier that drives the loudspeaker. The action of a

loudspeaker in a living room has been considered in Sec. 11.2R. Perspective views of all of the elements in the disk phonograph sound reproducing system are shown in Fig. 13.19.

B. *Stereophonic Disk Phonograph Sound Reproducing System.*—A complete two-channel stereophonic disk phonograph sound reproducing system is
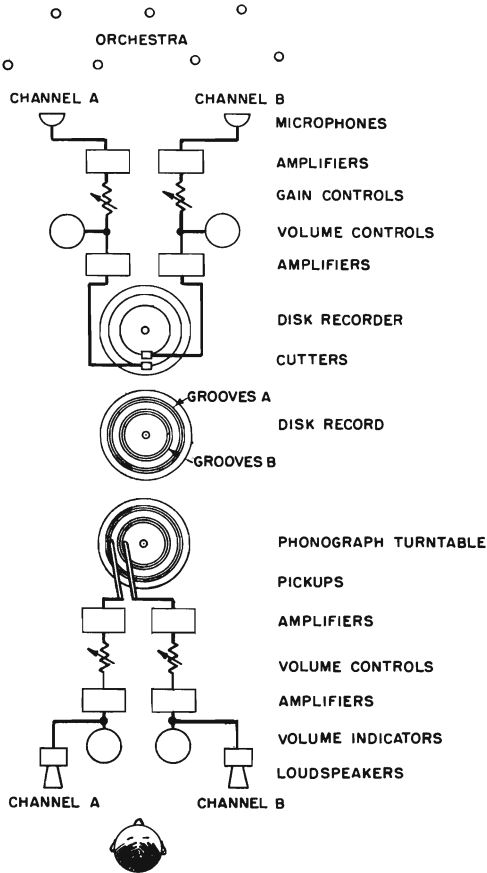
Fig. 13.20. Schematic arrangement of the apparatus in a complete stereophonic disk phonograph sound reproducing system.

shown in Fig. 13.20. Each channel consists of the elements of the system shown in Figs. 13.15, 13.16, 13.17, 13.18, and 13.19. In recording an orchestra or other sound sources spread out over an area or a sound source moving over an area, the sound is picked up by means of two microphones in channels A and B. The distance between the microphones and the distance from the microphones to the sources of sound are selected to provide

the most realistic sound reproduction when the loudspeakers are separated by a distance of about ten feet in an average living room. Two-channel stereophonic disks have been quite extensively commercialized.

13.7. Sound Motion-Picture Reproducing System.[6]—A. *Single-Channel Sound Motion-Picture Reproducing System.*—A complete sound motion-picture recording and reproducing system is shown in Fig. 13.21. The first element is the acoustics of the set. The factors which influence
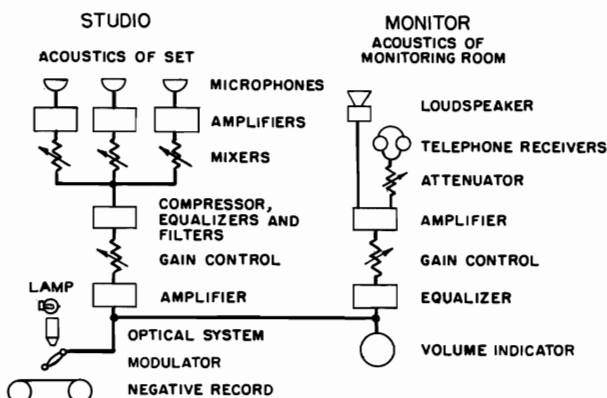
FIG. 13.21. Schematic arrangement of the apparatus in a complete system for sound recording on motion-picture film.

the collection of sound have been discussed in Sec. 11.3. The output of the microphones is amplified and fed to attenuators termed mixers. If more than one microphone is used, as for example, a soloist accompanying an orchestra, one microphone for the singer and one for the orchestra, the output of the two may be adjusted for the proper balance. A low-pass filter is usually used to reduce ground noise above the upper limits of repro-duction. 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 accen-tuate the high frequencies to compensate for the film transfer loss at high frequencies. A compressor is used to reduce the volume range (see Sec. 9.11). The following attenuator controls the over-all volume. The output of the amplifier feeds 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). (See Sec. 9.9B.) The monitoring system is also connected to the output of the recording amplifier. An equalizer is used to adjust the frequency characteristic to simulate that of the ultimate reproduction. If the monitor-

[6] Kellogg, E. W., "History of Motion Pictures," *Jour. Soc. Mot. Pic. Tel. Eng.*, Vol. 64, No. 6, p. 291, 1955; Vol. 64, No. 7, p. 356, 1955; Vol. 64, No. 8, p. 422, 1955.

ing is carried out in a room a loudspeaker is used. When the monitoring and mixing is carried out on the set, headphones are used for monitoring.

During the past decade, in the recording of sound for sound motion pictures the direct optical recording of the sound on film has been replaced by first
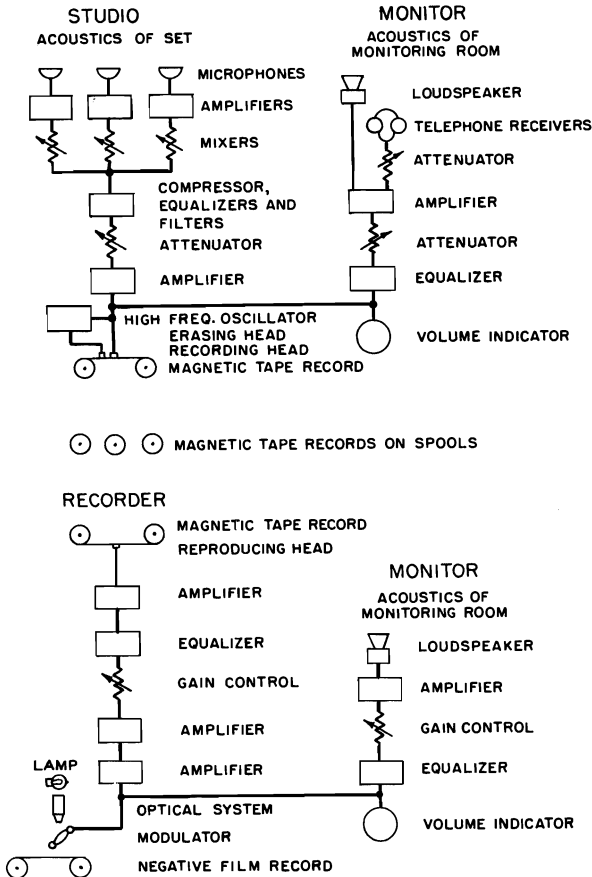
FIG. 13.22. Schematic arrangement of the apparatus in a complete magnetic tape sound recording and reproducing system and film sound recording system.

recording on magnetic tape by the system shown in Fig. 13.22. The procedures in recording on magnetic tape for sound motion pictures are similar to those for recording on photographic film and will not be repeated. The magnetic tape record is reproduced and the output recorded on photographic film by the system shown in Fig. 13.22.

Perspective views of all the elements in a sound motion picture recording system are shown in Fig. 13.23. The motion picture camera is interlocked with the magnetic tape recorder so that synchronism of the sound and picture will be preserved. The magnetic tape recorder is in turn interlocked
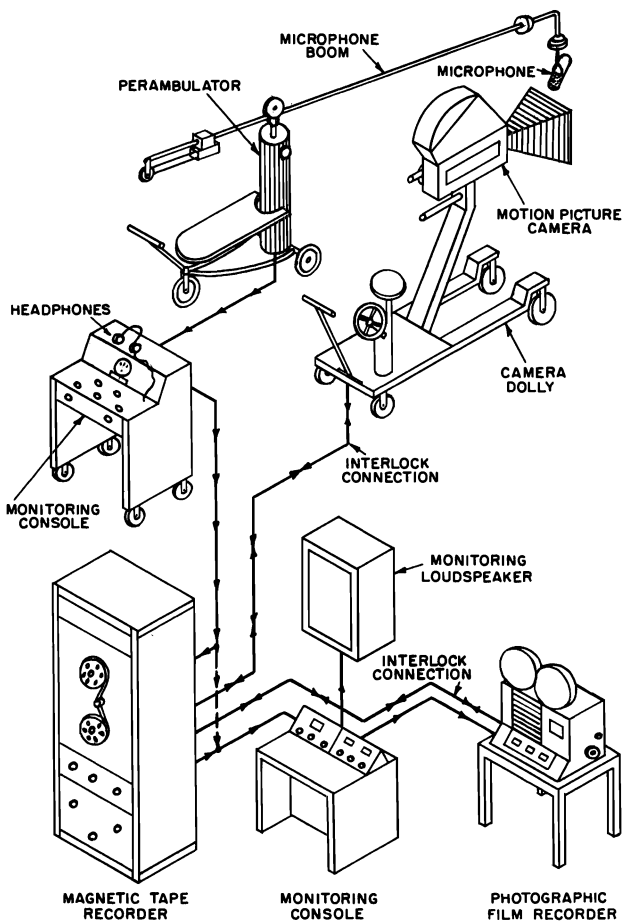
FIG. 13.23. Perspective view of a complete sound motion-picture recording system.

with the photographic film recorder when the sound is rerecorded on photographic film. If the recording is made directly on the photographic film the magnetic tape reproducer is by-passed.

The processes[7] in the film laboratory for the mass production of motion picture positive prints is shown in Fig. 13.24. The negative record is

[7] Crabtree, J. I., *Jour. Soc. Mot. Pic. Tel. Engs.*, Vol. 64, No. 1, p. 13, 1955.

developed as shown in Fig. 13.24. Then the required number of positive prints of both the picture and sound are printed from the negative record as shown in Fig. 13.24. These positive records are developed. These positive records are used for sound reproduction and picture projection in the theater. Fig. 13.25. See Sec. 9.8C1.

The variable density or variable area record is reproduced in the theater by pulling it past a slit illuminated by a light and a suitable optical system. The resultant variations in light, due to the variable density or variable area on the film, fall upon the photoelectric cell and are converted into the corresponding electrical variations (see Sec. 9.9C1). These are then amplified

LABORATORY
PROCESSING — PRINTING

THEATRE

UNDEVELOPED NEGATIVE
RECORD ON FILM

LAMP

OPTICAL SYSTEM

POSITIVE RECORD

NEGATIVE DEVELOPER

PHOTO CELL

AMPLIFIER

DEVELOPED NEGATIVE
RECORD ON FILM

EQUALIZER AND FILTERS

GAIN CONTROL

PRINTER

PRINTED POSITIVE
RECORD ON FILM

AMPLIFIER   ATTENUATOR

MONITOR
LOUDSPEAKER

POSITIVE DEVELOPER

DIVIDING NETWORK

STAGE LOUDSPEAKERS

POSITIVE FILM RECORDS
ON REELS

ACOUSTICS OF THEATRE

FIG. 13.24. The steps in the process for the production of motion-picture positive film from the negative film.
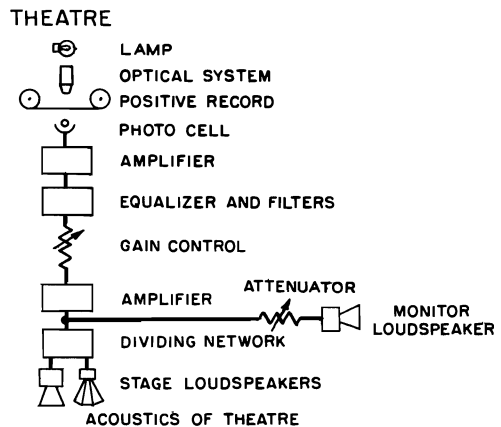
FIG. 13.25. Schematic arrangement of the apparatus in a system for the reproduction of sound from motion-picture film.

and fed to equalizers and filters. A low-pass filter is used to cut out the ground noise due to film above the upper limit of reproduction. An equalizer is used to adjust the frequency characteristic to that suitable for the best reproduction in the theater. The attenuator is used for adjusting the level of reproduction. The output of the power amplifier feeds the stage loudspeakers and monitoring loudspeaker. The monitoring loudspeakers and the attenuator are located in the projection booth. As a matter of fact, the entire system, save for the stage loudspeakers, is located in the projection booth. A dividing network and a two-channel loudspeaker system are shown in Fig. 13.25. Of course, any type of suitable loudspeaker described in Chapter VII may be used. The action of a sound motion-picture reproducer in a theater has been discussed in Sec. 11.2G and will not be repeated here.

Perspective views of the elements of a theater sound motion-picture reproducing system are shown in Fig. 13.26.
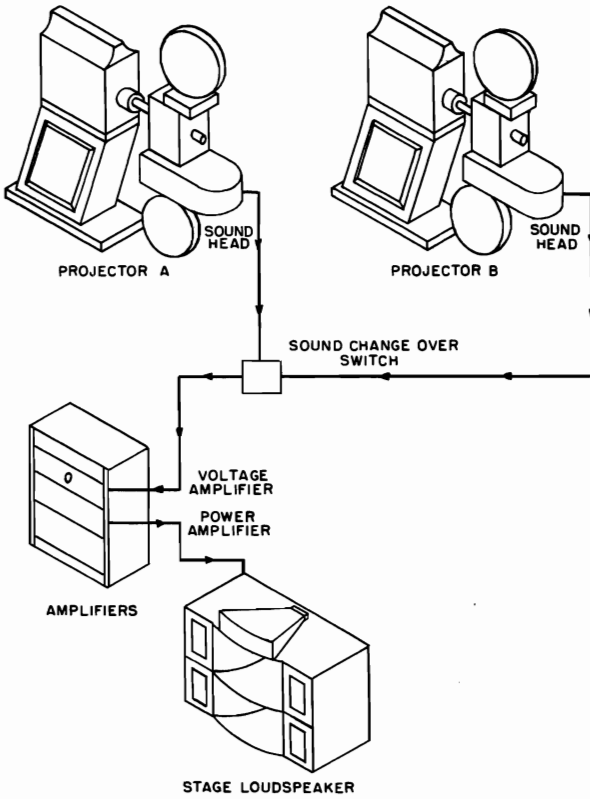
FIG. 13.26. Perspective view of a complete theater-type sound motion-picture reproducing system.
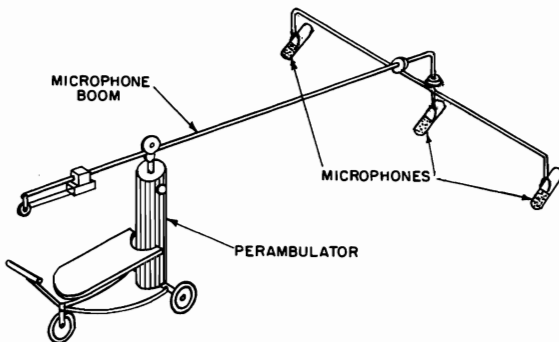
FIG. 13.27. Perspective view of the sound pickup system for recording stereophonic motion-picture sound.

B. *Multiple-Channel Sound Motion-Picture Reproducing System* (*Stereophonic System*).[8]—Stereophonic sound reproduction employing multiple channels has been introduced on wide commercial scale in connection with sound motion pictures. In one system four separate channels are used.
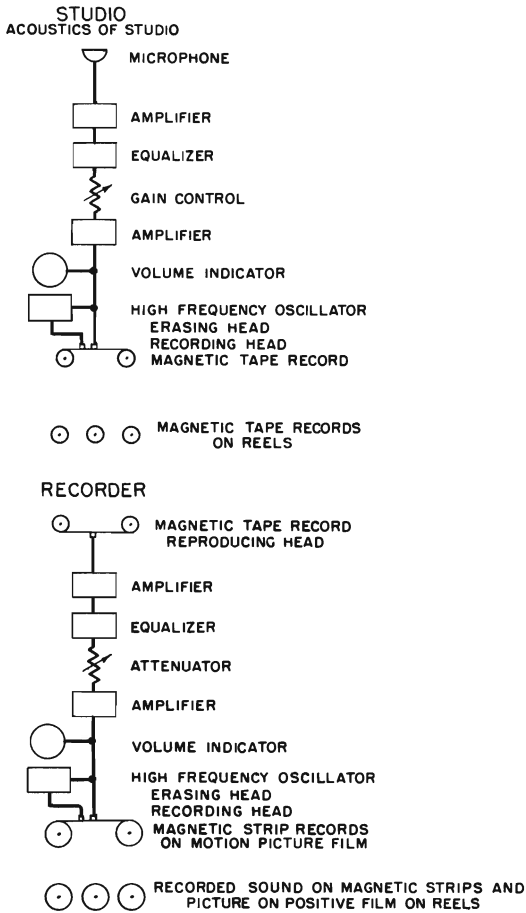
FIG. 13.28. One channel of a stereophonic system for recording and rerecording the sound in a sound motion-picture system.

The information is recorded on four separate strip magnetic tracks cemented to the positive motion picture film (see Sec. 9.10 and Fig. 9.62). Two of the tracks are placed outside the sprocket holes and the other two tracks are placed inside the sprocket holes. Three of the channels are used for

[8] Developments in Stereophony, *Jour. Soc. Mot. Pic. Tel. Eng.*, Vol. 61, No. 3, Part II, 1953. This part contains papers by several authors on stereophonic sound.

stereophonic reproduction. The fourth channel may be used for control or for auxiliary information. In recording three microphones are placed on the boom, as shown in Fig. 13.27. Three channels of the type shown in Fig. 13.28 are used to record the action in perspective. The program material is recorded on magnetic tape, as shown in Fig. 13.28. The informa-
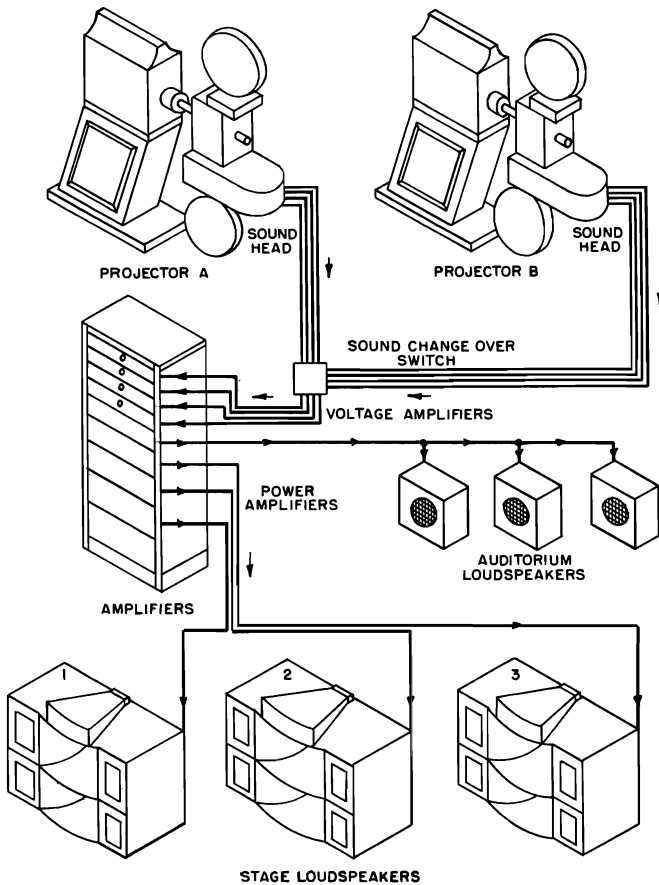
FIG. 13.29. Perspective view of the elements of a theater-type stereophonic sound reproducing system.

tion is rerecorded on the tracks of the positive motion-picture film. See Sec. 9.10C and Fig. 9.62.

The system for reproducing the recorded information in the four tracks is shown in Fig. 12.29. A wide screen is used in projecting the picture. Three stage loudspeakers are located behind the screen. The fourth channel is used to reproduce sound over loudspeakers located in the auditorium.

13.8. Radio Sound Reproducing System.—A complete radio broadcasting system is shown in Fig. 13.30. In the transmitting system, the sound wave is converted into the corresponding electrical wave by the microphones. The electrical outputs of the microphones are amplified by
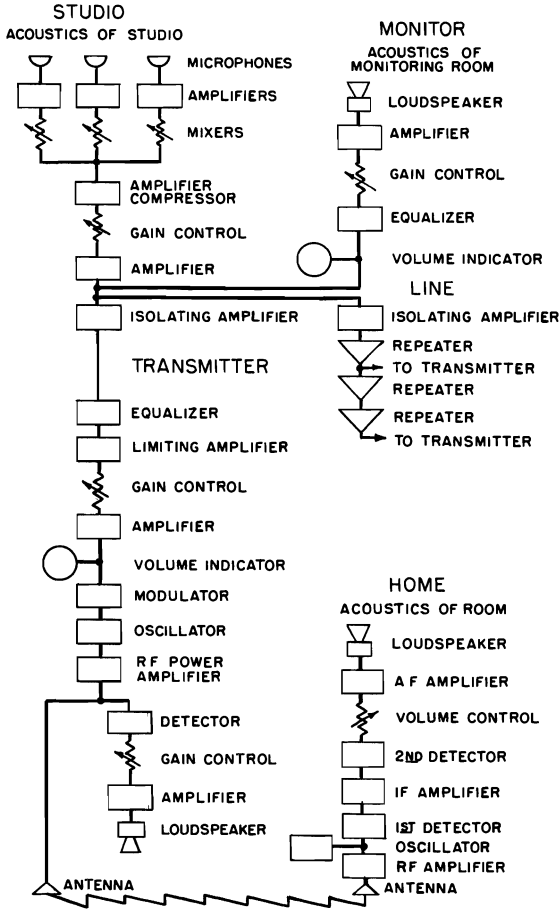
FIG. 13.30. The arrangement of the apparatus in a complete amplitude-type radio broadcasting and receiving system.

means of vacuum-tube amplifiers and fed to mixers. The microphones and source of sound to be picked up are located in the studio (Sec. 11.3C and Fig. 11.35). 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. 11.3B). The output of the compressor amplifier is coupled to the gain control. The gain control is followed by an amplifier. The output of the amplifier is fed to the monitoring system and the equalizer. The monitoring system is located in the control room. A volume indicator is provided to ensure that the proper level is fed to the transmitter. The equalizer is used to compensate for any deficiencies in the system and the line running to the transmitter. At the transmitter, a limiting amplifier is used to limit the signal and thereby prevent overloading of the transmitter. The limiting amplifier is followed by a gain control for controlling the input to the transmitter amplifier. The amplifier is followed by the modulator
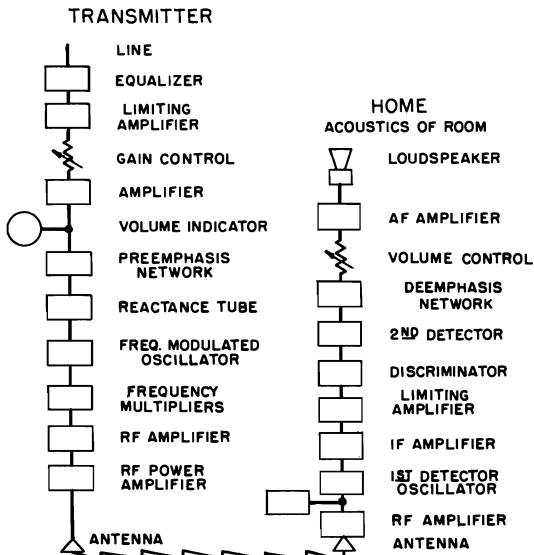
FIG. 13.31. The arrangement of the apparatus in a frequency modulation transmitting and receiving system.

and radio-frequency oscillator. In the standard broadcast band, the frequency of the oscillator lies somewhere between 550 to 1700 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 radio-frequency power amplifier is coupled to the transmitting antenna. The modulated radio-frequency wave is transmitted in the ether in all directions.

In the receiving system a very small portion of the radio-frequency energy 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 to the undulations in the original
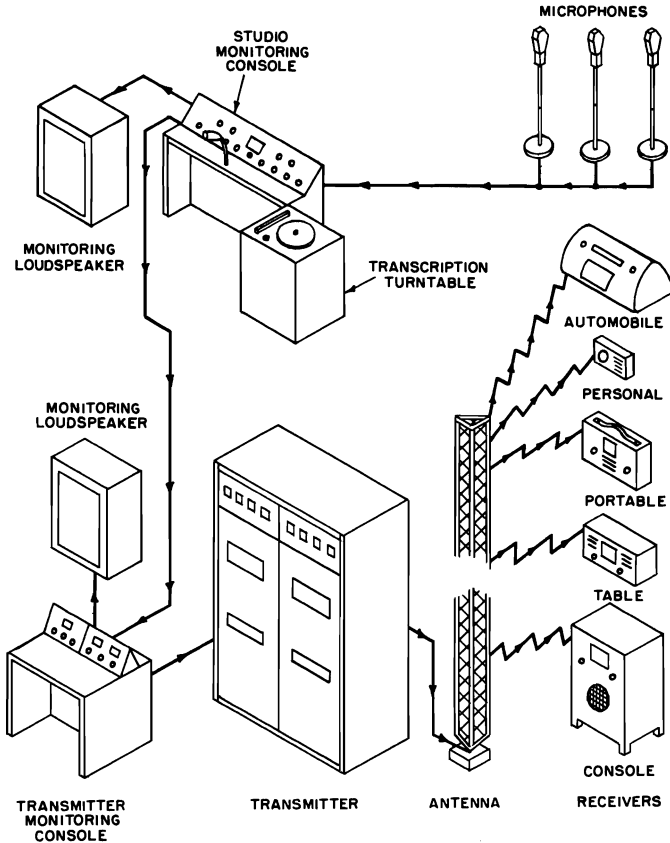
FIG. 13.32. Perspective view of a complete radio broadcasting and receiving system.

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. 11.2R and Fig. 11.27).

A frequency modulation radio transmitter and receiver is shown in Fig. 13.31.   In the frequency modulation radio transmitter, the frequency of the radio-frequency oscillator is varied to correspond to audio-frequency signal.   Some form of frequency discriminator and detector is used in the receiver to convert the frequency modulations into the corresponding audio variations.

A perspective view of the elements of a radio sound reproducing system is shown in Fig. 13.32.   Except for the type of modulation the elements in the amplitude and frequency modulated transmitters are the same.

13.9. Television Sound Reproducing System.[9,10]—The elements of the sound channel in a television system are essentially the same as those of a frequency modulated radio broadcasting system.   Fig. 13.33.   The picture and sound variations are picked up and converted into the corresponding electrical variations and transmitted simultaneously.   The studio, sets, and techniques for television pickup are described in Sec. 11.3I.   At the receiver the sound and picture are reproduced simultaneously.   The sound portion of the television receiver is similar to the frequency modulation radio receiver described in Sec. 13.8.   The reproduction of the sound in the room by a television receiver is similar to that of a radio receiver or phonograph (see Sec. 11.2R).

A perspective view of the elements of a television system are shown in Fig. 13.33.

13.10. Dictating Machines.—A dictating machine consists of a sound recorder and reproducer used for dictating letters and similar material.   The early machines employed a mechanical recorder and reproducer.   Machines developed and commercialized during the past two decades employ a recording system consisting of a microphone, amplifier, and cutter or embosser similar to the phonograph recording system.   The reproducing system consists of a pickup, amplifier, and loudspeaker similar to the phonograph reproducer.   Cylinder, disk, and belt records are employed as the storage medium.   Dictating machines employing magnetic tape recorders and reproducers have also been developed and commercialized.

13.11. Hearing Aids.—Tests made upon a representative cross section of the people in this country show that a very large percentage are hard of hearing.   Practically all of these people may obtain satisfaction from the use of a hearing aid.   A hearing aid is a complete sound reproducing system consisting of a microphone, amplifier, and receiver which increases the sound pressure or stimulus over that normally received by the ear.

During the past two decades, hearing aids employing vacuum tubes have been developed to a high order of quality combined with small size.   The development of subminiature vacuum tubes has made this possible.   The amplifiers are for the most part three-pentode type vacuum tubes with

[9] Hogan, J. V. L., "The Early Days of Television," *Jour. Soc. Mot. Pic. Tel. Eng.*, Vol. 63, No. 5, p. 169, 1954.

[10] Jensen, A. G., "The Evaluation of Modern Television," *Jour. Soc. Mot. Pic. Tel. Eng.*, Vol. 63, No. 5, p. 174, 1954.
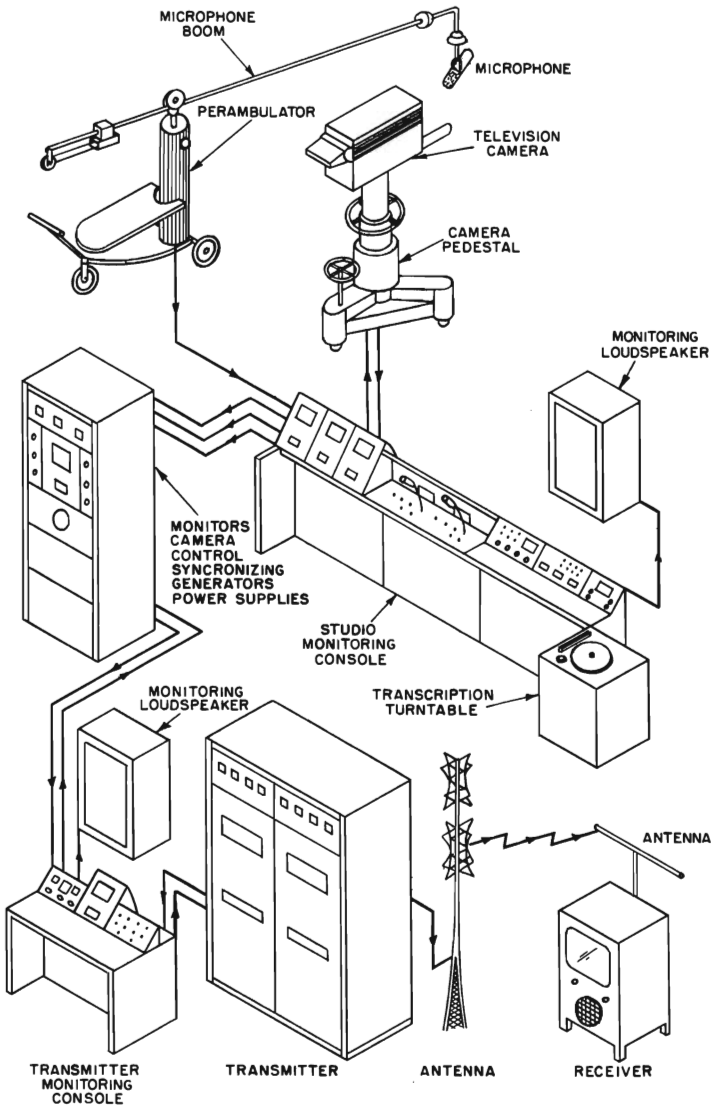
Fig. 13.33. Perspective view of a complete television system.

resistance-capacitance coupling. The microphone used with the vacuum tube hearing aid is of the diaphragm type using either a Rochelle salt crystal or a barium-titanate ceramic as a transducer, see Sec. 8.2C. The air receiver is a magnetic type similar to the standard telephone receiver, see Sec. 9.13.

The receiver is made very small and unobtrusive so that it may be placed in the outer ear cavity and coupled directly to the ear canal. The bone conduction receiver is also of the magnetic type with a mechanical transformer that couples the vibrating system to the mastoid bone, see Sec. 9.13. The electrical impedance of the receivers is relatively low, being about 100 ohms. Therefore, a transformer is used to couple the output of the power vacuum tube to the receiver.

During the past two years, transistor-type hearing aids have been developed and commercialized. The advantage of the transistor amplifying
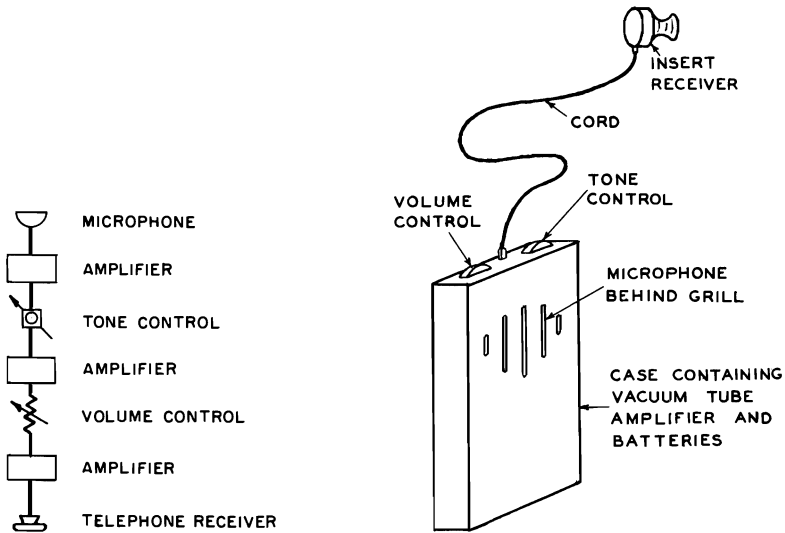
Fig. 13.34. Schematic arrangement of the apparatus in a hearing aid.

Fig. 13.35. Perspective view of a hearing aid.

system is the low battery-power consumption. In addition, only one battery is required as contrasted to A and B batteries in vacuum-tube hearing aids. The input electrical impedance of a transistor is relatively low. Therefore, the microphone which has been developed for transistor hearing aids is a diaphragm type employing a magnetic transducer, see Sec. 8.2E. The electrical impedance of this unit can be made so that it matches the input electrical impedance of the transistor. The receivers used with the transistor hearing aids are the same as those used with the vacuum-tube hearing aid.

The arrangement of the elements in a hearing aid system is shown in Fig. 13.34. Either vacuum tubes or transistors may be used for the amplifiers.

A perspective view of a hearing aid is shown in Fig. 13.35. In the transistor-type hearing aid the microphone, amplifier, and batteries can be housed in a case of only two cubic inches.

13.12. Sound Reproducing System Components.—Magnetic tape, radio, and disk phonograph sound reproducers have been described in the preceding sections.   The components of a sound reproducing system for the reproduction of magnetic tape, radio, and disk phonograph records are shown
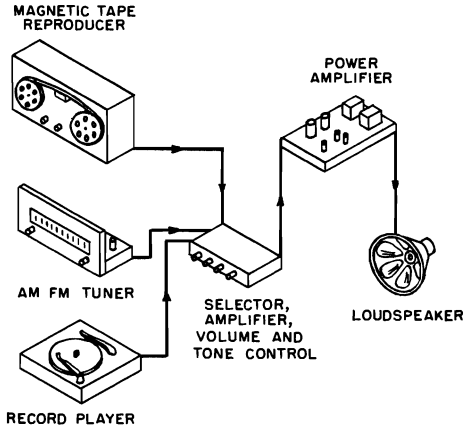
Fig. 13.36.   Perspective view of the components of a sound reproducing system.

in Fig. 13.36.   The components may be placed in a single cabinet and termed a combination sound reproducer.   The components may be used in a custom-type sound reproducing system in which one or more of the components may be housed in separate cabinets or the components may be placed in special compartments as the part of a house or apartment.

14

MEANS FOR THE COMMUNICATION OF INFORMATION

14.1. Introduction.—Speech always has been and still remains the most important method for the transmission of information between individuals. Manual signals involving the arms, hands, face, eyes, and mouth are secondary to the voice. To augment the process of communication man devised sound generators in the form of noise makers, sound signalling devices, and musical instruments and visible signalling, as exemplified by fire, smoke, and flags. In a further expansion of communication, orthography was evolved, using symbolic notation which made it possible to record and store information and to transport information by symbolic recordings to a distant point. The advent of the printing press about 400 years ago supplied a means for the mass dissemination of information in the form of the printed page. Still further, photography was developed which provided a process for recording a scene in the form of a picture. A further development of photography was the motion picture, in which a series of photographs of a changing scene are taken and when projected depict a reproduction of the original scene. The telegraph was the first electrical system for the coded transmission of information over large distances by the use of wires for immediate decoding into the message at the recording end. The invention of wireless telegraphy at the turn of the century provided a system for the transmission of coded messages through the ether, thereby eliminating the use of a wire transmission line. The transmission of pictures and the printed page by facsimile, either by wire or through the ether, was developed and introduced about three decades ago. The inventions and developments in the field of sound and scene reproduction during the past half century, as exemplified by the telephone, phonograph, sound systems, radio, sound motion picture, and television have made it possible to pick up sound or sound and scene and reproduce it or them, either at the same point or some other point either at the same time or some subsequent time. The impact of the telephone, phonograph, radio, sound motion pictures, and television upon the dissemination of information, art, and culture has been tremendous. However, the list will not end with the communication systems that have been enumerated. There will be additional systems and adjuncts in the field of communication which have not been developed that will ultimately improve, modify,

augment, or displace existing communication systems. It is the purpose of this chapter to describe existing means and systems for the communication of information between individuals and to outline the possibilities of further developments.

The diagram[1] of Fig. 14.1 depicts two individuals, namely, a sender and a receiver and the means and systems which may be used by the sender to transmit information to the receiver. This diagram illustrates in schematic form the subject matter of this chapter. The various means and systems for the communication of information between individuals will now be described.

14.2. Existing Means and Systems for the Transmission of Information.[2]—The existing means and systems for the communication of information between individuals are as follows: voice, manual signals, sound generator, semaphore, orthography, typography, photography, telegraph, teletype, motion picture, telephone, phonograph, radio, sound systems, facsimile, sound motion picture, and television. It is the purpose of the sections which follow to describe these communication means and systems.

14.3. Voice.—Speech is man's primary and most important process of communication. The primary use of the voice is in direct transmission. The human voice of the sender of Fig. 14.1, controlled by the brain, produces sound waves which carry the information through air to the ear of the receiver. The ear of the receiver converts the impinging sound waves into impulses which are sent to the brain for the decoding of the message. The sound waves produced by the sender also impinge upon the ear of the sender. This is a feedback loop which provides the sender with information for controlling and monitoring his transmission.

14.4. Manual Signals.—Manual signals involving motion of the arms, hands, face, eyes, and mouth are secondary to the voice. Nevertheless, these forms of expression play a very important part in the process of communication. As a matter of fact, very important transactions are carried out by the wave of the hand, the wink of the eye, etc. Manual signals are sent through the ether and picked up by the eye of the receiver. The eye converts the impinging light waves into impulses which are sent to the brain for decoding of the message.

14.5. Feedback.—Feedback, as used in the diagram of Fig. 14.1 and in this chapter, means any information that is received by the sender as a result of the information which has been transmitted. This may involve sound or manual signals produced by the receiver, or sound or light signals produced by some mechanism actuated by the sender. Feedback plays an important part in the process of communication of information. This will be further illustrated in other parts of this chapter.

14.6. Sound Generator.—The term, sound generator, as used in this book is any system which produces sound waves in accordance with the

---

[1] For the lack of a better term, ether is used to designate the medium for transmission of light waves through space in Fig. 14.1.

[2] Olson, Harry F., *Jour. Acous. Soc. Amer.*, Vol. 26, No. 5, p. 637, 1954.
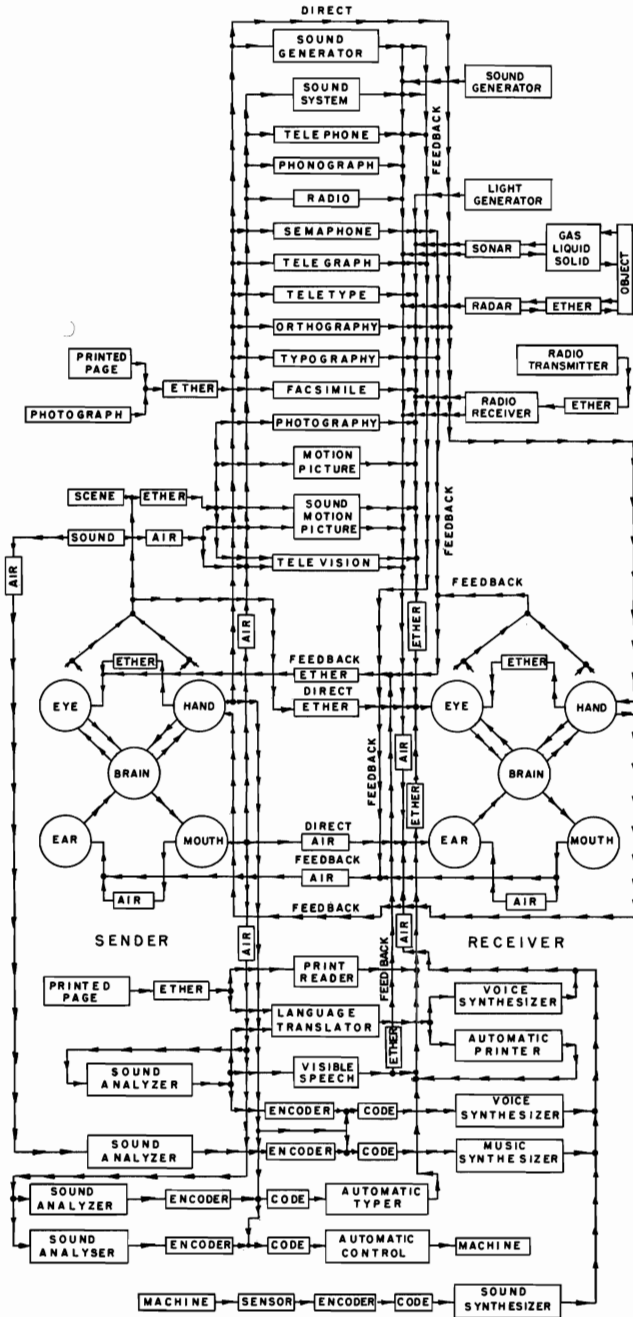
FIG. 14.1.   Systems for the communication of information between two individuals.

manual actuation by the sender. Sound generators include the following: musical instruments, horns, whistles, bells, buzzers, sirens, and guns.

In most of these systems feedback to the ear of the sender plays a very important part in producing the desired results. This is particularly true of the production of music by means of musical instruments.

The sound produced by the sound generator is picked up by the ear of the receiver and sent to the brain for decoding and interpretation.

14.7. Semaphore.—A semaphore is a signalling system manually actuated by the sender and produces visible signals which are received by the eye of the receiver. A semaphore may be any object or contrivance that is moved by the hand or a light system moved or interrupted by the hand. Semaphores employing both movable objects and interrupted and moving lights are used in the control of all types of traffic on land, sea, and in the air, as well as for the transmission of information by a code system.

14.8. Orthography.—The term, orthography, as used in this chapter, is the process of recording information in the form of symbolic notation inscribed upon some medium. In the highest form of the development of orthography, the symbols represent the sounds of speech. This means of communication made it possible to record and store information and to transport information by symbolic recordings to a distant point.

A development of orthography for the blind is the representation of the language in Braille. The characters are coded in configurations of raised dots. The tactile sense is employed in the interpretation of the Braille code.

14.9. Typography.—Typography is any system which employs printing with type. The printing press and the typewriter are the outstanding examples of typography. The invention of the printing press supplied a means for the mass distribution of information in the form of the printed page.

14.10. Photography.—Photography is the process of producing images on a sensitized surface by the action of light. Photography provides a process for recording a scene in the form of a picture which can be duplicated in large quantities. By the use of the electrotype, the picture can be printed on a printing press. Photography provides a system for the mass distribution of pictures.

14.11. Telegraph.—The telegraph was the first electrical system for the coded transmission of information over long distances by the use of wires for immediate decoding at the receiving end. The coded message is reproduced as sound at both the sending and receiving end. Feedback to the sender provides control and check of the message being transmitted. Telegraphy made it possible to transmit messages over large distances in a relatively short time. It was the first step in the immediate transmission of information over any distance. The invention of the wireless telegraphy produced a means for the transmission of coded messages through the ether, thereby eliminating the use of a wire transmission line.

14.12. Teletype.—A teletype is a printing telegraph system. The message at the sending end is typed on a machine similar to a conventional

typewriter.   The message is sent in the form of electrical impulses over a wire line.   The electrical impulses are used at the receiving end to type the message in a form similar to that produced by a conventional typewriter. As contrasted to the telegraph, the teletype does not require decoding at the receiving point.

14.13. Motion Picture.—The motion picture process consists of taking a series of separate photographs of objects in successive positions in a scene. The photographs are recorded on a strip of light-sensitive film.   Thus, there is produced a series of photographs on the film depicting the different stages of the action.   In reproduction, the series of photographs are projected on a screen, thus producing a series of images.   Due to the persistence of vision, the effect is the production of picture with continuous motion.   The advent of the motion picture added the element of motion to the still picture.

14.14. Telephone.—The telephone is the oldest sound reproducing system. From the standpoint of sound reproduction, the telephone consists of the microphone or transmitter, which converts the sound vibrations into the corresponding electrical variations, and the telephone receiver, which converts the electrical variations into the corresponding, and hence original, sound vibrations.   The electrical variations are sent over a wire.   Much more equipment is needed for a complete telephone system so that the different subscribers can be connected to each other at will.   There are 55,000,000 telephones in this country.   In other words, almost every family in this country has a telephone instrument, which means that any person can talk to any other person in a very short interval of time.

14.15. Phonograph.—Phonograph is a term used to designate a sound reproducing system in which the information is recorded on and reproduced from a record.   The records have been in many forms, starting with the cylinder.   This was displaced by the disk record which is still in use today.

The disk phonograph consists of either acousto-mechanical or acousto-electro-mechanical recording means for producing an undulation in the groove in the disk record which corresponds to the original sound vibrations and either mechano-acoustic or mechano-electro-acoustic means for producing sound vibrations from the groove which corresponds to the undulations in the groove of the record.

The phonograph was the first sound reproducing system which made it possible for all the people of the world to hear statesmen, actors, artists, operas, symphonies and all manner of musical aggregations where only a relative few had been able to hear them first hand.   The phonograph is used in. every country and clime.   There are 40,000,000 phonographs in this country.   More than 5 billion disk phonograph records have been sold.

During the first two decades after the commercialization of the phonograph, recording and reproducing were made by acoustical and mechanical means.   After the advent of the vacuum tube, recording and reproducing have been made by the combination of acoustical, mechanical, and electrical means.

The disk sound recording and reproducing system has also been used for home and commercial applications as well as for the prerecorded records.

Magnetic tape phonographs or reproducers have been developed to a high order of excellence during the past decade. In the magnetic tape system, the information is recorded on a ferromagnetic coating on a plastic base tape in the form of magnetized elements corresponding to the electrical input. In reproduction, the stored magnetic information on the tape is converted into the corresponding electrical variations. The tape system has been applied to recording and reproducing sound for home and commercial applications. The commercial applications include sound motion pictures, television signals, master recordings, transcriptions, and memory systems for computors.

14.16. Radio.—A radio broadcasting system consists of a transmitter and a receiver. At the transmitter, the sound vibrations in the studio which represent the program material are converted into the corresponding electrical variations by the microphone. These electrical variations modulate a radio-frequency system which is coupled to the antenna where the resultant signal is radiated. A small part of the signal from the transmitter is picked up by the radio receiver antenna. This signal is amplified and demodulated into the original audio electrical signal. The audio electrical signal is fed to the loudspeaker which converts the electrical variations into original sound vibrations.

Two types of radio broadcasting are in use today, namely, amplitude- and frequency-modulated systems.

The radio, like the phonograph, is a home-type instrument. There are 3200 broadcasting stations, 125,000,000 home-type receivers and 35,000,000 automobile receivers in this country. Practically every family owns a radio receiver, which means that they can hear famous persons in government, in industries, in the schools, in the theater, and in all manner of musical aggregations.

In addition to conventional broadcasting, there are radio communication systems of all types, involving stationary land and mobile land, water, and air systems.

14.17. Sound Systems.—Sound system is a term used to designate the combination of a microphone, amplifier, and loudspeaker and auxiliary equipment. Sound systems include hearing aids, public address, sound reinforcing, electric megaphone, intercommunicating, announce, paging, and call systems.

14.18. Facsimile.—The term, facsimile, is used to designate a system for the transmission of graphic material in the form of a photograph, drawing, or printed page. In existing commercial facsimile systems, the information is transmitted bit by bit in a sequential manner. This is done by dividing the original into a large number of elementary areas. Then signals are transmitted that correspond to the relative light shades of these areas. Such shades are reproduced in graphic form at the receiving point. The elementary areas are reproduced in a sequence corresponding to that at the

pickup at the transmitter. The fidelity of reproduction depends upon the correspondence in amplitude of the number of these elementary areas in the picture.

14.19. Sound Motion Picture.—The addition of sound to the motion picture made this type of expression complete. It was the first commercial system in which picture and sound were synchronised and reproduced at the same time. In some of the early systems the sound was recorded and reproduced by means of a disk record which was synchronized with picture. During the past twenty-five years, the photographic method for recording and reproducing the sound has been universally employed. The photographic reproduction of sound is accomplished by recording information in the form of a narrow band, termed the sound track, located between the sprocket holes and the picture. The recording system produces photographic undulations in the sound track which correspond to the original sound. In reproduction, these photographic undulations are converted into the corresponding electrical variations which, in turn, are converted into sound by the loudspeaker.

Recently, stereophonic sound systems employing multiple sound channels have been introduced on a wide commercial scale. In one system, four separate magnetic sound tracks are used. Two of the tracks are outside the sprocket holes and the other two are inside the sprocket holes. The output from the three separate sound channels and the control channel are recorded on the four magnetic tracks. In reproduction, four separate magnetic head, amplifier, and loudspeaker systems are used to reproduce the sound in perspective. A wide screen is used to project the picture. The three loudspeakers are located behind the screen.

There are 18,000 motion picture theaters in this country. The average weekly attendance is 90,000,000 persons.

14.20. Television.—Television is the latest commercial system in which picture and sound are reproduced at the same time. The fundamental objective of a television system is to extend the sense of sight and hearing beyond their natural limits. At the pickup point the camera scans the scene and thereby divides the scene into elementary projected areas. The light reflected from the elementary areas is picked up by the camera in the scanning process and converted into the corresponding electrical variations. At the receiving end the electrical variations are converted into the corresponding light variations and depicted upon some sort of screen in the pattern similar to that at the pickup. The gradations of the light produced upon the screen are proportional to the reflected light intensities of the scene being televised, thereby producing a picture corresponding to the televised scene.

14.21. Undeveloped Systems for the Transmission of Information.—In the over-all plan[3] of the systems for the transmission of information, there are many that have not been developed or commercialized. These

[3] Peterson, Gordon E., *Jour. Speech and Hearing Disorders*, Vol. 18, p. 116, 1953.

include the following: visible speech, speech synthesizer, phonetic typewriter, print reader, language translator, music synthesizer, and speech control of machines. It is the purpose of the sections which follow to describe these communication systems.

14.22. Visible Speech.[4]—Visible speech is an electronic system which converts spoken words into visible patterns that someone can learn to read. The fundamental element of a visible speech machine is a multiple-frequency analyzing system. The output of the frequency analyzing system is fed to a light reproducing system in the form of a screen with light persistence. The pattern on the screen depicts the speech sounds as bands of illumination. Each band corresponds to a frequency band. The intensity of light along the beam depicts the amplitude. The length of a light band represents the time that the sound persists. In this manner, sound is depicted as a function of the frequency, amplitude, and time. Each vowel or consonant sound produces a unique and distinguishable pattern on the screen. Under these conditions speech can be read from the screen. Some of the uses of the visible speech apparatus are as follows: visual hearing for the deaf, teaching the deaf to speak, and speech correction.

14.23. Speech Synthesizer.[5,6]—A speech synthesizer is a machine that converts speech information in the form of a code into the sounds of speech. The sounds of speech can be described as follows: the fundamental frequency with all the overtones, the formant frequency bands which emphasize some of the frequency regions, and the hiss sounds. Thus, the code for a speech sound will carry the information corresponding to the fundamental frequency, the formant regions, and the hiss sounds. This code can be set by hand and converted into speech sounds. The coded message can also be obtained from a machine that analyzes the sounds of speech into the fundamental frequency, the formant regions, and the hiss sounds. It has been established that intelligible speech can be obtained from a coded message having a band width of 100 cycles. The advantage of coding the sounds of speech is to reduce the band width required for transmission or recording.

14.24. Phonetic Typewriter.—A phonetic typewriter is a machine that converts the sounds of speech, picked up by the microphone of the machine, into the corresponding typed words on paper. The important element of a phonetic typewriter is an analyzing system which will analyze the sounds of speech and tabulate these sounds and store them in a memory device. Another element is a correlating, selecting, and actuating system which types the words corresponding to the speech sounds stored in the memory. The tools for developing a phonetic typewriter are rapidly becoming available and the development of a phonetic typewriter is now within the realm of possibility.

When the analyzing system for the phonetic typewriter has been

[4] Potter, R. K., *Jour. Acous. Soc. Amer.*, Vol. 18, No. 1, p. 1, 1946.

[5] Dudley H., *Jour. Acous. Soc. Amer.*, Vol. 11, No. 2, p. 169, 1927.

[6] Olson and Belar, *Jour. Acous. Soc. Amer.*, Vol. 27, No. 3, p. 595, 1955.

developed, it may be used to produce a record for actuating the speech synthesizer. The advantage of the coded record is that many times as much speech information may be stored in a given record as contrasted to that required for the conventional reproduction of speech.

14.25. Print Reader.[7]—A print reader is a machine that converts the information on a printed page into the corresponding speech sounds. An optical-electronic-acoustic system has been developed which scans the letters of a printed page and "speaks out" the letters. This development shows that the printed page contains enough information so that the information can be deduced by the machine. A further extension is a machine that will "speak out" the words instead of the letters. With the electronic tools that are becoming available, the development of such a machine is within the realm of possibility. The principal problem is the storage or memory system because the storage system must be capable of storing a tremendous amount of information. Following that, a means must be developed which will provide easy and rapid access to the memory. When such a machine has been developed, a newspaper, magazine, or book can be "read" by listening instead of by reading.

14.26. Language Translator.[8]—A language translater is a machine that translates from one language to another language. When a print reader has been developed, then another step is the conversion from the printed page into the sounds of a different language. This too requires an even greater memory system because the sequence of words will not be the same. When an automatic printer has been developed, the printed page of one language can be converted into the printed page of another language. When the phonetic typewriter has been developed, another step in the translation machine will be the translation of one language spoken into another language printed by the machine. Still another step will be the translation of one language spoken into the machine into another language spoken by a machine.

14.27. Music Synthesizer.[9]—A music synthesizer is a machine that converts the information stored in a code into the sounds of music. Music producing machines employing a coded record to actuate the sound producing means have been in use for centuries. Examples are music boxes, player pianos, orchestras, and bands. The sound producing elements in these machines have been of the conventional type. It has been demonstrated that an electronic system can produce any sound that has ever been produced or occurred in nature or any sound that can be conceived. This opens an entirely new field for the production of music. The music can be set up in the form of a coded record which can be set up by hand.

It has been demonstrated that even the greatest artists can be copied by a musical engineer employing an electronic system and a coded record

[7] Zworykin, Flory, and Pike, *Electronics*, Vol. 22, No. 6, p. 80, 1944; p. 80, June 1949.

[8] Mechanical Translation, Massachusetts Institute of Technology, Cambridge, Mass., 1954.

[9] Olson and Belar, *Jour. Acous. Soc. Amer.*, Vol. 27, No. 3, p. 595, 1955.

so that it is impossible to distinguish the synthesized rendition from the original.

The synthesizer provides the musician with a new tool for producing music. The operation does not require manual dexterity, as in the case of conventional instruments. Once a certain selection or effect has been obtained, it can be retained forever.

With the advent of the music synthesizer, the composer will not be hampered in the writing of compositions by the limitations of existing musical instruments because the music synthesizer can produce any tone or musical complex that can be imagined or conceived.

It is also possible to analyze and code music produced by conventional means. The advantage of this process is that the amount of record material required is only a fraction of that required for conventional recording. For example, an entire symphony could be recorded in code form on a three-inch record.

14.28. Control of Machines by Speech.—A system which converts the sounds and words of speech into a code may be used to actuate the controls of a machine. These machines may be of all types used in homes, farms, offices, and factories. The control may range from a very simple one which starts or stops the machine, to complex controls that go beyond the speed and scope of manual controls.

14.29. Machine and Object Sensor.—The physical or mechanical aspects of a machine or an operation may be sensed and converted into a code. This code may be used to actuate a sound or speech synthesizer. Thus, the various mechanical or physical aspects of machine or an operation will be converted into characteristic signals or speech sounds which can be correlated with the complexion of the machine or operation. An example of a development in this field is sonar[10] which senses the distance and direction of an object or obstacle under water. A similar device has been developed for use in air as an aid to the blind.

14.30. List of Undeveloped Systems for the Transmission of Information.—Communication systems which have not been developed or commercialized have been described in the preceding sections. One or more of the following elements are involved in this system: analyzers, encoders, coders, memories, decoders, and synthesizers. Thirteen communication systems employing these elements are depicted in Fig. 14.2. The following is a summary of the systems which have not been developed or commercialized:

System 1 of Fig. 14.2 is a phonetic typewriter. A phonetic typewriter is a machine that converts the sounds of speech into the corresponding typed words on paper.

System 2 of Fig. 14.2 is an electronic music synthesizer. An electronic music synthesizer is a machine that produces music from a coded record. The coded record is produced by a musical engineer.

[10] Olson, "Elements of Acoustical Engineering," D. van Nostrand Company, Princeton, N.J., 2nd Edition, p. 519, 1947.

System 3 of Fig. 14.2 is an electronic speech synthesizer.   An electronic speech synthesizer is a machine that produces speech from a coded record. The coded record is produced by a speech engineer.

System 4 of Fig. 14.2 is a machine which reproduces music through the
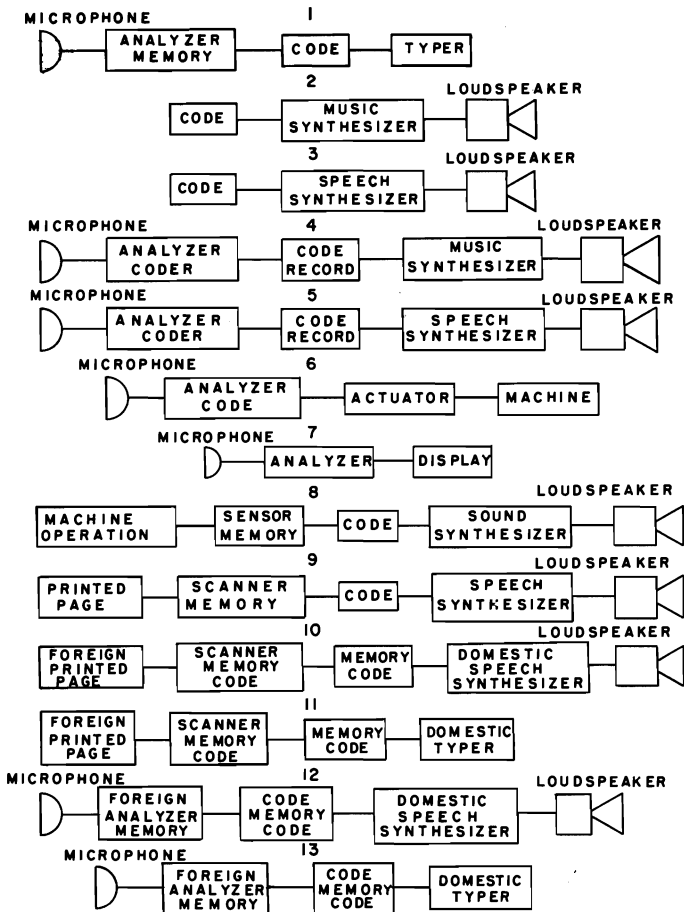
FIG. 14.2.   Systems for the communication of information that have not been developed or commercialized.

intermediary of a coded record.   Music is picked up by a microphone and a coded record is produced.   The coded record is used to actuate the music synthesizer to reproduce the original music.   The advantage of the coded record is that a much longer musical selection can be stored in a given record as contrasted to that required for the conventional reproduction of sound.

System 5 of Fig. 14.2 is a machine which reproduces speech through the intermediary of a coded record. The operation of the system is similar to the music system 4 of Fig. 14.2.

System 6 of Fig. 14.2 is a machine that converts the sounds of speech into mechanical operations.

System 7 of Fig. 14.2 is a machine that converts the sounds of speech into visible patterns that can be read.

System 8 of Fig. 14.2 is a machine that scans or senses the various physical or mechanical aspects of a machine or an operation and converts this information into characteristic signals or speech sounds which can be correlated with the complexion of the machine or the operation.

System 9 of Fig. 14.2 is a machine that scans the printed page and converts the information into the corresponding sounds of speech.

System 10 of Fig. 14.2 is a machine that scans a foreign printed page and converts the information into the corresponding sounds of speech in the domestic language.

System 11 of Fig. 14.2 is a machine that scans a foreign printed page and converts the foreign printed page into the domestic typed page.

System 12 of Fig. 14.2 is a machine that picks up a foreign language spoken into the machine and converts the foreign language to the corresponding domestic language and reproduces it by means of a loudspeaker.

System 13 of Fig. 14.2 is a machine that picks up a foreign language spoken into the machine and converts the foreign language to the corresponding domestic language typed on paper.

14.31. Conclusion.—The existing systems for the communication of information between individuals have been described in this chapter. The impact of these systems upon the dissemination of information, art, and culture has been tremendous. However, the list of systems does not end with the systems in use today. This chapter has described additional systems which have not been developed, that will ultimately improve, augment, or displace existing means for the communication of information between individuals. Considerable fundamental research will be required before these systems can be developed. Nevertheless, much progress in the development of the elements for these new systems has already been made. These elements include analyzers, synthesizers, encoders, decoders, coders, memories, codes, etc. As these elements are brought to a higher stage of development, the systems of Fig. 14.2 will become realities.

15

UNDERWATER SOUND

15.1. Introduction.—There are four general methods for the transmission of signals underwater—namely, optical, magnetic, electrical, and acoustical. Water is very opaque to infrared and ultraviolet light and is not particularly transparent for visible light. Magnetic transmission and detection may be used over relatively short distances. Electromagnetic or radio waves are rapidly attenuated in passing through sea water because it is a good conductor of electricity. Subaqueous signaling by means of sound waves is far superior to the other methods mentioned above because water is a good medium for the transmission of sound waves. Therefore, except for certain specific applications, the acoustic method is almost universally used for the transmission of intelligence in water. It is the purpose of this chapter to describe systems for generating and detecting sound waves in water and some applications of these systems.

15.2. Sound Waves in Water.—Sound waves in water have been produced, transmitted, and detected over the frequency range from 2 cycles to 50 megacycles. This is a tremendous frequency range covering a band of twenty-five octaves.

The energy flow in a spherical wave, in the absence of dissipation, decreases inversely as the square of the distance from the sound source. Dissipation due to viscosity introduces additional attenuation. The intensity of sound[1] in a spherical sound wave is given by

$$I = \frac{P\epsilon^{-\alpha r}}{4\pi r^2} \qquad\qquad 15.1$$

where I = intensity, in ergs per square centimeter,
$\quad P$ = power output of the sound source, in ergs per second,
$\quad r$ = distance from the sound source, in centimeters,
$\quad \alpha = \dfrac{2}{3}\dfrac{\mu k^2}{\rho c}$
$\quad \mu$ = viscosity, $\mu = .0114$ for water,
$\quad k = \dfrac{2\pi}{\lambda},$

[1] Lamb, "Dynamical Theory of Sound," Edward Arnold, London, 1931.

λ = wavelength, in centimeters,

ρ = density, in grams per cubic centimeter, and

c = velocity of propagation, in centimeters per second.

For 1 megacycle, $\alpha = 10^{-4}$. A sound wave of this frequency would travel 100 meters before it was attenuated to $1/\epsilon$. The attenuation under actual conditions is greater than that predicted by the classical theory. The amount may range from a small deviation to a factor of many times in attenuation. The magnitude of the anomalous attenuation depends upon the condition of the water, as, for example, air bubbles of certain sizes and concentrations will produce tremendous attenuations. In the case of transmission over large distances additional attenuation may be due to inhomogeneities in the water. These may be due to temperature gradients which produce reflections and refractions.

In the above considerations it has been assumed that the wave is of a spherical nature. In shallow water the wave may be confined between the surface and the bottom in which case the propagation is similar to that in a cylindrical wave. Under these conditions, in the absence of dissipation, the energy falls off universely as the distance. It has been found that tremendous ranges may be obtained at the low frequencies in shallow water.

The amount of sound energy which a subaqueous loudspeaker may produce is limited by cavitation at the diaphragm. In general, cavitation occurs at the diaphragm when the pressure, in the rarefaction cycles, is near zero. For a loudspeaker operating near the surface of the water the pressure is that due to the atmosphere. Under these conditions the rarefaction pressure is 10^6 dynes per square centimeter.

The output of a subaqueous loudspeaker in water is given by

$$P = r_{AW}\, \dot{x}^2 = \frac{p^2}{r_{AW}} \qquad\qquad 15.2$$

where P = output, in ergs,

r_{AW} = acoustical resistance, in acoustical ohms,

$\dot{x}$ = volume current, in cubic centimeters per second, and

p = sound pressure, in dynes per square centimeter.

If the ultimate acoustical resistance has been obtained, the value of the acoustical resistance for 1 square centimeter is

$$r_{AW} = \rho c = 144{,}000 \qquad\qquad 15.3$$

where ρ = density of water, in grams per cubic centimeter, and

c = velocity of sound in water, in centimeters per second.

For a peak sound pressure of 10^6 dynes per square centimeter the power output per square centimeter is 3.6×10^6 ergs or .36 watt. If the loudspeaker is lowered to a depth of 33 feet the output will be quadrupled.

The preceding presents the idealized picture of sound transmission in water. Transmission of sound in the sea does not conform to this simple description. There are many factors[2] that influence the transmission of sound in the sea. A few of the factors are as follows: The surface of the sea, the bottom of the sea, the inhomogeneity of the sea.

The surface of the sea affects the transmission by the reflection produced at the surface. Since the change in acoustical impedance from the sea to the air is about 3000, the reflection is practically perfect. If the sea were perfectly plane, account of the surface reflection could be accounted for by the establishment of an image transmitter of opposite phase above the surface. However, the sea is never quiet, with the result that there are waves which disturb the plane surface condition.

In shallow water the bottom of the sea introduces another troublesome boundary condition. Unlike the surface of the sea, the shape of the bottom is not known. In addition, the absorption and reflection varies over wide limits.

Sea water is not homogeneous. There are both vertical and horizontal variations in the velocity of propagation of sound in the sea. Vertical variations are perhaps more pronounced and produce refraction of the sound waves. See Sec. 1.11. The bending of the sound waves by refraction reduces the efficiency in echo ranging. In the case of severe refraction, the distance of satisfactory operation may be seriously reduced.

The surface reflection, the bottom reflection, and the reflection due to variations in the sea produce reflections. This lends to what appears to be reverberation in the sonar receiver.

The reflection and refraction effects also produce variation in the received sound in both passive and echo systems. The roll and pitch of the sending and receiving vessel produce fluctuations in the received signal.

The anomalies of the sea outlined above are of great importance in both undersea warfare as well as in commercial navigation. The application of some of the newer principles in the fields of communication such as correlation techniques and recognition procedures should be of value in solving many of the problems in the transmission of sound in the sea.

As in all transmitting media, noise produced in the medium influences the reception of information. The different types of noise includes the ambient noise in the sea due to natural causes and noise produced by ships and other sound sources. The ambient noise in the sea is a function of the state of agitation of the sea and the wind velocity. Various undersea life also produces discrete sounds and noises. In general, underwater signals are received from hydrophones suspended below the ship. Under these conditions the noise produced by the ship influences the reception of underwater signals. The principal source of noise in water, produced by a ship, is due to cavitation at the propellers. As in all other underwater phenomena, noise in the sea is a very complex subject.

[2] A Survey Report on Basic Problems of Underwater Acoustics Research, Committee on Undersea Warfare, National Research Council, Washington, D.C., 1950.

15.3. Direct Radiator Dynamic Projector.[3,4]—The direct radiator dynamic projector is a projector designed to operate under water, in which a diaphragm is driven by a voice coil located in a magnetic field (Fig. 15.1).

The theory of the direct radiator dynamic projector, except for the high impedance of the medium, is the same as that of the air direct radiator dynamic loudspeaker described in Sec. 6.2. The higher acoustical impedance of the medium makes it expedient to incorporate some constructional features which differ from the corresponding air-type loudspeaker.

Since the acoustical impedance of water is about 3400 times that of air the diaphragm of the direct radiator projector is relatively small compared to that of an air direct loudspeaker employing a comparable driving system.
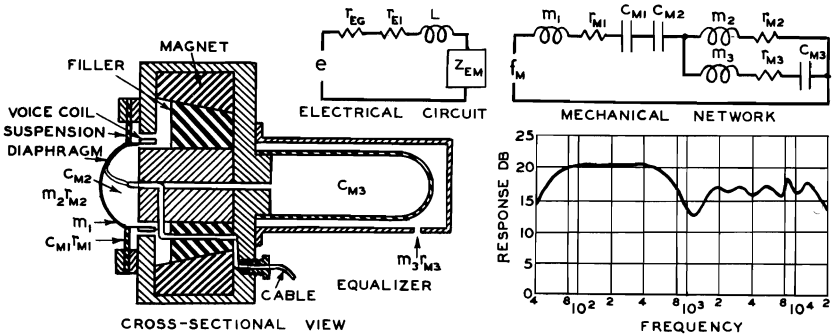
Fig. 15.1. Cross-sectional view, electrical circuit, mechanical network, and pressure response frequency characteristic of a direct radiator dynamic projector. In the electrical circuit: z_{EM} = the motional electrical impedance. L and r_{E1} = the inductance and electrical resistance of the voice coil. r_{EG} = the electrical resistance of the electrical generator. e = the voltage of the electrical generator. In the mechanical network: m_1 = the mass of the diaphragm and coil. r_{M1} and C_{M1} = the mechanical resistance and compliance of the suspension system. C_{M2} = the compliance of the air chamber behind the diaphragm. m_2 and r_{M2} = the mass and mechanical resistance of the water load. m_3 and r_{M3} = the mass and mechanical resistance of the aperture in the pressure equalizer. C_{M3} = the compliance of the pressure equalizer.

In the case of a projector it is not necessary to use a large baffle or cabinet. The back of the diaphragm can be terminated in a relatively small volume of air because of the relatively low acoustical impedance of air. As in the air direct loudspeaker the response is independent of the frequency in the frequency region where the acoustical radiation resistance is proportional

[3] The American Standard Terminology Z24.1–1951 of the American Standards Association defines the terms projector and hydrophone as follows: "An underwater sound projector is a transducer used to produce sound in water. Note: Where no confusion will result, the underwater sound projector may be shortened to projector. A hydrophone is an electroacoustic transducer which responds to water-borne sound waves and delivers essentially equivalent electrical waves." Accordingly, the terms projector and hydrophone will be used in this book in line with the American Standard Terminology.

[4] Olson, Hackley, Morgan, and Preston, *RCA Review*, Vol. 8, No. 4, p. 698, 1947.

to the square of the frequency and when the system is mass controlled (see Sec. 6.2). In order to obtain mass control down to a relatively low frequency, a relatively limp suspension system must be employed. A limp suspension will not support any appreciable differential pressure between the two sides of the diaphragm. Since the pressure in water increases about .44 pound per foot of depth, some means must be provided to maintain uniform pressure on the two sides of the diaphragm if operation at any appreciable depth is desired. The equalizing means, in one form of the subaqueous dynamic loudspeaker, may consist of a limp rubber bag connected to the air space behind the diaphragm. It will be seen that this system automatically provides equal pressure on the two sides of the diaphragm.

An air bubble in the water in close proximity to the diaphragm provides a shunt series resonant acoustical circuit. The result is that very little energy can be radiated at the resonant frequency of the bubble. The presence of a bubble produces a serious dip in the response frequency characteristic. Since the compensating chamber is an air bubble, means must be provided to prevent the deleterious effects of the bubble in the response range. This is accomplished by enclosing the limp compensating chamber within a rigid case. A small aperture provides communication between the inside and outside of the case. Referring to the mechanical circuit it will be seen that, if the mass of the water in this aperture is made sufficiently large, the resonant frequency of m_3 and C_{M3} will occur below the desired response frequency range.

The direct radiator dynamic projector shown in Fig. 15.1 is designed to operate with maximum efficiency in the audio-frequency range. The diaphragm is about 2 inches in diameter. The air-gap flux is supplied by a permanent magnet. The flux density in the air gap is 15,000 gausses. The performance of the system may be deduced from the mechanical circuit and the constants of the system. The response frequency characteristic of the subaqueous loudspeaker is shown in Fig. 15.1. The variations in the response in the high-frequency range are due to the lack of rigidity of the suspension system. It may be mentioned, in passing, that the response frequency range is considerably greater than that of an air loudspeaker with a comparable driving system.

15.4. Condenser Hydrophone.[5]—A condenser hydrophone is a hydrophone which depends for its operation on variations in electrical capacitance. A typical subaqueous condenser hydrophone shown in Fig. 15.2, consists of a thick diaphragm separated by a small distance from an insulated plate. The mechanical circuit of the vibrating system is shown in Fig. 15.2. The clamped plate has been considered in Sec. 3.5A. The effective mass of the plate is one-third of the total mass. The compliance of the diaphragm can be obtained from the effective mass and the resonant frequency. The water load can be obtained from Sec. 5.8 assuming the effective area of the diaphragm is equal to a piston having one-third the area of the diaphragm.

[5] Olson, Hackley, Morgan, and Preston, *RCA Review*, Vol. 8, No. 4, p. 698, 1947.

As outlined in Sec. 8.2, a uniform pressure-input, voltage-output relationship is obtained when the controlling mechanical impedance in the vibrating system is a compliance. Since the mass of the water load on the diaphragm is very large, a correspondingly small compliance must be provided in order to obtain compliance control. The performance of the vibrating system may be obtained from a consideration of the mechanical circuit of the mechanical system. Employing a plate type diaphragm of Monel, about one-eighth inch in thickness and one and one-half inches in diameter, the fundamental resonant frequency occurs at about 12,000 cycles. The effect of the elements due to the space between the diaphragm and back plate upon the response is very small. The electrical circuit used with the con-
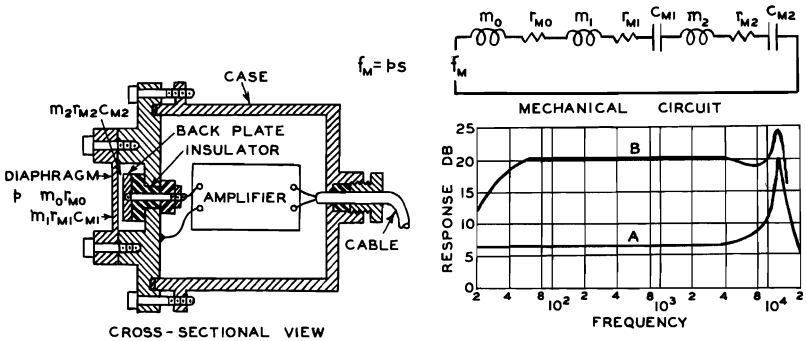
FIG. 15.2. Cross-sectional view, mechanical circuit, and voltage response frequency characteristic of a condenser hydrophone. In the mechanical circuit: m_0 and r_{M0} = the mass and mechanical resistance of the water load upon the diaphragm. m_1, r_{M1}, and C_{M1} = the mass, mechanical resistance, and compliance of the diaphragm. m_2, r_{M2}, and C_{M2} = the mass, mechanical resistance, and compliance of the air film between the diaphragm and the back plate. In the graph: A = the open-circuit voltage response frequency characteristic of the condenser hydrophone. B = the output voltage response characteristic of the combination of a two-stage amplifier and condenser hydrophone for an output impedance of 250 ohms. 0 db = 1 microvolt per dyne per square centimeter.

denser hydrophone is the same as that of the air condenser microphone shown in Fig. 8.5.

The open-circuit voltage response frequency characteristic of the condenser hydrophone, and the output voltage characteristic of the condenser hydrophone and amplifier combination is shown in Fig. 15.2. It will be seen that the response is uniform below the fundamental resonant frequency.

15.5. High-Frequency Direct Radiator Dynamic Projector and Hydrophone.—The response of the projector described in Sec. 15.3 falls off quite rapidly above 15,000 cycles. In the frequency above 10,000 cycles it is possible to employ a smaller diaphragm because the amplitude for moderate power requirements is relatively small. Furthermore, the suspension system can be made very stiff and rigid and still retain mass control. A typical high-frequency, dynamic projector is shown in Fig. 15.3. The

diaphragm in a typical unit is about three-fourths inch in diameter. The response frequency characteristic is shown in Fig. 15.3. Relatively good output is obtained from 10 to 80 kilocycles.

The same unit may be used as a hydrophone over the same frequency range.
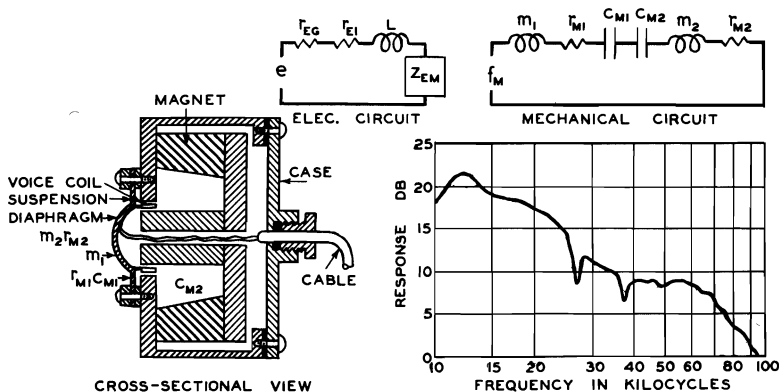
FIG. 15.3. Cross-sectional view, the electrical circuit, the mechanical circuit, and the pressure response frequency characteristic of a high-frequency direct radiator dynamic projector. In the electrical circuit: z_{EM} = the motional electrical impedance. L and r_{E1} = the inductance and electrical resistance of the voice coil. r_{EG} = the electrical resistance of the generator. e = the voltage output of the generator. In the mechanical circuit: m_1 = the mass of the diaphragm. r_{M1} and C_{M1} = the mechanical resistance and compliance of the suspension system. C_{M2} = the compliance of the air chamber behind the diaphragm. m_2 and r_{M2} = the mass and mechanical resistance of the water load.

15.6. Magnetic Projector.—The magnetic projector is a projector designed to operate under water consisting of a resonant diaphragm driven by forces resulting from magnetic reactions. The magnetic projector shown in Fig. 15.4 is the unpolarized armature type. The force[6] on the armature, in dynes, is given by

$$f_M = \frac{C^2 i^2}{4\pi A a^2} \qquad 15.4$$

where i = current in the coil, in abamperes,

$C = 2\pi n A$,

A = area of the center pole, in square centimeters,

n = number of turns in the coil, and

a = spacing, in centimeters.

[6] Olson, "Dynamical Analogies," p. 128, D. Van Nostrand Company, Princeton, N.J., 1943.

If the current in the coil is sinusoidal, then the expression for the current can be written

$$i = i_{max} \sin \omega t \qquad 15.5$$

where i_{max} = amplitude of the current, in abamperes,

$\omega = 2\pi f$,

f = frequency, in cycles per second, and

t = time, in seconds

Substituting equation 15.5 for the current in 15.4, the force on the armature, in dynes, is

$$f_M = \frac{C^2}{4\pi A a^2} i^2_{max} \sin^2 \omega t \qquad 15.6$$

$$= \frac{C^2}{4\pi A a^2} i^2_{max} (\tfrac{1}{2} - \tfrac{1}{2} \cos 2 \omega t) \qquad 15.7$$

Equation 15.7 shows that there is a steady force and an alternating driving force of twice the frequency of the impressed current.
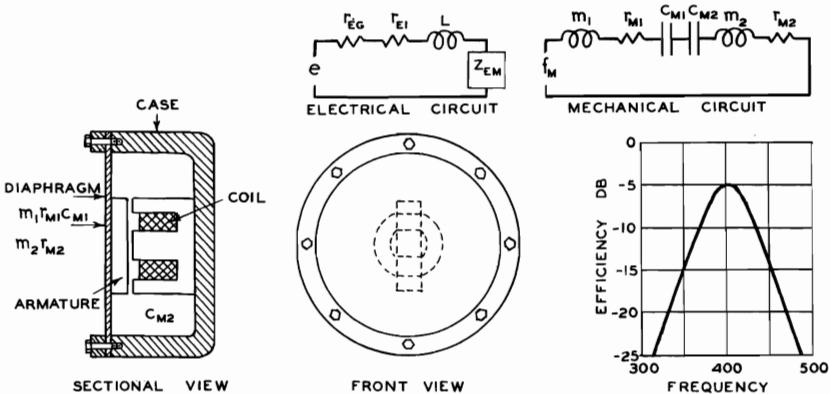
Fig. 15.4. Cross-sectional and front views, electrical circuit, mechanical circuit, and efficiency frequency characteristic of a magnetic projector. In the electrical circuit: z_{EM} = the motional electrical impedance. L and r_{E1} = the inductance and electrical resistance of the coil. r_{EG} = the electrical resistance of the electrical generator. e = the voltage output of the electrical generator. In the mechanical circuit: m_1, r_{M1}, and C_{M1} = the mass, mechanical resistance, and compliance of the diaphragm. C_{M2} = the compliance of the air chamber behind the diaphragm. m_2 and r_{M2} = the mass and mechanical resistance of the water load.

The performance of the system can be determined from a consideration of the mechanical circuit of Fig. 15.4. The effective mass of a clamped plate is one-third the total mass of the plate (see Sec. 3.5A). The effective mechanical impedance load of the water upon the diaphragm may be obtained from Sec. 5.8, assuming that the effective area of the diaphragm is equal to a circular piston having one-third the area of the diaphragm.

The efficiency frequency characteristic of a typical magnetic projector is shown in Fig. 15.4.

The motional electrical impedance[7] of the system is

$$z_{EM} = \frac{2\pi^2 n^4 A^2 i^2}{a^2 z_M}$$ 15.8

where z_{EM} = motional electrical impedance, in abohms,

n = number of turns,

A = area of the center pole, in square centimeters,

i = current, in abamperes,

a = spacing, in centimeters, and

z_M = total mechanical impedance, in mechanical ohms.

At resonance the electrical impedance, z_{EM}, becomes an electrical resistance.

The efficiency, in per cent, is

$$\mu = \frac{r_{ED}}{r_{EM} + r_{ED}} \times 100$$ 15.9

where r_{ED} = damped electrical resistance of the coil, in abohms, and

r_{EM} = motional electrical resistance, in abohms.

From equations 15.8 and 15.9 it will be seen that the efficiency increases with the power input. This characteristic is typical of unpolarized driving systems.

15.7. Magnetostriction Projector.—A magnetostriction projector is a projector in which a diaphragm is driven by the mechanical forces generated in a ferromagnetic rod possessing magnetostrictive properties. A cross-sectional view of a magnetostriction projector is shown in Fig. 15.5. A rod of nickel is coupled to the diaphragm. The polarizing magnetic flux in the rod is supplied by a permanent magnet. The coil surrounding the nickel rod supplies the alternating magnetic flux. The resultant magnetic flux in the rod is the sum of the polarizing and alternating flux. The variation of magnetic flux in the nickel rod causes it to vary in length. The electrical circuit and the mechanical network of a magnetostriction projector are shown in Fig. 15.5.

The driving force,[8] in dynes, generated in the rod is given by

$$f_M = \frac{4\pi N i K}{R} \sin \omega t$$ 15.10

---

[7] Olson, "Dynamical Analogies," p. 130, D. Van Nostrand Company, Princeton, N.J., 1943.

[8] Olson, "Dynamical Analogies," p. 143, D. Van Nostrand Company, Princeton, N.J., 1943.

where $K =$ constant representing the dynamical Joule magnetostriction effect,

$R =$ reluctance of the magnetic circuit,

$N =$ number of turns in the coil,

$i =$ current, in abamperes,

$\omega = 2\pi f$,

$f =$ frequency, in cycles per second, and

$t =$ time, in seconds.

The motional electrical impedance[9] of the system is given by

$$z_{EM} = \frac{16\pi^2 N^2 K^2}{R^2 z_M} \qquad 15.11$$

where $z_{EM} =$ motional electrical impedance, in abohms, and

$z_M =$ total mechanical impedance load upon the rod, including the effective mechanical impedance of the rod, in mechanical ohms.

The normal electrical impedance of the coil is

$$z_{EN} = z_{E1} + z_{EM} \qquad 15.12$$

where $z_{E1} =$ damped electrical impedance of the coil, in abohms.

The damped electrical impedance of the coil comprises an electrical resistance in series with an inductance. The damped electrical impedance and motional electrical impedance are effectively in series as depicted by the electrical circuit of Fig. 15.5.

The lumped constant representation of the system as depicted in the mechanical network of Fig. 15.5 is valid in the region at and near the resonant frequency of the rod. The mass,[10] m_1 in Fig. 15.5 is given by

$$m_1 = \frac{\rho l A}{2} \qquad 15.13$$

where $\rho =$ density of the rod material, in grams per cubic centimeter,

$l =$ length of the rod, in centimeters, and

$A =$ cross-sectional area of the rod, in square centimeters.

The compliance,[11] C_{M1}, in Fig. 15.5 is given by

$$C_{M1} = \frac{8l}{\pi^2 Q A} \qquad 15.14$$

[9] Olson, "Dynamical Analogies," p. 145, D. Van Nostrand Company, Princeton, N.J., 1943.

[10] Olson, "Dynamical Analogies," p. 145, D. Van Nostrand Company, Princeton, N.J., 1943.

[11] Olson, "Dynamical Analogies," p. 146, D. Van Nostrand Company, Princeton, N.J., 1943.

where A = cross-sectional area of the rod, in square centimeters,

   l = length of the rod, in centimeters, and

   Q = Young's Modulus, in dynes per square centimeter.

The performance of the system may be obtained from the constants of the system, the electrical circuit and the mechanical network.

The efficiency, in per cent, is

$$\mu = \frac{r_{EM}}{r_{ED} + r_{EM}} \times 100 \qquad\qquad 15.15$$

where r_{ED} = damped electrical resistance, in abohms, and

   r_{EM} = motional electrical resistance, in abohms.
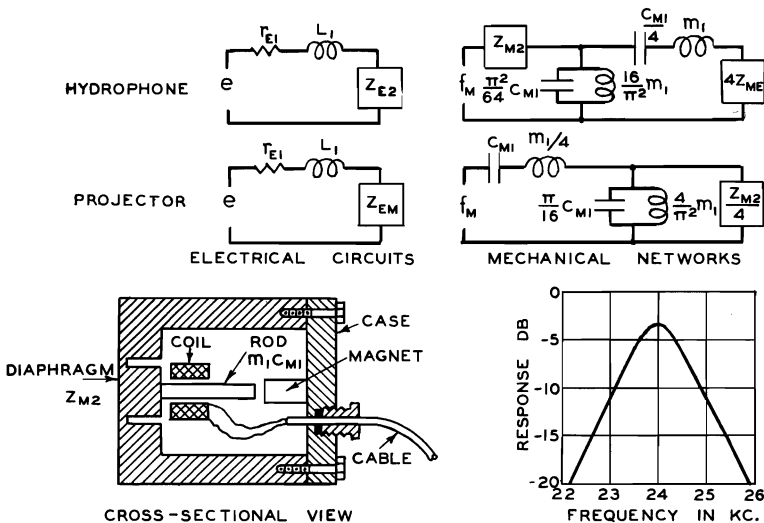
FIG. 15.5.   Cross-sectional view, electrical circuits, mechanical networks, and response frequency characteristics of a magnetostriction projector or hydrophone. In the electrical circuit: z_{E2} = the electrical load when used as a hydrophone. z_{EM} = the electrical motional impedance when used as a projector. L_1 and r_{E1} = the inductance and electrical resistance of the coil. e = the open-circuit voltage output when used as a hydrophone or the voltage input when used as a projector. In the mechanical network: z_{ME} = the mechanical impedance due to the electrical system. z_{M2} = the mechanical impedance of the diaphragm and water load. m_1 = the mass of the rod. C_{M1} = the compliance of the rod. f_M = the driving force.

The efficiency ranges from 10 to 50 per cent depending upon the band width, the type of rod, and the frequency.

A typical response frequency characteristic is shown in Fig. 15.5.

Magnetostriction projectors of the type shown in Fig. 15.5 are suitable for frequency band response in the frequency range from 10 to 50 kilocycles.

A single-rod system is shown in Fig. 15.5.   Projectors using a large

number of rods coupled to a large diaphragm is the usual arrangement. The action is essentially the same as that of a single-unit system.

In Fig. 15.5 a single rod is coupled to a small diaphragm. In the echo direction and ranging sonar a highly directional projector-hydrophone must be employed in order to obtain the required angular accuracy. The required order of directivity can be obtained if the diameter of the diaphragm corresponds to several wavelengths of the medium. See Sec. 2.13. A projector-hydrophone with a large diaphragm is shown in Fig. 15.6. The system
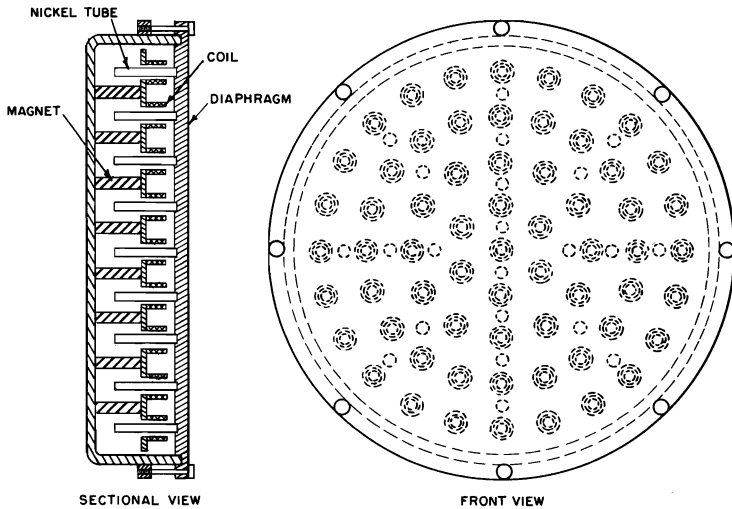
FIG. 15.6. Cross-sectional view of a magnetostriction echo ranging projector-hydrophone consisting of a large number of nickel tube transducers·coupled to a large diaphragm.

consists of a diaphragm driven by a large number of magnetostriction tubes. All of the rods vibrate with the same phase and as a consequence all parts of the diaphragm vibrate with the same phase. Under these conditions the directivity pattern is that of a circular piston. See Sec. 2.13.

15.8. Magnetostriction Hydrophone.—A magnetostriction hydrophone is a hydrophone in which a voltage is generated in a coil surrounding a rod having magnetostrictive properties. A cross-sectional view of a magnetostriction hydrophone is shown in Fig. 15.5. The mechanical network of the vibrating system and electrical circuit is also shown in Fig. 15.5. The mass m_1 and the compliance C_{M1} are given by equations 15.13 and 15.14.

The driving force, f_M, in dynes, is

$$f_M = pS \qquad 15.16$$

where p = sound pressure, in dynes per square centimeter, and
S = area of the diaphragm, in square centimeters.

The mechanical impedance[12] due to the electrical system is

$$z_{ME} = \frac{16\pi^2 N^2 K^2}{z_E R^2}$$

15.17

where N = number of turns in the coil,

K = magnetostriction constant,

R = reluctance of the magnetic circuit, and

$z_E = z_{E1} + z_{E2}$,

z_{E1} = electrical impedance of the coil, in abohms, and

z_{E2} = electrical impedance of the external circuit, in abohms.
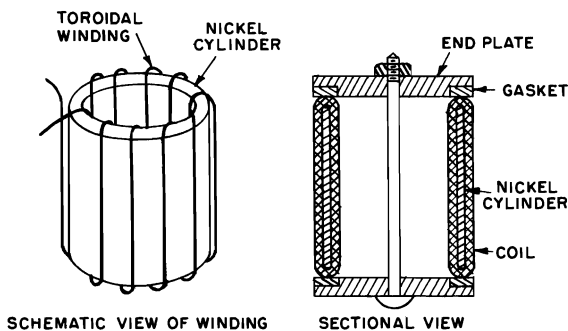
FIG. 15.7. Schematic perspective and cross-sectional views of a cylindrical magnetostriction hydrophone.

The voltage,[13] in abvolts, developed in the coil due to deformation of the rod is

$$e = \frac{4\pi N K}{R} \dot{x}$$

15.18

where N = number of turns in the coil,

R = reluctance of the magnetic circuit,

K = constant representing the dynamical Villari magnetostriction effect, and

$\dot{x}$ = velocity at the point of application of the force to the rod, in centimeters per second.

The internal voltage developed may be obtained from equation 15.18. The voltage developed across the external electrical impedance z_{E2} may be obtained from a consideration of the electrical circuit.

The voltage response frequency characteristic is usually the same as that of the projector shown in Fig. 15.5.

A cylindrical tube of magnetostrictive material may be used as a radial sound radiator or receiver, see Fig. 15.7. The coil is wound in a toroidal

[12] Olson, "Dynamical Analogies," p. 163, D. Van Nostrand Company, Princeton, N.J., 1943.

[13] Olson, "Dynamical Analogies," p. 162, D. Van Nostrand Company, Princeton, N.J., 1943.

fashion over the tube and the polarizing and alternating flux paths are circumferential in the tube. The fundamental resonant frequency of a radiator of this type without any external load is given by

$$f = \frac{c}{2\pi R}$$
  15.19

where c = velocity of sound in the magnetostrictive material, and
  R = radius of the tube.

Transducers of this type have been used as both hydrophones and projectors.

15.9. Quartz Crystal Projector.[14]—A quartz crystal projector is a projector in which a quartz crystal is driven by mechanical forces generated in the crystal due to converse piezoelectric properties. A cross-sectional view of the quartz crystal projector is shown in Fig. 15.8. The electrical circuit and the mechanical network of the quartz crystal projector are shown in Fig. 15.8.

The driving force,[15] in dynes, generated in the crystal is given by

$$f_M = \frac{KQAe}{l}$$
  15.20

where K = constant of the crystal, 6.4×10^{-8} for quartz,
  Q = Young's modulus, in dynes per square centimeter,
  A = cross-sectional area of the crystal,
  e = applied voltage, in statvolts, and
  l = effective length of the crystal, in centimeters.

The motional electrical impedance[16] of the system is

$$z_{EM} = \frac{l^2}{K^2 Q^2 A^2} z_M$$
  15.21

where z_{EM} = motional electrical impedance, in statohms, and
  z_M = total mechanical impedance of the vibrating system, in mechanical ohms.

The normal electrical impedance of the crystal system is

$$z_{EN} = \frac{z_{EM}}{1 + j\omega C_{E1} z_{EM}}$$
  15.22

where z_{EM} = motional electrical impedance, equation 15.21, and
  C_{E1} = electrical capacitance of the crystal in the absence of motion, in abfarads.

The damped electrical impedance and the motional electrical impedance are effectively in parallel as depicted by the electrical network of Fig. 15.8.

[14] Olson, Hackley, Morgan, and Preston, *RCA Review*, Vol. 8, No. 4, p. 698, 1947.
[15] Olson, "Dynamical Analogies," p. 149, D. Van Nostrand Company, Princeton, N.J., 1943.
[16] Olson, "Dynamical Analogies," p. 150, D. Van Nostrand Company, Princeton, N.J., 1943.

The lumped constant representation of the system as depicted in the mechanical network of Fig. 15.8 is valid in the region at and near the resonant frequency of the crystal. The mass,[17] m_1, in Fig. 15.8 is given by

$$m_1 = \frac{\rho l A}{2}$$ 15.23

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

l = length of the crystal, in centimeters, and

A = cross-sectional area of the crystal, in square centimeters.
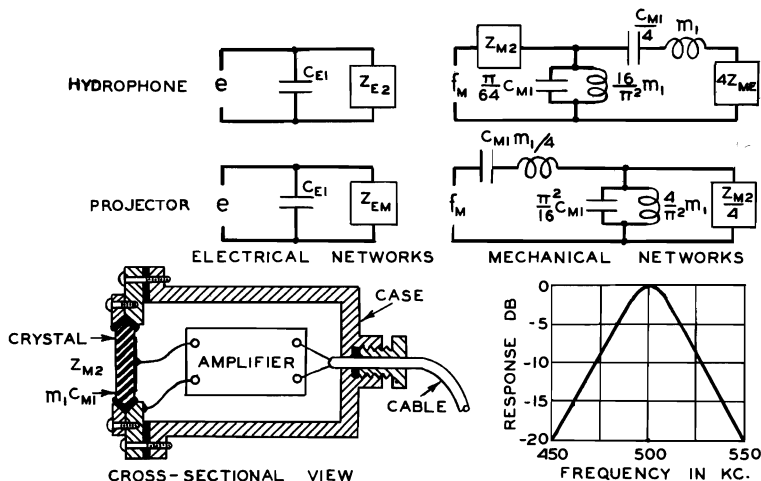
FIG. 15.8. Cross-sectional view, electrical networks, mechanical networks, and response frequency characteristic of a quartz crystal projector or hydrophone. In the electrical circuit: z_{E2} = the electrical impedance of the load when used as a hydrophone. z_{EM} = the electrical motional impedance when used as a projector. C_{E1} = the electrical capacitance of the crystal. e = the voltage output when used as a hydrophone or the voltage input when used as a projector. In the mechanical network: z_{ME} = the mechanical impedance due to the electrical system. z_{M2} = the mechanical impedance of the water load upon the crystal. m_1 = the mass of the crystal. C_{M1} = the compliance of the crystal. f_M = the driving force.

The compliance,[18] C_{M1}, in Fig. 15.8 is given by

$$C_{M1} = \frac{8l}{\pi^2 Q A}$$ 15.24

where A = cross-sectional area of the crystal, in square centimeters,

l = length of the crystal, in centimeters, and

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

[17] Olson, "Dynamical Analogies," p. 151, D. Van Nostrand Company, Princeton, N.J., 1943.

[18] Olson, "Dynamical Analogies," p. 151, D. Van Nostrand Company, Princeton, N.J., 1943.

The performance of the system may be obtained from the constants of the system, the electrical circuit and the mechanical circuit.

The efficiency, in per cent, is

$$\mu = \frac{r_{EM}}{r_{ED} + r_{EM}} \times 100 \qquad\qquad 15.25$$

where r_{ED} = electrical resistance in the absence of a load in statohms, and
r_{EM} = material electrical resistance with the normal load, in statohms.

The efficiency of the quartz crystal projector is very high, being very close to 100 per cent.

A typical response frequency characteristic is shown in Fig. 15.8.

Quartz crystal projectors of the type shown in Fig. 15.8 are suitable for frequency band response in the frequency range from 100 kilocycles to 100 megacycles. It appears that the quartz crystal projector is the only system suitable for high conversion in the ultra-high-frequency range.

15.10. Quartz Crystal Hydrophone.[19]—A quartz crystal hydrophone is a hydrophone in which a voltage is generated in a crystal having converse piezoelectric properties. A cross-sectional view of a quartz crystal hydrophone is shown in Fig. 15.8. The mechanical network of the mechanical system and the electrical network are also shown in Fig. 15.8. The mass m_1 and the compliance C_{M1} are given by equations 15.23 and 15.24.

The driving force, f_M, in dynes, is

$$f_M = pS \qquad\qquad 15.26$$

where p = sound pressure, in dynes per square centimeter, and
S = area of the diaphragm, in square centimeters.

The mechanical impedance[20] due to the electrical circuit is

$$z_{ME} = \frac{K^2 Q^2 A^2}{l^2} z_E \qquad\qquad 15.27$$

where K = constant of the crystal, 6.4×10^{-8} for quartz,
Q = Young's modulus, in dynes per square centimeter,
l = length of the crystal, in centimeters,
A = area of the electrode, in square centimeters,

$$z_E = \frac{z_{E1} z_{E2}}{z_{E1} + z_{E2}}, \qquad\qquad 15.28$$

$$z_{E1} = \frac{1}{j\omega C_{E1}}, \qquad\qquad 15.29$$

where C_{E1} = electrical capacitance of the generator, in statfarads, and
z_{E2} = electrical impedance of the external load, in statohms.

[19] Olson, Hackley, Morgan, and Preston, *RCA Review*, Vol. 8, No. 4, p. 698, 1947.
[20] Olson, "Dynamical Analogies," p. 167, D. Van Nostrand Company, Princeton, N.J., 1943.

The generated electromotive force,[21] e, in statvolts, across the electrical impedances z_{E1} and z_{E2} is

$$e = \frac{KQA\dot{x}}{l}\, z_E \qquad\qquad 15.30$$

where $\dot{x} =$ velocity of the crystal, in centimeters per second.

The voltage response frequency characteristic is usually the same as that of the projector shown in Fig. 15.8.

15.11. Quartz Crystal Sandwich Projector and Hydrophone.— A quartz crystal sandwich projector is a projector consisting of two blocks of metal cemented to the two sides of a relatively thin quartz crystal (Fig. 15.9). One of the metal blocks is terminated in water and the other is terminated in air. The maximum output occurs when the over-all effective length is one-half wavelength. Under these conditions the center of the crystal coincides with a velocity node and a pressure maximum. Maximum efficiency is obtained at this frequency. The band width of transmission depends upon the density of the material of the blocks. Response fre-
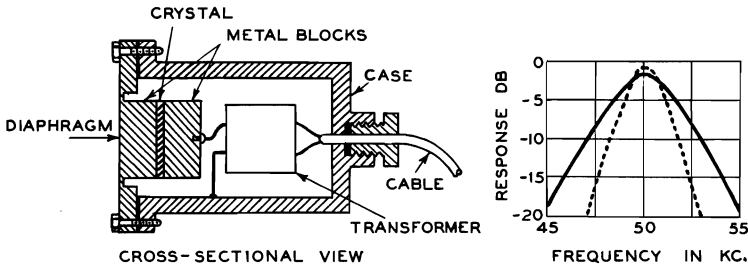
FIG. 15.9. Cross-sectional view and response frequency characteristic of a quartz crystal sandwich projector or hydrophone. The solid-line response frequency characteristic depicts the response for aluminum blocks and the dotted-line response frequency characteristic depicts the response for steel blocks.

quency characteristics for steel and aluminum blocks are shown in Fig. 15.9. As would be expected, blocks with a lower density yield a wider frequency band. The quartz crystal sandwich projector conserves quartz since only a thin layer is used as compared to the large block for the all-quartz subaqueous loudspeaker. In addition, the electrical impedance is lower. With metal blocks of the same density as quartz, the shape of the response frequency characteristic is approximately the same as a solid block of quartz.

The system shown in Fig. 15.9 may also be used as a hydrophone. The shape of the response frequency characteristic is approximately the same for the hydrophone as that shown in Fig. 15.9 for the projector.

[21] Olson, "Dynamical Analogies," p. 169, D. Van Nostrand Company, Princeton, N.J., 1943.

15.12. Rochelle Salt Crystal Projector and Hydrophone.—A Rochelle salt crystal projector is a projector in which the crystal is driven by mechanical forces generated in the crystal due to converse piezoelectric properties. One form of a Rochelle salt crystal projector is shown in Fig. 15.10.   Y- or X-cut crystals are cemented to the metal back plate.   The front face of the crystal assembly is covered with a rubber diaphragm.   In order to insure good mechanical coupling between the crystals and the rubber the thin space between the rubber and the face of the crystal assembly is filled with castor oil.   Resonance occurs when the effective over-length of the crystals and back plate is one-half wavelength.   A typical response frequency characteristic is shown in Fig. 15.10.

The system shown in Fig. 15.10 may also be used as a hydrophone.   The shape of the response frequency characteristic is approximately the same for the hydrophone as that shown in Fig. 15.10 for the projector.

There has been another type crystal[22] developed which may displace
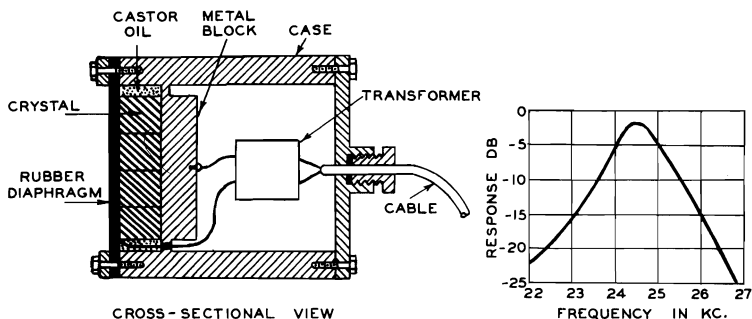
FIG. 15.10.   Cross-sectional view and response frequency characteristic of a Rochelle salt crystal projector or hydrophone.

Rochelle salt for many applications.   This crystal is ammonium dihydrogen phosphate and has been given the abbreviation ADP.

ADP crystals provide high electromechanical coupling.   They are free from nonlinear response and hysteresis effects.   ADP has no water of crystallization and does not dehydrate.   It is stable up to temperatures as high as 100° centigrade.   Rochelle salt dehydrates below 35 per cent humidity and disintegrates at temperatures above 55° centigrade.   For these reasons, the ADP crystals may displace Rochelle salt in many piezoelectric applications.

15.13. Barium Titanate Hydrophone.—A barium titanate hydrophone is a hydrophone in which a voltage is generated in a polarized barium titanate ceramic.   A cross-sectional view of a barium titanate hydrophone is shown in Fig. 15.11.   The ceramic element consists of a cylinder of barium titanate. The cylinder is equipped with electrodes on the outside and inside surfaces.

[22] Mason, W. P., *Bell Labs. Record*, Vol. 24, No. 7, p. 257, 1946.

These electrodes are connected to the two conductors of the cable. The cylinder is prepolarized by the application of 20,000 volts per centimeter of thickness. The unit then behaves somewhat in the manner of piezoelectric transducers. The ceramic unit is sealed against the sea by a rubber watertight casing. The hydrophone of Fig. 15.11 is nondirectional in the plane normal to the axis of the cylinder. The same unit may be used as a projector. In this case it radiates with equal efficiency in all directions in the plane normal to the axis of the cylinder. Other types of barium titanate transducers may also be employed as projectors or hydrophones in a manner similar to that of piezoelectric crystals for these applications because the actions are alike in many respects.

FIG. 15.11. Cross-sectional view of a barium titanate ceramic hydrophone.

15.14. Passive Sonar.[23]—A passive sonar is an underwater sound system used to detect sounds in the water. In general, the detection system is the human ear. The passive sonar was the first underwater sound detection system. The earliest passive sonar was an all acoustic system in the form of an underwater stethoscope. The diaphragm of the stethoscope was immersed in the water. By the use of a large diaphragm a reasonable match in acoustical impedance of the water and the air in the tubes which led to the ear could be obtained. Later, an underwater carbon microphone was used. However, the self-noise of the microphone precluded the possibility of the detection of faint underwater sounds. With the advent of the vacuum tube, all of the hydrophones described in this chapter could be used for the detection of underwater sounds. Passive systems range in complexity from a single small hydrophone, amplifier, and telephone receivers to large directional hydrophone arrays, amplifiers, and means for both listening and recording. The nondirectional passive sonar can only be used to detect the presence of a sound. The directional passive sonar can provide additional information in the form of the direction of the sound.

15.15. Echo Depth Sounding Sonar.—The depth of the sea may be determined by means of sonic echo sounding. The schematic arrangement of the apparatus for determining the depth of the sea is shown in Fig. 15.12. A high-frequency oscillator operating at a fixed frequency in the range from 10 to 100 kilocycles is coupled to a power amplifier. The projector and hydrophone are mounted flush with the hull of the vessel and directed

---

[23] SONAR is a term derived from *SO*und *N*avigation *A*nd *R*anging. It embraces all types of underwater sound equipment used on ships for locating and tracking submarines, for depth sounding, for underwater communication and as a navigational aid.

downward. The sequence switch connects the power amplifier to the projector for about a few milliseconds. The pulse sent out by the projector is reflected from the ocean floor and received by the hydrophone. The output of the hydrophone is coupled to the vertical amplifier by means of the sequence switch. The output of the vertical amplifier is fed to the recorder. At the time the pulse is sent out, the recording pen begins to move to the right at a constant rate. When the reflected sound impulse is received by the hydrophone it is converted into the corresponding electrical impulse. This impulse is amplified by the vertical amplifier and fed to the record mechanism. The pen is actuated by this impulse and places a dot upon the paper. The distance of the dot from the base line on the paper is proportional to the depth. .Graph paper calibrated in fathoms is used for the
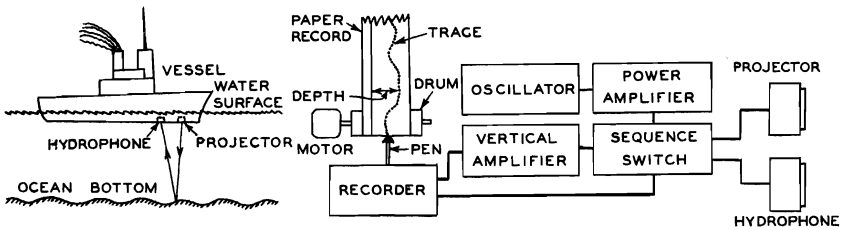
FIG. 15.12.   Echo depth sounding sonar system.

record paper. The process is repeated every few seconds. In this way, a continuous record of the depth of the sea is obtained. Depths from 5 feet to several thousand feet may be measured. Both magnetostriction or crystal projectors and hydrophones have been used for sonic depth indicators.

15.16. Echo Direction and Ranging Sonar.[24]—The position of submerged submarines may be determined by means of sonic echo ranging equipment. The general arrangement and the schematic diagram of echo direction and ranging equipment are shown in Fig. 15.13. The oscillator operates at a fixed frequency somewhere between 15 and 50 kilocycles, the customary frequency being about 25 kilocycles. The sequence switch connects the power amplifier to the projector-hydrophone used as a projector for a few milliseconds. Then the sequence switch connects the amplifier to the projector-hydrophone used as a hydrophone. If a reflected sound impulse is received, it will be indicated as a "pip" on the cathode-ray tube or other visual indicating means and reproduced as a "ping" on the air loudspeaker. The horizontal sweep system drives the cathode-ray spot to the right starting at zero when the sound impulse is sent out. The horizontal axis of the tube is calibrated in yards. The diameter of the diaphragm of the projector-hydrophone is usually about 5 to 10 wavelengths, see Sec. 15.7. Since the same unit is used for both sending and receiving, the ordinates of the directional patterns shown in Fig. 2.13 must be squared. It will be seen that under these conditions the directional pattern of a

[24] Evans, R. J., *Electronics*, Vol. 19, No. 8, p. 88, 1946.

diaphragm 6 wavelengths in diameter is quite narrow and it is possible to determine the direction of the submerged submarine quite accurately. The combination of direction and range gives the position of the submerged submarine.

In the detection of submarines by echo ranging sonar, use can be made of the Doppler effect to determine the velocity of the target with respect to the
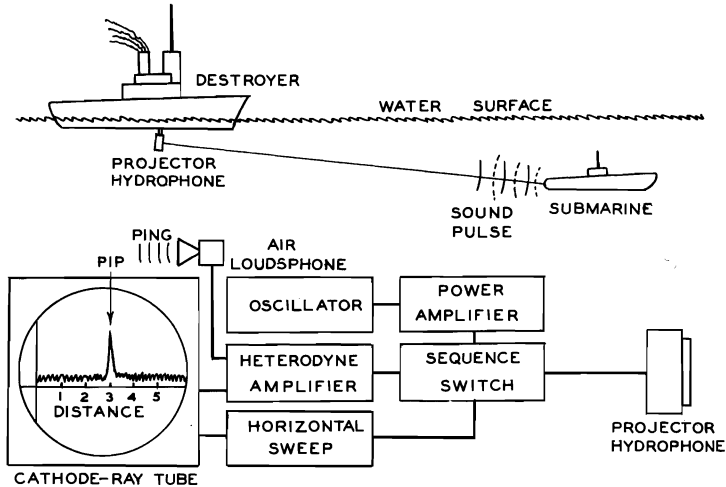
FIG. 15.13. Echo direction and ranging sonar system.

transmitting point, see Sec. 1.10. The frequency f_e, in cycles per second, of the returning echo is given by

$$f_e = f_s \left[\frac{u + 2 \left(V_s \cos \theta_s - V_t \cos \theta_t \right)}{u} \right] \qquad 15.31$$

where f_s = frequency of the source, in cycles per second,

u = velocity of sound propagation in the water, in centimeters per second,

V_s = speed of the source in the water, in centimeters per second,

V_t = speed of the target in the water, in centimeters per second,

θ_s = angle between the direction of motion of the source and the line joining the source and the target, and

θ_t = angle between the direction of motion of the target and the line joining the source and the target.

If the velocity of the source is known, the velocity of the target relative to the source can be obtained from a comparison of the frequency of the returning echo with the frequency of the outgoing pulse.

15.17. Scanning Echo Direction and Ranging Sonar.—An echo ranging direction and ranging sonar system for submarine detection, employing a highly directional projector-hydrophone, has been described in Sec. 15.14 and depicted in Fig. 15.14. The searching process is carried out

by turning the projector over a total angle of 360°. In an extension of the system, an electronically operated scanning echo direction and ranging sonar has been developed in which a plan view of the field and the position of the submarine are depicted on a cathode-ray tube, see Fig. 15.14. The combination projector-hydrophone consists of a series of elements arranged in cylindrical form. In transmitting, all the elements of the projector are
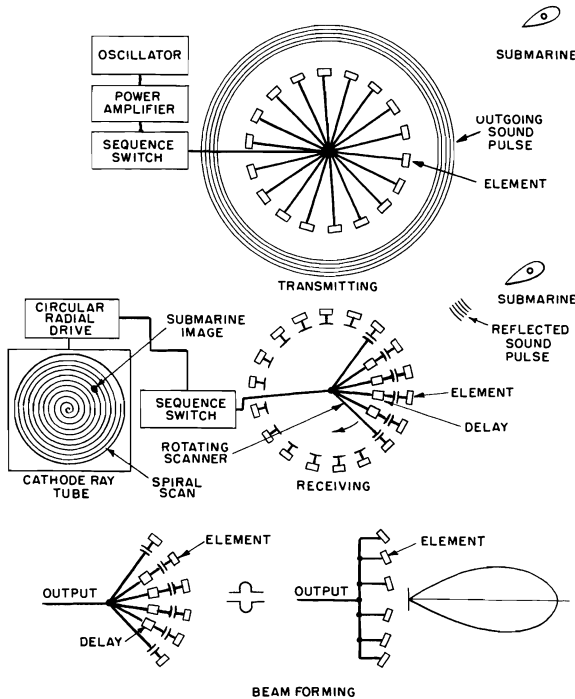
FIG. 15.14. Scanning echo direction and ranging sonar showing the apparatus arrangement in transmitting and receiving and the beam forming system.

connected to the power amplifier and a pulse of high-frequency waves is sent out in all directions in the horizontal plane. The system may operate at any frequency range. However, in general, it is of the order of 20 kilocycles. The pulse may be of 5 to 20 milliseconds duration. As soon as the pulse has been transmitted, the sequence switch converts the system to receiving. In the receiving condition, a scanning switch connects a group of elements to the indicating system. A circular line array of this type exhibits a very broad directivity pattern, see Sec. 2.11. Such a system would be unsuitable because high directivity is required in receiving in order to obtain the direction of the submarine with the required accuracy. In order to obtain high directivity, a delay system is combined with the scanner which converts the

circular-type array to a line array, see Sec. 2.6. The conversion from a circular-line array to a straight-line array is graphically depicted in Fig. 15.14. The scanning and delay system then provides a highly directional rotating receiving beam. When the switch is made from transmitting to receiving, the beam of the cathode-ray tube begins a spiral scan starting at the center of the tube. The beam scanning system and the cathode-ray indicating system are interlocked with respect to angle. The radial distance on the cathode-ray tube is proportional to time and, therefore, to the distance of the target. When an electrical pulse enters from the scanning system produced by a reflected sound pulse the indicating system, the intensity of the beam of the cathode-ray tube, is increased, thereby producing a bright spot of high light intensity. This spot depicts the position of the submarine with respect to distance and angle. The sequence switch carries out a continuous series of sending and receiving programs. The time interval of this sequence can be adjusted for large or small distances of search.
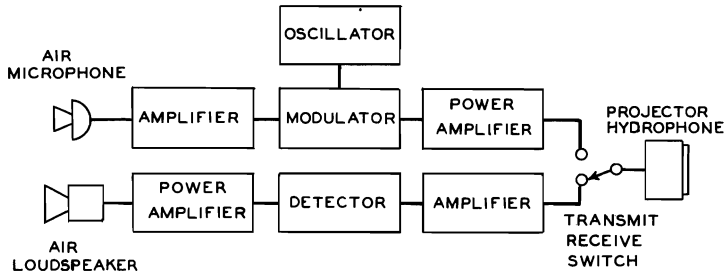
FIG. 15.15. Communication sonar system.

15.18. Communication Sonar.—Signaling from ship to ship, particularly by submerged submarines, is another application of underwater sound equipment. The sound signal in the water may be a replica of the original voice or telegraph frequency or it may be a modulated ultrasonic frequency. In general, the latter system is used because the use of a high frequency makes it possible to obtain a sharp directional pattern with a relatively small diaphragm. A schematic view of a voice modulated ultrasonic underwater communication system is shown in Fig. 15.15. In general, the same underwater transducer is used as both a projector and hydrophone. Both magnetostriction or crystal transducers have been used for the projector-hydrophone. The frequency band width of the projector-hydrophone must be sufficiently broad to accommodate the modulated signal. The system shown in Fig. 15.15 is the conventional amplitude modulated system. Of course, frequency, phase, or single side band amplitude modulation with the carrier suppressed may also be used. A carrier frequency of from 10 to 100 kilocycles has been used. Ranges up to 20 miles have been obtained under good conditions.

16

ULTRASONICS

16.1. Introduction.—The term ultrasonics is used to designate the general subject of sound in the frequency region above 15 kilocycles per second. Many of the effects of ultrasonic vibrations have been known for about 25 years. However, it is only within the last few years that there has been any appreciable growth in the applications of ultrasonics. One of the reasons being that in the early stages of the science of ultrasonics the efficiency of conversion of electrical to acoustical energy was low. During World War II many new developments were made in all manner of ultrasonic transducers. The outstanding results of these developments were improvements in the form of high efficiency and increased power handling capacity. These improvements have made it possible to handle materials on a relatively large scale. Developments in the electronic field of switching, gating, and display have made it possible to use ultrasonics for non-destructive testing. Techniques developed in underwater sound have been extended to the field of medicine in both analysis and treatment. It is the purpose of this chapter to describe some of the common ultrasonic generators and a few of the salient applications of ultrasonics.

16.2. Ultrasonic Generators.[1,2,3,4,5,6,7,8,9,10,11,12,13,14]—Ultrasonic energy may be applied to gases, liquids, and solids to produce desired changes or effects. The systems for the production of ultrasonic vibrations are termed ultrasonic generators. For many of these applications, in liquids and solids, the underwater projectors described in Chapter XV on Under-

[1] Pierce, G. W., *Proc. Inst. Rad. Eng.*, Vol. 17, No. 1, p. 42, 1929.
[2] Salisbury and Porter, *Rev. Sci. Inst.*, Vol. 10, No. 4, p. 142, 1939.
[3] Salisbury and Porter, *Rev. Sci. Inst.*, Vol. 10, No. 9, p. 269, 1939.
[4] Wood, "Supersonics," Brown Univ. Press, Providence, R.I., 1939.
[5] Bergmann, "Ultrasonics," John Wiley and Sons, New York, N.Y., 1938.
[6] St. Clair, H. W., *Rev. Sci. Inst.*, Vol. 12, No. 5, p. 250, 1941.
[7] Young, V. J., *Electronics*, Vol. 17, No. 3, p. 122, 1944.
[8] Carlin, "Ultrasonics," McGraw-Hill Book Company, New York, N.Y., 1949.
[9] Bergmann, "Der Ultraschall," 5th Ed., S. Hirsel Verlag, Zurich, 1949.
[10] Vigoureux, "Ultrasonics," Chapman and Hall, Ltd., London, 1950.
[11] Richardson, "Ultrasonic Physics," Elsevier Press, Inc., Houston, Texas, 1952.
[12] Mattiat, Oskar, *Jour. Acous. Soc. Amer.*, Vol. 25, No. 2, p. 291, 1953.
[13] Crawford, "Ultrasonic Engineering," Academic Press, New York, N.Y., 1955.
[14] Hueter and Bolt, "Sonics," John Wiley and Sons, New York, N.Y., 1955.

water Sound are particularly suitable as ultrasonic generators. For some ultrasonic applications in gases, loudspeakers, and microphones described in Chapters VI, VII, and VIII may be used. For the production of high-intensity sound waves in air, whistles, and sirens are the most common ultrasonic generators. The Galton whistle[15] is similar to a small organ pipe. Ultrasonic energy of up to 10 watts may be produced with a frequency range up to 40 kilocycles with whistles of this design. In the Hartman generator[16,17] a jet of air moving with a speed greater than the velocity of sound impinges upon a small cavity thereby producing ultrasonic vibrations of high intensity. The Hartman generator has produced outputs of
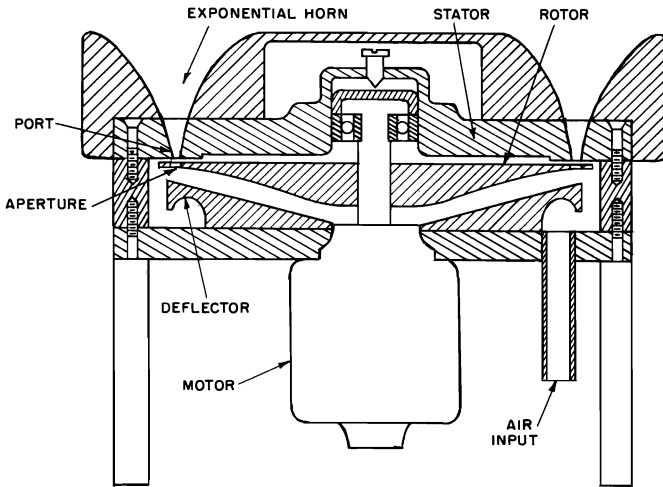
FIG. 16.1.   Cross-sectional view of an ultrasonic siren.

50 watts and frequencies as high as 120 kilocycles. The ultrasonic siren[18] consists of a high pressure source of air modulated by a rotary interrupter at ultrasonic frequencies, see Fig. 16.1. The stator is equipped with one hundred tapered holes located on a circle six inches in diameter. The tapered holes are coupled to the throat of a ring-type exponential horn. The rotor located next to the stator consists of a disk of duraluminum with holes corresponding to those in the stator. The rotor is driven by a motor. Compressed air is fed to the chamber behind the rotor. The rotor throttles the air stream and produces sound waves of a frequency corresponding to the rate of interruption. Ultrasonic waves up to 34 kilocycles can be produced with the ultrasonic siren. Employing an air pressure of three

[15] Nyborg, Schilling, and Thorpe, *Jour. Acous. Soc. Amer.*, Vol. 19, No. 1, p. 287, 1947.
[16] Hartman, J., *Jour. Sci. Inst.*, Vol. 16, No. 12, p. 1946, 1946.
[17] Monson and Binder, *Jour. Acous. Soc. Amer.*, Vol. 25, No. 5, p. 1007, 1953.
[18] Nyborg, Schilling, and Thorpe, *Jour. Acous. Soc. Amer.*, Vol. 19, No. 1, p. 287, 1947.

atmospheres, two kilowatts of acoustical power can be obtained at an efficiency of 20 per cent.

The theory and operation of magnetostriction underwater projectors have been outlined in Sec. 15.7. A magnetostriction ultrasonic generator[19] suitable for the processing of materials is shown in Fig. 16.2. The element consists of a consolidated stack of thin nickel laminations. Magnetostriction generators operate with the highest efficiency in the lower frequency portion of the ultrasonic band because the electrical losses produced by eddy currents increase with frequency. Ultrasonic generators of the type shown in Fig. 16.2 have been built for the 20 kilocycle region with an efficiency of 60 per
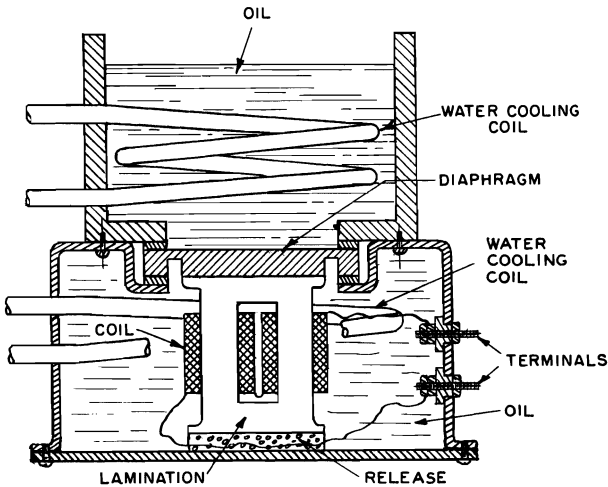
FIG. 16.2. Cross-sectional view of a magnetostriction ultrasonic generator.

cent. Magnetostriction ultrasonic generators are mechanically rugged and capable of producing large acoustical outputs. The electrical impedance can be made any value suitable for coupling directly to a vacuum tube.

The theory and operation of piezoelectric underwater projectors have been outlined in Secs. 15.9, 15.11, and 15.12. A piezoelectric ultrasonic generator employing an X-cut quartz crystal is shown in Fig. 16.3. The crystal is plated on the two sides. The electrical input is applied to these two electrodes. The maximum ultrasonic power output occurs at the resonant frequency of the crystal and the load. The maximum acoustical power output that can be obtained is about 40 watts per square centimeter in the frequency range from 300 to 1000 kilocycles. The efficiency of conversion of electrical to acoustical energy of the quartz transducer is very high. Efficiencies of the order of 90 per cent are quite common. Relatively high temperatures do not reduce the efficiency of quartz transducers. The

[19] Camp, Leon, *Jour. Acous. Soc. Amer.*, Vol. 20, No. 5, p. 616, 1948.

electrical impedance of the quartz crystal is relatively high. Therefore, a transformer must be used to couple the crystal to a vacuum tube amplifier.
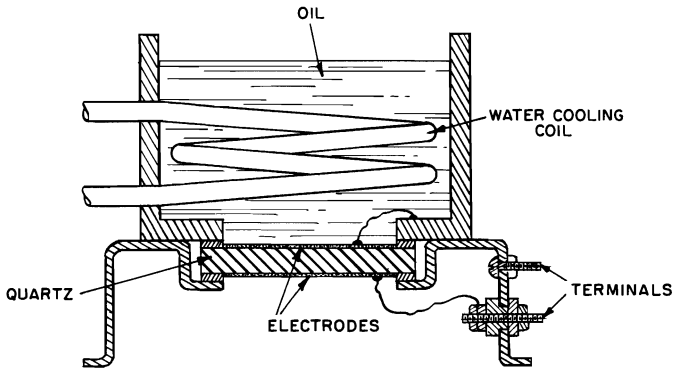
FIG. 16.3. Cross-sectional view of a quartz crystal ultrasonic generator.

Other piezoelectric crystals beside quartz may also be used in ultrasonic generators, as for example, Rochelle salt and ammonium dihydrogen phosphate. However, because of the excellent mechanical qualities, quartz
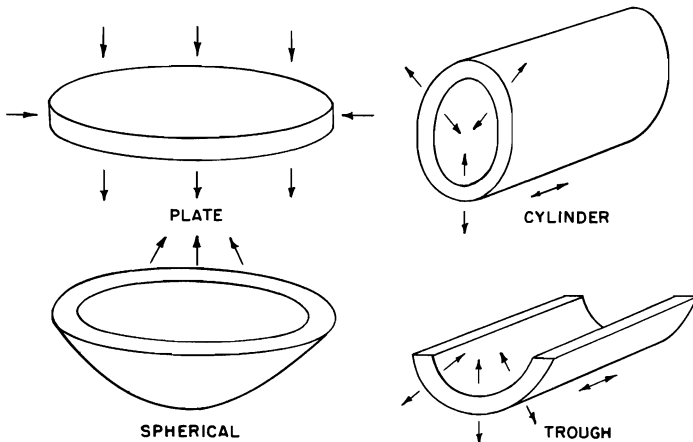
FIG. 16.4. Ceramic transducers. Arrows indicate modes of vibration.

crystals are extensively used in ultrasonic generators in the high-frequency ultrasonic region.

Barium titanate is an electrostrictive material which by prepolarization assumes properties which are similar to piezoelectric materials. Barium titanate ceramics of four different shapes[20] are shown in Fig. 16.4. The

Ceramics may be fabricated in the shapes shown in Fig. 16.4 by casting, pressing, or extruding before the material is fixed in a furnace.   The change in physical dimensions is proportional to the applied voltage as long as the latter is less than the prepolarizing voltage.   Silver electrodes are fired onto the major faces.   Prepolarization perpendicular to these faces may be obtained by subjecting it to an electric field of 20,000 volts per centimeter of thickness at temperatures above the Curie point of 120° centigrade and maintaining it until the material is cooled to room temperature.   Bars and disks prepared in this way behave in a manner similar to quartz crystals but with different mechanical and electrical properties.   The outstanding difference is the low electrical impedance.   The voltage required is about 1/100 that of quartz.   For example, an acoustical power of one watt per square centimeter at a frequency of 100 kilocycles can be achieved by the
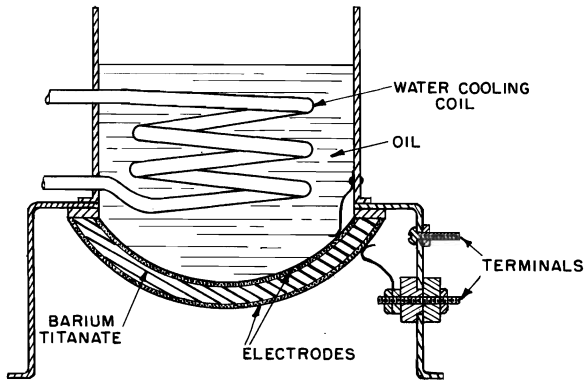
FIG. 16.5.   Cross-sectional view of a focusing barium titanate ceramic ultrasonic generator.

application of 100 volts.   A focusing barium titanate ultrasonic generator is shown in Fig. 16.5.   It is operated in a thickness vibration.   For a frequency of 400 kilocycles the thickness of the bowl is $\frac{1}{4}$ inch.

The ultrasonic generators shown in Figs. 16.2, 16.3, and 16.5 are all coupled to an oil reservoir.   The material to be treated is usually placed in a thin wall, glass container and lowered into the oil.   There is considerable heat generated in the oil by the action of the high intensity sound waves. Therefore, in order to maintain the temperature at a normal value, the oil chamber is equipped with a cooling coil through which cold water is circulated.   If the material to be treated is a liquid, a continuous process may be employed by passing the liquid through a pipe immersed in the oil.   In some processes, the material may be substituted for the oil and the transfer of ultrasonic energy is transferred directly from generator to the material.

In high power systems, the dissipation of energy in the transducer may be high and some means must be provided for carrying away the generated heat energy in order to obviate overheating of the transducer.   As an

example, the magnetostriction transducer in Fig. 16.2 is cooled by being immersed in oil. The oil chamber is equipped with cooling coils through which cold water is circulated. In order to prevent radiation of vibrational energy into the coil, the end of the magnetostriction vibrator is terminated in a low acoustical impedance, as for example, as "foam" type material.

16.3. Cavitation.[21,22,23,24,25]—If a sound wave is impressed upon a liquid and the intensity is increased, a point will be reached where cavitation occurs. Cavitation is the formation of a gas bubble in the liquid during the rarefaction cycle. The sound pressure required to produce cavitation in water has been considered in Sec. 15.2. When the compression cycle occurs the gas bubble collapses. During the collapse tremendous pressures are produced. The pressure may be of the order of several thousand atmospheres. Thousands of these small bubbles are formed in a small volume of the liquid. It is quite generally agreed that it is cavitation that produces most of the biological, detergent, mechanical, and chemical effects in the application of high intensity sound to various mediums.

16.4. Dispersion Due to Ultrasonics.[26,27,28,29,30]—Dispersion in chemistry means the breaking down of a liquid or solid particle into smaller sizes or finer texture and distributing them in another medium.

In Chemistry the term system is applied to the whole mixture. Each of the substances comprising the system is called a component. A mixture of two substances is termed a two-component system. The form in which the component exists is called a phase, as, for example, gas, liquid, or solid. A colloidal solution is a two-component system in which a finely divided substance is uniformly distributed through the other. These systems may be classified according to the fineness of dispersion, as, for example, mechanical suspensions, colloidal solutions, and molecular solutions.

A list of two-component systems is given below:

Solid	Solid
Solid	Liquid
Solid	Gas
Liquid	Solid
Liquid	Liquid
Liquid	Gas
Gas	Solid
Gas	Liquid

[21] Williard, G. W., *Jour. Acous. Soc. Amer.*, Vol. 25, No. 4, p. 669, 1953.
[22] Galloway, William J., *Jour. Acous. Soc. Amer.*, Vol. 26, No. 5, p. 849, 1954.
[23] Mellen, Robert H., *Jour. Acous. Soc. Amer.*, Vol. 26, No. 3, p. 356, 1954.
[24] Connolly and Fox, *Jour. Acous. Soc. Amer.*, Vol. 26, No. 5, p. 843, 1954.
[25] Also references 4, 5, 8, 9, 10, 11, 12, and 14.
[26] Solner, K., *Jour. Phys. Chem.*, Vol. 42, p. 1071, Nov. 1938.
[27] Freundlick and Gillings, *Faraday Soc. Trans.*, Vol. 35, p. 319, Feb. 1939.
[28] Bergmann, "Ultrasonics," John Wiley and Sons, New York, N.Y., 1938.
[29] Herzfeld, K. F., *J. Chem. Phys.*, Vol. 9, p. 513, July, 1941.
[30] Crawford, "Ultrasonic Engineering," Academic Press, New York, N.Y., 1955.

All of the above may be obtained in colloidal dimensions. However, liquid + solid and liquid + liquid have received the most attention. Intense sound fields may be used to bring about mixtures in the above systems.

16.5. Emulsification Due to Ultrasonics.[31,32,33,34,35,36,37,38,39]—An emulsion is a suspension of fine particles or globules of a liquid in a liquid. Emulsions are generally produced by violent agitation. This suggests that ultrasonics may be used to produce emulsions.

If two immiscible liquids, such as water and gasoline, are placed in a container and subjected to intense sound vibrations it has been found that an emulsion will be formed.

The action of ultrasonics in producing emulsification can also be applied to the production of alloys of iron and lead, aluminum and lead, aluminum and cadmium, etc., which are not miscible in the liquid state. It is possible to keep the metals mixed by the application of supersonics up to the point of solidification. New bearing materials have been made in this way.

Ultrasonics have also been applied to photographic emulsions with an improvement in homogeneity and stability.

Ultrasonics have been applied to molten zinc, tin, and aluminum. It was found that solidification occurred more quickly. In addition, the structure in the solidified state was found to be finer grained.

The homogenization of milk, that is, the reduction in size of the fat particles so that cream does not form while the milk stands, can be carried out by means of the application of ultrasonics.

16.6 Coagulation Due to Ultrasonics.[40,41,42,43,44,45]—In spite of the fact that ultrasonics have strong dispersive effects on liquid emulsions their effect on gas and solids and gas and liquids is the opposite—namely, coagulation. The solid and liquid particles in mist, dust, and smoke agglomerate when these mixtures are subjected to intense sound waves. The particles in a small smoke attack have been coagulated and precipitated. The action depends in some degree on the wavelength and intensity.

Degassing of molten metals by the application of ultrasonics is another example of coagulation. Small bubbles form at first which join to form

[31] Wood and Loomis, *Phys. Rev.*, Vol. 29, p. 373, 1927.
[32] Chambers, L. A., *Jour. of Dairy Sci.*, Vol. 19, p. 29, 1936.
[33] Bondy and Sollner, *Faraday Soc. Trans.*, Vol. 32, p. 556, March 1946.
[34] Sokoloff, S., *Acta Physicochimico*, Vol. 3, p. 939, 1936.
[35] Schmid and Ehret, *Zeit. f. Electrochem*, Vol. 43, p. 869, Nov. 1937.
[36] Bergmann, "Ultrasonics," John Wiley and Sons, New York, N.Y., 1938.
[37] Wood, "Supersonics," Brown Univ. Press, Providence, R.I., 1939.
[38] Hiedmann, Egon A., *Jour. Acous. Soc. Amer.*, Vol. 26, No. 5, p. 831, 1954.
[39] Hueter and Bolt, "Sonics," John Wiley and Sons, New York, N.Y., 1955.
[40] Sollner and Bondy, *Faraday Soc. Trans.*, Vol. 32, p. 616, April 1936.
[41] Number of Authors, *Faraday Soc. Trans.*, Vol. 32, p. 104, August 1936.
[42] Behr, A., *Metal Ind. Lon.*, p. 422, Dec. 31, 1943.
[43] Sorenson, Ch., *Ann Phys. Lpz.*, Vol. 27, p. 70, 1936.
[44] Brandt and Freund, *Z. Phys.*, Vol. 94, p. 348, 1936.
[45] Hiedmann, Egon A., *Jour. Acous. Soc. Amer.*, Vol. 26, No. 5, p. 831, 1954.

larger ones. The larger ones rise to the surface and are expelled. This use of ultrasonics should lead to an improvement in castings where the presence of bubbles is very objectionable.

16.7. Chemical Effects of Ultrasonics.[46,47,48,49,50,51,52,53]—A large number of experiments have been conducted on the effect of intense sound waves upon chemical reactions. Certain types of chemical reactions have been speeded by the application of intense sound waves. However, in some cases it is difficult to isolate the thermal effects due to the sound and the effects due to the sound alone. Another chemical effect is the breaking down of molecules. For example, a chain molecule of starch has been broken into six fragments. The application of intense sound waves to speed up the aging of whiskey has been suggested. The explanation is that in the aging process there is a gradual change in the structure of complex molecules which could be accomplished in a relatively short time with the application of sound.

16.8. Biological Effects of Ultrasonics.[54,55,56,57,58,59,60,61,62,63,64]— Ultrasonics have a very destructive effect upon small living organisms. Small fish have been killed by high-power echo ranging and sounding devices.

Ultrasonics have been used in the extraction of antibodies secreted in the cells of pathogenic bacteria. These antibodies are used in serums for immunization against typhoid and other diseases. The bacterial cell walls are broken down by the application of supersonic waves and the antibodies are set free. The cell walls of the bacteria are separated from the antibodies by centrifuging.

It appears that bacteria can be destroyed by ultrasonics. The bacteria in milk have been reduced by the application of supersonics. This indicates that milk can be sterilized by ultrasonics.

Another application in medicine is the use of sound to produce stimulation within the body. Therapeutic effects of a different nature but similar to those produced by heat and radio-frequency diathermy may be obtained.

[46] Porter and Young, *Jour. Amer. Chem. Soc.*, Vol. 60, p. 1497, 1938.
[47] Benthe H., *Zeit. f. Phys. Chem.*, Vol. 163, p. 161, Feb. 1933.
[48] Schmitt, Johnson, and Olson, *Jour. Amer. Chem. Soc.*, Vol. 51, p. 370, 1929.
[49] Barrett and Porter, *Jour. Amer. Chem. Soc.*, Vol. 63, p. 3434, Dec. 1941.
[50] Weissler, Alfred, *Jour. Acous. Soc. Amer.*, Vol. 25, No. 4, p. 651, 1953.
[51] Yeager and Hovorko, *Jour. Acous. Soc. Amer.*, Vol. 25, No. 3, p. 443, 1953.
[52] Hueter and Bolt, "Sonics," John Wiley and Sons, New York, N.Y., 1955.
[53] Crawford, "Ultrasonic Engineering," Academic Press, New York, N.Y., 1955.
[54] Wood and Loomis, *Phys. Rev.*, Vol. 29, p. 373, 1927.
[55] Harvey, E. N., *Amer. Jour. Physiology*, Vol. 91, p. 284, 1929.
[56] Harvey and Loomis, *Jour. of Bacteriology*, Vol. 17, p. 373, 1929.
[57] Harvey, E. N., *Biological Bulletin*, Vol. 59, p. 306, 1930.
[58] Flosdoff and Chambers, *Jour. of Immunology*, Vol. 28, p. 297, 1935.
[59] Bergmann, "Ultrasonics," John Wiley and Sons, New York, N.Y., 1938.
[60] Wood, "Supersonics," Brown Univ. Press, Providence, R.I., 1939.
[61] Gregs, E. C., O. Glasser's "Medical Physics," p. 1591.
[62] Lynn, Zwemer, and Chick, *Science*, Vol. 96, p. 119, 1942.
[63] *Electronics*, Vol. 16, No. 5, p. 154, 1943.
[64] Danner, Ackerman, and Fring, *Jour. Acous. Soc. Amer.*, Vol. 26, No. 5, p. 731, 1954.

As in the case of chemical effects the biological effects are somewhat obscure but very interesting.

16.9. Medical Applications of Ultrasonics.[65]—The applications of ultrasonics in the medical field have involved analysis and treatment. The developments in the medical field appear to be very promising.

The effect of ultrasonics on tissues[66] has been investigated. The heating and mechanical effects have been isolated. The conclusion is that there is an effect outside of the heating effect.

The effects of the changes produced by high intensity sound upon the central nervous system[67,68] has been investigated. The results show that nerve cells are particularly sensitive to ultrasonics, while blood vessels and nerve fibers are much more resistant.

A study of the therapeutic[69] effect of ultrasonics shows that the heat which is produced plays the major role. However, ultrasonics also produces a mechanical effect. Ultrasonics has been used to produce deep-seated heating in the treatment of arthritis.

The cerebral ventricular geometry[70,71] has been portrayed by means of ultrasonic techniques. The head is immersed in water. An underwater projector sends an ultrasonic wave through the head. A hydrophone picks up the transmitted sound. A frequency of 2.5 megacycles was used. A scanning system together with a facsimile-type recorder presents the ultrasonogram in the form of a picture showing the cerebral ventricular geometry. This method provides a means for the detection of brain tumors similar to that of the X-ray.

Recent work on tumor detection employs ultrasonic waves and echo-ranging techniques with cathode-ray presentation. The pulses are sent into the body and the echos return in different intensities depending upon the difference in acoustical impedance of the malignant and nonmalignant tissues and in different times depending upon the depths of the reflecting boundaries.

A small version of the ultrasonic drill has been developed for use by dentists in drilling teeth. The advantages of the ultrasonic drill is reduction in pain and improved definition of the drilled area.

16.10. Thermal Effects of Ultrasonics.—There is a considerable temperature rise in the supersonic field in a liquid. A rise of several degrees per minute can be obtained. The generation in heat is due to dissipation of the sound by absorption in the liquid. The generation of heat by the action of supersonics obscures the effects which can be attributed to sound alone because many chemical and biological phenomena observed when

[65] Herrick and Krusen, *Jour. Acous. Soc. Amer.*, Vol. 26, No. 2, p. 236, 1954.

[66] Fry, Wolff, Tucker, and Fry, *Jour. Acous. Soc. Amer.*, Vol. 22, No. 6, p. 867, 1950.

[67] Wall, Tucker, Fry, and Mosberg, *Jour. Acous. Soc. Amer.*, Vol. 25, No. 2, p. 281, 1953.

[68] Fry, William J., *Jour. Acous. Soc. Amer.*, Vol. 25, No. 1, p. 1, 1953.

[69] Lehman, Justus F., *Jour. Acous. Soc. Amer.*, Vol. 25, No. 1, p. 17, 1953.

[70] Hueter and Bolt, *Jour. Acous. Soc. Amer.*, Vol. 23, No. 2, p. 160, 1951.

[71] Hueter and Bolt, "Sonics," John Wiley and Sons, New York, N.Y., 1955.

supersonics are applied are also obtained by the application of heat. The practical value of heating by supersonics remains to be seen.

16.11. Ultrasonics as a Detergent.—Ultrasonics may be used to clean and wash various substances. Tests have been made of a supersonic washing machine in which clothes mixed with the conventional water and soap solutions are subjected to high intensity sound waves. It has been found that clothes can be cleaned as effectively in this way as by conventional means.

16.12. Ultrasonic Cleaning and Degreasing.[72,73,74,75]—The use of ultrasonics for cleaning and degreasing surfaces has found widespread use in industry. Cavitation reduces the surface tension of the clinging dirt and thereby produces a cleaning action on all surfaces and recesses. Cavitation mulsifies greases and oils and thereby assists in the removal of such coatings.

16.13. Ultrasonic Drilling.[76,77,78]—The drilling of glass, ceramics, and metals is now being done by means of ultrasonics. An ultrasonic generator for use in drilling is shown in Fig. 16.6. The tip sets up cavitation
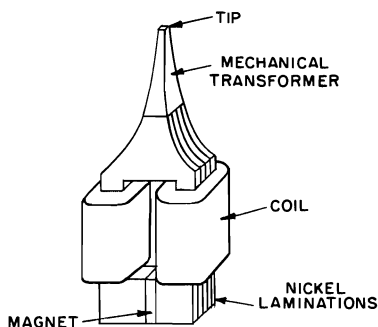
PERSPECTIVE VIEW

FIG. 16.6. Perspective view of an ultrasonic drill.

in a surrounding liquid-borne abrasive slurry. The forces produced by the cavitation bubbles propel the abrasive slurry against the material being drilled. The result is that glass, ceramics, and metals are penetrated in the matter of a few seconds. The point may be any shape as contrasted to circular drills. The ultrasonic drill shown in Fig. 16.6 employs a magneto-striction transducer. The system operates as a half-wave resonator. The

[72] Mattiat, Oskar, *Jour. Acous. Soc. Amer.*, Vol. 26, No. 2, p. 241, 1954.
[73] Kearney, Thomas, *Jour. Acous. Soc. Amer.*, Vol. 26, No. 2, p. 244, 1954.
[74] Hueter and Bolt, "Sonics," John Wiley and Sons, New York, N.Y., 1955.
[75] Crawford, "Ultrasonic Engineering," Academic Press, New York, N.Y., 1955.
[76] Mattiat, Oskar, *Jour. Acous. Soc. Amer.*, Vol. 26, No. 2, p. 241, 1954.
[77] Hueter and Bolt, "Sonics," John Wiley and Sons, New York, N.Y., 1955.
[78] Crawford, "Ultrasonic Engineering," Academic Press, New York, N.Y., 1955.

dimensions of the resonator are usually such that resonance occurs at 20 kilocycles. The amplitude at the drilling tip is increased by the use of a mechanical transformer in the form of a tapered rod.

16.14. Ultrasonic Soldering.[79,80,81]—Aluminum is very difficult to solder because of the oxide which is formed on the surface. When cavitation is induced in the molten solder applied to the surface, the resultant forces break down the metal oxides formed on the surface of the parts being soldered. In this way, the solder is exposed to the pure base metal in a nonoxidising atmosphere. The ultrasonic energy may be applied to a pot of molten solder or to the tip of a soldering iron. Aluminum may be soldered by dipping the parts in the molten solder in the pot or by the application of the soldering iron. The soldering of aluminum may be carried out without the use of flux.

16.15. Testing of Materials by Means of Ultrasonics.[82,83,84,85,86,87,88] —A number of systems have been devised for testing materials, particularly metals, for flaws such as hollows, cracks, or other defects of homogeneity.

One of the methods employs the distortion of sand patterns on a steel plate when it is caused to vibrate under the influence of sound. This system can only be applied to plates in which the sand pattern is known for a perfect plate.

Another system which is particularly useful in that it can be used to detect flaws in a piece of metal of almost any shape is analogous to the echo ranging or depth sounding devices described in Secs. 15.15 and 15.16. A quartz crystal projector hydrophone (see Secs. 15.9 and 15.10) is placed in intimate contact with the metal object to be tested by using a film of oil between crystal and metal. A short pulse of very high-frequency sound (5 megacycles) is sent out by the crystal used as a projector. The reflected pulse is picked up by the crystal used as a hydrophone. The output of the hydrophone is amplified and applied to the screen of a cathode-ray tube. Since all these operations take place in fractions of milliseconds, the electronic switching, etc., is quite intricate and complex. The cathode ray depicts the outgoing pulse and all reflected pulses. From the dimensions and geometry of the piece under test and the velocity of sound in the material and pattern on the oscilloscope it is possible to determine the presence or absence of flaws. This is a very useful and powerful tool. It possesses advantages over X-ray testing in that the particular piece to be tested need not be moved to the apparatus to be tested since the test

[79] Mattiat, Oskar, *Jour. Acous. Soc. Amer.*, Vol. 26, No. 2, p. 241, 1954.
[80] Hueter and Bolt, "Sonics," John Wiley and Sons, New York, N.Y., 1955.
[81] Crawford, "Ultrasonic Engineering," Academic Press, New York, N.Y., 1955.
[82] Bergmann, "Ultrasonics," John Wiley and Sons, New York, N.Y., 1938.
[83] Andrews, A., *Electronics*, Vol. 17, No. 5, p. 122, 1944.
[84] Hayes, H. C., *Jour. Acous. Soc. Amer.*, Vol. 8, No. 4, p. 220, 1937.
[85] Firestone, F. A., *Jour. Acous. Soc. Amer.*, Vol. 17, No. 4, p. 363, 1946.
[86] *Jour. Appl. Phys.*, Vol. 15, No. 3, p. XIV, March, 1944.
[87] Hueter and Bolt, "Sonics," John Wiley and Sons, New York, N.Y., 1955.
[88] Crawford, "Ultrasonic Engineering," Academic Press, New York, N.Y., 1955.

equipment is quite small and portable. Furthermore, other intervening or adjacent components need not be removed to carry out the tests.

A system for detecting flaws in tires by the use of ultrasonics has been developed. The tire is immersed in water and the transmission of an ultrasonic wave through the tire is obtained by a projector and hydrophone combination. Since the characteristic acoustical impedance of rubber and water is practically the same, there will be very little attenuation or other anomalies in the transmission of the ultrasonic wave except in the case of a flaw or defect in the rubber.

16.16. Ultrasonic Delay Lines and Filters.—Delay lines[89],[90] for the storage of pulses one microsecond in length and for periods up to 2000 microseconds have been developed. These delay lines are used for the storage of radar pulses from one pulse to the next. Both mercury and solid lines have been used. Quartz crystal transducers are used for the transmitter and the receiver.

Ultrasonic band-pass filters[91] for use in the intermediate frequency amplifiers in radio receivers consist of mass and compliance elements. Magnetostriction transducers are used for the transmitter and the receiver. The outstanding characteristic is the very high attenuation over a very narrow frequency range at the upper and lower cutoff frequencies. For example, a band-pass filter with a pass band of 6 kilocycles at 100 kilocycles shows 45 decibels attenuation in 1 kilocycle at the cutoff frequency.

[89] Arenberg, David L., *Jour. Acous. Soc. Amer.*, Vol. 20, No. 1, 1948.
[90] Hueter and Bolt, "Sonics," John Wiley and Sons, New York, N.Y., 1955.
[91] Burns, Jr., Leslie L., *RCA Review*, Vol. 13, No. 1, p. 34, 1952.

INDEX