ELEMENTS OF

Acoustical Engineering

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By

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PREFACE TO THE SECOND EDITION

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 have been made in acoustical engineering since the issuance of the first edition of this book. Developments and improvements in radio receivers, phonographs, records, sound motion picture, intercommunicating, and sound systems, sound pickup technics, hearing aids, acoustical treatment and a multitude of other devices and systems have kept pace with the ever increasing public interest. In these applications acoustics plays a major role in public acceptance. World War II stimulated activity in both subaqueous and air acoustics and, as a result, many new principles and systems were evolved. In view of the progress in acoustics since the first edition, it appeared advisable to prepare a new edition of this book.

In preparing new material and in revising existing material in the second edition, the same principles were followed as in the first edition. 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 Underwater Sound and Supersonics and Ultrasonics have been added. The new edition contains 539 pages as compared to 344 pages in the first edition. The first edition contained 197 illustrations. The new edition contains 342 illustrations of which 50 are revised and 145 are new. As in the first edition, most of the illustrations contain several parts so that a complete theme is depicted in a single illustration.

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June 1947 HARRY F. OLSON

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ELEMENTS OF ACOUSTICAL ENGINEERING

CHAPTER I

SOUND WAVES

1.1. Introduction. — A knowledge of the elements of acoustics is becoming increasingly important to any profession depending in any manner upon acoustics. Modern civilization is becoming more critical of sound reproduction. The radio receiver, phonograph, sound motion picture or sound reinforcing system of a few years ago is not acceptable today. Auditoriums and studios must exhibit proper acoustical qualities. Reduction of noise in all types of machinery and appliances is demanded by the consumers. Acoustics, one of the oldest divisions of physics, appeared to be a decadent science a few years ago. Today it is an important and necessary branch of Applied Science and its application to every phase of modern civilization is in its infancy.

The widespread interest in the phonograph, radio broadcasting, television, sound motion pictures, sound reinforcing, architectural acoustics and noise problems has stimulated research and developments in these fields. Acoustics were involved in almost every type of communication system used in World War II. Accelerated by the war, tremendous advances were made in underwater sound. The industrial applications of ultrasonics and supersonics is beginning to unfold a new field in the application of sound.

During the early stages progress in the development of acoustical devices was made by the trial-and-error method. Later, by the extension and application of scientific knowledge, results have been obtained that could not have been accomplished by other means. A major portion of the problems in acoustics is concerned with vibrating systems. Comparisons between these problems and those of electricity considered from a dynamical viewpoint have led to impedance methods in acoustics. By a judicious application of dynamical theory and experimental research, the science of acoustics has developed into a wide field of interesting phenomena with countless useful applications.

In this book, the author has attempted to outline the essentials of acoustics from the standpoint of the engineer or applied scientist. 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 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.

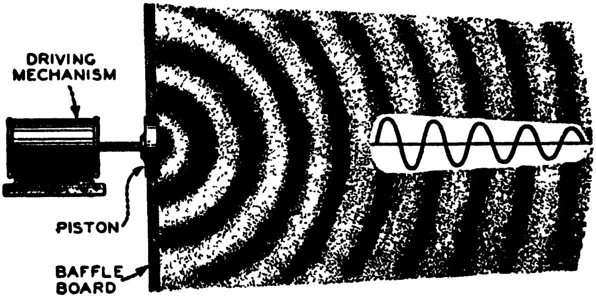

FIG. 1.1. Production of sound waves by a vibrating piston.

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 froms 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 $\frac{\partial \rho'}{\partial t}\Delta x\,\Delta y\,\Delta z$ 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 $\rho'\ \Delta x\ \Delta y\ \Delta z\ \frac{\partial u}{\partial t}$. The mean pressures on the faces perpendicular to x are

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

where p_0' = pressure in the medium.

The difference is a force $\frac{\partial p_0}{\partial x}\ \Delta x\ \Delta y\ \Delta z$ 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

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

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

$$\frac{\partial}{\partial t} d\phi = -\frac{1}{\rho'} dp_0' \qquad 1.12$$

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

$$\frac{\partial \phi}{\partial t} = -\frac{p}{\rho} \qquad 1.13$$

where p = excess pressure.

From equations 1.9, 1.10, 1.11 and 1.13

$$\frac{\partial^2 \phi}{\partial t^2} - \frac{\gamma p_0}{\rho}\left(\frac{\partial^2 \phi}{\partial x^2} + \frac{\partial^2 \phi}{\partial y^2} + \frac{\partial^2 \phi}{\partial z^2}\right) = 0 \qquad 1.14$$

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

$$\frac{\gamma p_0}{\rho} = c^2 \tag{1.15}$$

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

TABLE 1.1. YOUNG'S MODULUS $\mathcal{Q}$, 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	$\mathcal{Q}$	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	12 7 × 10^{11}	.33	1 8	8400	150 × 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	5 2 × 10^{11}	.33	22 4	1500	340 × 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
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	$\mathcal{Q}$	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	$\mathcal{Q}$	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, ll 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)

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, ¼ to ½ of the above values for c.

Plastics

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

PLASTICS (*continued*)

Substance	$\mathcal{Q}$	σ	ρ	c	ρc
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	5×10^8	.	.95	70	67×10^4
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

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	1504	15.5×10^4

GASES

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,

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

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

$$s = -\frac{1}{c^2}\frac{\partial \phi}{\partial t} = -\frac{jkA}{cr} \epsilon^{jk(ct-r)} \qquad 1.37$$

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

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

where $\tan\theta = 1/kr$.

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

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

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

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

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

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

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

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

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

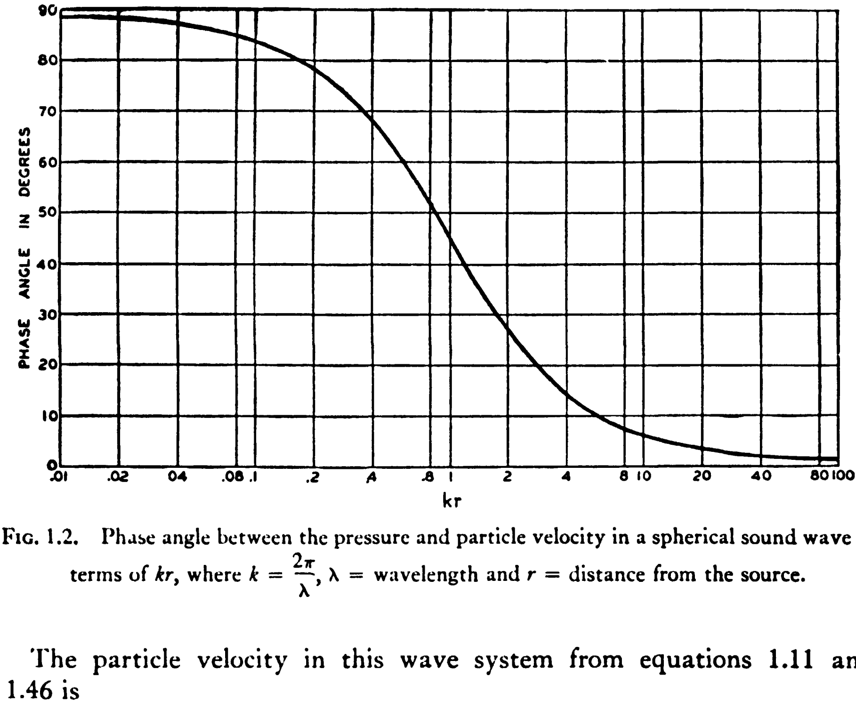

Fig. 1.2. Phase angle between the pressure and particle velocity in a spherical sound wave in terms of kr, where $k = \frac{2\pi}{\lambda}$, λ = wavelength and r = distance from the source.

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

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

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

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

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.

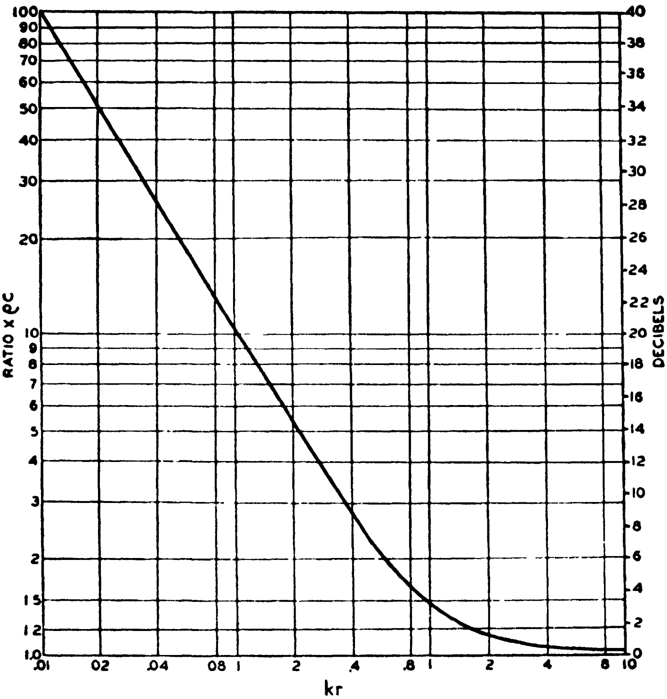

FIG. 1.3. Ratio of the absolute magnitude of the particle velocity to the pressure in a spherical sound wave in terms of kr, where $k = \frac{2\pi}{\lambda}$, λ = wavelength and r = distance from the source.

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

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

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

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

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

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

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.

[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.

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,
v_0 = velocity of the observer,
v_s = velocity of the source, and
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.

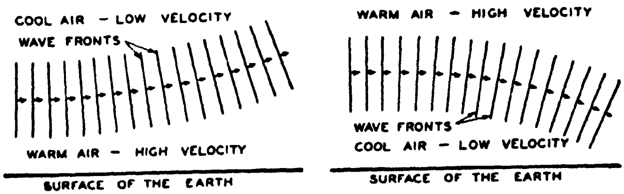

FIG. 1.4. The refraction of a sound wave in air.

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.

[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, New York, N. Y., 1930.

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.

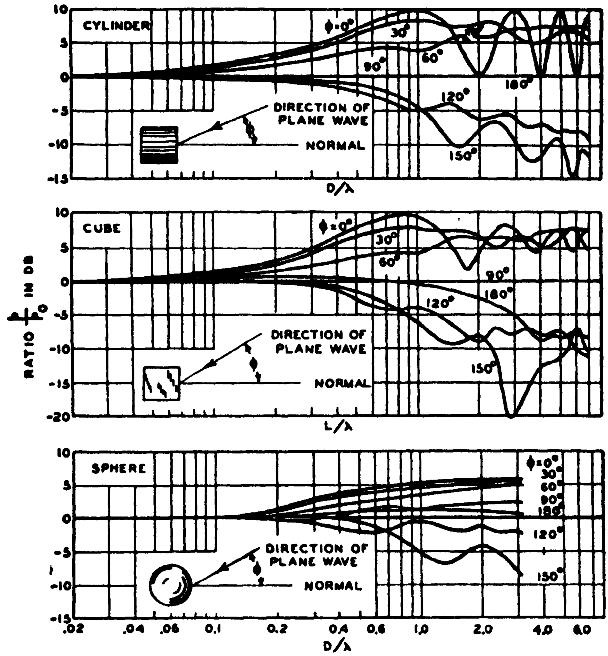

FIG. 1.5. The diffraction of a sound wave by a cylinder, cube and sphere. (After Muller, Black and Dunn.)

The diffraction of sound by microphones and loud speakers is important in the performance of these instruments. The diffraction [3] of sound by a sphere, a cube and a cylinder as a function of the dimensions is shown in Fig. 1.5. 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.

Another example of diffraction of sound is illustrated by the zone plate shown in Fig. 1.6. The path lengths of the sound from the source to the focus vary by an integral wavelength. As a consequence, all the pencils

[3] Muller, Black and Dunn, *Jour. Acous. Soc. Amer.*, Vol. 10, No. 1, p. 6, 1938.

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.

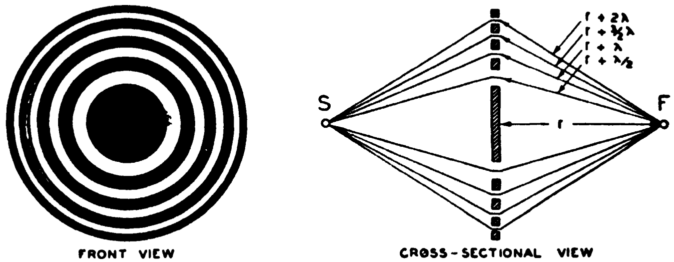

Fig. 1.6. Zone plate. The source *S* and the focus *F* are equidistant from the zone plate.

1.12. Acoustical Reciprocity Theorem.[4,5,6,7] — 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', p'' and particle velocities v' and v''. Multiply equation 1.4 by the p and v of the other set.

$$v''\frac{dv'}{dt} - v'\frac{dv''}{dt} + \frac{1}{\rho}v''\,\mathrm{grad}\,p_0' - \frac{1}{\rho}v'\,\mathrm{grad}\,p_0'' = 0 \qquad 1.60$$

If p and v vary as a harmonic of the time, equation 1.60 becomes

$$\frac{1}{\rho}v''\,\mathrm{grad}\,p_0' - \frac{1}{\rho}v'\,\mathrm{grad}\,p_0'' = 0 \qquad 1.61$$

There is the following relation:

$$v\,\mathrm{grad}\,p = \mathrm{div}\,vp - p\,\mathrm{div}\,v \qquad 1.62$$

From equations 1.9 and 1.10

$$\frac{1}{\gamma p_0}\frac{\partial p}{\partial t} + \mathrm{div}\,v = 0 \qquad 1.63$$

[4] Rayleigh, "Theory of Sound," Macmillan and Company, London, 1926.
[5] Ballentine, S., *Proc.*, I.R.E., Vol. 17, No. 6, p. 929, 1929.
[6] Olson, H. F., *RAC Review*, Vol. 6, No. 1, p. 36, 1941.
[7] Olson, "Dynamical Analogies," D. Van Nostrand Company, New York, N. Y., 1943.

From equations 1.61, 1.62 and 1.63,

$$\operatorname{div}(v''p' - v'p'') = 0 \qquad 1.64$$

The relation of equation 1.64 is for a point. Integration of equation 1.64 over a region of space gives

$$\int\int (v''p' - v'p'')ds = 0 \qquad 1.65$$

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

$$\int\int (v''p' - v'p'')_n ds = 0 \qquad 1.66$$

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. 1.7, equation 1.66 becomes

$$p'v'' = p''v' \qquad 1.67$$

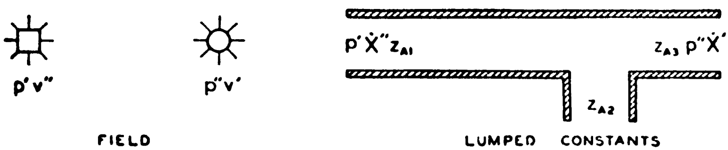

Fig. 1.7. Reciprocity in field and lumped constant acoustical systems.

where p', p'' and v', v'' are the pressures and particle velocities depicted in the free field acoustical system of Fig. 1.7.

The above theorem is applicable to all acoustical problems. However, it can be restricted to lumped constants [8] 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

$$\sum_{j=1}^{n} p_j'\dot{X}_j'' = \sum_{j=1}^{n} p_j''\dot{X}_j' \qquad 1.68$$

[8] Olson, "Dynamical Analogies," D. Van Nostrand Company, New York, N. Y., 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 1.68 becomes

$$p'\dot{X}'' = p''\dot{X}' \qquad 1.69$$

where p', p'' and $\dot{X}'$, $\dot{X}''$ are the pressures and volume currents depicted in the acoustical system of lumped constants in Fig. 1.7.

1.13. Acoustical Principle of Similarity.[9] — 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 Fig. 1.5. Other examples, are the directional characteristics of various sound sources Figs. 2.3 to 2.18 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

$$\delta x + \delta\xi = \delta x + \frac{\partial \xi}{\partial x}\delta x \qquad 1.70$$

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

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

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

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

$$\mathcal{Q} = \frac{\frac{F}{S}}{\frac{\partial \xi}{\partial x}} \qquad 1.71$$

where $\mathcal{Q}$ = 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

$$F = \mathcal{Q}S\,\frac{\partial \xi}{\partial x} \qquad 1.72$$

The force acting across the second face of the element is

$$F + \delta F = \mathcal{Q}S\,\frac{\partial \xi}{\partial x} + \frac{\partial}{\partial x}\left(\mathcal{Q}S\,\frac{\partial \xi}{\partial x}\right)\delta x \qquad 1.73$$

$$= \mathcal{Q}S\,\frac{\partial \xi}{\partial x} + \mathcal{Q}S\,\frac{\partial^2 \xi}{\partial x^2}\,\delta x \qquad 1.74$$

The resultant force on the element is

$$\delta F = \mathcal{Q}S\,\frac{\partial^2 \xi}{\partial x^2}\,\delta x \qquad 1.75$$

The acceleration of momentum of the element is

$$S\rho\delta x\,\frac{\partial^2 \xi}{\partial t^2} \qquad 1.76$$

where ρ = density, in grams per cubic centimeter.

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

$$\frac{\partial^2 \xi}{\partial t^2} = \frac{\mathfrak{Q}}{\rho} \frac{\partial^2 \xi}{\partial x^2} \tag{1.77}$$

This is the wave equation for ξ. Equation 1.77 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 in a rod is

$$c = \sqrt{\frac{\mathfrak{Q}}{\rho}} \tag{1.78}$$

where $\mathfrak{Q}$ = 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

$$c = \sqrt{\frac{\mathfrak{Q}}{2\rho(\sigma + 1)}} \tag{1.79}$$

where $\mathfrak{Q}$ = 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).

CHAPTER II

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 loud speakers. 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
$\dot{\xi}_0$ = 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

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

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

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

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

$$P_T = 2\pi\rho c k^2 A^2 \tag{2.6}$$

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 loud speaker, which acts as a simple source, suspended in space at a large distance from any walls or boundaries. *B* would correspond to a loud speaker placed on the floor in the center of

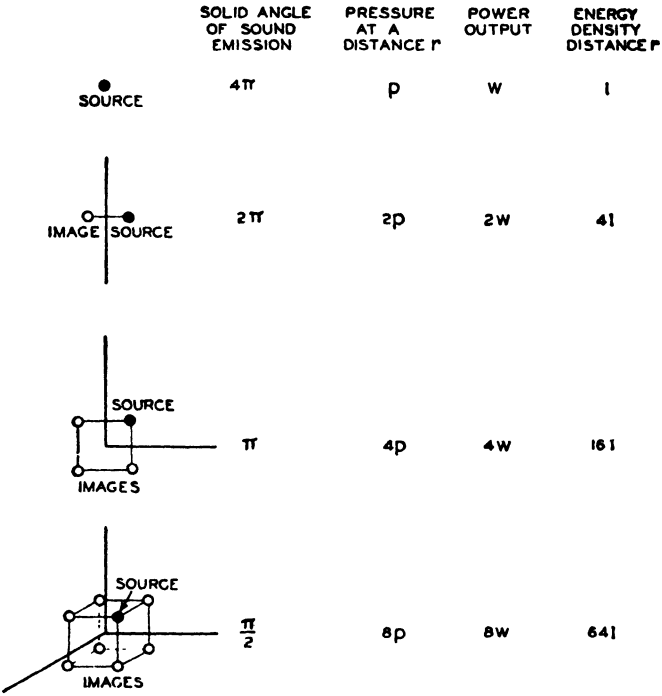

FIG. 2.1. The pressure, total power output and energy density delivered by a point source operating in solid angles of 4π, 2π, π and $\pi/2$ steradians.

the room. *C* would correspond to a loud speaker placed on the floor along a wall, and *D* would correspond to a loud speaker 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.

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 sepa-

[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, New York, N. Y., 1926.

rated 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

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

The pressure from equation 2.7 is

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

Retaining the real parts of equation 2.8

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

At a very large distance

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

At a very small distance

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

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,

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

Retaining the real parts of equation 2.12

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

At a very large distance

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

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

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

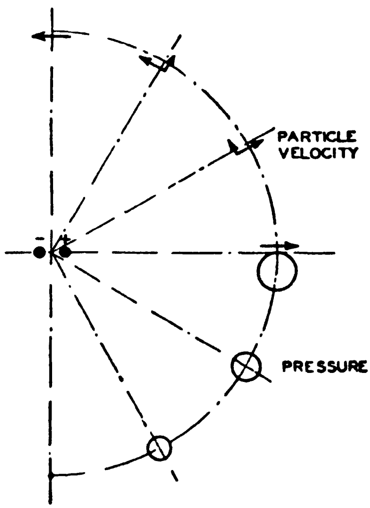

Fig. 2.2. The pressure and particle velocity at a constant distance from a doublet source. The magnitude of the pressure is indicated by the circle. The particle velocity has two components: a radial and a transverse component. The direction and magnitude of these two components are indicated by vectors.

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 loud speaker mounted in a small baffle. (Dimensions of the baffle are small compared to the wavelength.) If the response of such a loud speaker 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 loud speaker, 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 loud speaker, the result will be a sine directional characteristic.

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

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

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.

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 loud speaker falls off rapidly with frequency when the dimensions of the baffle are small compared to the wavelength (see Sec. 6.8).

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

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

[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.

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.

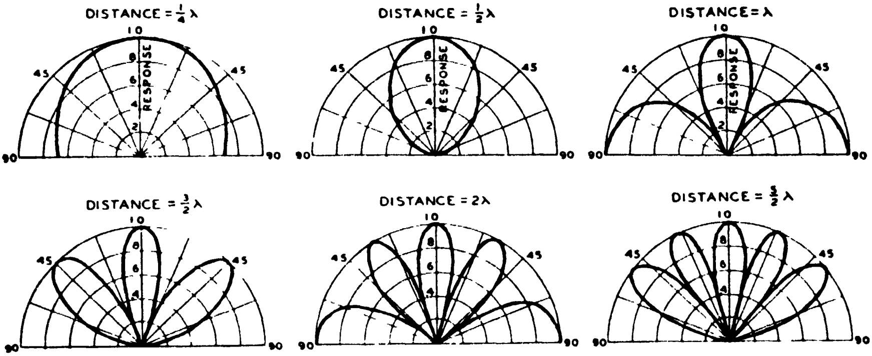

FIG. 2.3. Directional characteristics of two separated equal small sources vibrating in phase as a function of the distance between the sources and the wavelength. The polar graph depicts the pressure, at a fixed distance, as a function of the angle. The pressure for the angle 0° is arbitrarily chosen 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.

2.5. 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

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

The directional characteristics of a continuous line source are shown in

Fig. 2.4. The directional characteristics are symmetrical about the line 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.

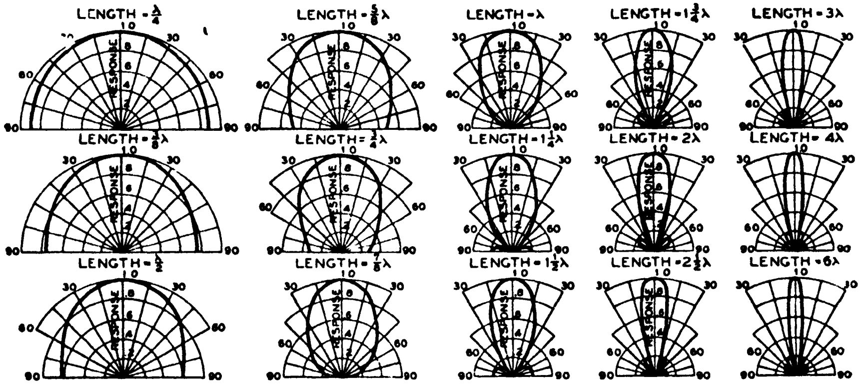

FIG. 2.4. Directional characteristics of a line source as a function of the length and the wavelength. The polar graph depicts the pressure, at a large fixed distance, as a function of the angle. The pressure for the angle 0° is arbitrarily chosen as unity. The direction corresponding to the angle 0°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

2.6. 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

$$R_\alpha = \frac{\sin^2\left(\dfrac{\pi l}{2\lambda}\sin\alpha\right)}{\left(\dfrac{\pi l}{2\lambda}\sin\alpha\right)^2} \qquad 2.25$$

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. 2.5. Comparing the directional characteristics of Fig. 2.5 with those

[8] Menges, Karl, *Akus. Zeit.*, Vol. 6, No. 2, p. 90, 1941.

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.

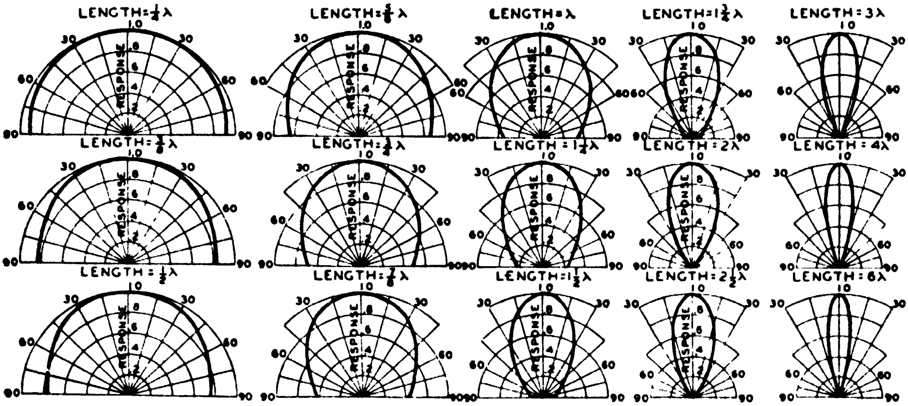

FIG. 2.5. Directional characteristics of a tapered line source as a function of the length and the wavelength. The volume current output along the line vanes linearly from a maximum at the center to zero at the two ends. The polar graph depicts the pressure, at a fixed distance, as a function of the angle. The 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.

2.7. 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

$$R_\alpha = \frac{\int_{-\frac{d}{2}}^{+\frac{d}{2}} f(x)\, \epsilon^{-j\left(\frac{2\pi x}{\lambda}\right)\sin\alpha}\, dx}{\int_{-\frac{d}{2}}^{+\frac{d}{2}} f(x)\, dx} \qquad 2.26$$

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 2.25.

2.8. 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,

$$R_\alpha = \frac{1}{2m+1}\left| \sum_{k=-m}^{k=m} \cos\left[\frac{2\pi R}{\lambda}\cos(\alpha + k\theta)\right] + j\sum_{k=-m}^{k=m} \sin\left[\frac{2\pi R}{\lambda}\cos(\alpha + k\theta)\right]\right| \qquad 2.27$$

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 [9] 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 of the chords is the same. In this case the result takes the form,

$$R_\alpha = \frac{1}{2m+1}\left| \sum_{k=-m}^{k=m} \cos\left\{\frac{2\pi R}{\lambda}\cos(\alpha + k\theta)\right\}\frac{\sin\left[\frac{\pi d}{\lambda}\sin(\alpha + k\theta)\right]}{\frac{\pi d}{\lambda}\sin(\alpha + k\theta)} + j\sum_{k=-m}^{k=m} \sin\left\{\frac{2\pi R}{\lambda}\cos(\alpha + k\theta)\right\}\frac{\sin\left[\frac{\pi d}{\lambda}\sin(\alpha + k\theta)\right]}{\frac{\pi d}{\lambda}\sin(\alpha + k\theta)}\right| \qquad 2.28$$

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.

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

The directional characteristics for an arc of 60°, 90° and 120° are shown in Figs. 2.6, 2.7 and 2.8. The interesting feature of the directional char-

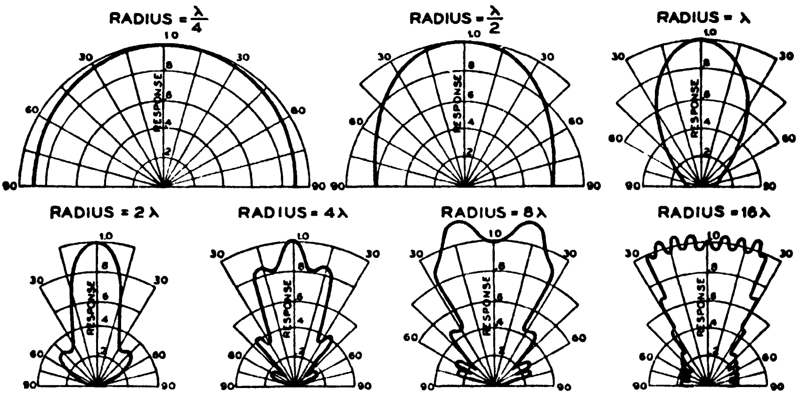

FIG. 2.6. Directional characteristics of a 60° arc as a function of the radius and the wavelength. The polar graph depicts the pressure, at a large fixed distance, as a function of the angle in the plane of the arc. The pressure for the angle 0° is arbitrarily chosen as unity.

acteristics of an arc is that the directional characteristics are very broad for wavelengths large compared to the dimensions, and are narrow for

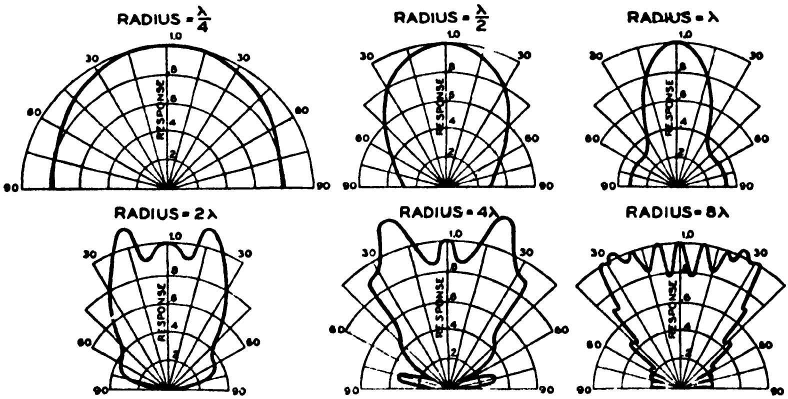

FIG. 2.7. Directional characteristics of a 90° arc as a function of the radius and the wavelength. The polar graph depicts the pressure at a large fixed distance, as a function of the angle in the plane of the arc. The pressure for the angle 0° is arbitrarily chosen as unity.

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.

2.9. Circular Ring Source. — The directional characteristics [10,11] of a circular ring source of uniform strength and the same phase at all points on the ring is

$$R_\alpha = J_0\left[\left(\frac{2\pi R}{\lambda}\right)\sin\alpha\right] \tag{2.29}$$

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.

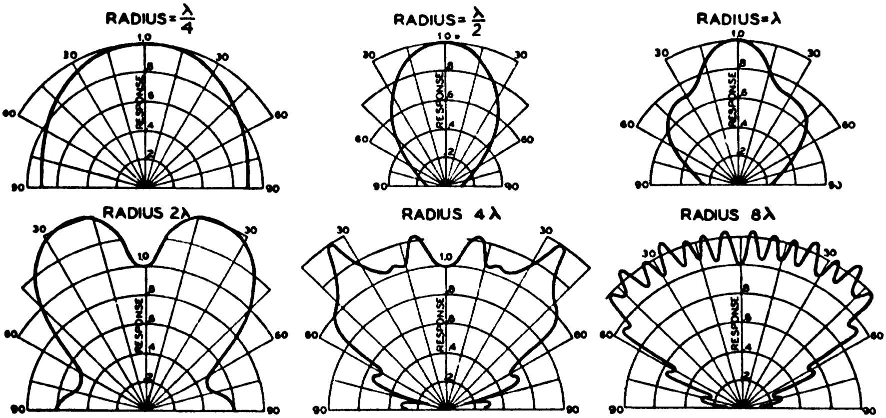

FIG. 2.8. Directional characteristics of a 120° arc as a function of the radius and the wavelength. The polar graph depicts the pressure, at a large fixed distance, as a function of the angle in the plane of the arc. The pressure for the angle 0° is arbitrarily chosen as unity.

The directional characteristics of a circular-ring source as a function of the diameter and the wavelength are shown in Fig. 2.9. The shapes are 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.

[10] Stenzel, H., *Elek. Nach. Tech.*, Vol. 4, No. 6, p. 1, 1927.
[11] Wolff, I. and Malter, L., *Jour. Acous. Soc. Amer.*, Vol. 2, No. 2, p. 201, 1930.

2.10. Plane Circular Surface Source. — The directional characteristics[12,13] of a circular surface source with all parts of the surface vibrating

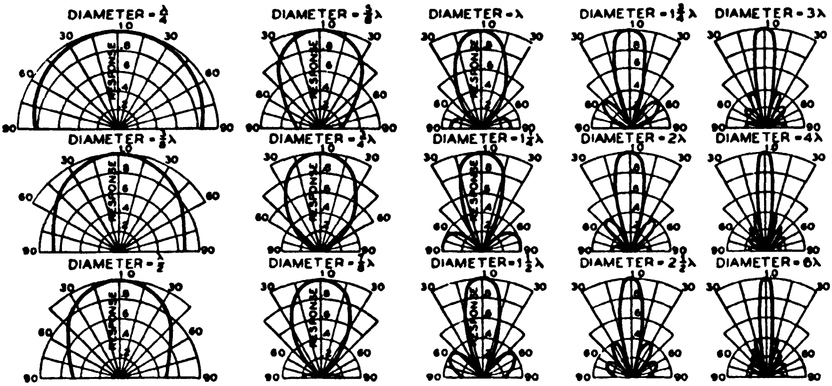

FIG. 2.9. Directional characteristics of a circular line or ring source as a function of the diameter and wavelength. The polar graph depicts the pressure, at a large fixed distance, as a function of the angle. The pressure for the angle 0° is arbitrarily chosen as unity. The direction corresponding 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.

with the same strength and phase are

$$R_\alpha = \frac{2J_1\left(\dfrac{2\pi R}{\lambda}\sin\alpha\right)}{\dfrac{2\pi R}{\lambda}\sin\alpha} \qquad 2.30$$

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 circle, in centimeters,
α = angle between the axis of the circle and the line joining the point of observation and the center of the circle, and
λ = wavelength, in centimeters.

The directional characteristics of a plane circular surface source as a function of the diameter and wavelength are shown in Fig. 2.10. The characteristic is somewhat broader than that of the uniform line of length

[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.

equal to the diameter of the circle, but has approximately the same form. The amplitudes of the secondary lobes are smaller than those of the uniform line.

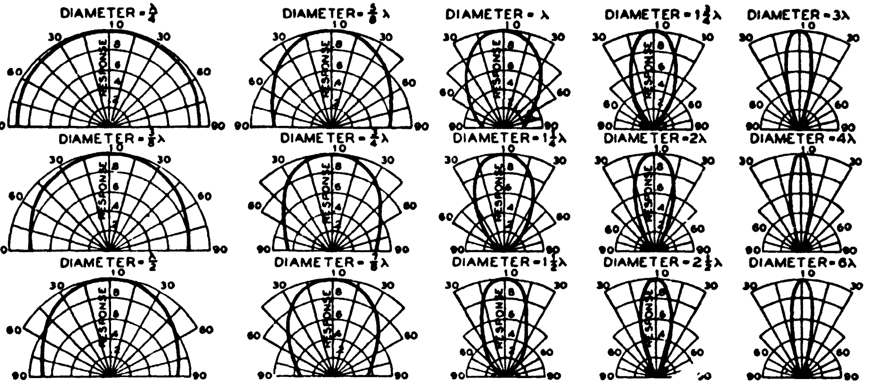

FIG. 2.10. Directional characteristics of a circular piston source as a function of the diameter and wavelength. The polar graph depicts the pressure, at a large fixed distance, as a function of the angle. The pressure for the angle 0° is arbitrarily chosen as unity. The direction corresponding to the 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.

2.11. Nonuniform Plane Circular Surface Source.[14] — The integratio 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 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.

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

[14] 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.

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 2.25).

2.13. 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

$$R_\alpha = \frac{\sin\left(\frac{\pi l_a}{\lambda}\sin\alpha\right)}{\frac{\pi l_a}{\lambda}\sin\alpha} \cdot \frac{\sin\left(\frac{\pi l_b}{\lambda}\sin\beta\right)}{\frac{\pi l_b}{\lambda}\sin\beta} \qquad 2.31$$

where l_a = length of the rectangle,

l_b = width of the rectangle,

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

2.14. 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 phase 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 [15,16] of an exponential horn is depicted in Fig. 2.11. 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 prac-

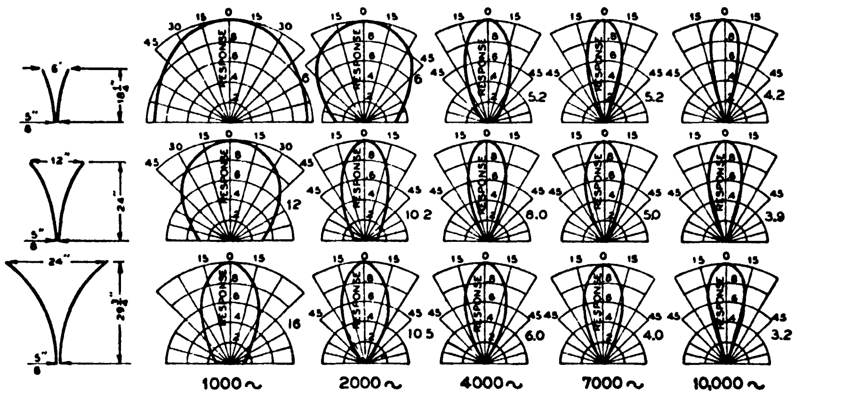

FIG. 2.11. The directional characteristics of a group of exponential horns, with a constant flare and throat diameter of ⅝ inch as a function of the mouth diameter. The number at the right of each polar diagram indicates the diameter of a circular piston which will yield the same directional characteristic. The polar graph depicts the pressure, at a fixed distance, as a function of the angle. The 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.

tically 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. 2.12 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

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

[16] 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. 2.11 and 2.12 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.

directional characteristics are approximately the same as those of a vibrating piston of the same size as the mouth. Above this frequency the directional 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

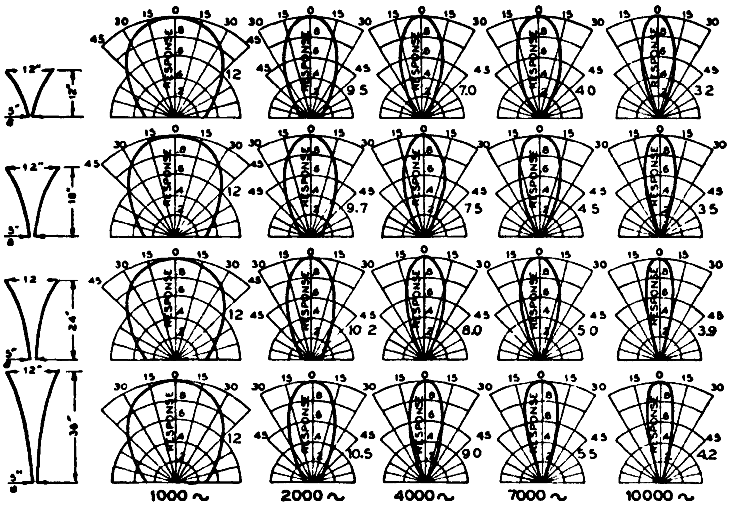

FIG. 2.12. The directional characteristics of a group of exponential horns, with a mouth diameter of 12 inches and a throat diameter of 5/8 inch, as a function of the flare. The number at the right of each polar diagram indicates the diameter of a circular piston which will yield the same directional characteristic. The polar graph depicts the pressure, at a fixed distance, as a function of the angle. The 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.

the mouth. These results also show that the directional characteristics vary very slowly with frequency at these smaller wavelengths. Referring to Fig. 2.12, 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.

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. 2.13. 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. 2.13 are practically the same as those of a spherical surface source.

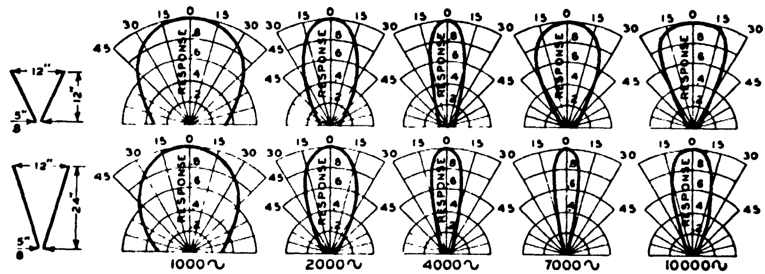

FIG. 2.13. The directional characteristics of two conical horns with mouth diameters of 12 inches and throat diameters of 5/8 inch and lengths of 12 inches and 24 inches. The polar graph depicts the sound pressure, at a fixed distance, as a function of the angle. The sound pressure for the angle 0° is 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.

C. *Parabolic Horns.* — In the parabolic horn the sectional area is proportional to the distance from the apex. This horn may be constructed as shown in Fig. 2.14 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. 2.14. The source at the mouth is essentially a curved-line source described in Sec. 2.8. 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. 2.14 with the 90° arc source of Fig. 2.7 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. 2.11, 2.12, 2.13 and 2.14 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.

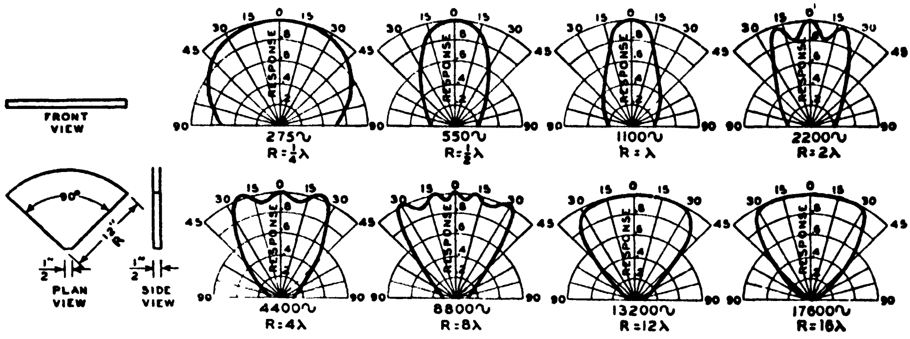

FIG. 2.14. The directional characteristics of a parabolic horn of the shape and the dimensions shown in the sketches on the left. The patterns were obtained in the plane midway between and parallel to the two parallel sides. The polar graph depicts the sound pressure, at a fixed distance, as a function 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. 2.7.

2.15. 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 loud speaker [17,18,19] 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. 2.15*A*. 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 loud speaker may be

[17] Wente, E. C., and Thuras, A. L., *Jour. A. I. E. E.*, Vol. 53, No. 1, p. 17, 1934.
[18] Hilliard, J. K., *Tech. Bull. Acad. Res. Council*, March, 1936.
[19] Olson, H. F., *RCA Review*, Vol. 1, No. 4, p. 68, 1937.

predicted theoretically[19] from the directional characteristics of an individual horn and the geometrical configuration of the assembly of horns. Assume that the point of observation is located on the *OY* axis, Fig. 2.15*B*, 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

$$\theta = \frac{d}{\lambda} 360^\circ \qquad 2.32$$

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

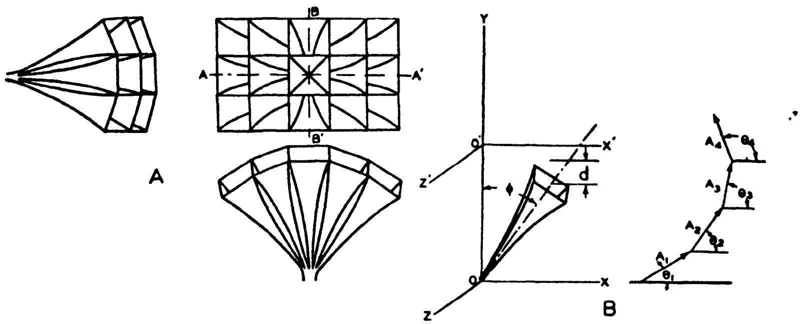

FIG. 2.15. *A*. A spherical radiating surface consisting of 15 individual exponential horns *B*. Geometry for predicting the directional characteristics of a cluster of small horns.

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 2.32, are added vectorially as shown in Fig. 2.15*B*. 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. 2.15*A* are shown in Figs. 2.16 and 2.17. 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 char-

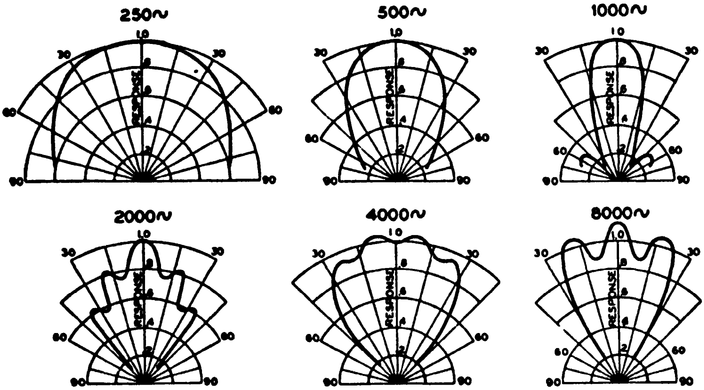

FIG. 2.16. Directional characteristics of the 15-cell cellular horn shown in Fig. 2.15*A* in a plane containing the line *B*–*B*′ and the axis of the center horn. The polar graph depicts the pressure, at a fixed distance, as a function of the angle. The pressure for the angle 0° is arbitrarily chosen.

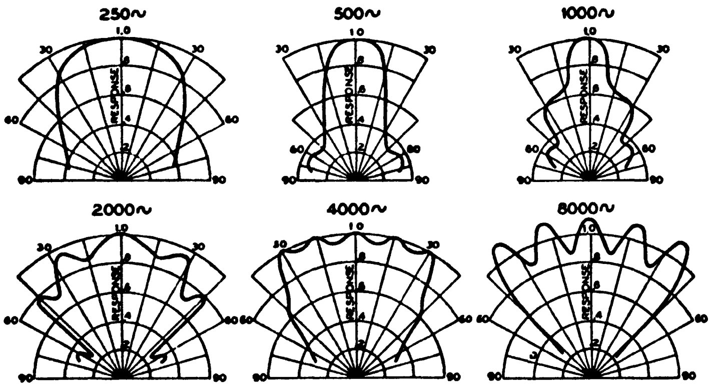

FIG. 2.17. Directional characteristics of the 15-cell cellular horn shown in Fig. 2.15*A* in a plane containing the line *A*–*A*′ and the axis of the center horn. The polar graph depicts the pressure, at a fixed distance, as a function of the angle. The pressure for the angle 0° is arbitrarily chosen.

acteristics 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. 2.15 in the plane containing the line AA' and the axis is $87\frac{1}{2}°$. This may be compared to the arc of Fig. 2.7. 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. 2.6 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 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.

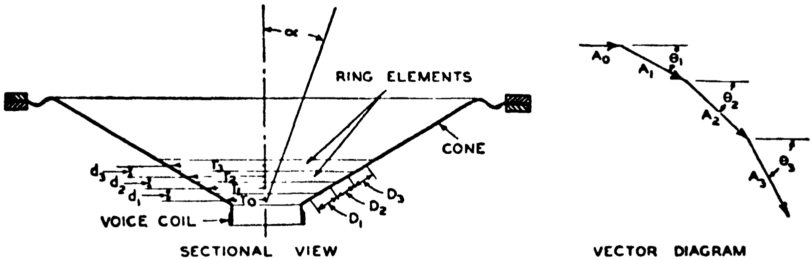

FIG. 2.18. Geometry for obtaining the directional pattern of a cone type radiator.

2.16. Cone Surface Source.[20] — The directional characteristics[21] of a paper or felted paper cone used in the direct radiator type loud speaker 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. 2.18. The dimension of the ring along the cone should be a small 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 . . . A_n where the A's are the amplitudes of the individual rings.

[20] Carlisle, R. W., *Jour. Acous. Soc. Amer.*, Vol. 15, No. 1, p. 44, 1943.

[21] 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.

The phase angle of the amplitude of the first ring is

$$\theta_0 = 0 \tag{2.33}$$

The phase angle of the amplitude of the second ring is

$$\theta_1 = 2\pi\left(\frac{d_1}{\lambda_A} - \frac{D_1}{\lambda_P}\right)\cos\alpha \tag{2.34}$$

The phase angle of the amplitude of the third ring is

$$\theta_2 = 2\pi\left(\frac{d_1 + d_2}{\lambda_A} - \frac{D_1 + D_2}{\lambda_P}\right)\cos\alpha \tag{2.35}$$

The phase angle of the amplitude of the nth ring is

$$\theta_n = 2\pi\left(\frac{d_1 + d_2 \ldots d_n}{\lambda_A} - \frac{D_1 + D_2 \ldots D_n}{\lambda_P}\right)\cos\alpha \tag{2.36}$$

where $d_1, d_2, \ldots$ = axial distances shown in Fig. 2.18 in centimeters, and
$D_1, D_2, \ldots$ = distances along the cone shown in Fig. 2.18 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

$$A_n = 2\pi r_n D_n J_0\left(\frac{2\pi r_n}{\lambda_A}\sin\alpha\right) \tag{2.37}$$

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

$$R_\alpha = \frac{\sum_{K=0}^{K=n} A_K\cos\theta_K - j\sum_{K=0}^{K=n} A_K\sin\theta_K}{\sum_{K=0}^{K=n} A_K} \tag{2.38}$$

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

A consideration of equation 2.38 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).

CHAPTER III

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 end, nodes will occur at these points. The fundamental frequency of the string is given by

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

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

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

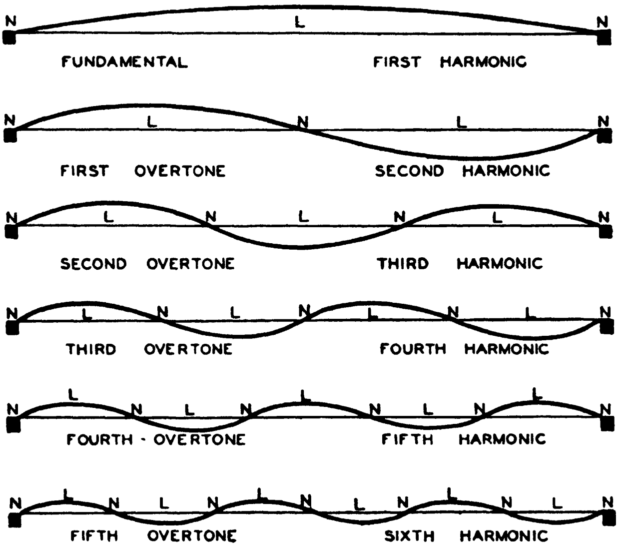

FIG. 3.1. Modes of vibration of a stretched string. The nodes and loops are indicated by N and L.

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.

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

[1] Rayleigh, "Theory of Sound," Macmillan and Company, London, 1926.
[2] Crandall, "Theory of Vibrating Systems and Sound," D. Van Nostrand Company, New York, N. Y., 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.

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

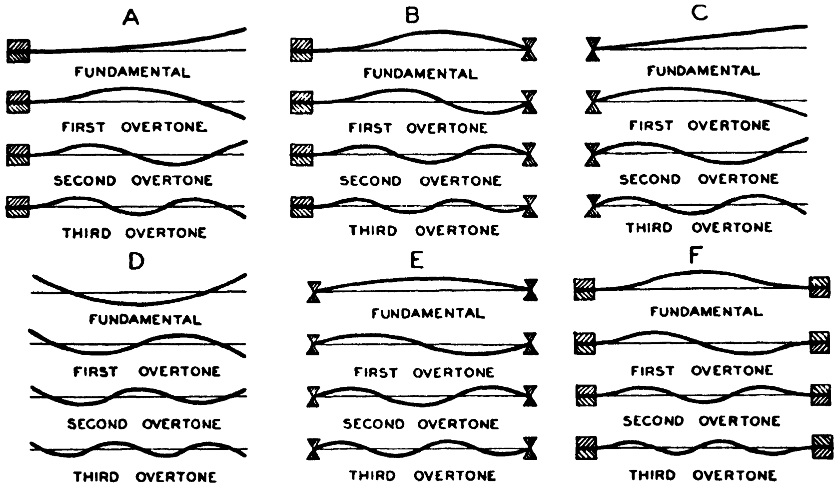

FIG. 3.2. Modes of transverse vibrations of bars. *A*. A bar clamped at one end and free at the other. *B*. A bar clamped at one end and supported at the other. *C*. A bar supported at one end and free at the other. *D*. A bar free at both ends. *E*. A bar supported at both ends. *F*. A bar clamped at both ends.

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. 3.2*A*). The fundamental frequency is given by

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

where l = length of the bar, in centimeters,

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

$\mathfrak{Q}$ = Young's modulus, in dynes per square centimeter, see Table 1.1, and

K = radius of gyration.

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
a_1 = inside radius of the pipe, in centimeters.

The modes of vibration of a bar clamped at one end are shown in Fig. 3.2*A*. 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. 3.2*D*). The fundamental frequency is given by

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

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. 3.2*D*. 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, .5, .723, .9266	8 933f_1

C. *Bar Clamped at Both Ends.* — Consider a bar rigidly clamped at both ends (Fig. 3.2*F*). 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. 3.2*E*). The fundamental frequency is given by

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

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. 3.2*B*). 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. 3.2*B*.

F. *Bar Supported at One End and Free at the Other.* — Consider a bar supported at one end and free at the other (Fig. 3.2*C*). 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. 3.2*C*.

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. 3.3*A*, 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.

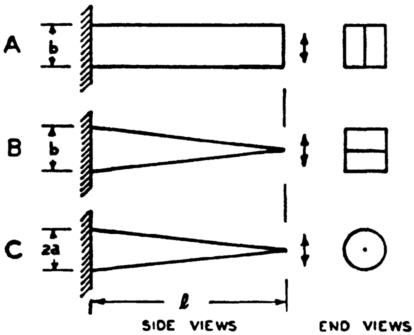

FIG. 3.3. Tapered cantilever bars, that is, bars clamped at one end and free at the other. *A*. A wedge-shaped bar vibrating in a direction normal to the two parallel sides. *B*. A wedge-shaped bar vibrating in a direction parallel to the two parallel sides. *C*. A conical bar.

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

$$f = \frac{.85}{l^2}\sqrt{\frac{\mathcal{Q}b^2}{12\rho}} \tag{3.8}$$

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

$$f = \frac{1.39}{l^2}\sqrt{\frac{\mathcal{Q}a^2}{4\rho}} \tag{3.9}$$

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 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

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

where m = mass, in grams per square centimeter of area,
R = radius of the membrane, in centimeters, and
T = tension, in dynes per centimeter.

The fundamental vibration is with the circumference as a node and a maximum displacement at the center of the circle (Fig. 3.4*A*) The fre-

[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, New York, N.Y., 1926.

quencies of the next two overtones with nodal circles are

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

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

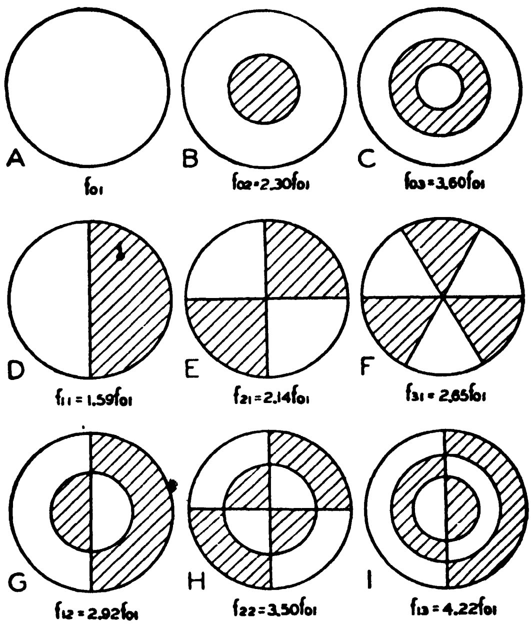

FIG. 3.4. Modes of vibration of a stretched circular membrane. Shaded segments are displaced in opposite phase to unshaded.

and are shown in Figs. 3.4*B* and 3.4*C*. 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. 3.4*D*, 3.4*E*, and 3.4*F*. 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. 3.4*G*, is

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

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

$$f_{22} = 3.50f$$

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

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

The stretched circular membrane is used in the condenser microphone (see Sec. 8.2*B*). 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, New York, N. Y., 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.

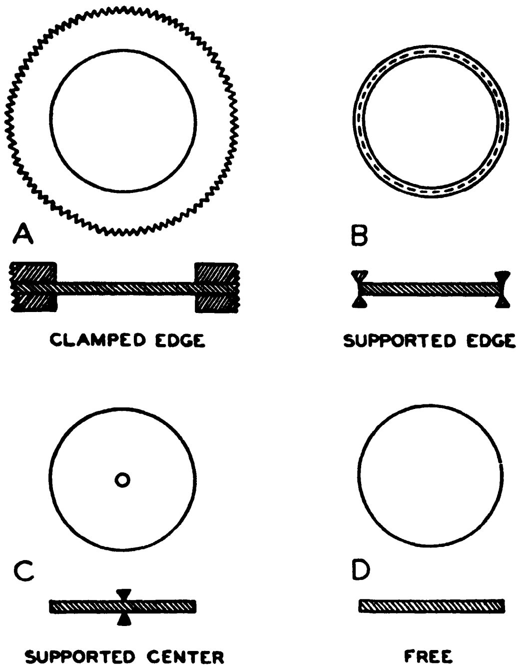

FIG. 3.5. Circular plates. *A.* A circular plate clamped at the edge. *B.* A circular plate supported at the edge. *C.* A circular plate supported at the center. *D.* A free circular plate.

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

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

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
$\mathfrak{Q}$ = 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. 3.6*A*).

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

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

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

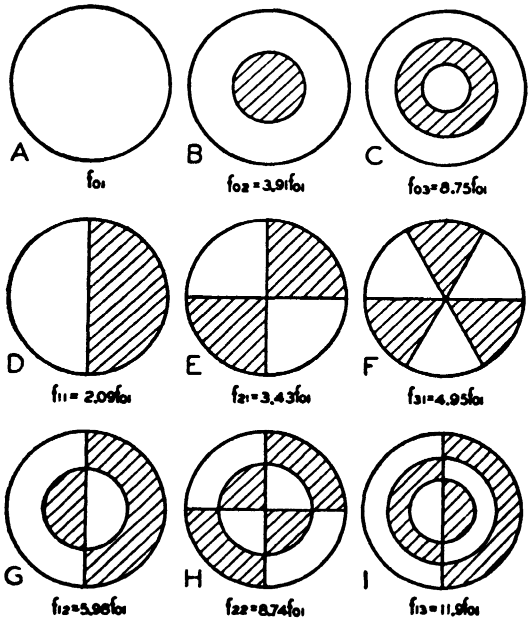

FIG. 3.6. Modes of vibration of a clamped circular plate. Shaded segments are displaced in opposite phase to unshaded.

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. 3.6*D*, 3.6*E* and 3.6*F*.

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. 3.6*G*, is

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

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

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

The frequency of two nodal circles and one nodal diameter, Fig. 3.6*I*, 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. 9.2*A*). It is used in carbon microphones (see Sec. 8.2*A*). It is used in the subaqueous condenser microphone (Sec. 13.4) and the magnetic subaqueous loud speaker (Sec. 13.6). 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.

In telephone receivers, microphones and loud speakers 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. 3.5*D*). For a vibration with nodal circle, as depicted in Fig. 3.4*B*, the frequency is

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

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
$\mathfrak{Q}$ = Young's modulus, in dynes per square centimeter (see Table 1.1).

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

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

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

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

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

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

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,

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

where l = length of the rod, in centimeters,
ρ = density of the material, in grams per cubic centimeter (see Table 1.1),
$\mathcal{Q}$ = 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.

[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.

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.

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.

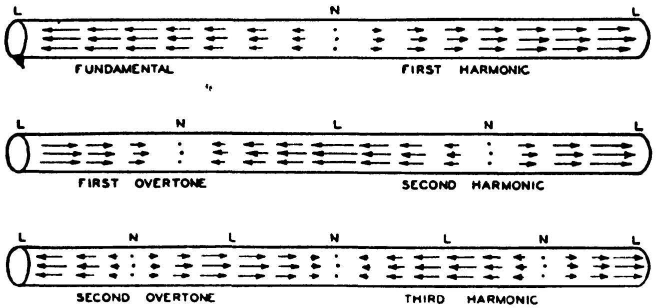

FIG. 3.7. Modes of longitudinal vibrations of a free rod. The nodes and loops are indicated by N and L.

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. 13.7 and 13.8).

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 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

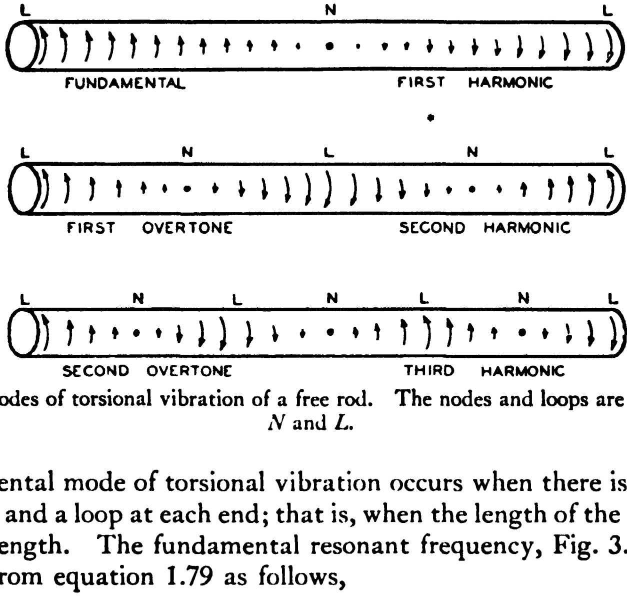

FIG. 3.8. Modes of torsional vibration of a free rod. The nodes and loops are indicated by N and L.

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,

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

where l = length of the rod, in centimeters,
ρ = density, in grams per cubic centimeter (see Table 1.1),
$\mathfrak{Q}$ = Young's modulus, in dynes per square centimeter (see Table 1.1),
σ = Poisson's ratio (see Table 1.1),

[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.

c = velocity of propagation of torsional waves, in centimeters per second, see equation 1.79, and
λ = wavelength of the torsional wave, in centimeters.

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.

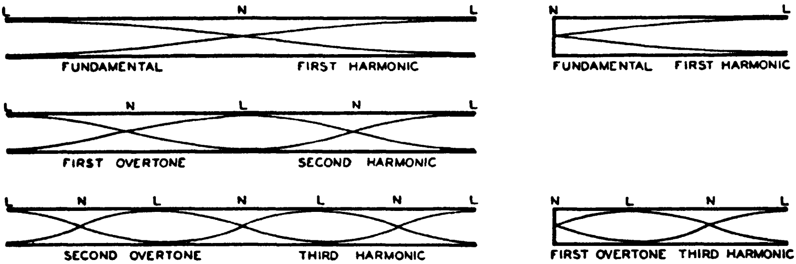

FIG. 3.9. Modes of vibration of the air column in a pipe open at both ends and in a pipe closed at one end and open at the other end. The velocity nodes and loops are indicated by N and L.

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

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

where l = length of the pipe, in centimeters,
c = velocity of sound, in centimeters per second (see Table 1.1), and
λ = wavelength, in centimeters.

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

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

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 correction has been omitted. Rayleigh[23] 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 correction 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.

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

CHAPTER IV

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.

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

[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.

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.

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.

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

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 the 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 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.

[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.

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. — An element or circuit parameter in an electrical system defines a distinct activity in its part of the circuit. In an electrical system these elements are resistance, inductance and capacitance. 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

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

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

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

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

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

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

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

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

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

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

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

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), is given by

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

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), is defined as

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

where M = inertance, in grams per (centimeter),
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[5] may be expressed as

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

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

The inertance of a circular tube is

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

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

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

Equation 4.11 may be written

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

[5] See Sec. 5.6.

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

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

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

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

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

$$p = c^2 \rho s \tag{4.15}$$

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

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

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

$$V - V' = X \tag{4.17}$$

where X = volume displacement, in cubic centimeters.

From equations 4.15, 4.16, and 4.17, the pressure is

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

Acoustical capacitance C_A is defined as

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

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

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

where V = volume, in cubic centimeters.

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.

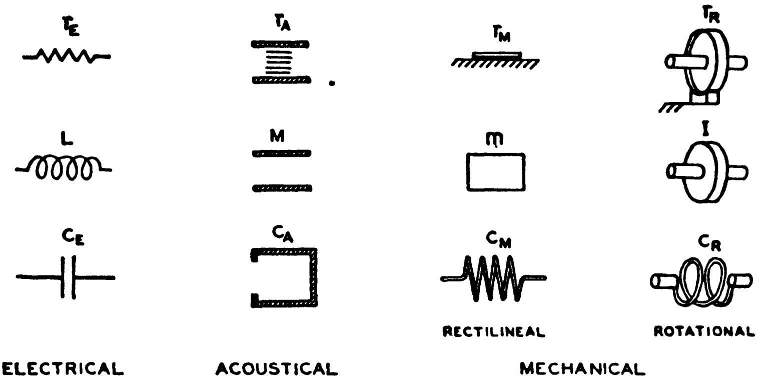

FIG. 4.1. Graphical representation of the three basic elements in electrical, mechanical rectilineal, mechanical rotational and acoustical systems.

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

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, New York, N. Y., 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	Symnol	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 × 10^8	e	$M^{1/2}L^{3/2}T^{-2}$	Force	Dynes	f_M	MLT^{-2}
Charge or Quantity	Coulombs × 10^{-1}	q	$M^{1/2}L^{1/2}$	Linear Displacement	Centimeters	x	L
Current	Amperes × 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 × 10^9	z_E	LT^{-1}	Mechanical Impedance	Mechanical Ohms	z_M	MT^{-1}
Electrical Resistance	Ohms × 10^9	r_E	LT^{-1}	Mechanical Resistance	Mechanical Ohms	r_M	MT^{-1}
Electrical Reactance	Ohms × 10^9	x_E	LT^{-1}	Mechanical Reactance	Mechanical Ohms	x_M	MT^{-1}
Inductance	Henries × 10^9	L	L	Mass	Grams	m	M
Electrical Capacitance	Farads × 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 Centimeters 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}

CHAPTER V

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,

[1] Crandall, "Vibrating Systems and Sound," D. Van Nostrand Company, New York, N. Y., 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, New York, N. Y., 1926.
[5] Lamb, "Dynamical Theory of Sound," E. Arnold, London, 1931.
[6] Rayleigh, "Theory of Sound," Macmillan and Company, London, 1926.

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.

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.

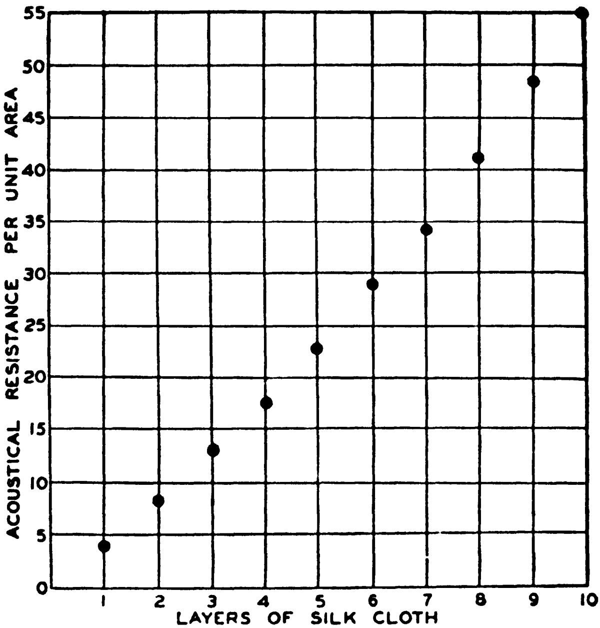

FIG. 5.1. The acoustical resistance, per square centimeter, of sheer silk cloth as a function of the number of layers.

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 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 loud speakers 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. 4.5*D*, 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 and 5.10. 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 and 5.10 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. 1.3*D*, 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.

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

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

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

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

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

where r_A = acoustical resistance of the boundary, in acoustical ohms,
C_A = acoustical capacitance of the volume, in cubic centimeters per second,
$\omega = 2\pi f$, and
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

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

where R = radius of piston, in centimeters,
ρ = density, in grams per cubic centimeter,
c = velocity of sound, in centimeters per second,
$k = 2\pi/\lambda$,
λ = wavelength, in centimeters,
$\omega = 2\pi f$, and
f = frequency, in cycles per second.

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[7] Rayleigh, "Theory of Sound," Macmillan and Company, London, 1926.
[8] Crandall, "Vibrating Systems and Sound," D. Van Nostrand Company, New York, N. Y., 1926.
[9] Stewart and Lindsay, "Acoustics," D. Van Nostrand Company, New York, N. Y., 1930.

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

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

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

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

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

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 direct radiator loud speaker. 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.

[10] Watson, "Theory of Bessel Functions," Cambridge Press, London, 1922.
[11] Jahnke and Emde, "Tables of Function," Teubner, Berlin, 1928.

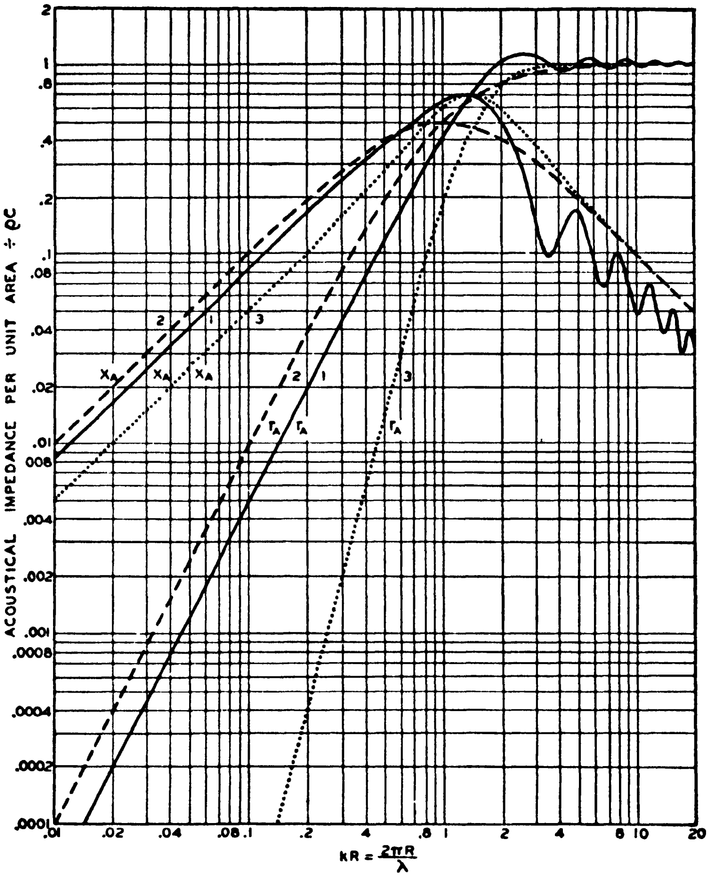

FIG. 5.2. The acoustical resistance, r_A, and acoustical reactance, x_A, load per unit area divided by ρc, as a function of kR for the following radiators: 1. a piston of radius R set in an infinite baffle; 2. a pulsating sphere of radius R; 3. an oscillating sphere of radius R. Note: The ordinate scale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

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

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

The acoustical impedance per unit area is

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

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

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

where R = radius of the sphere, in centimeters,
ρ = density, in grams per cubic centimeter,
$k = \frac{2\pi}{\lambda}$,
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

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

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 loud speaker without a baffle.

5.11. 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.

5.12. 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

$$x_A = \frac{\rho l}{\pi R^2}\omega \qquad 5.20$$

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.

The acoustical impedance, in acoustical ohms, of the entire system is

$$z_A = \frac{2\rho c}{\pi R^2}\left[1 - \frac{J_1(kR)}{kR}\right] + j\frac{\omega\rho}{\pi R^4 k^3}K_1(2kR) + \frac{j\rho l}{\pi R^2}\omega \qquad 5.21$$

5.13. 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 loud speaker 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 loud speaker 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.

5.14. Fundamental Horn Equation.[12, 13, 14, 15, 16, 17, 18, 19] — 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

$$S\Delta x \tag{5.22}$$

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

$$\Delta x \frac{\partial(S\rho' u)}{\partial x} \tag{5.23}$$

where u = component of the particle velocity along the axis, and
ρ' = density of the medium.

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.

$$\frac{\partial \rho'}{\partial t} S\Delta x = -\Delta x \frac{\partial(S\rho' u)}{\partial x} \tag{5.24}$$

or

$$S\frac{\partial \rho'}{\partial t} + \frac{\partial(S\rho' u)}{\partial x} = 0 \tag{5.25}$$

From equations 1.19 and 1.6

$$-\rho\ddot{\phi} = c^2\dot{\rho}' \tag{5.26}$$

[12] Webster, A. G., *Jour. Nat. Acad. Sci.*, Vol. 5, p. 275, 1919.
[13] Stewart, G. W., *Phys. Rev.*, Vol. 16, No. 4, p. 313, 1920.
[14] Goldsmith and Minton, *Proc. Inst. Rad. Eng.*, Vol. 12, No. 4, p. 423, 1924.
[15] Slepian and Hanna, *Jour. Amer. Inst. Elec. Eng.*, Vol. 43, p. 393, 1924.
[16] Ballantine, G., *Jour. Frank. Inst.*, Vol. 203, No. 1, p. 85, 1927.
[17] Crandall, "Vibrating Systems and Sound," D. Van Nostrand Company, New York, N. Y., 1926.
[18] Stewart and Lindsay, "Acoustics," D. Van Nostrand Company, New York, N. Y., 1930.
[19] Olson and Massa, "Applied Acoustics," P. Blakiston's Son and Company, Philadelphia, Pa., 1934.

From equation 1.11

$$u = \frac{\partial \phi}{\partial x} \tag{5.27}$$

Substituting equations 5.26 and 5.27 in 5.25 the result may be written as

$$\ddot{\phi} - c^2 \frac{\partial \phi}{\partial x} \frac{\partial}{\partial x} (\log S) - c^2 \frac{\partial^2 \phi}{\partial x^2} = 0 \tag{5.28}$$

Equation 5.28 is the wave equation for the axial motion in a tube of varying section.

5.15. Infinite Cylindrical Horn (Infinite Pipe). — The equation expressing the cross-sectional area as a function of the distance along the axis is

$$S = S_1 \tag{5.29}$$

where S_1 = cross section of the pipe, in square centimeters.

The general horn equation for the infinite pipe from equations 5.28 and 5.29 is

$$\ddot{\phi} - c^2 \frac{\partial^2 \phi}{\partial x^2} = 0 \tag{5.30}$$

The velocity potential, pressure and volume current are

$$\phi = A\epsilon^{jk(ct-x)} \tag{5.31}$$

$$p = kc\rho A\epsilon^{jk(ct-x)} \tag{5.32}$$

$$U = S_1 k A\epsilon^{jk(ct-x)} \tag{5.33}$$

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

$$r_A = \frac{\rho c}{S_1} \tag{5.34}$$

$$x_A = 0 \tag{5.35}$$

5.16. Infinite Parabolic Horn.[20] — The equation expressing the cross-sectional area as a function of the distance along the axis is

$$S = S_1 x \tag{5.36}$$

[20] Olson and Wolff, *Jour. Acous. Soc. Amer.*, Vol. 1, No. 3, p. 410, 1930.

The general horn equation for the parabolic horn is

$$\ddot{\phi} - \frac{c^2}{x}\frac{\partial \phi}{\partial x} - c^2 \frac{\partial^2 \phi}{\partial x^2} = 0 \qquad 5.37$$

The velocity potential, pressure and volume current are,

$$\phi = A\,[J_0(kx) - jY_0(kx)]\,\epsilon^{j\omega t} \qquad 5.38$$

$$p = -i\omega\rho A\,[J_0(kx) - jY_0(kx)]\,\epsilon^{j\omega t} \qquad 5.39$$

$$U = ASk\,[-J_0'(kx) + jY_0'(kx)]\,\epsilon^{j\omega t} \qquad 5.40$$

The real and imaginary components of the acoustical impedance, in acoustical ohms, at the throat are

$$r_A = \frac{\rho c}{S_1}\,\frac{2}{\pi k x_0 [J_1^2(kx_0) + Y_1^2(kx_0)]} \qquad 5.41$$

$$x_A = \frac{\rho c}{S_1}\,\frac{J_0(kx_0)J_1(kx_0) + Y_0(kx_0)Y_1(kx_0)}{J_1^2(kx_0) + Y_1^2(kx_0)} \qquad 5.42$$

where J_0, J_1 = Bessel functions of the first kind of order zero and one,
Y_0, Y_1 = Bessel functions[21] 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.

5.17. Infinite Conical Horn. — The equation expressing the cross-section area as a function of the distance along the axis is,

$$S = S_1 x^2 \qquad 5.43$$

The general horn equation for the conical horn is

$$\ddot{\phi} - \frac{2c^2}{x}\frac{\partial \phi}{\partial x} - c^2 \frac{\partial^2 \phi}{\partial x^2} = 0 \qquad 5.44$$

The velocity potential, pressure and volume current are

$$\phi = \frac{A}{x}\,\epsilon^{j(\omega t - kx)} \qquad 5.45$$

[21] Jahnke and Emde, "Tables of Functions," Teubner, Berlin, 1928.

$$p = -\frac{j\omega\rho A}{x}\epsilon^{j(\omega t - kx)} \tag{5.46}$$

$$U = -\frac{AS(1 + jkx)\epsilon^{j(\omega t - kx)}}{x^2} \tag{5.47}$$

The real and imaginary components of the acoustical impedance, in acoustical ohms, at the throat are

$$r_A = \frac{\rho c}{S_1}\frac{k^2x_1^2}{1 + k^2x_1^2} \tag{5.48}$$

$$x_A = \frac{\rho c}{S_1}\frac{kx_1}{1 + k^2x_1^2} \tag{5.49}$$

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.

5.18. Infinite Exponential Horn. — The equation expressing the cross-sectional area as a function of the distance along the axis

$$S = S_1\epsilon^{mx} \tag{5.50}$$

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

$$\ddot{\phi} - c^2m\frac{\partial\phi}{\partial x} - c^2\frac{\partial^2\phi}{\partial x^2} = 0 \tag{5.51}$$

The velocity potential, pressure and volume current are

$$\phi = \epsilon^{-(m/2)x}\left[A\epsilon^{-j\frac{\sqrt{4k^2 - m^2}}{2}x}\right]\epsilon^{j\omega t} \tag{5.52}$$

$$p = -j\omega\rho\epsilon^{-(m/2)x}\left[A\epsilon^{-j\frac{\sqrt{4k^2 - m}}{2}x}\right]\epsilon^{j\omega t} \tag{5.53}$$

$$U = -AS\left[\frac{m}{2} + j\frac{\sqrt{4k^2 - m^2}}{2}\right]\epsilon^{-\frac{m}{2}x - j\frac{\sqrt{4k^2 - m^2}}{2}x + j\omega t} \tag{5.54}$$

The real and imaginary components of the acoustical impedance, in acoustical ohms, at the throat are

$$r_A = \frac{\rho c}{S_1}\sqrt{1 - \frac{m^2}{4k^2}} \qquad 5.55$$

$$x_A = \frac{\rho c}{S_1}\frac{m}{2k} \qquad 5.56$$

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

$$x_A = \frac{\rho c}{S_1}\left(\frac{m}{2k} - \sqrt{1 - \frac{m^2}{4k^2}}\right) \qquad 5.57$$

5.19. Infinite Hyperbolic Horn.[22] — The equation expressing the cross-sectional area along the axis is

$$S = S_1 (\cosh \alpha + T \sinh \alpha)^2 \qquad 5.58$$

where T = family parameter, in the hyperbolic horn $T < 1$,

$\alpha = \frac{x}{x_0}$, 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

$$r_A = \frac{\rho c}{S_1}\frac{\sqrt{1 - \frac{1}{\mu^2}}}{1 - \frac{1 - T^2}{\mu^2}} \qquad 5.59$$

$$x_A = \frac{\rho c}{S_1}\frac{\frac{T}{\mu}}{1 - \frac{1 - T^2}{\mu^2}} \qquad 5.60$$

[22] Salmon, V., *Jour. Acous. Soc. Amer.*, Vol. 17, No. 3, p. 212, 1946.

where $\mu = kx_0 = \dfrac{f}{f_0}$

$k = \dfrac{2\pi}{\lambda}$,

f_0 = cutoff frequency, and

f = frequency under consideration.

Below the cutoff frequency, $\mu = 1$, the acoustical impedance is entirely reactive and

$$x_A = \frac{\rho c}{S_1} \frac{\dfrac{T}{\mu} - \sqrt{\dfrac{1}{\mu^2} - 1}}{1 - \dfrac{1 - T^2}{\mu^2}} \qquad 5.61$$

5.20. 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. 5.14, 5.15, 5.16, 5.17, 5.18 and 5.19. 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 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. 5.3. 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. 5.3.

5.21. Finite Cylindrical Horn. — The acoustical impedance, in acoustical ohms, at the throat of the finite cylindrical horn of Fig. 5.4 is

$$z_{A1} = \frac{p_1}{U_1} \qquad 5.62$$

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

$$z_{A2} = \frac{p_2}{U_2} \qquad 5.63$$

where p_2 = pressure at the mouth, in dynes per square centimeter, and
U_2 = volume current, in cubic centimeters per second.

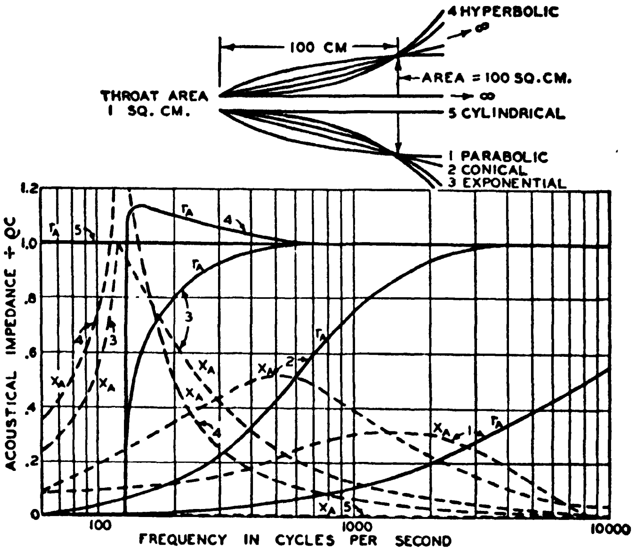

FIG. 5.3. Throat acoustical resistance r_A, and acoustical reactance x_A, frequency characteristics of infinite parabolic, conical, exponential, hyperbolic and cylindrical horns having a throat area of 1 square centimeter. The cross-sectional area of the parabolic, conical, exponential and hyperbolic horns is 100 square centimeters at a distance of 100 centimeters from the throat.

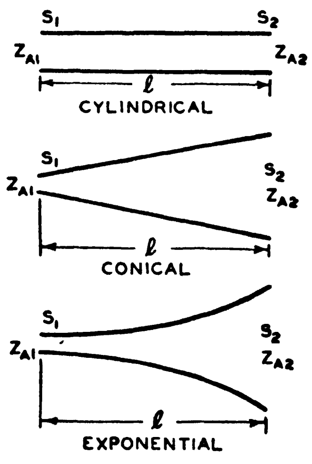

FIG. 5.4. Finite cylindrical, conical and exponential horns. z_{A1} = input acoustical impedance at the throat. S_1 = cross-sectional area at the throat, in square centimeters. x_{A2} = terminating acoustical impedance at the throat. S_2 = cross-sectional area at the mouth, in square centimeters. l = length, in centimeters.

From equations 5.32 and 5.33 the expressions for the pressures and volume currents at the throat and mouth are given by

At $x = 0$,

$$p_1 = kc\rho A\epsilon^{jkct} \qquad 5.64$$

$$U_1 = S_1kA\epsilon^{jkct} \qquad 5.65$$

At $x = l$,

$$p_2 = kc\rho A\epsilon^{jk(ct-l)} \qquad 5.66$$

$$U_2 = S_1kA\epsilon^{jk(ct-l)} \qquad 5.67$$

From equations 5.62, 5.63, 5.64, 5.65, 5.66 and 5.67 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

$$z_{A1} = \frac{\rho c}{S_1}\left(\frac{S_1 z_{A2}\cos(kl) + j\rho c\sin(kl)}{jS_1 z_{A2}\sin(kl) + \rho c\cos(kl)}\right) \qquad 5.68$$

where ρ = density of the medium, in grams per cubic centimeter,
$k = \dfrac{2\pi}{\lambda}$,
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. 5.5. 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.

5.22. Finite Conical Horn.—The acoustical impedance at the throat of a finite conical horn of Fig. 5.4 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

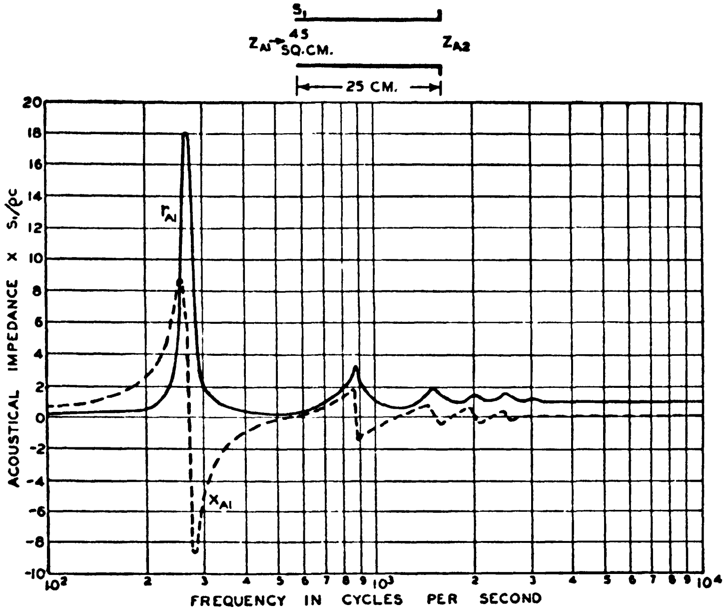

FIG. 5.5. The throat acoustical resistance and acoustical reactance frequency characteristics of a finite cylindrical horn. r_{A1} = acoustical resistance. x_{A1} = acoustical reactance. Note: The characteristics shown are the throat acoustical resistance and acoustical reactance multiplied by S_1 and divided by ρc.

horn and the acoustical impedance, z_{A2}, at the mouth is

$$z_{A1} = \frac{\rho c}{S_1}\left[\frac{jz_{A2}\dfrac{\sin k(l-\theta_2)}{\sin k\theta_2} + \dfrac{\rho c}{S_2}\sin kl}{z_{A2}\dfrac{\sin k(l+\theta_1-\theta_2)}{\sin k\theta_1 \sin k\theta_2} - \dfrac{j\rho c}{S_2}\dfrac{\sin k(l+\theta_1)}{\sin k\theta_1}}\right] \qquad 5.69$$

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,
$k = \dfrac{2\pi}{\lambda}$,

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. 5.6. 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.

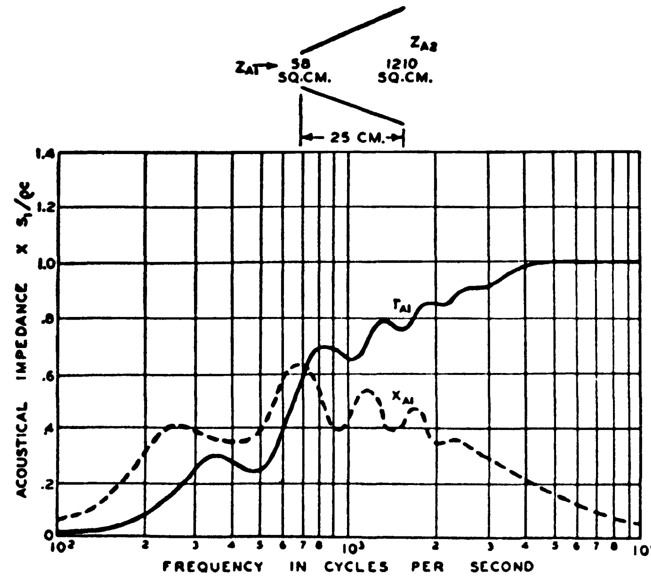

FIG. 5.6. The throat acoustical resistance and acoustical reactance frequency characteristics of a finite conical horn. r_{A1} = acoustical resistance. x_{A1} = acoustical reactance. Note: The characteristics shown are the throat acoustical resistance and acoustical reactance multiplied by S_1 and divided by ρc.

5.23. Finite Exponential Horn.[23]—The acoustical impedance at the throat of a finite exponential horn 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

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

constant of the horn and the acoustical impedance, z_{A2}, at the mouth is

$$z_{A1} = \frac{\rho c}{S_1}\left[\frac{S_2 z_{A2}\,[\cos\,(bl + \theta)] + j\rho c\,[\sin\,(bl)]}{jS_2 z_{A2}\,[\sin\,(bl)] + \rho c\,[\cos\,(bl - \theta)]}\right] \qquad 5.70$$

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

For $b = 0$, equation 5.70 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

$$z_{A1} = \frac{\rho c}{S_1}\left[\frac{z_{A2}\left(1 - \dfrac{ml}{2}\right) + j\dfrac{\rho c}{S_2}\dfrac{lm}{2}}{jz_{A2}\dfrac{lm}{2} + \dfrac{\rho c}{S_2}\left(1 + \dfrac{ml}{2}\right)}\right] \qquad 5.71$$

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:

$$\tan^{-1} jA = j\tanh^{-1} A = \tfrac{1}{2}j[\log_\epsilon (1 + A) - \log_\epsilon (1 - A)] \qquad 5.72$$

$$\log_\epsilon (-1) = \pm j\pi(2K + 1),\ K = \text{any integer} \qquad 5.73$$

$$\cos(A \pm jB) = \cos A \cosh B \mp j \sin A \sinh B \qquad 5.74$$

$$\sin jA = j \sinh A \qquad 5.75$$

The resistive and reactive components of the acoustical impedance of a finite exponential horn are shown in Fig. 5.7. 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. 5.7 shows that there is a sudden change in acoustical 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. 5.6, 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

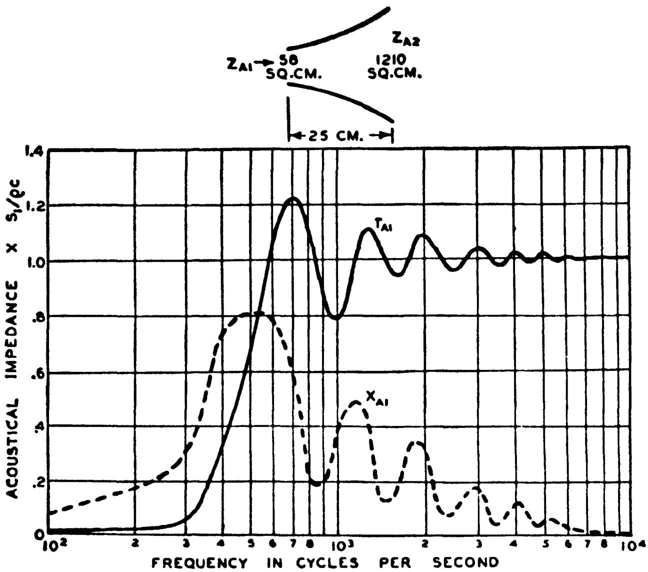

FIG. 5.7. The throat acoustical resistance and acoustical reactance frequency characteristics of a finite exponential horn. r_{A1} = acoustical resistance. x_{A1} = acoustical reactance. Note: The characteristics shown are the throat acoustical resistance or acoustical reactance multiplied by S_1 and divided by ρc.

for its almost universal use. In view of its widespread use it is interesting to examine some of the other characteristics of exponential horns.

5.24. Throat Acoustical Impedance Characteristics of Finite Exponential Horns.[24] — 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. 5.8. These results may be applied to horns of a different flare by multiplying

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

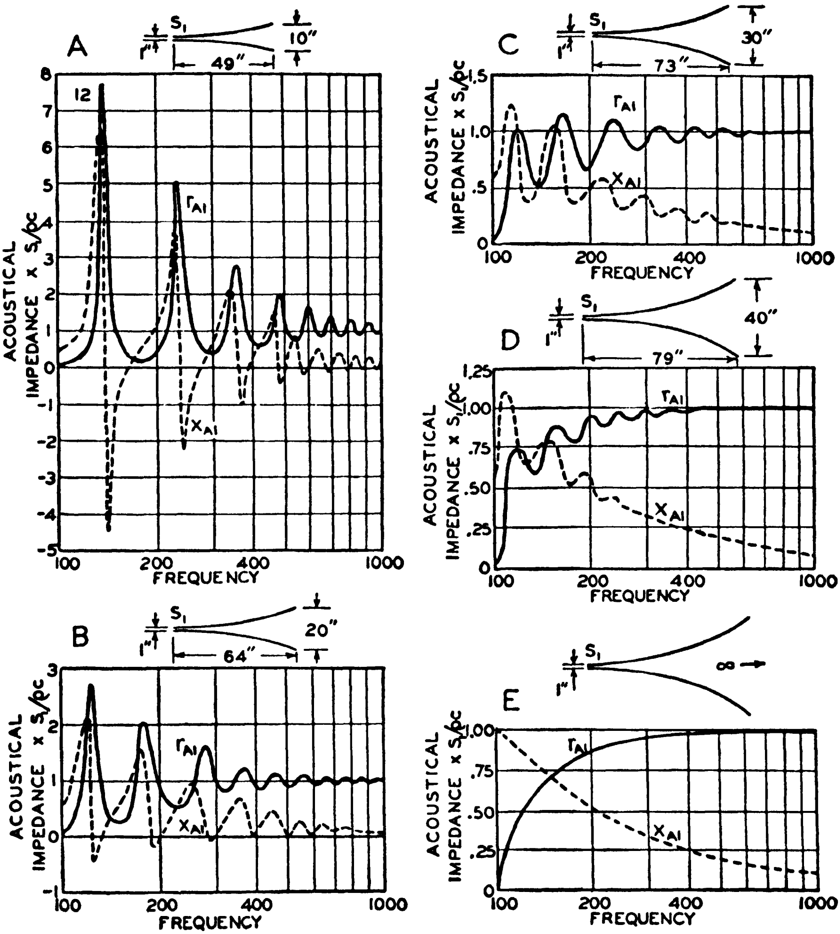

FIG. 5.8. The throat acoustical resistance and acoustical reactance frequency characteristics of a group of exponential horns, with a flare cutoff of 100 cycles and a throat diameter of 1 inch, as a function of the mouth diameter. S_1 = the throat diameter in square centimeters. r_{A1} = acoustical resistance. x_{A1} = acoustical reactance. Note: The characteristics shown are the throat acoustical resistance or acoustical reactance multiplied by S_1 and divided by ρc.

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

$$2\omega = mc \tag{5.76}$$

where $\omega = 2\pi f$,
f = frequency, in cycles per second, and
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. 5.8*A*. 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. 5.8*B*.

Fig. 5.8*C* 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. 5.8*D*, shows a deviation in acoustical resistance of only a few per cent from that of the infinite horn of Fig. 5.8*E*.

These results show that as the change in acoustical impedance in passing from the mouth to the free atmosphere becomes smaller by employing a 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 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. 5.9. 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. 5.8 and 5.9 cover the range from 100 to 1000 cycles, the lower value being the flare cutoff frequency. The finite horn, of course, transmits below this frequency because the acoustical resistance is not zero. However, save for the case where the throat is

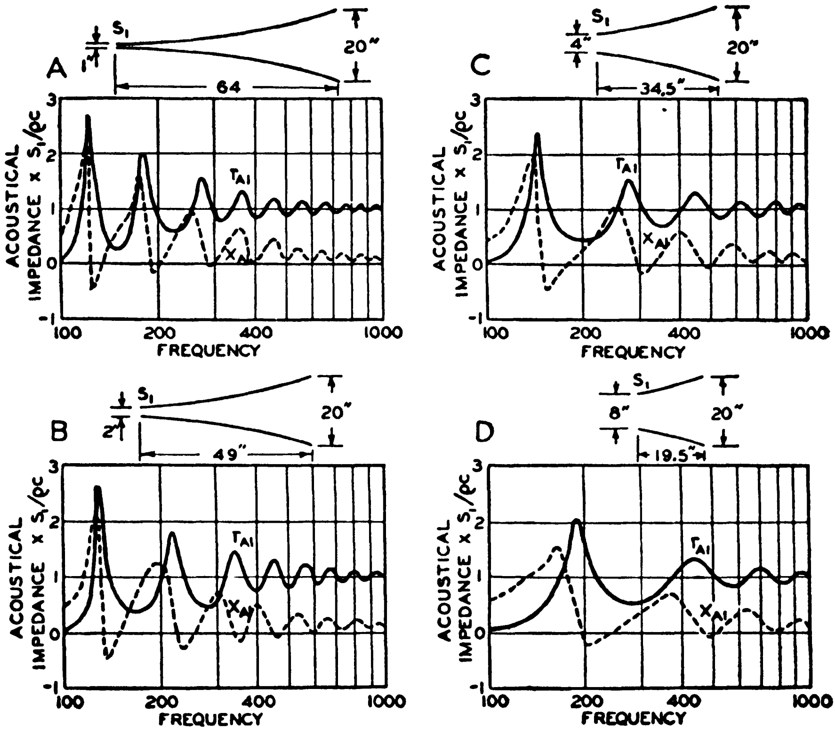

Fig. 5.9. The throat acoustical resistance and acoustical reactance frequency characteristics of a group of exponential horns, with a flaie cutoff of 100 cycles and a mouth diameter of 20 inches, as a function of the throat mouth diameter. S_1 = the throat diameter, in square centimeters. r_{A1} = acoustical resistanee. x_{A1} = acoustical reactance. Note: The characteristics shown are the throat acoustical resistance or acoustical reactance multiplied by S_1 and divided by ρc.

comparable to the mouth, as for example, Fig. 5.9*D*, the value of the acoustical resistance, at and below the flare cutoff frequency, is quite small.

5.25. 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

$$\frac{\rho c}{S_2} \tag{5.77}$$

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.

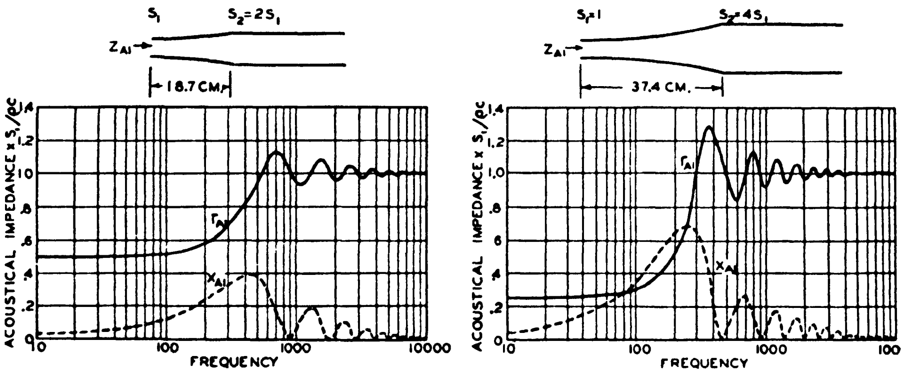

Fig. 5.10. The throat acoustical resistance and acoustical reactance frequency characteristics of two exponential connectors with a flare cutoff of 100 cycles. The mouth of the horn is connected to an infinite pipe. r_{A1} = acoustical resistance. x_{A1} = acoustical reactance. Note: The characteristics shown are the acoustical resistance or acoustical reactance multiplied by S_1 and divided by ρc.

Equation 5.77 is the mouth acoustical impedance of the exponential horn. Equation 5.70 then becomes

$$z_{A1} = \frac{\rho c}{S_1}\left[\frac{\cos (bl + \theta) + j \sin (bl)}{\cos (bl - \theta) + j \sin (bl)}\right] \tag{5.78}$$

For $b = 0$, equation 5.78 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

$$z_{A1} = \frac{\rho c}{S_1}\left[\frac{1 + j\dfrac{lm}{2} - \dfrac{lm}{2}}{1 + \dfrac{lm}{2} + j\dfrac{lm}{2}}\right] \qquad 5.79$$

Below the frequency corresponding to $b = 0$, b is imaginary. This portion of the range may be evaluated by employing equations 5.72, 5.73, 5.74 and 5.75.

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. 5.10. 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.

5.26. A Horn Consisting of Manifold Exponential Sections.[25] — The efficiency of a horn loud speaker 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 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. 5.11. 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 5.70. 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 5.70. For the frequency corresponding to $b = 0$ of the intermediate horn the acoustical impedance at the throat of the intermediate horn becomes

[25] Olson, H. F., *Jour. Soc. Mot. Pic. Eng.*, Vol. 30, No. 5, p. 511, 1938.

indeterminate. The expression can be evaluated as shown in Sec. 5.23 on the finite exponential horn. Next, the throat acoustical impedance at the throat of the small horn is obtained by again employing equation 5.70. The mouth acoustical impedance of the small horn is the throat acoustical

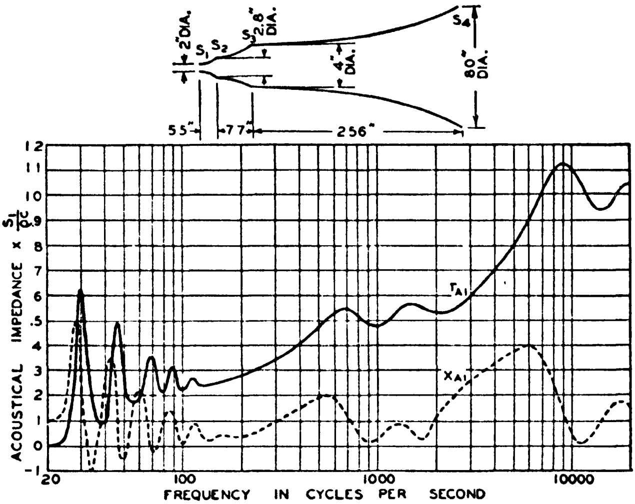

FIG. 5.11. The throat acoustical resistance and acoustical reactance frequency characteristics of a multiple flare exponential horn of three sections. The cutoffs due to flare of the three horns are 25, 100 and 1400 cycles. r_{A1} = acoustical resistance. x_{A1} = acoustical reactance. Note: The characteristics shown are the throat acoustical resistance or acoustical reactance multiplied by S_1 and divided by ρc. S_1 = area at the throat of the small horn in centimeters.

impedance just obtained for the intermediate horn. The acoustical impedance characteristic of Fig. 5.11 shows three distinct steps depicting the surge acoustical impedance of each section.

5.27. 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 5.68 by setting $z_{A2} = \infty$. The input acoustical impedance of the pipe closed at the far end is

$$z_{A0} = -\frac{j\rho c}{\pi R^2} \cot kl \tag{5.80}$$

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 5.80.

$$z_{AT} = \frac{\rho c}{\pi R^2}\left[1 - \frac{J_1(kR)}{kR}\right] + j\frac{\omega\rho}{2\pi R^4 k^3} K_1(2kR) - j\frac{\rho c}{\pi R^2}\cot kl \qquad 5.81$$

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

$$\frac{p_0}{p} = \sqrt{\left[\cos kl - \frac{\pi R^2}{\rho c} x_A \sin kl\right]^2 + \frac{(\pi R^2)^2}{(\rho c)^2} r_A{}^2 \sin^2 kl} \qquad 5.82$$

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.

The characteristics of Fig. 5.12 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.

5.28. Sound Transmission in Tubes.[26,27,28] — The effect of viscosity upon the characteristics of small holes and slits was considered in Secs. 5.3 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 [29] expressing the sound transmission in a tube is

$$A = A_0\epsilon^{-\alpha x} \qquad 5.83$$

[26] Crandall, " Vibrating Systems and Sound," D. Van Nostrand Company, New York, N. Y., 1926.
[27] Lamb, " Theory of Sound," E. Arnold, London, 1931.
[28] Rayleigh, " Theory of Sound," Macmillan and Company, London, 1926.
[29] Mason, W. P., *Phys. Rev.*, Vol. 31, No. 2, p. 283, 1928.

where A = amplitude (pressure or volume current) at a distance x centimeters from the amplitude A_0,

$$\alpha = \frac{\gamma'}{Rc}\sqrt{\frac{\omega\mu}{2\rho}},$$

R = radius of the tube, in centimeters,
c = velocity of sound, in centimeters per second,
$\omega = 2\pi f$,

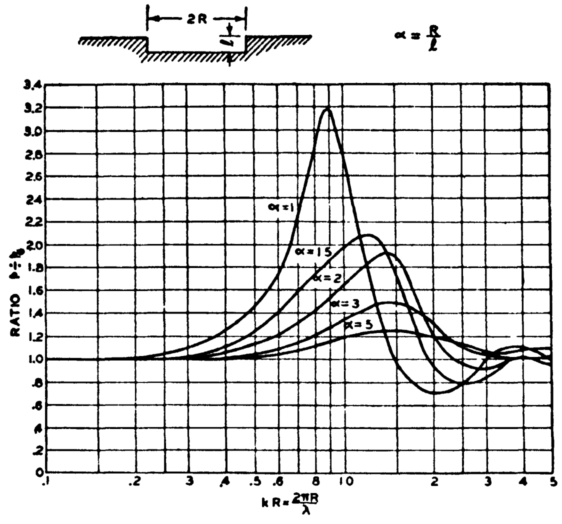

FIG. 5.12. Ratio of the pressure at the bottom of a cylindrical cavity to the free space pressure in the incident sound wave as a function of kR.

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. 5.13.

5.29. Transmission from One Pipe to Another Pipe of Different Cross-sectional Area.[30] — Consider two pipes of cross sections S_1 and S_2 joined as shown in Fig. 5.14. Assume that sound travels from pipe S_1 to pipe S_2.

[30] Stewart and Lindsay, "Acoustics," D. Van Nostrand Company, New York, N. Y., 1930.

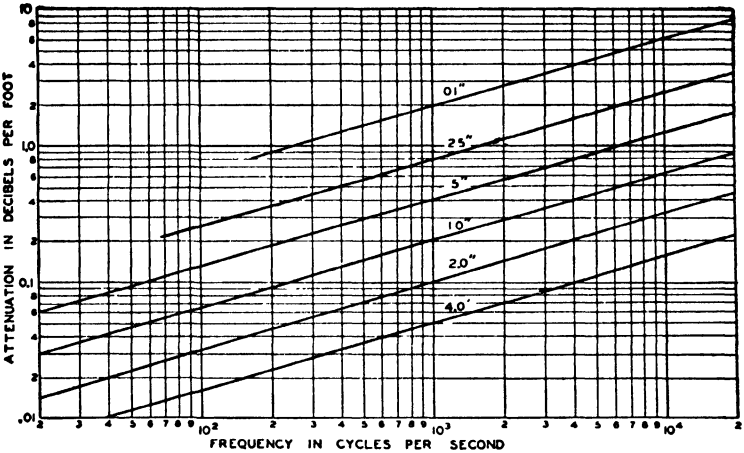

FIG. 5.13. The attenuation of a sound wave, in decibels per foot, as a function of the frequency, in cycles per second, in pipes of various diameters and filled with dry air at 20° Centigrade.

The boundary conditions are

1. Continuity of pressure;
2. Continuity of volume current.

The condition for pressure may be written

$$p_1 + p_1' = p_2 \qquad 5.84$$

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.

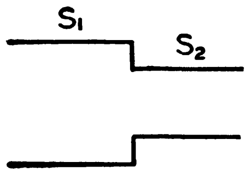

FIG. 5.14. Two connected pipes of cross-sectional areas S_1 and S_2.

The condition for volume current may be written

$$U_1 - U_1' = U_2 \qquad 5.85$$

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

$$r_{A1} = \frac{\rho c}{S_1} = \frac{p_1}{U_1} \qquad 5.86$$

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

$$r_{A2} = \frac{\rho c}{S_2} = \frac{p_2}{U_2} \tag{5.87}$$

where S_2 = cross-sectional area of the second pipe, in square centimeters.

Expressing equation 5.85 in terms of pressure

$$p_1 S_1 - p_1' S_1 = p_2 S_2$$

or

$$p_1 - p_1' = \frac{p_2 S_2}{S_1} \tag{5.88}$$

Eliminating p_1' from equations 5.84 and 5.88,

$$p_2 = \frac{p_1(2S_1)}{S_1 + S_2} = \frac{2p_1}{1 + \frac{S_2}{S_1}} \tag{5.89}$$

Expressing in terms of volume current,

$$U_2 = \frac{2U_1}{1 + \frac{S_1}{S_2}} \tag{5.90}$$

Equations 5.89 and 5.90 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

$$p_1' = \left(\frac{S_1 - S_2}{S_1 + S_2}\right) p_1 \tag{5.91}$$

The reflected volume current in terms of the incident volume current is

$$U_1' = \frac{(S_1 - S_2)}{S_1 + S_2} U_1 \tag{5.92}$$

Equations 5.91 and 5.92 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

$$P_{12} = \frac{4r_{A1}\, r_{A2}}{(r_{A1} + r_{A2})^2} \tag{5.93}$$

5.30. Transmission Through Three Pipes.[31] — Consider three pipes of cross sections S_1, S_2 and S_3 as shown in Fig. 5.15. 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.

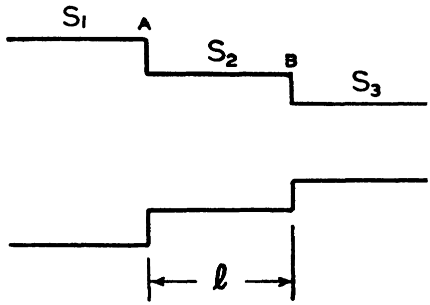

FIG. 5.15. Three pipes of cross-sectional areas S_1, S_2 and S_3. The pipe S_2 is of finite length l.

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

$$p_1 + p_1' = p_2 + p_2' \tag{5.94}$$

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

$$U_1 - U_1' = U_2 - U_2' \tag{5.95}$$

where U_1 = incident volume current in S_1, in cubic centimeters per second,
U_1' = reflected volume current in S_1, in 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.

At the boundary B the conditions for the pressure may be written

$$p_2\epsilon^{-jkl} + p_2'\epsilon^{jkl} = p_3 \tag{5.96}$$

[31] Stewart and Lindsay, "Acoustics," D. Van Nostrand Company, New York, N. Y., 1930.

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,
$k = \dfrac{2\pi}{\lambda}$,
λ = wavelength, in centimeters.

At the boundary B the conditions for the volume current may be written

$$U_2\epsilon^{-jkl} - U_2'\epsilon^{jkl} = U_3 \qquad 5.97$$

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 5.94, 5.95, 5.96 and 5.97,

$$p_1 = \frac{p_3}{2}\left[\left(\frac{S_3}{S_1} + 1\right)\cos kl + j\left(\frac{S_2}{S_1} + \frac{S_3}{S_2}\right)\sin kl\right] \qquad 5.98$$

The ratio of the power transmitted in S_3 to the incident flow of power in S_1 is

$$P_{13} = \frac{4\dfrac{S_3}{S_1}}{\left(\dfrac{S_3}{S_1} + 1\right)^2 \cos^2 kl + \left(\dfrac{S_2}{S_1} + \dfrac{S_3}{S_2}\right)^2 \sin^2 kl} \qquad 5.99$$

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

$$P_{13} = \frac{4\dfrac{S_3}{S_1}}{\left(\dfrac{S_2}{S_1} + \dfrac{S_3}{S_2}\right)^2} = \frac{4S_3S_2}{(S_2^2 + S_1S_3)^2} \qquad 5.100$$

Equation 5.95 shows that $P = 1$ if $S_2^2 = S_1S_3$. That is, if $\sin kl = \pm 1$ and providing the area of S_2 is a geometric mean S_1 and S_3, the transmission is unity.

5.31. Transmission from One Medium to Another Medium.[32] — The problem of transmission from one medium to another medium as shown in Fig. 5.16 is the same as the problem transmission from one pipe to another pipe of different cross-sectional area. The boundary between the two media is assumed to be plane and parallel to the wave front 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. 5.16 is

$$P_{12} = \frac{4r_{A1}\,r_{A2}}{(r_{A1} + r_{A2})^2} \qquad 5.101$$

where r_{A1} = acoustical resistance of medium 1, and
r_{A2} = acoustical resistance of medium 2.

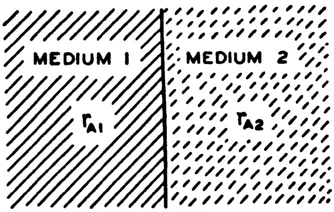

FIG. 5.16. Two media of acoustic resistances r_{A1} and r_{A2}.

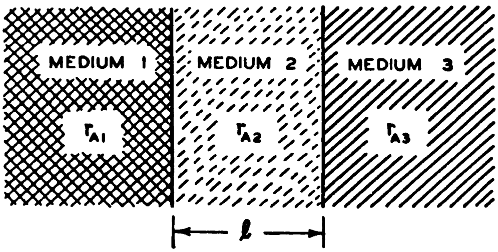

FIG. 5.17. Three media of acoustic resistances r_{A1}, r_{A2} and r_{A3}. The length of the medium 2 is l.

5.32. Transmission Through Three Media.[33] — The problem of transmission through the three media of Fig. 5.17 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

$$P_{13} = \frac{4\dfrac{r_{A1}}{r_{A3}}}{\left(\dfrac{r_{A1}}{r_{A3}} + 1\right)^2 \cos^2 kl + \left(\dfrac{r_{A1}}{r_{A2}} + \dfrac{r_{A2}}{r_{A3}}\right)^2 \sin^2 kl} \qquad 5.102$$

where r_{A1} = acoustical resistance of the medium 1,
r_{A2} = acoustical resistance of the medium 2, between 1 and 3,

[32] Stewart and Lindsay, "Acoustics," D. Van Nostrand Company, New York, N. Y., 1930.

[33] Stewart and Lindsay, "Acoustics," D. Van Nostrand Company, New York, N. Y., 1930.

r_{A3} = acoustical resistance of the medium 3,
l = length of the medium 2, in centimeters,
$k = \dfrac{2\pi}{\lambda}$
λ = wavelength in the medium 2, in centimeters.

5.33. 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 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,[34]

$$\frac{\text{db}}{\text{ft}} = 12.6\alpha^{1.4}\frac{P}{A} \qquad 5.103$$

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 5.103 holds for square ducts and rectangular ducts in which the ratio between the two sides is not greater than two.

The general subject [35, 36] of tubes lined with absorbing material, with both rigid and vibratile walls, has been considered theoretically and experimentally.

5.34. 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. 5.18. The resonant frequency, in cycles per second, is given by

$$f_r = \frac{1}{2\pi\sqrt{LC_E}} \qquad 5.104$$

[34] Sabine, H. J., *Jour. Acous. Soc. Amer.*, Vol. 12, No. 1, p. 53, 1940.
[35] Sivian, L. J., *Jour. Acous. Soc. Amer.*, Vol. 9, No. 2, p. 135, 1937.
[36] Molloy, C. T., *Jour. Acous. Soc. Amer.*, Vol. 16. No. 1, p. 31, 1944.

where L = inductance, in henries, and
C_E = electrical capacitance, in farads.

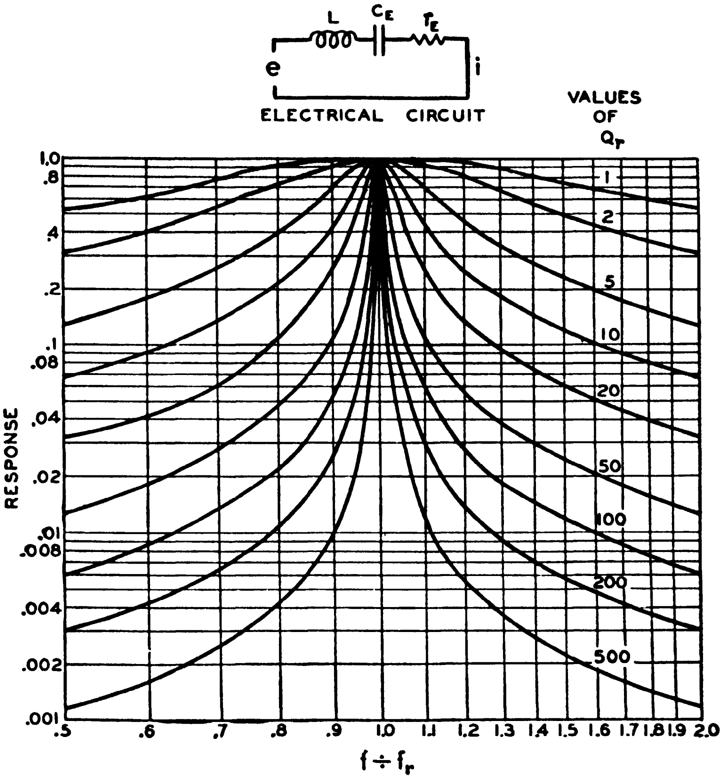

FIG. 5.18. The current response characteristics of a simple series circuit as a function of the ratio $f \div f_r$, where f_r = the resonant frequency, and f = the frequency under consideration. The numbers on the characteristics refer to the value of $\mathcal{Q}_r$, $\mathcal{Q}_r = \dfrac{2\pi f_r L}{r_E}$. 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 in the circuit is given by

$$i = \frac{e}{r_E + j\left(\omega L + \dfrac{1}{j\omega C_E}\right)} \qquad 5.105$$

where r_E = electrical resistance, in ohms,
e = driving voltage, in volts, and
i = current, in amperes.

The quantity $\mathcal{Q}_r$ is given by

$$\mathcal{Q}_r = \frac{\omega_r L}{r_E} \qquad 5.106$$

where $\omega_r = 2\pi f_r$.

The current response characteristics as function of the ratio $f \div f_r$ for various values of $\mathcal{Q}_r$ are shown in Fig. 5.18.

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 Chap. IV).

CHAPTER VI

DIRECT RADIATOR LOUD SPEAKERS

6.1 Introduction. — A loud speaker is an electroacoustic transducer designed to radiate acoustical energy into a room or open air. There are two general types of loud speakers in use today, namely: the direct radiator and the horn type loud speaker. The diaphragm of the direct radiator loud speaker is coupled directly to the air. The diaphragm of the horn loud speaker is coupled to the air by means of a horn. The direct radiator loud speaker will be considered in this chapter and the horn loud speaker will be considered in the following chapter.

The almost universal use of the direct radiator loud speaker 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 loud speaker. 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 loud speaker 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 loud-speaker 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 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 loud speaker, 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.

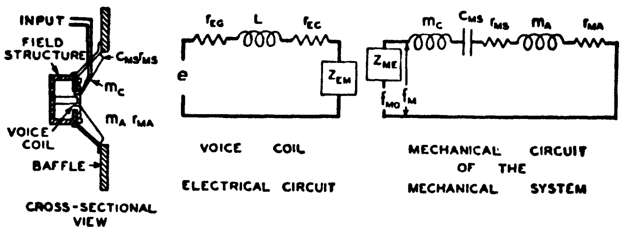

FIG. 6.1. Cross-sectional view of a single-coil, single-cone, direct radiator, dynamic, loud-speaker mechanism mounted in a baffle. In the voice coil circuit: e = the internal voltage of the generator. r_{EG} = the internal electrical resistance of the generator. r_{EC} and L = the electrical resi tance 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_{MO} = the mechanomotive force of the mechanical generator.

6.2. Single-Coil, Single-Cone Loud Speaker.—The simple dynamic loud speaker 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 loud speaker are shown in Fig. 6.1. The 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

$$z_{MT} = r_{MS} + r_{MA} + jx_{MC} + jx_{MA} - jx_{MS} \tag{6.2}$$

where r_{MS} = mechanical resistance of the supension system, in mechanical ohms,

r_{MA} = mechanical resistance of the air load, in mechancial 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. 5.8 and Fig. 5.2.

The motional electrical impedance,[1] in abohms, of the mechanical system is

$$z_{EM} = \frac{(Bl)^2}{z_{MT}} \tag{6.3}$$

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 loud speaker 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,

$$\mu = \frac{r_{ER}}{r_{EC} + r_{EM}} \times 100 \tag{6.4}$$

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.

[1] Olson, "Dynamical Analogies," D. Van Nostrand Company, New York, N. Y., 1943.

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

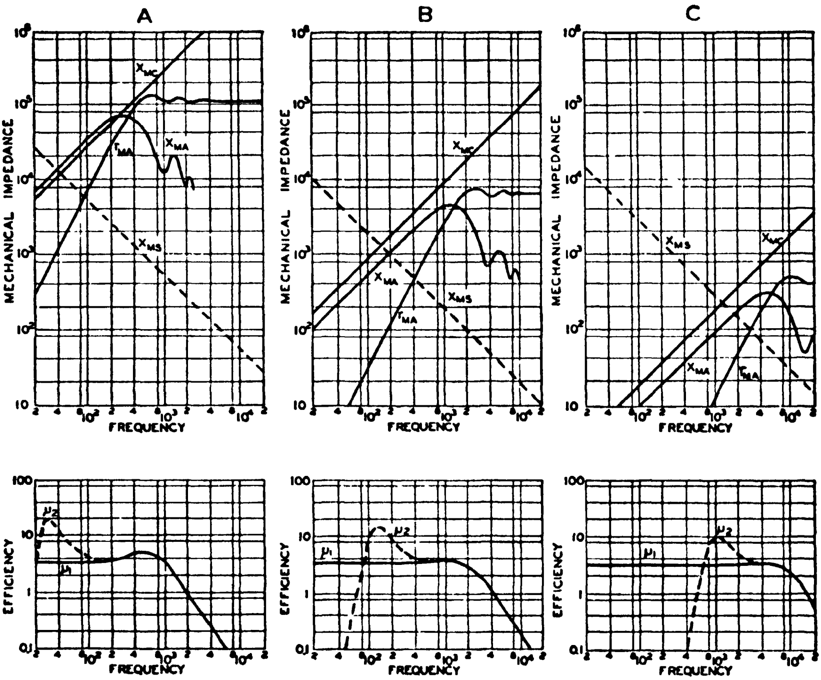

Fig. 6.2. The mechanical impedance frequency characteristics of three direct radiator loud speakers having 1-inch, 4-inch and 16-inch diameter cones. x_{MC} = the mechanical reactance due to the cone and coil. x_{MS} = the mechanical reactance due to the suspension system. x_{MA} = the mechanical reactance 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.

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 loud speaker is

$$\mu = \frac{(Bl)^2 r_{MA}}{(Bl)^2(r_{MS}+r_{MA})+r_{EC}[(r_{MC}+r_{MA})^2+(x_{MA}+x_{MC}-x_{MS})^2]} \times 100 \qquad 6.5$$

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

$$\mu = \frac{(Bl)^2 r_{MA}}{r_{EC}(x_{MA} + x_{MC})^2} \times 100 \tag{6.6}$$

In terms of the resistivity and density of the voice coil, equation 6.6 becomes

$$\mu = \frac{B^2 r_{MA} m_1}{\rho K_r (x_{MA} + x_{MC})^2 10^3} \times 100 \tag{6.7}$$

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.

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.

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 loud speakers 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

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; TEMPERATURF, 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
Bismuth	9 7	119 0	116.0
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
Mercury	13.5	95.7	1290.0

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 loud speaker 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 loud speaker is given by equation 6.3. The normal electrical impedance, in abohms, of voice coil is given by

$$z_{EN} = z_{EM} + z_{ED} \qquad 6.8$$

where z_{EM} = motional electrical impedance, in abohms, and
z_{ED} = electrical impedance of the voice coil in the absence of motion, that is blocked, in abohms.

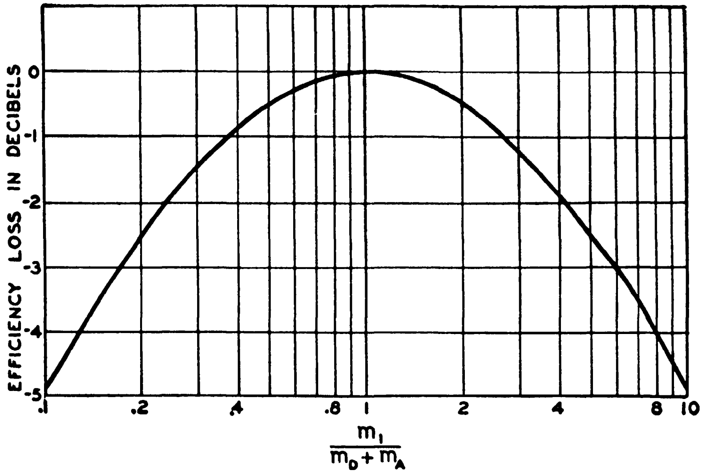

FIG. 6.3. The efficiency loss in a direct radiator loud speaker as a function of the ratio $\frac{m_1}{m_D + m_A}$, where m_1 = the mass of the voice coil, m_D = the mass of the diaphragm, m_A = the mass of the air load. The maximum efficiency is arbitrarily depicted as 0 db.

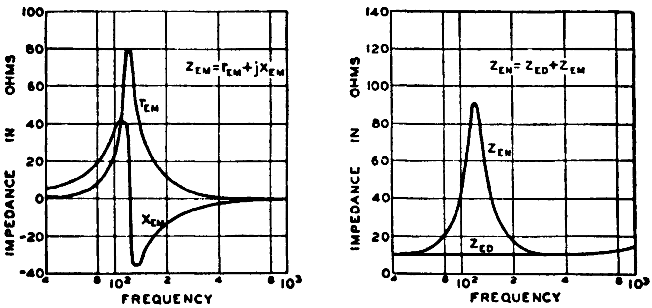

FIG. 6.4. The electrical impedance characteristics of the voice coil in a direct radiator loud speaker. z_{EN} = the normal electrical impedance. z_{ED} = the damped electrical impedance. z_{EM} = the motional electrical imedpance. r_{EM} = the resistive component of the motional electrical impedance. x_{EM} = the reactive component of the motional electrical impedance.

The components of the motional electrical impedance are shown in Fig. 6.4. 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 driving force,[2] in dynes, applied to the mechanical system is

$$f_M = Bli \qquad 6.9$$

where B = flux density in the air gap, in gausses,
l = length of the conductor, in centimeters, and
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.

The mechanical impedance,[3] in mechanical ohms, due to the electrical circuit is

$$z_{ME} = \frac{(Bl)^2}{z_{ET}} \qquad 6.10$$

where $z_{ET} = r_{EC} + j\omega L + r_{EG}$,
r_{EC} = damped electrical resistance of the voice coil, in abohms,
L = damped inductance of the voice coil, in abhenries, and
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 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

$$f_{M0} = f_M + \frac{f_M z_{ME}}{z_{MT}} \qquad 6.11$$

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 loud speaker at the higher frequencies. The electrical impedance characteristics of the vacuum tube

[2] Olson, " Dynamical Analogies," D. Van Nostrand Company, New York, N. Y., 1943.

[3] Olson, "Dynamical Analogies," D. Van Nostrand Company, New York, N. Y., 1943.

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. 6.5. In the case of a large, heavy voice coil

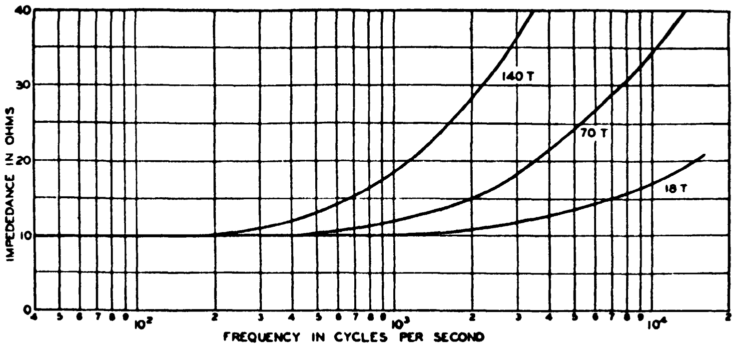

FIG. 6.5. The electrical impedance characteristics of 1½-inch diameter voice coils of 140, 70 and 18 turns and all having 10 ohms d-c resistance.

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 loud speaker 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 loud speaker 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 loud speaker. Measured directional characteristics of direct radiator loud speakers, having the constants given in Fig. 6.2, are shown in Figs. 6.6 and 6.7. Employing the mechanical circuit and the electrical circuit of Fig. 6.1 and the data of Fig. 6.2, the total output of the loud speaker 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.

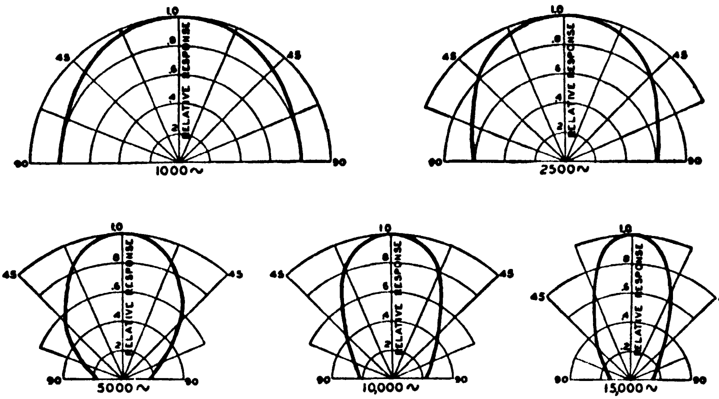

Fig. 6.6. Directional characteristics of a dynamic, direct radiator loud speaker, with a 110° cone 4 inches in diameter, mounted in a large baffle.

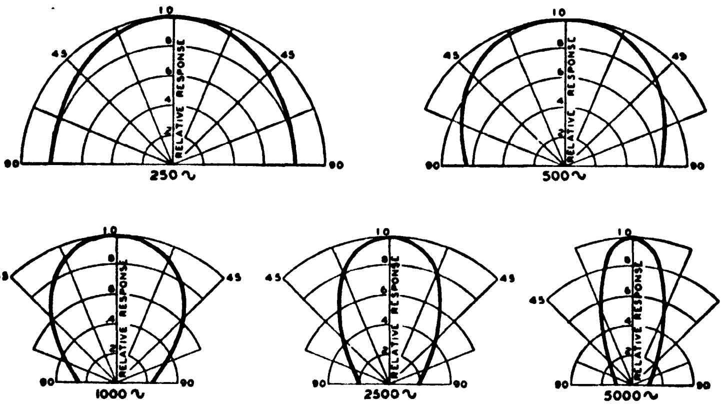

Fig. 6.7. Directional characteristics of a dynamic, direct radiator loud speaker with a 110° cone 16 inches in diameter, mounted in a large baffle.

The power output of a loud speaker may be obtained from the directional pattern and the response frequency characteristic by considering the sound flow through a spherical surface in which the loud speaker is located at the center (see Sec. 10.3*D*1). 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

$$P = \frac{10^{-7}}{\rho c} \int \int p^2 dS \qquad 6.12$$

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.

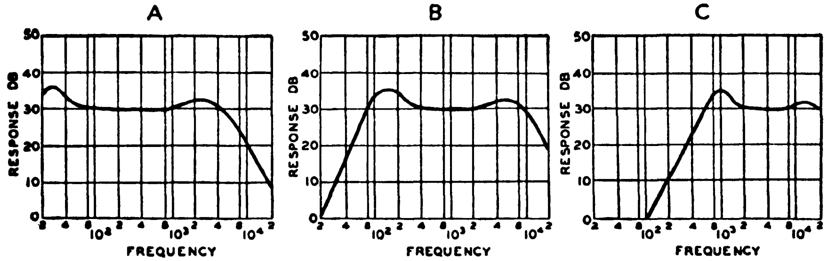

FIG. 6.8. Pressure response frequency characteristics of the loud speakers of Fig. 6.2 having cone diameters of 1 inch, 4 inches and 16 inches.

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. 6.6 and 6.7. From these data, the pressure on the axis may be determined from equation 6.12. The computed response frequency characteristics of the loud speakers of Fig. 6.2 are shown in Fig. 6.8. 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

$$P = (r_{MA}\dot{x}^2)10^{-7} \qquad 6.13$$

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 loud speaker in the region where 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.

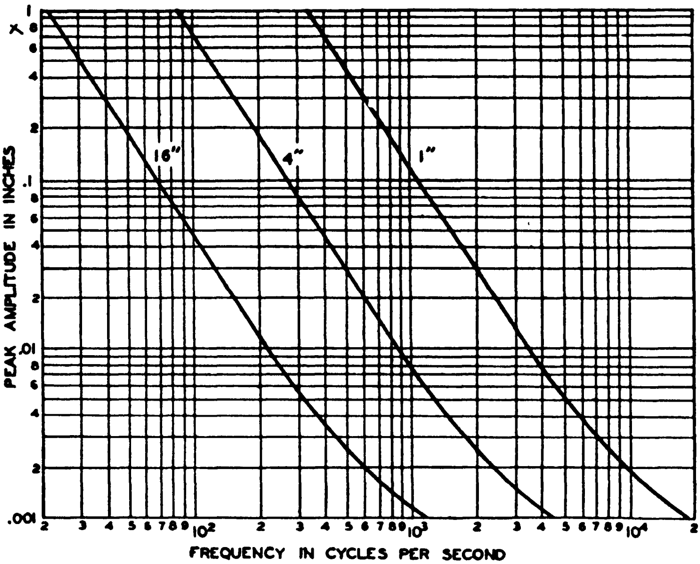

FIG. 6.9. The amplitude frequency characteristics of vibrating pistons of various diameters, mounted in an infinite wall, for 1-watt output on one side.

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. 6.9. 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 Figs. 6.6 and 6.7. It will be seen 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. 6.10. 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

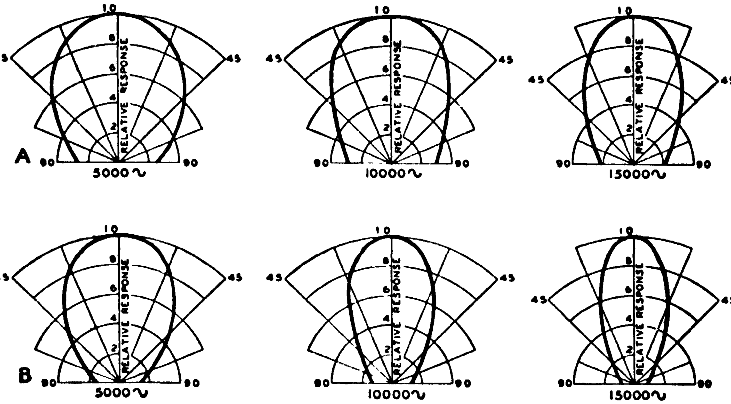

FIG. 6.10. Directional characteristics of dynamic direct radiator loud speakers with cones 4 inches in diameter for two different cone angles. Row *A*. 130° cone. Row *B*. 100° cone.

be broadest for the cone with the widest angle. The preceding observations with regard to cone type vibrators may be substantiated by theoretical considerations as outlined in Sec. 2.16.

The characteristics of Figs. 6.2 and 6.8 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 single-cone single-coil loud speaker. 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

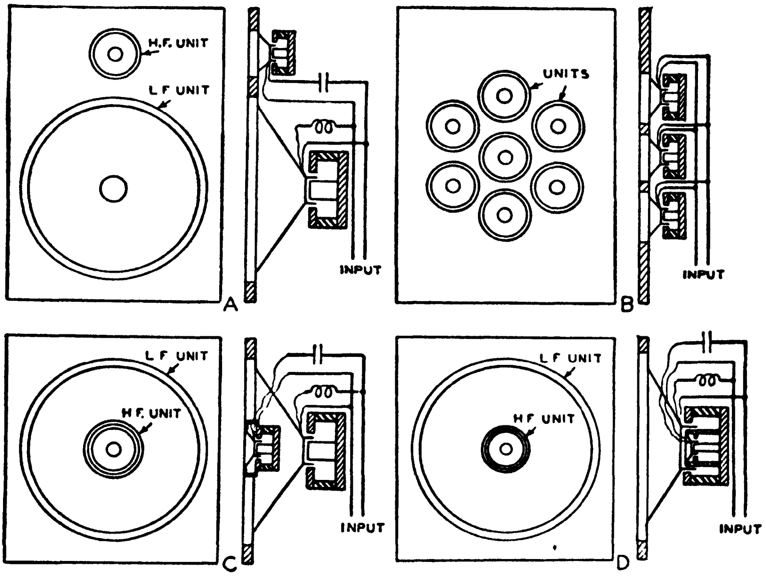

Fig. 6.11. Multiple single-cone, single-coil, direct radiator, dynamic, loud-speaker systems. *A*, *C* and *D*. Large low-frequency unit, small high-frequency unit and filter system. *B*. Seven small units connected in parallel.

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 loud-speaker 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 Loud Speaker.—Several arrangements for obtaining uniform response, broad directional pattern, adequate power handling capacity and tolerable efficiency are shown in Fig. 6.11.

The systems of Fig. 6.11*A*, *C* and *D* consist of a large diameter heavy cone driven by large voice coil for the low-frequency range and a small 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. 6.5, 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. 6.11*A* 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. 6.11*C* but is not eliminated. However, a disadvantage of the system of Fig. 6.11*C* 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. 6.11*A* and *C* referred to above have been eliminated in Fig. 6.11*D*. In this system[4] the large cone is geometrically a continuation of the small cone. Therefore, in the overlap region the two cones vibrate together as a single cone. In this way phase and diffraction effects are eliminated.

Referring to Figs. 6.2 and 6.8 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. 6.9. An arrangement of seven small loud-speaker units mounted in a flat baffle with the voice coils connected in parallel is shown in Fig. 6.11*B*. The voice coils of the loud speakers 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. 2.15).

[4] Olson and Preston, *RCA Review*, Vol. 7, No. 2, p. 155, 1946.

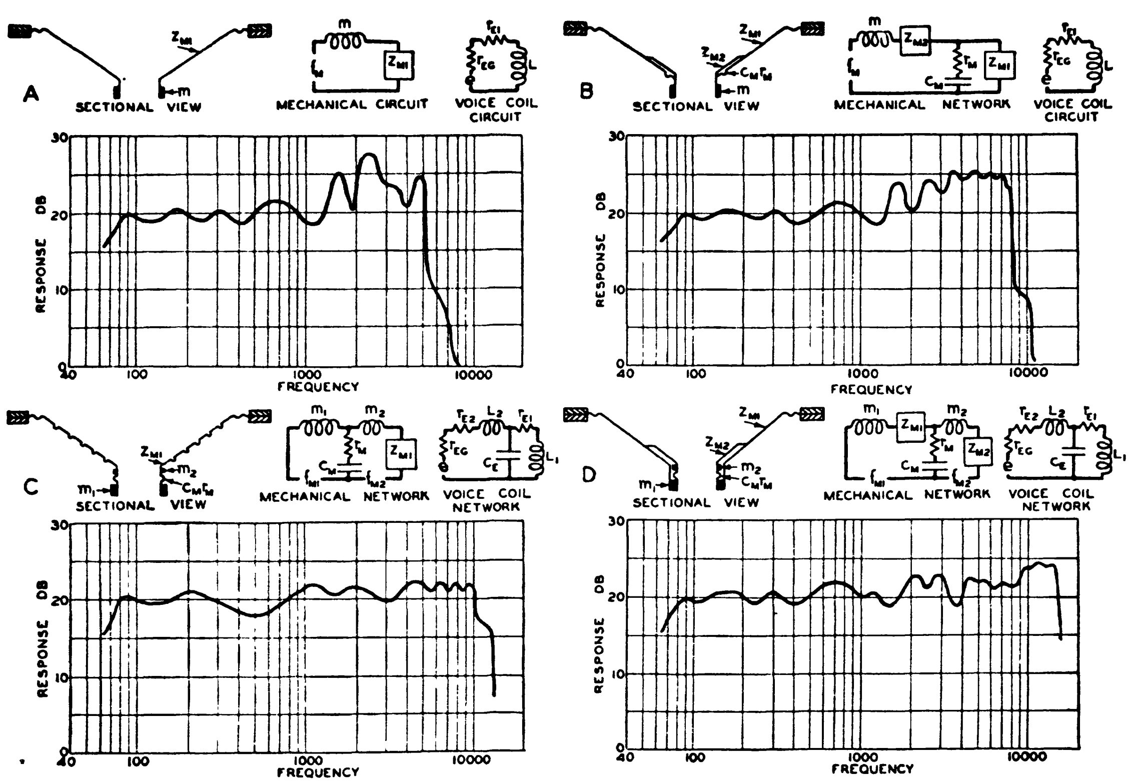

FIG. 6.12. *A*. Cross-sectional view of a single-cone, single-coil loud speaker with the voice coil electrical circuit and the mechanical circuit of the mechanical system. In the voice coil electrical circuit: e = the internal voltage of the generator. r_{EG} = the internal electrical resistance of the 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 loud speaker 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 loud speaker 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 vew of a double-cone, double-coil loud speaker 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} = the force generated in the small coil. The graph shows the pressure response frequency characteristic.

The frequency range of a direct radiator loud speaker 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 Loud Speaker.[5] — A typical single-coil, double-cone loud speaker, Fig. 6.12*B*, 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 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 mechani-

[5] Olson, H. F., *Jour. Acous. Soc. Amer.*, Vol. 10, No. 4, p. 305, 1939.

cal 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 loud speaker is shown in Fig. 6.12*A*. The voice coil and large cone of Fig. 6.12*B* is the same as that of Fig. 6.12*A*. 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 Loud Speaker.[6] — The double-coil, single-cone loud speaker, Fig. 6.12*C*, 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 complicance, 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 loud speaker. 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 loud speaker is shown in Fig. 6.12*C*.

[6] Olson, H. F., *Proc. Inst. Rad. Eng.*, Vol. 22, No. 1, p. 33, 1934.

6.6. Double-Coil, Double-Cone Loud Speaker.[7] — The double-coil, double-cone loud speaker, Fig. 6.12*D*, 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 loud speaker. Both parts 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 complicance, 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, and the large cone, z_{M1}, remain stationary. This system is equivalent to two separate loud speakers. 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 loud speaker is shown in Fig. 6.12*D*. 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 handling capacity as a direct radiator loud speaker may be obtained at the low frequencies.

6.7. Mechanical Networks for Controlling the High-Frequency Response of a Loud Speaker. — 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 attentuate 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

[7] Olson, H. F., *Jour. Acous. Soc. Amer.*, Vol 10, No. 4, p. 305, 1939.

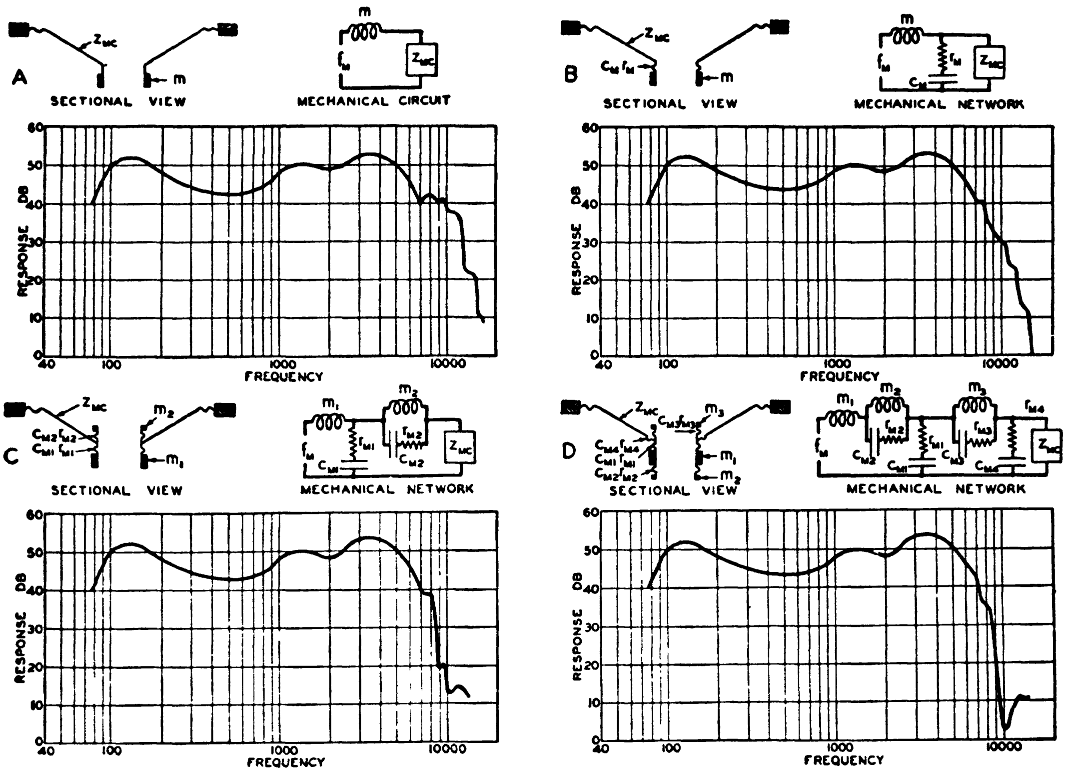

Fig. 6.13. *A*. Cross-sectional view of a conventional single-coil direct radiator, dynamic loud speaker and the mechanical circuit of the mechanical system. In the mechanical circuit: m = the mass of the voice coil. z_{MC} = the mechanical impedance of the cone and suspension system. The graph shows the pressure response frequency characteristic. *B*. Cross-sectional view of the loud speaker 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 loud speaker in *B* with a mass connected by a corrugation to the front of the cone. In the mechanical network: m_1 = the mass of the voice coil. C_{M1} and r_{M1} = the compliance and mechanical resistance of the corrugaton 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 loud speaker 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 and mechanical resistance of the corrugation connecting the mass m_2 to the coil. 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.

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 loud speaker.

A relatively light weight, 8-inch loud speaker was chosen for these tests. This type of loud speaker 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 loud speakers. The loud speaker 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 Loud Speaker.* — The response frequency characteristic of the conventional loud speaker, referred to above, is shown in Fig. 6.13*A*. The mechanical circuit of the mechanical system is also shown in Fig. 6.13*A*. 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 loud speaker is well adapted to illustrate the performance of mechanical networks for controlling the response at the higher frequencies.

B. *Loud Speaker with a Compliance Shunting the Cone Mechanical Impedance.* — One of the simplest means for attenuating the high-frequency response of a loud speaker is a compliance inserted between the voice coil and the cone. This compliance, C_M, may take the simple form of a bead or corrugation pressed into the voice coil form. The response characteristic of a conventional loud speaker with a compliance between the voice coil and cone is shown in Fig. 6.13*B*. In the mechanical network the compliance, C_M, shunts the cone mechanical impedance, z_{MC}. Comparing with Fig. 6.13*A* 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 consequence, the mechanical impedance does not increase directly with the frequency. In some loud speakers the mechanical impedance, z_{MC}, actually decreases with frequency at the higher frequencies.

C. *Loud Speaker 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 mechanical resistance in the compliance. An example of this system is shown in Fig. 6.13*C*. The mass and compliance are designated as m_2 and C_{M2}. Comparing with Fig. 6.13*A* 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. *Loud Speaker with a "T" Type Filter Connecting the Voice Coil Mass and the Cone Mechanical Impedance.* — This system, Fig. 6.13*D*, consists of two parallel resonant mechanical circuits, or a parallel resonant mechanical circuit, m_2 and C_{M2}, connected to the bottom of the voice coil of the system of Fig. 6.13*C*. The mechanical network is also shown in Fig. 6.13*D*. 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. 6.13*D*. Comparing with Fig. 6.13*A* 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 loud speaker at the high frequencies have been described. Some of these systems are in use in practically all loud speakers. 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. Loud-Speaker 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 loud speaker 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 Sec. 2.3). In this frequency range the power output for constant velocity is proportional to the fourth power of the frequency. 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 Sec 2.2). 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. 6.14*A* show "dips" in the response when the acoustical path from front to back is a wavelength. Using an irregular baffle, Fig. 6.14*B*, 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. 6.14*B*, show that the dip in the response frequency characteristic of the square baffle is eliminated by the use of an irregular baffle.

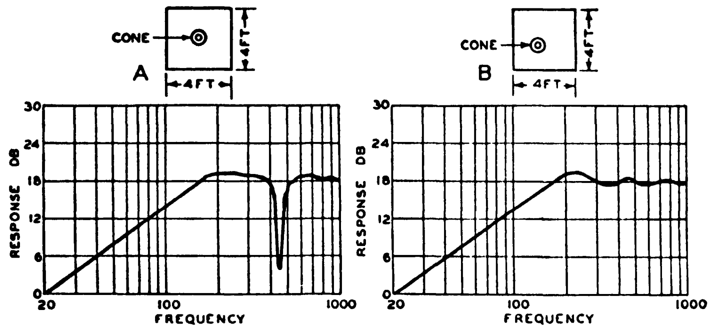

FIG. 6.14. Pressure response frequency characteristics of mass controlled, direct radiator, dynamic loud-speaker mechanisms, with 10-inch diameter cones, mounted in square baffles. In *A*, the loud-speaker mechanism is mounted in the center of the baffle. In *B*, the loud-speaker mechanism is mounted unsymmetrically to eliminate interference.

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 loud speaker will be independent of the frequency in the frequency range above the resonant frequency up to the frequency of ultimate mechanical resistance, and will be proportional to the fourth power of the frequency below the resonant frequency. The measured pressure response frequency characteristic of a direct radiator loud speaker having a fundamental resonance of 50, 100 and 200 cycles is shown in Fig. 6.15. It will be seen that the pressure 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 5.10). If the considerations are confined to the frequency range above the resonant frequency of the mechanism,

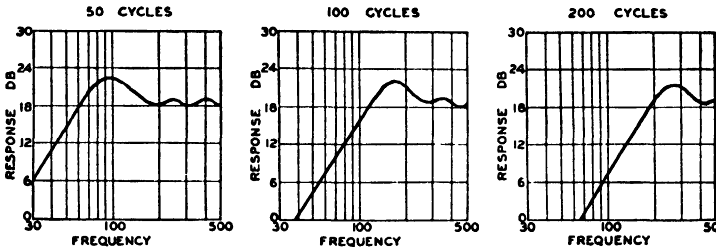

Fig. 6.15. Pressure response frequency characteristics of direct radiator, dynamic loud-speaker mechanisms, with 10-inch diameter cones, with resonant frequencies of 50, 100 and 200 cycles, mounted in very large baffles.

the velocity 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 inde-

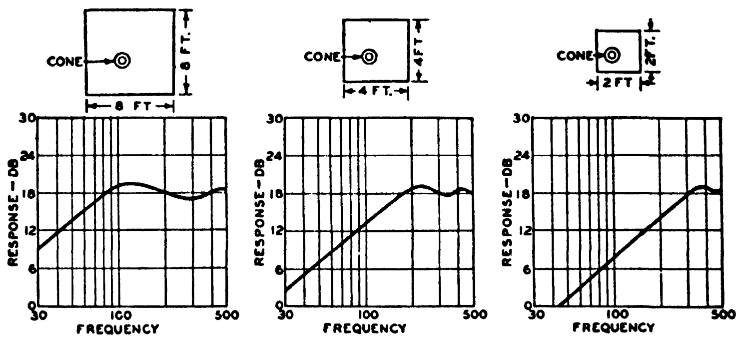

Fig. 6.16. Pressure response frequency characteristics of mass controlled, direct radiator, dynamic loud-speaker mechanisms, with 10-inch diameter cones, mounted in square baffles of 8, 4 and 2 feet on a side.

pendent of the frequency where the system behaves as a simple radiator. The experimental results of Fig. 6.16 substantiates these predictions for 2-, 4- and 8-foot baffles. Above the range where the system changes from doublet to singlet 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 loud speaker 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. 6.17 combines the loud-speaker mechanism of Fig. 6.15 with the baffles of Fig.

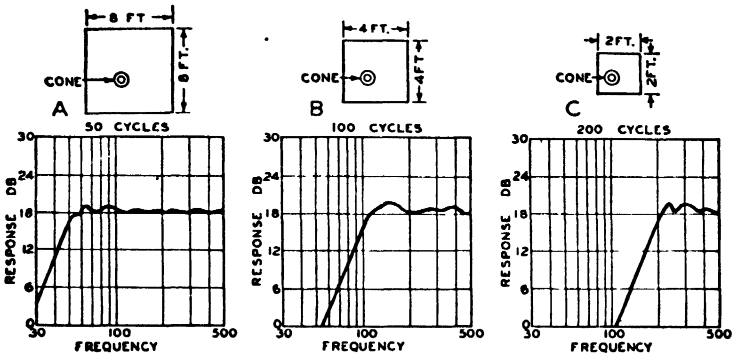

FIG. 6.17. Pressure response frequency characteristics of direct radiator, dynamic loud-speaker mechanisms, with 10-inch diameter cones, operating under the following conditions: *A*. Square baffle 8 feet on a side and a loud-speaker resonant frequency of 50 cycles. *B*. Square baffle 4 feet on a side and a loud-speaker resonant frequency of 100 cycles. *C*. Square baffle 2 feet on a side and a loud speaker with a resonant frequency of 200 cycles.

6.16. 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 Loud Speakers.—The most common housing for a direct radiator loud speaker is the conventional open-back cabinet which 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 loud speaker. 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 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.

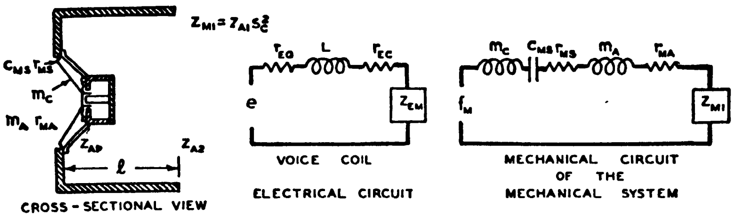

FIG. 6.18. Cross-sectional view of a single-coil, single-cone, direct radiator, dynamic loud-speaker mechanism mounted in open-back cabinet. In the voice coil circuit: e = the internal voltage of the generator. r_{EG} = the internal electrical resistance of the generator. r_{EC} and L = the electrical 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. $z_{A1} \doteq$ 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.

A cross-sectional view of a direct radiator loud speaker mounted in an open-back cabinet and the mechanical circuit of the mechanical system is shown in Fig. 6.18. The input acoustical impedance of a finite cylindrical pipe has been considered in Sec. 5.21. 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

$$z_{M1} = z_{A1}S_c^2 \qquad 6.14$$

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. 6.18 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 loud-speaker systems.

A. *Low Resonant Frequency, Different Cabinet Sizes.* — The pressure response frequency characteristics of a direct radiator loud-speaker mecha-

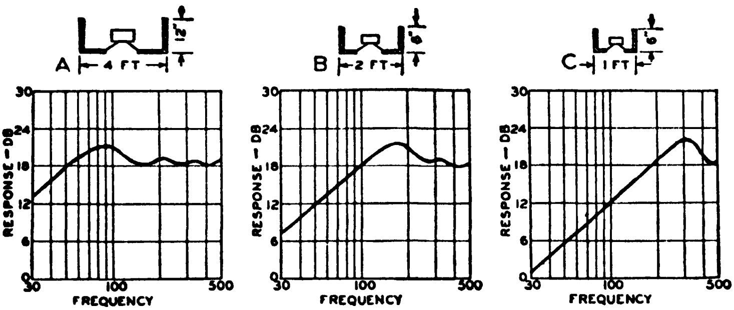

FIG. 6.19. Pressure response frequency characteristics of mass controlled, direct radiator, dynamic loud-speaker mechanisms with 10-inch diameter cones mounted in square open-back cabinets. *A*. Cabinet, 4 feet × 4 feet × 12 inches in depth. *B*. Cabinet, 2 feet × 2 feet × 8 inches in depth. *C*. Cabinet, 1 foot × 1 foot × 6 inches in depth.

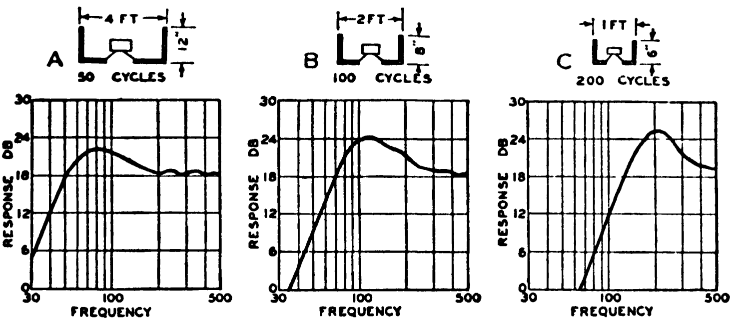

FIG. 6.20. Pressure response frequency characteristics of direct radiator, dynamic loud-speaker mechanisms, with 10-inch diameter cones, operating in open-back cabinets under the following conditions: *A*. Cabinet, 4 feet × 4 feet × 12 inches in depth and a loud-speaker resonant frequency of 50 cycles. *B*. Cabinet, 4 feet × 4 feet × 8 inches in depth and a loud-speaker resonant frequency of 100 cycles. *C*. Cabinet, 2 feet × 2 feet × 6 inches in depth and a loud-speaker resonant frequency of 200 cycles.

nism, having a resonant frequency of 20 cycles mounted in cabinets of various sizes, is shown in Fig. 6.19. The resonant frequencies at 80, 150 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.

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 frequency characteristics of combinations of various cabinets and mechanisms having different resonant frequencies are shown in Fig. 6.20. 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.

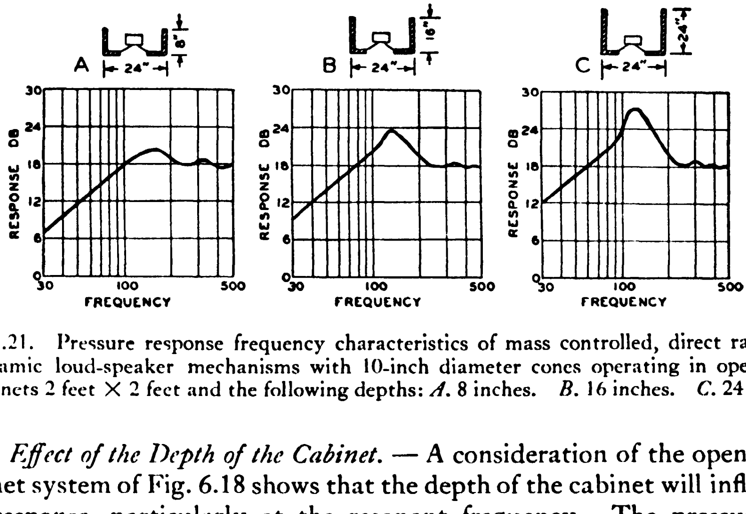

FIG. 6.21. Pressure response frequency characteristics of mass controlled, direct radiator, dynamic loud-speaker mechanisms with 10-inch diameter cones operating in open-back cabinets 2 feet × 2 feet and the following depths: *A*. 8 inches. *B*. 16 inches. *C*. 24 inches.

C. *Effect of the Depth of the Cabinet.* — A consideration of the open-back cabinet system of Fig. 6.18 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 loud-speaker mechanism mounted in 2-foot cabinets with depths of 8, 16 and 24 inches are shown in Fig. 6.21. 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 Loud Speaker. — A loud speaker mechanism with the back of the cone completely enclosed by the cabinet is shown in Fig. 6.22. At the low frequencies the system is a simple source (see 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. 6.22 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

$$C_{MB} = \frac{C_A}{S_c^2} \qquad 6.15$$

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.

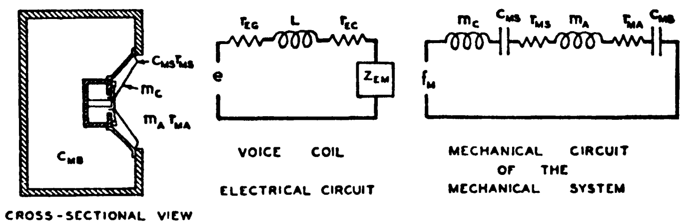

FIG. 6.22. Cross-sectional view of a single-coil, single-cone, direct radiator, dynamic loud-speaker mechanism mounted in closed-back cabinet. In the voice coil circuit: e = the internal voltage of the generator. r_{EG} = the internal electrical resistance of the generator. r_{EC} and L = the electrical 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.

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 loud-speaker mechanism having a resonant frequency of 150 cycles mounted in a 24-inch open-back cabinet is shown in Fig. 6.23*A*. The response falls off 18 db per octave below 150 cycles.

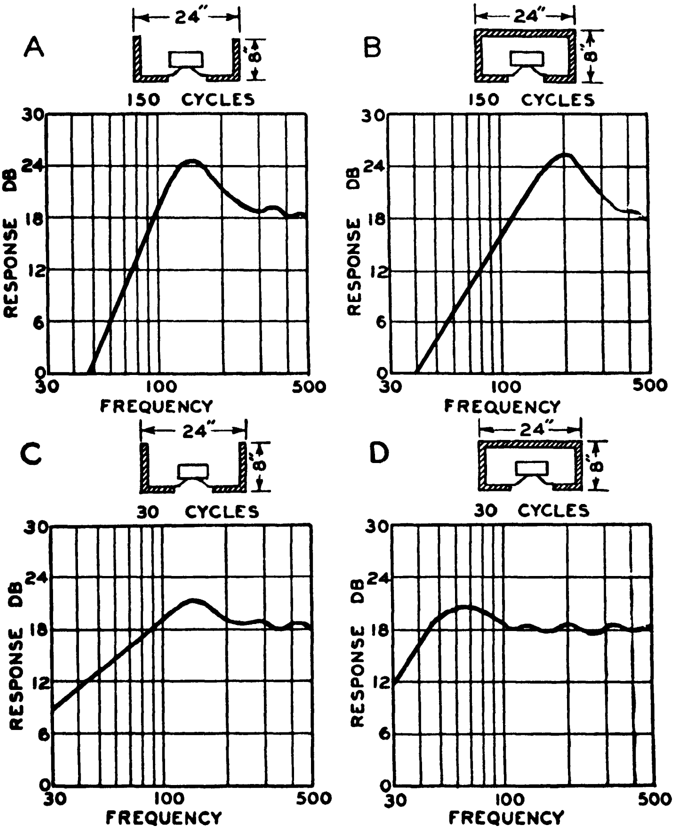

FIG. 6.23. Pressure response frequency characteristics of direct radiator, dynamic loud-speaker mechanisms with 10-inch diameter cones operating in open- and closed-back cabinets. *A*. Open-back cabinet, 2 feet × 2 feet × 8 inches in depth and a loud-speaker resonant frequency of 150 cyclés. *B*. Closed-back cabinet, 2 feet × 2 feet × 8 inches in depth and a loud-speaker resonant frequency of 150 cycles. *C*. Open-back cabinet, 2 feet × 2 feet × 8 inches in depth and a loud-speaker resonant frequency of 30 cycles. *D*. Closed-back cabinet, 2 feet × 2 feet × 8 inches in depth and a loud-speaker resonant frequency of 30 cycles.

The pressure response frequency characteristic of a loud-speaker mechanism, having a resonant frequency of 150 cycles, mounted in a completely enclosed 24-inch cabinet is shown in Fig. 6.23*B*. 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 loud-speaker mechanism having a resonance frequency of 30 cycles mounted in a 24-inch open-back cabinet is shown in Fig. 6.23*C*. In this case the response falls off 6 db per octave below the doublet-singlet transition frequency.

The pressure response frequency characteristic of a loud-speaker mechanism having a resonant frequency of 30 cycles mounted in a completely enclosed cabinet is shown in Fig. 6.23*D*. 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.

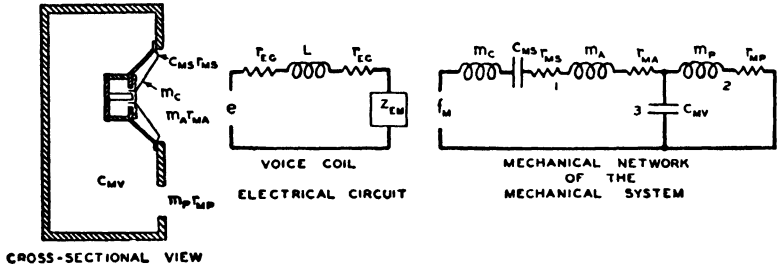

FIG. 6.24. Cross-sectional view of a single-coil, single-cone, direct radiator, dynamic loud-speaker mechanism mounted in closed-back cabinet. In the voice coil circuit: e = the internal voltage of the generator. r_{EG} = the internal electrical resistance of the generator. r_{EC} and L = the electrical 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_{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.

6.11. Acoustical Phase Inverter Loud Speaker. — The acoustical phase inverter loud-speaker[8] system consists of a direct radiator loud-speaker mechanism mounted in a completely closed cabinet save for a port coupling the cabinet volume to the air, Fig. 6.24. The phase of the velocities on the two sides of the cone differs by 180°. Referring to the mechanical

[8] Dickey, Caulton and Perry, *Radio Engineering*, Vol. 16, No. 10, p. 8, 1936.

network of Fig. 6.24, 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 branch 3. The phase angle will be reduced as mechanical resistance is introduced. However, the mechanical resistance in direct radiator loud-speaker 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

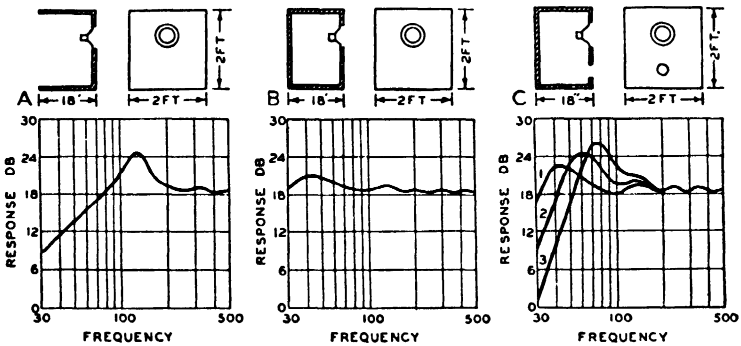

FIG. 6.25. Pressure response frequency characteristics of a direct radiator, dynamic loud-speaker mechanism with a 10-inch diameter cone and a resonant frequency of 30 cycles operating under the following conditions: *A*. Open cabinet, 2 feet × 2 feet × 18 inches in depth. *B*. Closed cabinet, 2 feet × 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.

mechanical resistance of a direct radiator loud speaker at the low frequencies. The pressure response frequency characteristic of a loud-speaker mechanism mounted in an open-back cabinet is shown in Fig. 6.25*A*. The pressure response frequency characteristic for the same mechanism mounted in a closed cabinet of the same dimensions is shown in Fig. 6.25*B*. 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. 6.25*C*. The low-frequency range is extended, the output is increased and cabinet resonance is eliminated by the phase inverter system.

6.12. Acoustical Labyrinth Loud Speaker.[9] — The acoustical labyrinth loud speaker consists of an absorbent walled conduit with one end tightly coupled to the back of the cone of a direct loud-speaker mechanism and the

[9] Olney, Benj., *Jour. Acous. Soc. Amer.*, Vol. 8, No. 2, p. 104, 1936.

other end opening in front or at the bottom of the cabinet within which it is folded (Fig. 6.26). 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 at the front of the cone. The radiation, then, 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

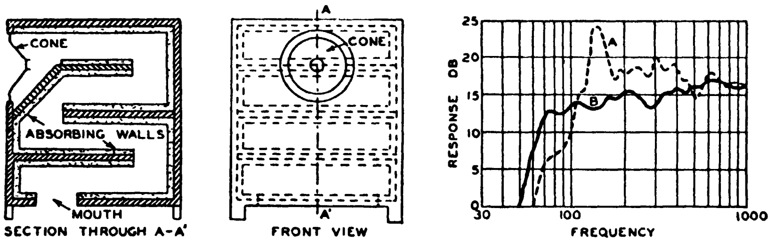

FIG. 6.26. Acoustical labyrinth loud speaker. The pressure response frequency characteristic of an acoustical labyrinth loud speaker is labeled *B* on the graph. The pressure response frequency characteristic of the corresponding open-back cabinet loud speaker is labeled *A* on the graph. (After Olney.)

suspension system upon the response may be eliminated by choosing the constants so that fundamental resonance of the loud speaker coincides with the quarter wavelength anti-resonance of the tube. The pressure response characteristic of a direct radiator loud speaker with and without a labyrinth is shown in Fig. 6.26. These characteristics show that the accentuated response due to cabinet resonance has been eliminated and that the low-frequency range has been extended.

6.13. Combination Horn and Direct Radiator Loud Speaker.[10] — One form of the combination horn and direct radiator loud speaker consists of a horn coupled to the back side of a direct radiator loud-speaker 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. 6.27).

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

[10] Olson and Hackley, *Proc. Inst. Rad. Eng.*, Vol. 24, No. 12, p. 1557, 1936.

horn. Therefore, the cone is coupled directly to the horn in this frequency range. In the system shown in Fig. 6.27 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 loud speaker. The use of a horn as a coupling means makes it possible to obtain large power

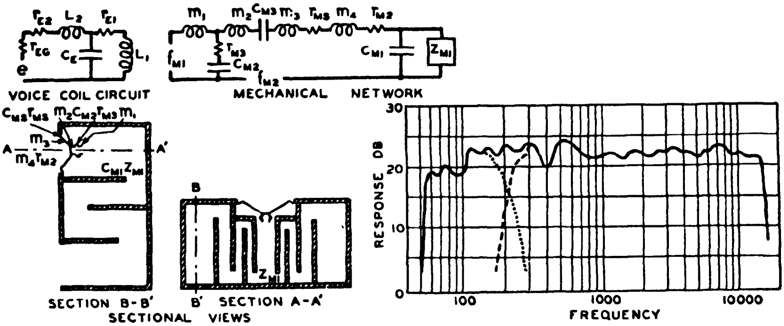

FIG. 6.27. Sectional views of the combination horn and double-voice coil, direct radiator, loud speaker. In the voice coil circuit: e = the internal voltage of the generator. r_{EG} = the internal electrical resistance of the generator. r_{E1} and L_1 = the electrical resistance and inductance of the large coil. r_{E2} and L_2 = the electrical resistance and inductance of the small coil. In the mechanicaln etwork: 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. C_{M3} = the 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 loud speaker. The overlap between the horn and direct radiator action is shown by the dotted and dashed characteristics. (After Olson and Hackley.)

outputs from a small diameter cone. In addition, the combination of a horn and a direct radiator loud-speaker 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 loud speaker with double voice coil driving system is shown in Fig. 6.27.

Another form of the combination horn and direct radiator loud speaker [11]

[11] Lansing, J. B., *Jour. Soc. Mot. Pic. Eng.*, Vol. 46, No. 3, p. 212, 1946.

consists of a direct radiator loud speaker with a large cone for the reproduction of the low-frequency range and a small horn loud speaker for the reproduction of the high-frequency range. Two different designs of this type of loud speaker are shown in Fig. 6.28. In Fig. 6.28*A* the center pole for the low-frequency loud speaker constitutes the small portion of the horn and a flared type cone in the direct radiator loud speaker provides a con-

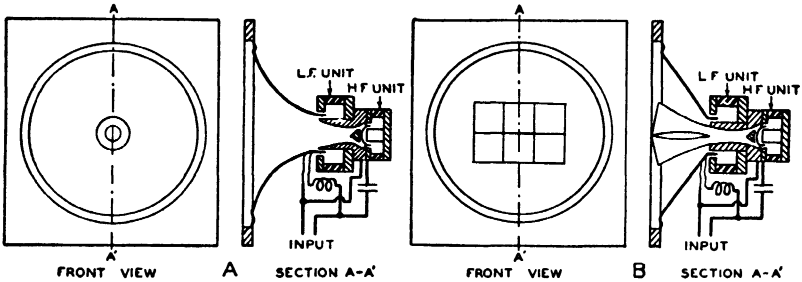

FIG. 6.28. Combination horn and direct radiator loud speakers. *A*. A direct radiator loud speaker is used for the reproduction of the low-frequency range; and a horn loud speaker, in which the pole and low-frequency cone form the horn, is used for the reproduction of the high-frequency range. *B*. A direct radiator loud speaker is used for the reproduction of the low-frequency range and a cellular horn loud speaker is used for the reproduction of the high-frequency range.

tinuation of the horn. In Fig. 6.28*B* 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.

6.14. Feedback Applied to a Loud Speaker. — 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 loud speaker. 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 loud speaker. It is not an easy proposition to employ feedback in this way because of the very special control required of phase shifts in the amplifier and loud-speaker system. Unless certain phase relations[12, 13] are maintained, oscillation will occur. Fig. 6.29 shows feedback applied to an amplifier and loud speaker. In Fig. 6.29*A* 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 loud speaker without feedback is shown in Fig. 6.29. The same loud speaker and amplifier with 15 db negative feedback from the pickup coil are also shown in Fig. 6.29. It will be

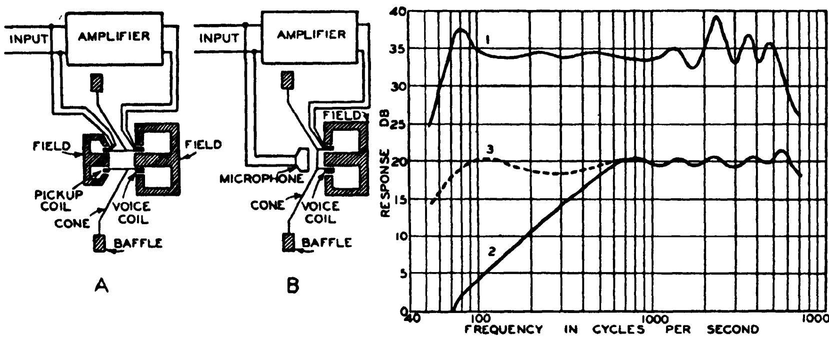

Fig. 6.29. Loud speaker and amplifier feedback systems. *A*. The output of the pickup coil is fed into the input side of the amplifier. *B*. The output of the microphone is fed into the input side of the amplifier. The graph shows the pressure response frequency characteristic of system *A*: 1. Without feedback. 2. With feedback. 3. With feedback and compensation.

seen that the response at the high frequencies is improved. This system tends to drive the cone at constant velocity for constant signal voltage 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, loud speaker and microphone is shown in Fig. 6.29*B*. If a pressure operated microphone having uniform sensitivity with respect to frequency is used the response characteristic of the loud speaker will become more uniform as the amount of feedback is increased.

6.15. Transient Response.[14] — The subject of transient response embraces a wide variety of physical phenomena. Electrical transients con-

[12] Nyquist, H., *Bell Syst. Tech. Jour.*, Vol. 11, No. 1, p. 126, 1932.
[13] Black, H. C., *Bell Syst. Tech. Jour.*, Vol. 13, No. 1, p. 1, 1934.
[14] Olson, "Dynamical Analogies," D. Van Nostrand Company, New York, N. Y., 1943.

cern 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 loud speaker 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 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 loud speaker to a suddenly applied unit force.

The mechanical circuit of the dynamic loud speaker at the low frequencies is shown in Fig. 6.1. The differential equation for the system of 6.1 is

$$m\ddot{x} + r_{MT}\dot{x} + \frac{x}{C_M} = f_{MO} \qquad 6.17$$

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

$$r_{MT} = r_{MS} + r_{MR} + r_{ME} \qquad 6.18$$

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

$$z_{ME} = r_{ME} = \frac{(Bl)^2}{r_{ET}} \qquad 6.19$$

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 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 [15, 16, 17] 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.

[15] Carson, "Electric Circuit Theory and Operational Calculus," McGraw-Hill Book Company, New York, N. Y., 1926.
[16] Bush, "Operational Circuit Analysis," John Wiley and Sons, New York, N Y., 1937.
[17] Berg, "Heaviside's Operational Calculus," McGraw-Hill Book Company, New York, N. Y., 1936.

Let

$$\alpha = \frac{r_{MT}}{2m}$$

$$\omega = \sqrt{\frac{1}{mC_M} - \alpha^2}$$

The indicial mechanical admittance may be written

$$A(t) = \frac{1}{m\omega} \frac{p\omega}{(p + \alpha)^2 + \omega^2} \tag{6.21}$$

From tables of operational formulas, the solution is

$$A(t) = \frac{1}{m\omega} \epsilon^{-\alpha t} \sin \omega t \tag{6.22}$$

Fig. 6.30 shows the effect of the electrical impedance of the vacuum tube upon the transient response of a loud speaker. In this case the loud speaker 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 loud speaker 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 loud speaker is shown by the uppermost left-hand graph of Fig. 6.31. 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 loud speaker operates from a high electrical impedance vacuum tube amplifier, the internal mechanical resistance of the loud speaker is the major factor influencing the transient response. Fig. 6.31 shows response of a 12-inch (10-inch diameter cone) loud speaker 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 loud speaker coupled to a generator with very high internal electrical impedance. For this type of operation it is customary to provide a large mechanical resistance, r_{MS}, the second and third conditions of Fig. 6.31.

Figs. 6.30 and 6.31 show that the " hangover " in properly designed and operated loud speakers 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 loud speakers 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.

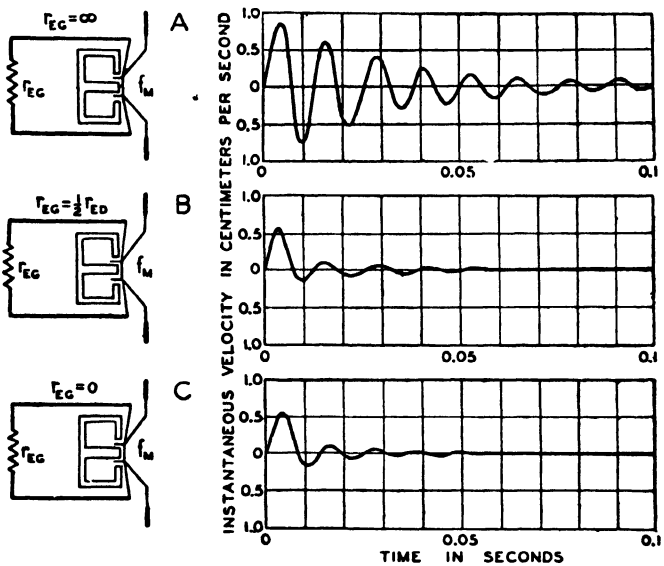

FIG. 6.30. The transient response of a direct radiator, dynamic loud speaker, with a 12-inch diameter cone, to a unit force for various types of electrical generators. *A*. Generator of very high electrical resistance. *B*. Generator having an electrical resistance of one half of the loud speaker electrical impedance. *C*. Generator of zero electrical impedance.

6.16. Distortion. — The general trend in all types of radio receivers and phonographs is more output without a corresponding increase in the size of the loud speaker. As a result, the maximum amplitude of the loud speaker is also increased. Many apparently peculiar activities are manifested by the loud speaker when the amplitude or excursion of the cone is large. 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 loud-speaker system.

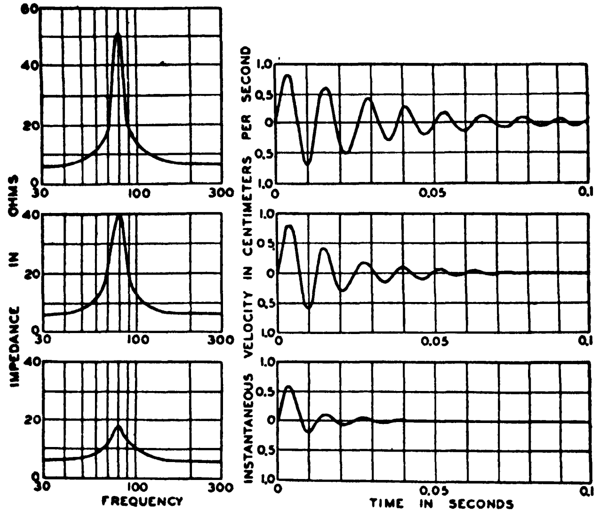

Fig. 6.31. The transient response of a direct radiator, dynamic loud speaker, with a 12-inch diameter cone, for different values of the suspension mechanical resistance. The electrical impedance frequency characteristic indicates the degree of internal damping.

A. *Nonlinear Suspension System.*[18] — The force displacement characteristic of a typical, direct radiator, loud-speaker, cone suspension system is shown in Fig. 6.32. It will be seen that for small amplitudes the suspension system is linear. However, for large amplitudes the suspension system is nonlinear.

The force deflection characteristic of the loud-speaker cone suspension system of Fig. 6.32 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. 6.32 may be obtained

[18] Olson, H. F., *Jour. Acous. Soc. Amer.*, Vol. 16, No. 1, p. 1, 1944.

from equation 6.23 as follows:

$$C_M = \frac{x}{f_M} = \frac{1}{\alpha + \beta x^2} \qquad 6.24$$

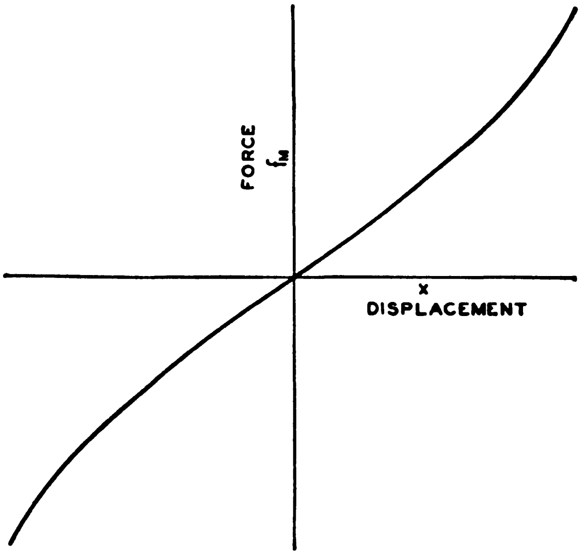

FIG. 6.32. Force displacement characteristic of the suspension system of a direct radiator loud speaker.

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:

$$m\ddot{x} + \alpha x + \beta x^3 = F \cos \omega t. \qquad 6.27$$

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 loud speaker with a nonlinear cone suspension system.

B. *Distortion Characteristics of a 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 loud speaker. The wave shape under these conditions is shown in Fig. 6.33. 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 loud speaker, 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 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. 6.34. 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 loud speakers. 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 loud speakers.

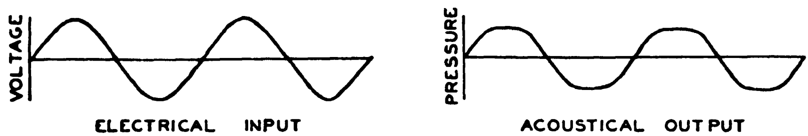

FIG. 6.33. The wave shapes of the electrical input and the sound pressure output of a loud speaker with a nonlinear suspension system.

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

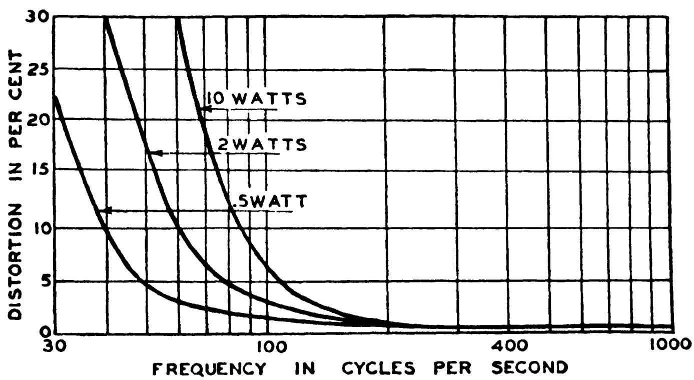

FIG. 6.34. Distortion frequency characteristics of a direct radiator, dynamic loud-speaker mechanism with a 10-inch diameter cone and a nonlinear suspension system for electrical inputs of 2, 5 and 10 watts.

also occur in nonlinear systems. Those familiar with the performance of loud speakers 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 loud speaker due to a nonlinear cone suspension system. As pointed out above, the amplitude of the cone of a direct radiator loud speaker 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.

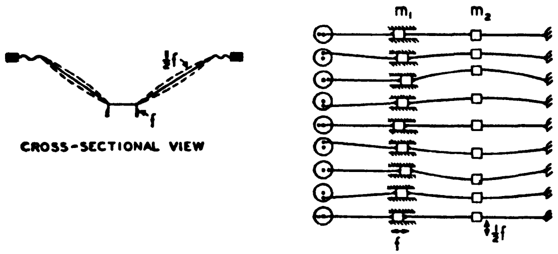

FIG. 6.35. A system consisting of a mass m_1 driven by a crank at a frequency f and a second mass m_2 supported by a spring coupled to m_1 vibrating with a frequency $\frac{1}{2}f$. The cross-sectional view of the cone shows a similar system and illustrates how subharmonics may be produced by a loud speaker.

Fig. 6.35 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. 6.35, will vibrate at a subharmonic frequency. The existence of subharmonics in direct radiator loud speakers is well known. However, in horn loud speakers the diaphragms are relatively small and quite rigid. Consequently the conditions for the production of subharmonics are not particularly favorable. Circular corrugations in the diaphragm or cone may be used to increase the stiffness and thereby reduce the tendency to break into subharmonic vibrations.

C. *Response Frequency Characteristics of a Direct Radiator Loud Speaker With a Nonlinear Suspension System.* — The velocity frequency characteristic of a loud speaker with a nonlinear suspension system may be obtained from the equation 6.29. A theoretical response frequency characteristic is shown in Fig. 6.36.

Suppose that a constant current is applied to the voice coil of the loud speaker and at a low-frequency point A of Fig. 6.36. 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. 6.37. The velocity frequency characteristic for an increase in frequency is shown in Fig. 6.37. The velocity frequency characteristic for a decrease in frequency is also shown in Fig. 6.37. These characteristics are quite similar to the theoretical characteristic of Fig. 6.36.

D. *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

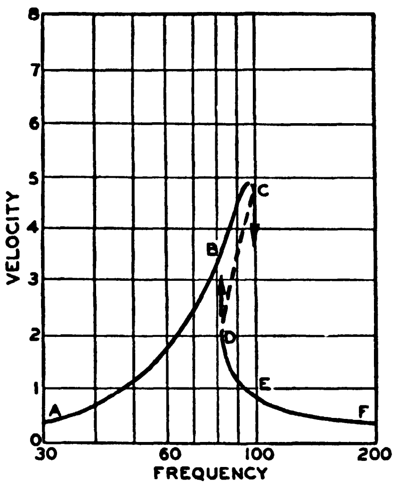

FIG. 6.36. Theoertical response frequency characteristic of a direct radiator loud-speaker mechanism with a nonlinear suspension system. The unstable portion of the response frequency characteristic is indicated by a dashed line.

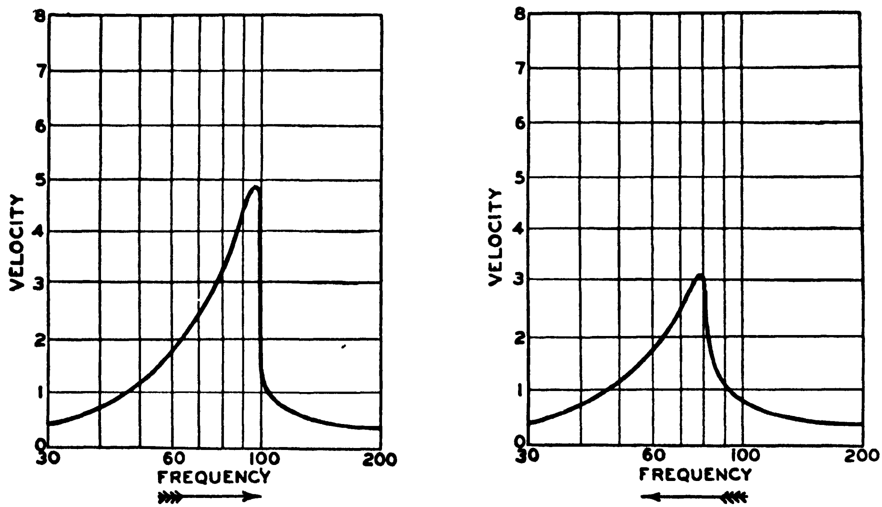

FIG. 6.37. Experimental response frequency characteristic of a direct radiator loud-speaker mechanism with a nonlinear suspension system for an increasing and a decreasing applied frequency.

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. 6.33.

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.

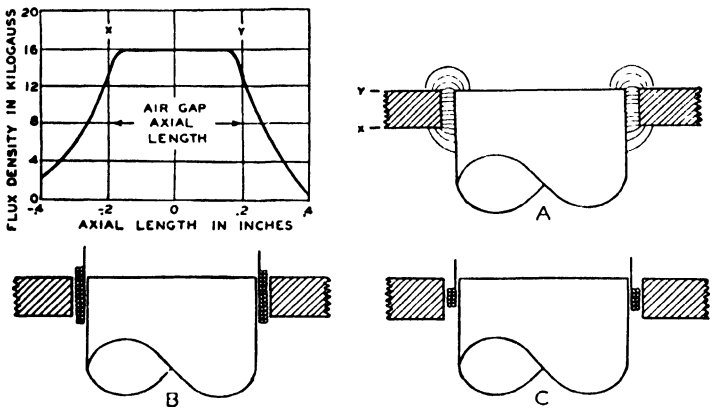

FIG. 6.38. Graph of the flux distribution in an air gap. *A*. Typical field map of the flux lines in an air gap. *B*. A voice coil longer than the air gap. *C*. A voice coil shorter than the air gap.

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 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. 6.38. 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. 6.38*B*, 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 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. 6.38*C*. The latter method is used for high-frequency loud speakers of high efficiency.

E. *Frequency-Modulation Distortion.*[19] — The amplitude of the cone in a direct radiator loud speaker 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 a 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.

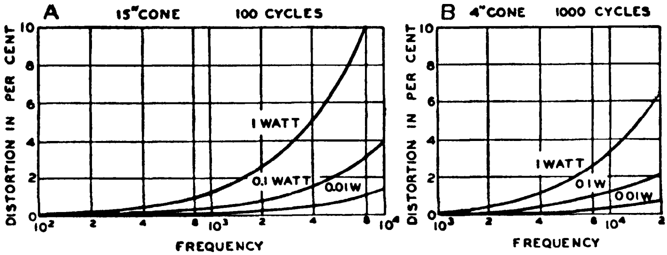

Fig. 6.39. *A*. Frequency modulation distortion characteristics of a 15-inch diameter cone with outputs of 1, .1 and .01 acoustical watts at 100 cycles for a second applied frequency over the range 100 to 10,000 cycles. *B*. Frequency modulation distortion characteristics of a 4-inch diameter cone with outputs of 1, .1 and .01 acoustical watts at 1000 cycles for a second applied frequency over the range 1000 to 10,000 cycles.

The square root of the ratio of the power in the sidebands to the total power in the sound wave, in per cent, is

$$D = 2900 \frac{f_2 \sqrt{p_1}}{f_1^2 d^2} \qquad 6.31$$

where f_2 = modulated frequency, in cycles,
f_1 = modulating freqüency, in cycles,
p_1 = acoustical output at f_1, in watts, and
d = cone diameter, in inches.

Frequency modulation distortion characteristics for a cone 15 inches in diameter and a cone 4 inches in diameter for acoustical outputs of 1, .1 and .01 watts are shown in Fig. 6.39.

[19] Beers and Belar, *Proc. I.R.E.*, Vol. 31, No. 4, p. 132, 1943.

F. *Air Nonlinear Distortion.* — In general, the distortion generated in the air between the cone of a direct radiator loud speaker and the listener is considered to be negligible. It is very much less than this type of distortion in a horn type loud speaker. However, if small distortions are of consequence, then some cons deration 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

$$\frac{p_{2r}}{p_{1r}} = \frac{(\gamma + 1)\omega p_{1r} r}{2\sqrt{2}\gamma p_0 c r_1} \log_e \frac{r}{r_1} \qquad 6.32$$

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,[20] 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 loud speaker the wave is diverging. At a distance equal to the radius of the cone the system can, from the stand-

[20] Thuras, Jenkins and O'Neil, *Jour. Acous. Soc. Amer.*, Vol. 6, No. 3, p. 173, 1935.

point 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

$$D = \frac{p_{2r}}{p_{1r}} 100 = \frac{(\gamma + 1)\omega p_{1r}}{2\sqrt{2}\gamma p_0 c} \left[.85r + \frac{r}{r_1} \log_e \frac{r}{r_1} \right] 100 \qquad 6.34$$

where p_{1r} = pressure at the observation point, in dynes per square centimeter.

The distortion frequency characteristics, for a distance of 3 meters (about 10 feet) and various pressures at the observation point, for direct radiator loud speakers with cones having diameters of 2 inches and 8 inches, are shown in Fig. 6.40.

6.17. Diaphragms, Suspensions, and Voice Coils. — The diaphragm or cone of practically all direct radiator loud speakers is made of paper. Typical cones, shown in Fig. 6.41, 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. 6.41. 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. 6.42*A*; the flared shape, Fig. 6.42*B*; the corrugated conical shape, Fig. 6.42*C*. The shapes of Fig. 6.42 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 propagation of the wave in the cone and thereby broadens the directional pattern.

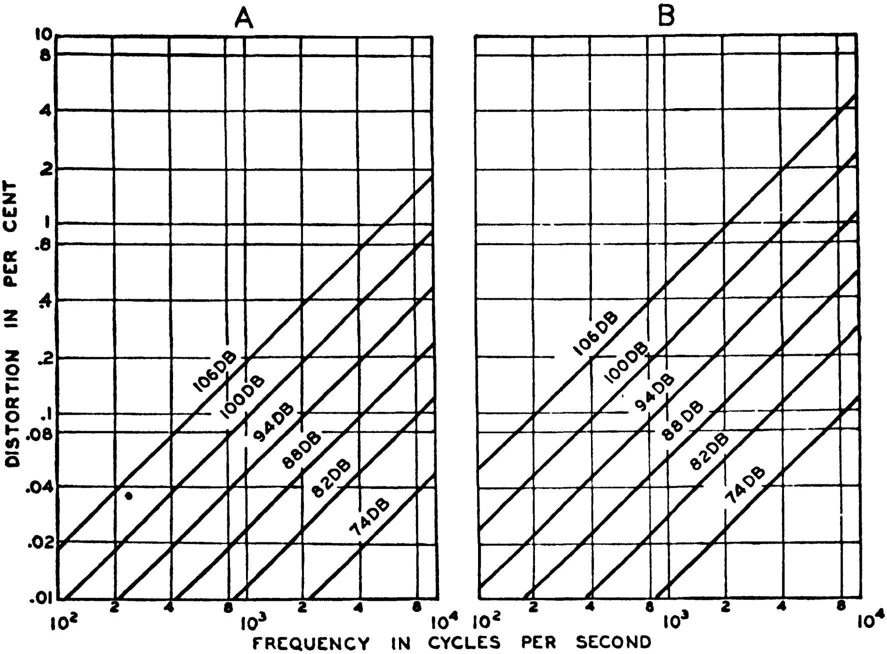

FIG. 6.40. Distortion frequency characteristics depicting the distortion generated in the air between the cone and the observer in a direct radiator loud speaker. *A*. Loud speaker with a cone 8 inches in diameter. *B*. Loud speaker with a cone 2 inches in diameter. Labels on characteristics indicate sound levels at observation point. 0 db = .0002 dyne per square centimeter. 100 db = 20 dynes per square centimeter. Distance 3 meters.

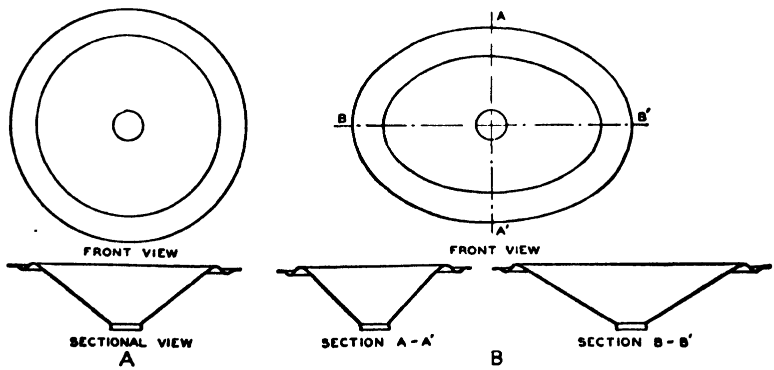

FIG. 6.41. Felted cones for direct radiator, dynamic loud speakers. *A*. Circular cone. *B*. Elliptical cone.

The three types of suspension systems, shown in Fig. 6.43, are in general use. The leather or kidskin suspension system shown in Fig. 6.43*A* is gradually going out of use. It has been displaced by the one piece felted cone and suspension system shown in Fig. 6.43*B*. The latter system is

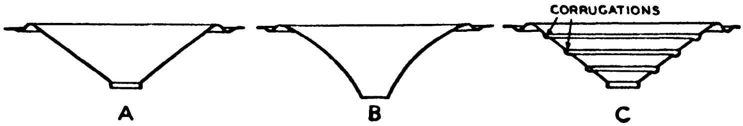

FIG. 6.42. Sectional views of felted cones for direct radiator, dynamic loud speakers: *A*. Conical shape. *B*. Flared shape. *C*. Conical shape with corrugations.

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. 6.43*C*. The reduction in 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).

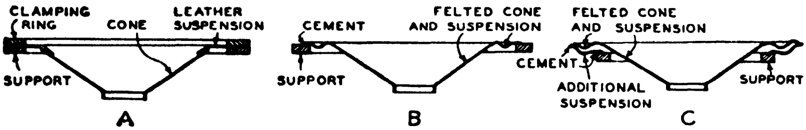

FIG. 6.43. Sectional views of cone suspension systems for direct radiator loud speakers. *A*. Leather suspension. *B*. Felted corrugated suspension. *C*. Folded or double-felted corrugated suspension system.

Centering suspensions for keeping the voice coil aligned in the air gap are shown in Fig. 6.44. An inside slotted centering suspension, usually made of fiber, is shown in Fig. 6.44*A*. An inside felted paper centering suspension with corrugations is shown in Fig. 6.44*B*. This type of suspension is usually employed where the amplitude is small. An outside felted paper suspension with corrugations is shown in Fig. 6.44*C*. The outside centering suspension can be made large in diameter and thereby obtain a very low value of stiffness.

Voice coil construction used in direct radiator loud speakers are shown in Fig. 6.45. A voice coil wound on a cylindrical paper form with round wire is shown in Fig. 6.45*A*. 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. 6.45*B*, *C* and *D*. Thermosetting cement is used to bind the entire assembly. The use of a self-supporting coil eliminates the cylindrical paper form and thereby reduces 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.

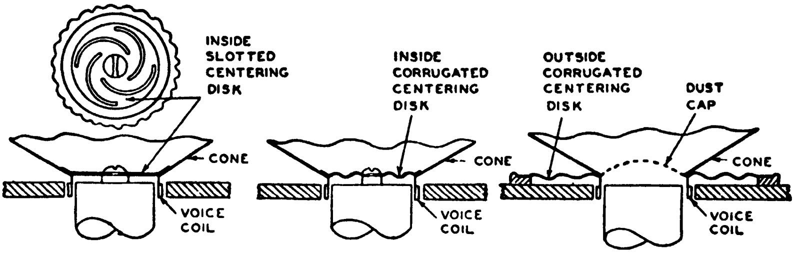

FIG. 6.44. Sectional views of voice coil centering systems for direct radiator dynamic loud speakers: *A*. Inside slotted fiber disk. *B*. Inside felted corrugated centering disk. *C*. Outside felted corrugated centering disk.

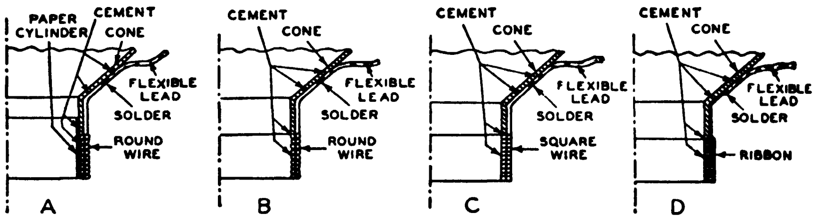

FIG. 6.45. Sectional views of voice coil constructions. *A*. Round enameled wire wound on a paper form. *B*, *C*, and *D*. Self-supporting voice coils held together with thermosetting cement. *B*. Round wire. *C*. Square wire. *D*. Edgewise wound ribbon.

6.18. 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. 6.46*A* and 6.46*B*. The radius of curvature of the wave front with the distributor being considerably less than that of the plain cone shows that the distributor broadens the radiation pattern. The vertical section, Fig. 6.46*C*, 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.

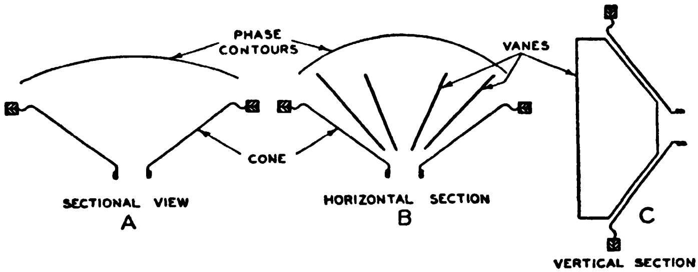

FIG. 6.46. High-frequency sound distributors for direct radiator loud speakers. *A*. The contour of equal phase for a plain cone. *B*. The horizontal cross-sectional view of a cone with a vane distributor and the contour of equal phase. *C*. The vertical cross-sectional view of a cone with a vane distributor.

6.19. Field Structures. — Five typical electromagnetic and permanent magnet field structures are shown in Fig. 6.47. The most widely used field structure for use with direct radiator loud speakers in a-c radio receivers and phonographs is depicted in Fig. 6.47*A*. During the past

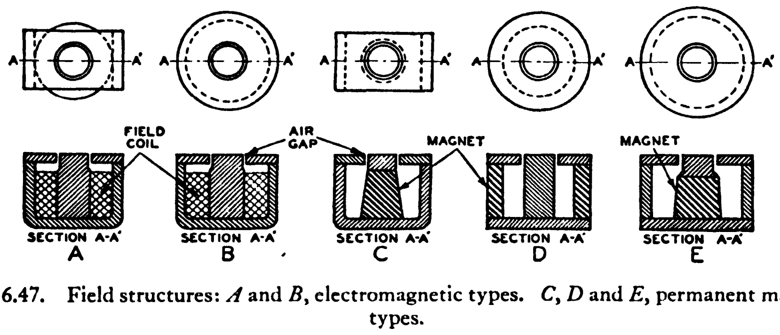

FIG. 6.47. Field structures: *A* and *B*, electromagnetic types. *C*, *D* and *E*, permanent magnet types.

few years materials for permanent magnets have been improved and it is now possible to obtain high flux densities in structures of reasonable size. These new alloys are combinations of aluminum, nickel, cobalt and iron and have been termed "Alnico." The structures *C* and *D* in Fig. 6.47

are most widely used in direct loud speakers for battery operated radio receivers and sound systems.

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 structures are used in loud speakers, 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 egs 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 fundamental equation of magnetic circuits is

$$\phi = \frac{M}{R} \qquad 6.35$$

where ϕ = total lines of flux, in maxwells,
M = magnetomotive force, in gilberts, and
R = reluctance, no unit.

The magnetomotive force, in gilberts, in an electromagnetic system, as shown in Fig. 6.48*A* is given by

$$M_T = 4\pi ni \qquad 6.36$$

where n = number turns, and
i = current, in abamperes.

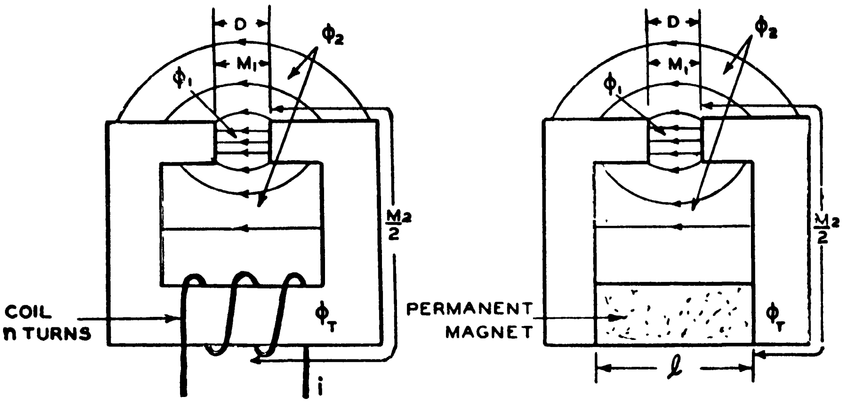

FIG. 6.48. Electromagnetic and permanent magnet field structures. M_1 = the magnetomotive drop in the air gap. M_2 = magnetomotive force drop in the iron. ϕ_1 = the total line in the air gap flux. ϕ_2 = the total lines in the leakage flux.

The magnetizing force, in oersteds, developed by the permanent magnet in the permanent magnet system of Fig. 6.48*B* may be obtained from the demagnetization curves of Fig. 6.49. 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

$$\phi_T = \phi_1 + \phi_2 \qquad 6.37$$

where ϕ_1 = lines in the air gap, in maxwells, and
ϕ_2 = lines in the leakage field, in maxwells.

The total magnetomotive force, in gilberts, developed by the energized coil or permanent magnet is

$$M_T = M_1 + M_2 \qquad 6.38$$

The number of lines in the air gap, in maxwells, is

$$\phi_1 = \frac{M_1}{R_1} \qquad 6.39$$

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

$$R_1 = \frac{l}{A_1} \tag{6.40}$$

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.

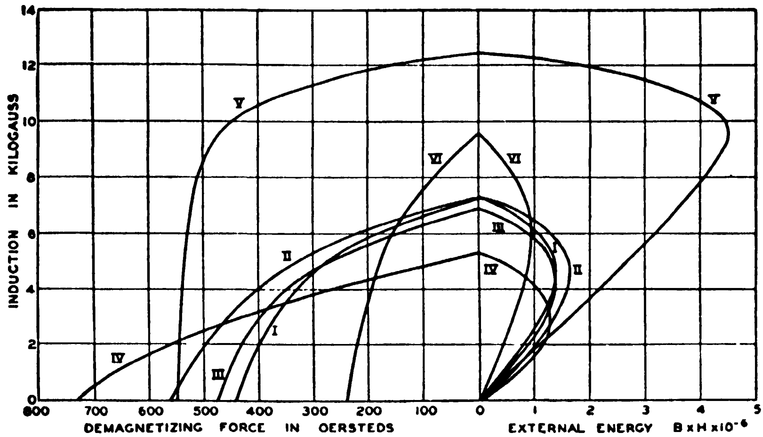

FIG. 6.49. Demagnetization and energy characteristics of permanent magnet materials. *I*. Alnico I. *II*. Alnico II. *III*. Alnico III. *IV*. Alnico IV. *V*. Alnico V. *VI*. 36 per cent cobalt.

The reluctance in the iron structure

$$R_2 = \frac{l}{\mu A_2} \tag{6.41}$$

where l = length of the path in the iron, in centimeters,

A_2 = cross-sectional area, in square centimeters, and

μ = permeability.

The curves [21] of Fig. 6.50 show the relation between the intrinsic flux density and the magnetizing force for different magnetic materials. The

[21] Kentner, A. E., *Gen. Elec. Rev.*, Vol. 45, No. 11, p. 633, 1942.

permeability may be obtained from the curves of Fig. 6.50 and the following relation

$$\mu = \frac{B}{H} \tag{6.42}$$

where B = flux density, in gausses, and
H = magnetizing force in oersteds.

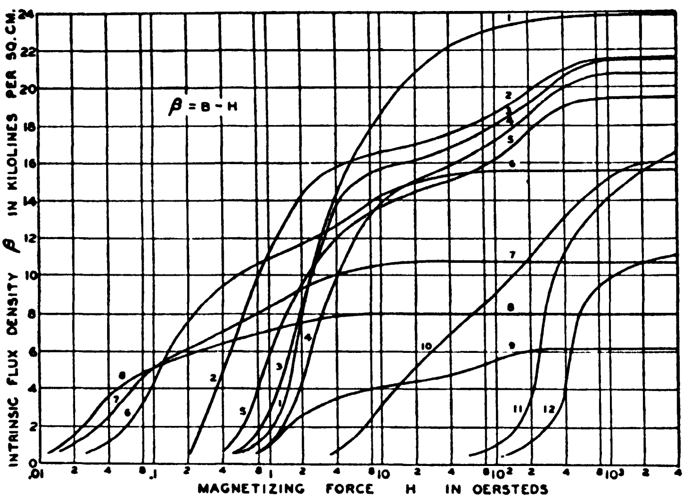

FIG. 6.50. D-c magnetization characteristics for various magnetic materials. 1. Permandur, 50 per cent cobalt and 50 per cent iron. 2. Electrolytic iron. 3. Armco iron. 4. Annealed cold drawn steel. 5. Medium hard silicon steel. 6. Nickel iron alloy, 47 per cent nickel. 7. Permalloy, 79 per cent nickel. 8. Allegheny, mumetal. 9. Pure nickel. 10. Cast iron. 11. Cobalt permanent magnet steel. 12. Alnico permanent magnet alloy. (After Kentner.)

The magnetomotive force drop in the iron is given by

$$M_2 = R_2 \phi_2 \tag{6.43}$$

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.

The air-gap flux density, in gausses, is given by

$$B = \frac{\phi_1}{A_1} = \frac{M_1}{l} \tag{6.44}$$

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 6.44 and the flux density, B. If the leakage lines, ϕ_2, are known then the total lines, ϕ_T, are given by equation 6.37. 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 6.43. Then the total magnetomotive, M_T, can be determined from equation 6.38. 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

$$l = \frac{M_T}{H} \qquad 6.45$$

where l = length, in centimeters,
M_T = total magnetomotive force required, in gilberts, and
H = demagnetizing force, in oersteds (see Fig. 6.49).

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.

CHAPTER VII

HORN LOUD SPEAKERS

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 loud speakers. At the present time, horn loud speakers seem to be the only satisfactory high efficiency system for large-scale sound reproduction. A horn loud speaker[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 overall 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 loud speaker 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 loud speaker. 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 loud speaker and to describe several horn loud-speaker systems.

7.2. Efficiency.[6,7,8] — The efficiency of a loud speaker 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. 140, 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 loud speaker 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 loud speaker 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. 5.24 and 2.14. 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 loud speaker. It is the purpose of this section to consider the effect of the various parameters, referred to above, upon the efficiency of a horn loud speaker 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 loud speakers 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 loud-speaker 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

$$z_{EM} = \frac{(Bl)^2}{z_M} \times 10^{-9} \qquad 7.1$$

[9] Olson, "Dynamical Analogies," D. Van Nostrand Company, New York, N. Y., 1943.

where B = flux density, in gausses,

l = length of wire in the voice coil, in centimeters, and

z_M = mechanical impedance of the vibrating system, in mechanical ohms, at the point f_M Fig. 7.1.

The efficiency, μ, in per cent, is

$$\mu = \frac{r_{EM}}{r_{ED} + r_{EM}} \times 100 \qquad 7.2$$

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.

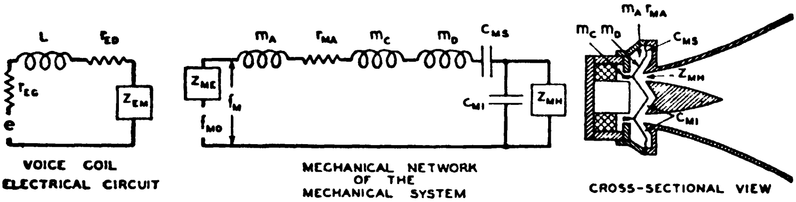

FIG. 7.1. Cross-sectional view of a horn loud speaker, the electrical circuit and mechanical network of the vibrating system. In the voice coil circuit: e = the internal voltage of the generator. r_{EG} = the internal electrical resistance of the generator. L = the inductance of the voice coil. r_{ED} = 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_{M0} = 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,

m_C = mass of the voice coil, in grams,

m_D = mass of the diaphragm, in grams,

r_{MA} = mechanical resistance load on the back of the diaphragm, in mechanical ohms,

C_{MS} = compliance of the suspension, in centimeters per dyne,

C_{M1} = compliance of the air chamber, in centimeters per dyne,

$z_{MH} = r_{MH} + jx_{MH}$ = mechanical impedance of the throat of the horn, in mechanical ohms,

r_{MH} = mechanical resistance of the throat of the horn, in mechanical ohms, and

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

$$z_M = r_{MH} = 42\,\frac{A_D^2}{A_T} \tag{7.4}$$

where A_D = area of the diaphragm, in square centimeters, and

A_T = area of the throat, in square centimeters.

Substituting equations 7.1 and 7.4 in equation 7.2

$$\mu = \frac{B^2}{\left(\dfrac{42A_D^2 r_{ED}}{l^2 A_T}\right)10^9 + B^2} \times 100 \tag{7.5}$$

The electrical resistance,[10] r_{ED}, in ohms, is given by

$$r_{ED} = \frac{K_r l}{S} \times 10^{-6} \tag{7.6}$$

where K_r = resistivity of the voice coil material, in microhms, per centimeter cube (see Table 6.1),

l = length of the conductor, in centimeters, and

S = area of the conductor, in square centimeters.

Then equation 7.5 becomes

$$\mu = \frac{B^2}{\left(\dfrac{42A_D^2 K_r}{lSA_T}\right) 10^3 + B^2} \times 100 \tag{7.7}$$

---

[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. 6.5, 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.

The mass of the coil, m_C, is

$$m_C = lS\rho \text{ grams} \tag{7.8}$$

where ρ = density, in grams per cubic centimeter (see Table 6.1). The efficiency may be written, employing equation 7.8, as

$$\mu = \frac{B^2}{\left(\dfrac{42A_D{}^2K_r\rho}{m_CA_T}\right)10^3 + B^2} \times 100 \tag{7.9}$$

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

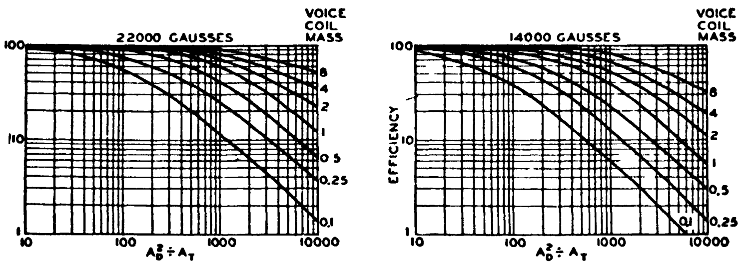

FIG. 7.2. The initial efficiency, in per cent, of a horn loud speaker as a function of $A_D{}^2/A_T$ for aluminum voice coils having masses of 0.1, 0.25, 0.5, 1.2, 4, and 8 grams and flux densities of 22,000 and 14,000 gausses. A_D and A_T are the areas of the diaphragm and throat, respectively, in square centimeters. The above graphs may be applied to a copper voice coil by multiplying the ratio $A_D{}^2 \div A_T$ by one half.

$A_D{}^2/A_T$ 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 loud speaker.

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

$$r_{EM} = \left(\frac{(Bl)^2 r_M}{r_M^2 + x_M^2}\right) 10^{-9} \text{ ohms} \tag{7.10}$$

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

$$x_M = \omega\,(m_D + m_C) \tag{7.11}$$

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

$$r_{MH} = 42\frac{A_D^2}{A_T} \tag{7.12}$$

where A_T = area of the throat, in square centimeters. Substituting equation 7.10 in 7.2, the efficiency, in per cent, becomes

$$\mu = \frac{(Bl)^2 r_{MH}}{r_{ED}(r_{MH}^2 + x_M^2)10^9 + (Bl)^2 r_{MH}} \times 100 \tag{7.13}$$

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 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, 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] — 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 loud speaker 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

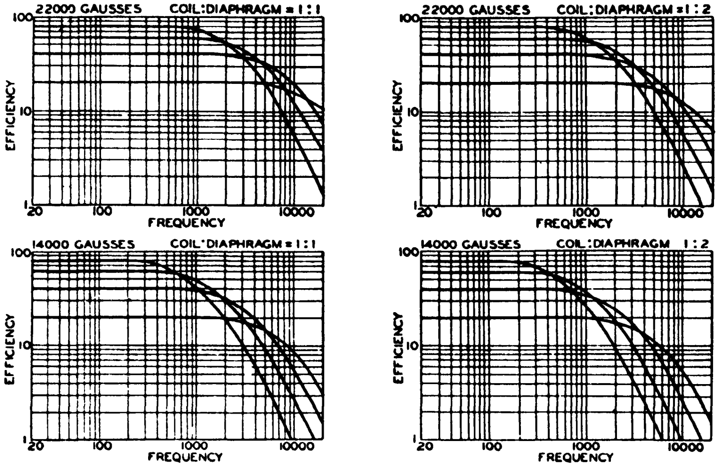

FIG. 7.3. The efficiency, in per cent, as a function of the frequency of horn loud speaker systems having ratios of voice coil mass to diaphragm mass of 1:2 and 1:1, flux densities of 22,000 and 14,000 gausses, and initial efficiencies of 20, 40, 60 and 80 per cent for an aluminum coil. The above graphs may be applied to a copper voice coil by multiplying the frequency by one half.

the 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

[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.

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

$$z_M = \frac{r_{MH}}{1 + j\omega r_{MH} C_{M1}} \tag{7.14}$$

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

$$C_{M1} = \frac{C_1}{A_D^2} = \frac{V}{\rho c^2 A_D^2} \tag{7.15}$$

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.

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 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 loud speakers, 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

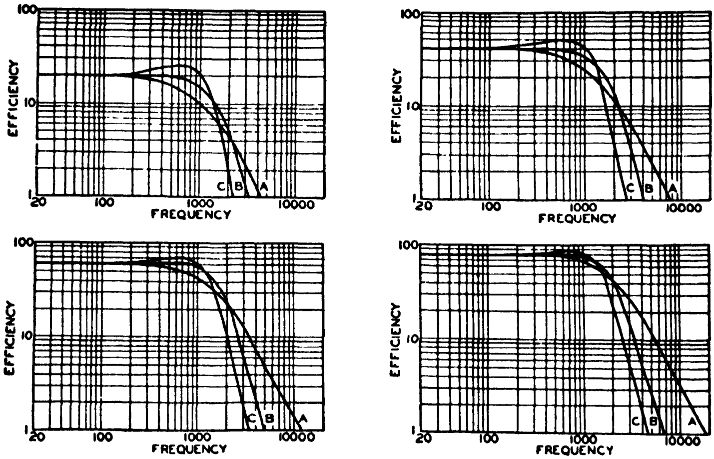

FIG. 7.4. The efficiency, in per cent, as a function of the frequency of a voice coil and diaphragm having a mechanical reactance of 1 mechanical ohm at 1000 cycles coupled to a throat of a horn having a mechanical resistance of 1 mechanical ohm and an air chamber having the following mechanical reactances: *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.

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

$$\text{Power} = \left(\frac{e}{|z_{ET}|}\right)^2 z_{EM} \qquad 7.16$$

where the motional electrical resistance, 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

$$z_{ET} = r_{ED} + r_{EG} + j\omega L + z_{EM} \tag{7.17}$$

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. 5.9*B* 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.

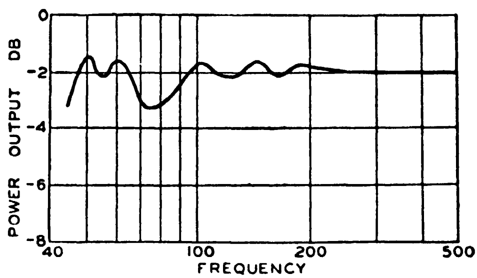

FIG. 7.5. Acoustical power output frequency characteristic of the horn (Fig. 5.9*B* with all dimensions multiplied by 2½) coupled to a 10½-inch diameter, 10-gram diaphragm driven by a 5-gram aluminum voice coil in a field of 20,000 gausses. Damped electrical resistance of voice coil, 20 ohms. Electrical impedance of vacuum tube through a transformer, 35 ohms.

E. *The Effect of the Voice Coil Temperature upon the Efficiency.*[16] — The effect of the temperature of the voice coil upon the efficiency is usually ignored in considerations of the performance of a loud speaker. In high

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

power loud speakers, 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.

The efficiency, in per cent, of a loud speaker, when the temperature correction is added, may be expressed

$$\mu = \frac{r_{EM}}{r_{ED0}(1 + \alpha t) + r_{EM}} \times 100 \tag{7.18}$$

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.

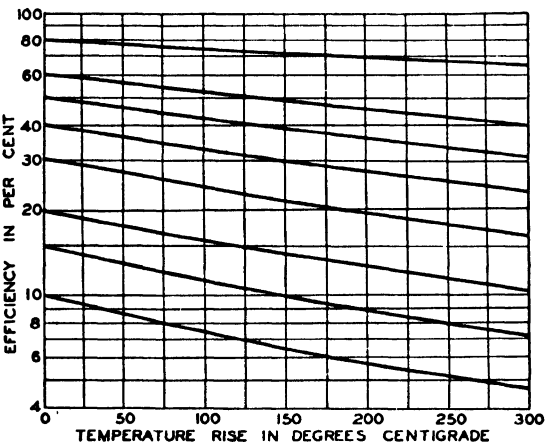

Fig. 7.6. The efficiency, in per cent, as a function of the temperature of a voice coil for various values of initial efficiency at 0° Centigrade.

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 loud speaker 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 loud speaker. 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.

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.

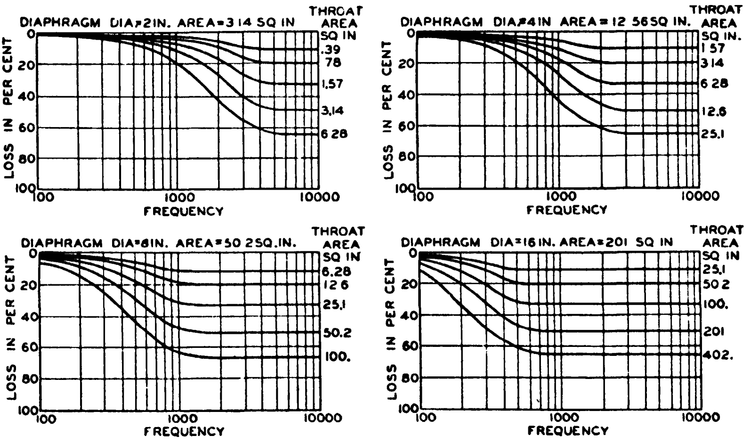

FIG. 7.7. Characteristics depicting the loss in per cent of the total sound radiation, due to the radiation of sound from the back of the diaphragm of a horn loud speaker for diameters of 2, 4, 8 and 16 inches and throat areas of 2, 1, ½, ¼, and ⅛ times the diaphragm area.

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 loud speaker 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 loud speakers.

A. *Distortion Due to Air Overload in the Horn.*[17, 18, 19] — 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 loud speakers 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.

The second harmonic distortion, at the mouth, in per cent of the fundamental, generated in an exponential horn is given by

$$D = \frac{p_2}{p_1} 100 = \frac{(\gamma + 1)p_1 \omega}{\sqrt{2}\gamma p_0 c m} (1 - \epsilon^{-mx/2})100 \qquad 7.20$$

[17] Rocard, *Comtes Rendus*, Vol. 196, p. 161, 1933.
[18] Thuras, Jenkins and O'Neil, *Jour. Acous. Soc. Amer.*, Vol. 6, No. 3, p. 173, 1935.
[19] Goldstein and McLachlin, *Jour. Acous. Soc. Amer.*, Vol. 6, No. 4, p. 275, 1935.

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. 5.18),
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[20] 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

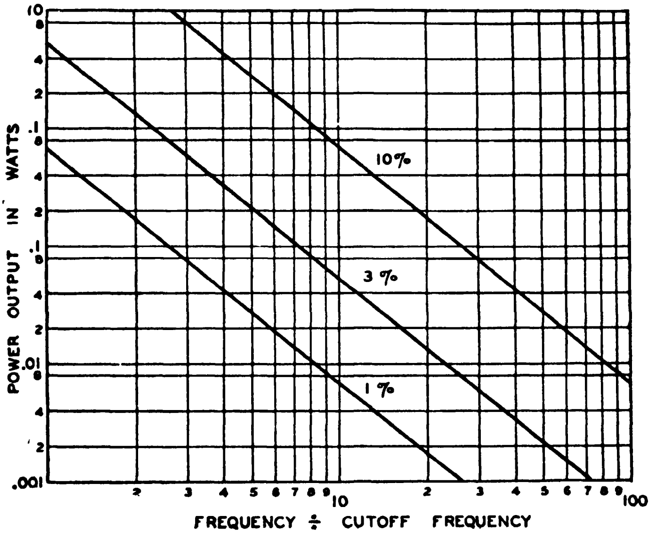

FIG. 7.8. The power output of infinite exponential horns, per square centimeter of throat area, for 1, 3 and 10 per cent distortion, as a function of the ratio of the frequency under consideration to the cutoff frequency.

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 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.

[20] Olson, H. F., *RCA Review*, Vol. 2. No. 2, p. 265, 1937.

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.*[21] — 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 loud speaker. 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

[21] Olson, H. F., *RCA Review*, Vol. 2, No. 2, p. 265, 1937.

the acoustical resistance of the horn (see Fig. 7.9). At the high frequencies 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

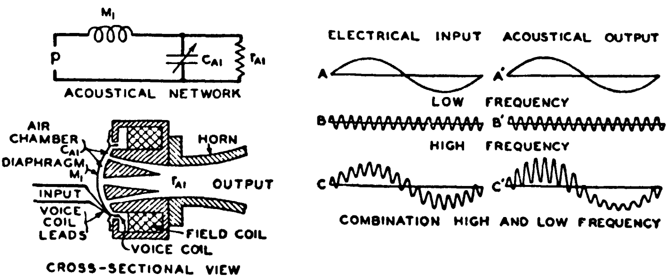

FIG. 7.9. A mechanism with an air chamber coupling the diaphragm to the horn. In the acoustical network: M_1 = the inertance of the diaphragm and voice coil. C_{A1} = the acoustical capacitance of the air chamber. r_{A1} = the acoustical resistance at the throat of the horn. p = the driving pressure. $p = \frac{Bli}{A}$. 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 acoustical network indicates the effect of the nonlinear element upon the system. 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.

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 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.*[22] — The outside suspension is another example of a variable circuit element in a mechanical 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. 6.16).

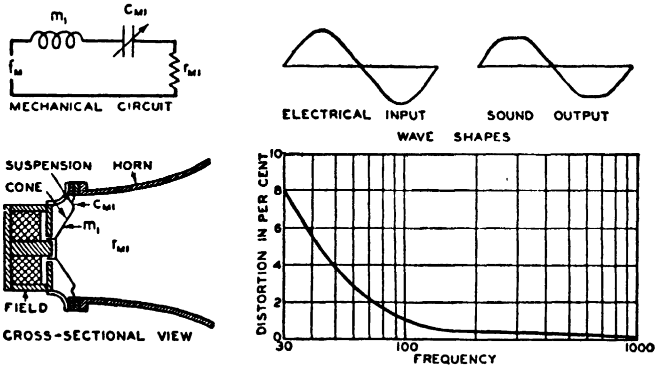

FIG. 7.10. Mechanism having a diaphragm with a nonlinear suspension system. In the mechanical circuit: m_1 = the mass of the diaphragm and voice coil. C_{M1} = the compliance of the diaphragm suspension system. r_{M1} = the mechanical resistance at the throat of the horn. f_M = the driving force. The 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.

In the case of a horn loud speaker 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. 6.16*B*). 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.

D. *Distortion Due to a Nonuniform Magnetic Field in the Air Gap.* — Inhomogeneity of the flux density through which the voice coil moves is

[22] Olson, H. F., *RCA Review*, Vol. 2, No. 2, p. 265, 1937.

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

$$f = Bli \qquad 7.22$$

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. 6.38. 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. 6.38. The latter method is usually used for high frequency loud speakers of high efficiency (also see Sec. 6.16*D*).

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. 6.16*B* that subharmonics are generated in vibrating systems with nonlinear elements. The existence of subharmonics in direct radiator loud speakers is well known. However, in horn loud speakers 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.*[23] — The maximum allowable distortion may determine the power rating for the loud speaker. However, in certain loud speakers the maximum allowable temperature of the voice coil determines the power rating. This is particularly true of high frequency loud speakers.

By making the efficiency a maximum, the dissipation in, and the resulting

[23] Olson, H. F., *RCA Review*, Vol. 2, No. 2 p. 265, 1937.

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 de-

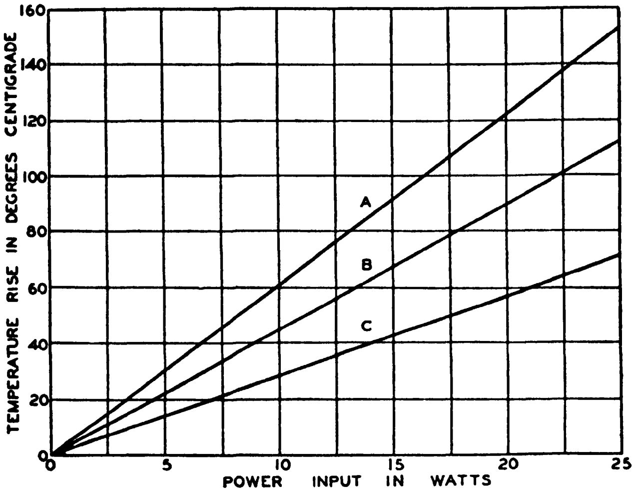

FIG. 7.11. The temperature rise as a function of the power delivered to a voice coil for air-gap clearances as follows: *A*. 0.021 inch, *B*. 0.015 inch, *C*. 0.009 inch. Coil 1½ inch in diameter and 0.25 inch in length.

creased. The temperature rises as a function of the power dissipated in the voice coil for various clearances between the voice coil and pole pieces 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.

G. *Power Handling Capacity and the Amplitude of the Diaphragm.*[24] — 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 loud speaker in which the diaphragm is terminated in an acoustical resistance is

$$P = \frac{\rho c (2\pi f)^2 d^2 A_D{}^2}{2A_H} 10^{-7} \qquad 7.23$$

[24] Massa, F., *RCA Review*, Vol. 3, No. 2, p. 196, 1938.

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.

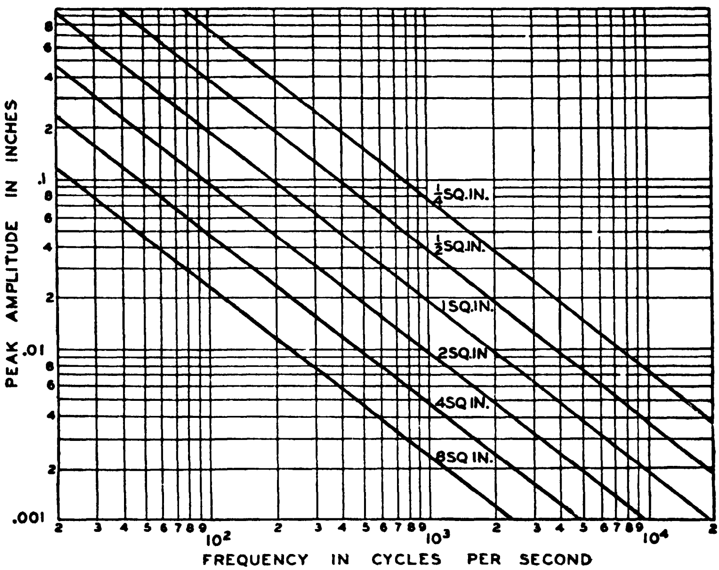

FIG. 7.12. The peak amplitude frequency characteristics of vibrating pistons of various areas in square inches, coupled to the throat of a horn having an area of 1 square inch, for 1 watt output.

7.4. Horn Loud Speaker 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 widely 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 flare horn coupled to a diaphragm, the efficiency can be increased by employing a multiple flare horn.

To obtain maximum efficiency in a horn loud speaker 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.

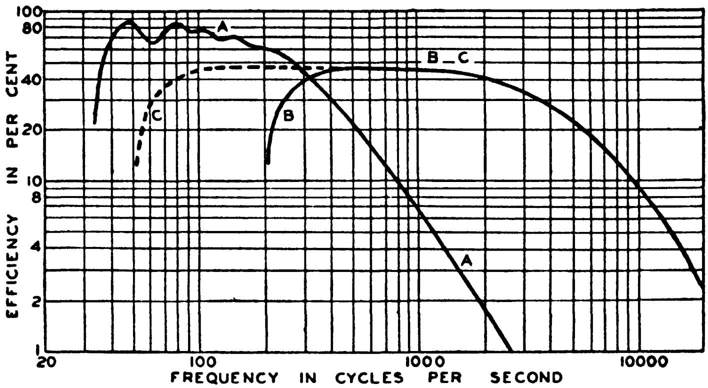

Fig. 7.13. *A*. Efficiency frequency characteristic of a horn loud speaker employing the horn of Fig. 5.9*D* with the dimensions multiplied by three and driven by 4 cones, 12 inches in diameter, with 5-gram copper voice coils operating in a field of 14,000 gausses. *B*. Efficiency frequency characteristic of a horn loud speaker employing the horn of Fig. 5.8*D* 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. 5.8*D* are multiplied by two.

The surge mechanical resistance of the exponential horn is independent of the frequency. However, the acoustical resistance[25] 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 efficiency frequency characteristic of the multiple flare horn described in Sec. 5.26 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 envelope of the family of characteristics shown in Fig. 7.3.

[25] Olson, H. F., *Jour. Soc. Mot. Pict. Eng.*, Vol. 30, No. 5, p. 511, 1938.

The two preceding horn loud speakers are suitable for high quality reproduction of speech and music. For certain types of announce installations it is desirable to project intelligible speech over very great distances (1 to 2 miles) under all manner of conditions. This requires acoustical out-

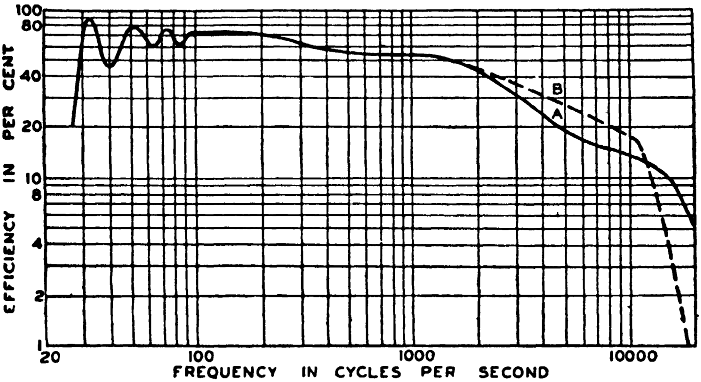

FIG. 7.14. Efficiency frequency characteristic of a diaphragm coupled to the horn of Fig. 5.11 and driven by an aluminum voice coil of one half the diaphragm mass in a field of 22,000 gausses. *A*. Without air chamber. *B*. With air chamber.

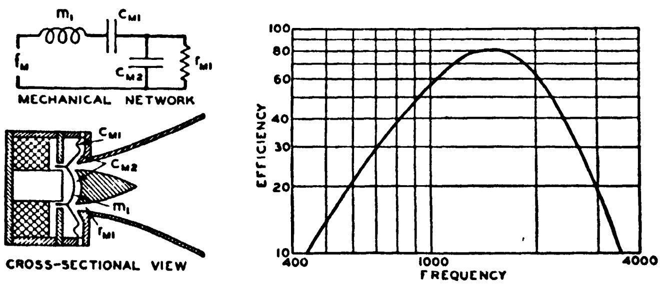

FIG. 7.15. Cross-sectional view and mechanical circuit of a loud speaker of 2 degrees of freedom. In the mechanical network: m_1 = the mass of the diaphragm and voice coil. C_{M1} = the compliance of the diaphragm suspension system. C_{M2} = the compliance of the air chamber. r_{M1} = the mechanical resistance at the throat of the horn. f_M = the driving force. The graph shows the efficiency frequency characteristic.

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 loud speaker of this capacity for the reproduction of the lower frequencies. A cross-sectional view of a high power announce loud speaker 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 loud speaker suitable for acoustical outputs of 500 to 1000 watts is shown in Fig. 7.15. Due to the large audio-power amplifier requirements, high loud-speaker efficiency is an extremely important economic factor.

B. *Multiple-Horn, Multiple-Channel System.* — The two-channel or "two-way" system [26, 27] is the most common example of a multichannel system. This loud speaker, Fig. 7.16, consists of a low-frequency folded horn unit for reproduction from 40 to 300 cycles and a multicellular horn unit for reproduction from 300 to 8000 cycles.

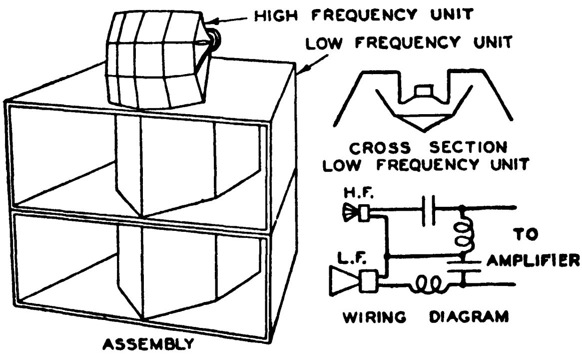

FIG. 7.16. A two-channel, theater, loud-speaker system consisting of a folded low-frequency horn unit and a multicellular horn high-frequency unit. The wiring diagram shows the electrical filter used to allocate the power, as a function of the frequency, to the two units.

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 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

[26] Wente and Thuras, *Jour. A.I.E.E.*, Vol. 53, No. 1, p. 17, 1934.
[27] Hilliard, J. K., *Tech. Bul. Acad. Res. Coun.*, March, 1936.

of this type of loud speaker were discussed in Sec. 2.15. 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 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 loud speaker without the filter are shown in Fig. 7.13, characteristics *B* and *A*.

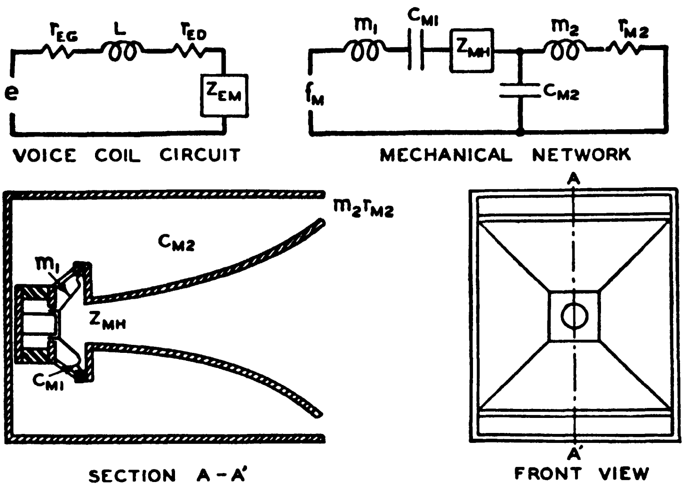

FIG. 7.17. Combination horn and phase inverter low-frequency loud speaker. In the voice coil circuit, e = the internal voltage of the electrical generator. r_{EG} = the electrical resistance of the electrical generator. L and r_{ED} = the damped inductance and electrical resistance of the voice coil. 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.

The low-frequency loud speaker 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 overall 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.

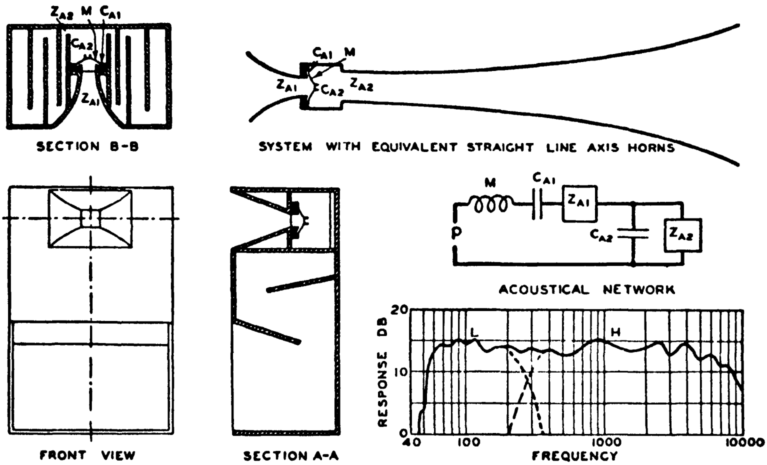

FIG. 7.18. Cross-sectional view of a compound horn loud speaker, the developed equivalent of the high- and low-frequency horns, and the acoustical network of the acoustical system. In the acoustical network: M = the inertance of the diaphragm. C_{A1} = the acoustical capacitance of the diaphragm suspension 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. $p = \frac{Bli}{A}\cdot$ 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 overall pressure response frequency characteristic.

C. *Compound Horn Loud Speaker.*[28] — The compound horn loud speaker 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. 7.18. The equivalent of the system is shown in Fig. 7.18. The functional acoustical network of the vibrating system is also shown in Fig.

[28] Olson and Massa, *Jour. Acous. Soc. Amer.*, Vol. 8, No. 1, p. 48, 1936.

7.18. 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, shunts out the low-frequency horn and radiation issues from the high-frequency horn. In the mid-range, radiation issues

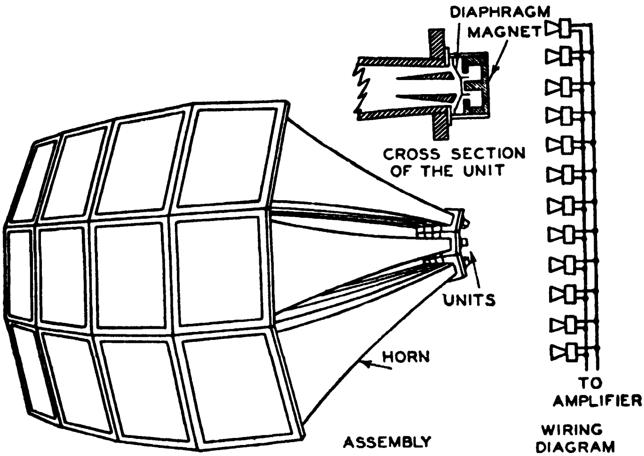

FIG. 7.19. A multiple-horn, single-channel, wide frequency range, loud-speaker system consisting of a cluster of multiflare horns, each coupled to a small diaphragm.

from both horns. The response frequency characteristic, Fig. 7.18, shows the response range of the two horns. The throats of the two horns may be chosen so that the efficiency characteristic of this loud speaker 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.

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. 7.19. A comparison of the efficiency characteristics of a multiflare horn loud speaker, Fig. 7.19, with a multichannel system, Fig. 7.19, 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.

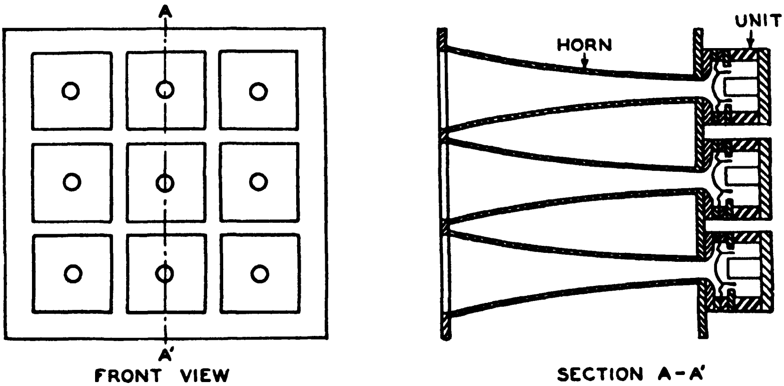

FIG. 7.20. A multiple-horn, single-channel system for high-power announce systems.

A multiple-horn, single-channel system loud speaker suitable for high power announce systems is shown in Fig. 7.20. This loud speaker performs the same function as the system shown in Fig. 7.15. The stresses in the diaphragm and 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 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. *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 7.18. 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. 7.21. A simple folded horn is shown in Fig. 7.21*A*. A folded horn with a ring shaped mouth is shown in Fig. 7.21*B*. 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. 7.21*C* is used for sending out radiation over 360° normal to the axis. It is customary to mount this loud speaker 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

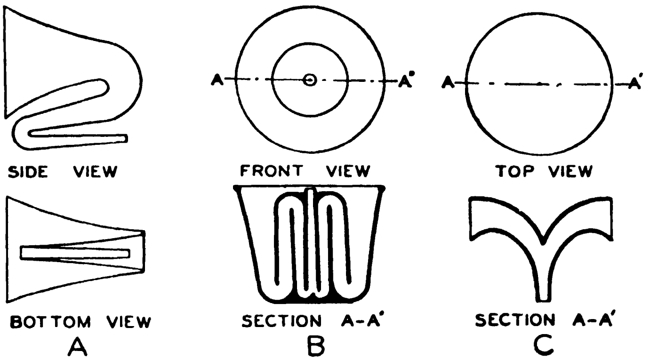

FIG. 7.21. Folded horns.

requires a large mouth horn for efficient reproduction of low-frequency sounds and sm:ll 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.

F. *Horn Loud-Speaker Mechanisms.* — The diaphragm, voice coil, magnet structure and air chamber of a horn loud-speaker 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 different methods for reducing interference in the air chamber are shown in Fig. 7.22*A*, *B*, *C* and *D* and Figs. 7.1, 7.9, 7.15 and 7.19. 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. 7.22*E*, without incurring any loss due to interference, notwithstanding the large size, because the dimensions are small compared to the wavelength.

G. *Diaphragms and Voice Coils.* – The diaphragms or cones of horn loud-speaker 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, 7.19, 7.20 and 7.22. Round, square and ribbon wire voice coil conductors are used as shown in Fig. 6.45.

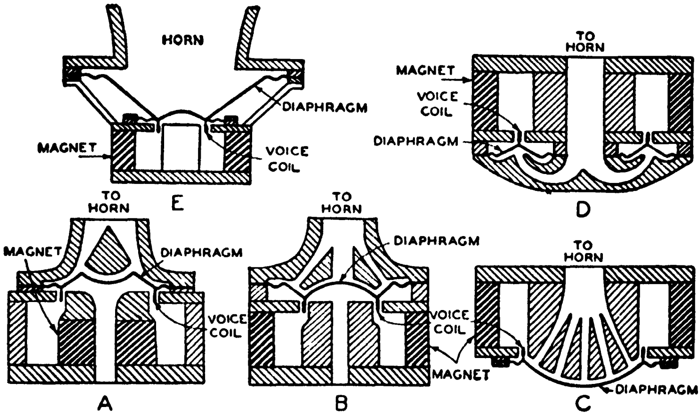

FIG. 7.22. Horn loud-speaker driving mechanisms. Mechanisms *A*, *B*, *C* and *D* depict various types of air chambers and diaphragms for coupling to a small throat horn. Mechanism *E* depicts a large diaphragm coupled to a large throat horn.

H. *Field Structures.* — Permanent magnet and electromagnetic field structures used in horn loud-speaker mechanisms are shown in Figs. 7.1, 7.9, 7.10, 7.15, 7.19, 7.20, 7.22 and 6.47. In general, it is customary to use higher flux densities in the gap in horn loud speakers than in direct radiator loud speakers. Soft iron may be used for the pole tips for flux densities up to 20,000 gausses (see Fig. 6.50). For flux densities from 20,000 to 23,000 gausses, a special alloy, Permandur [29] (see Fig. 6.50), is employed for the pole tip material in order to obtain these high densities with tolerable efficiency.

[29] Elmen, G. W., *Bell Syst. Tech. Jour.*, Vol. 15, No. 1, p. 113, 1936.

I. *Horn Walls. Vibration and Absorption.*[30] — 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.

[30] Phelps, W. D., *Jour. Acous. Soc. Amer.*, Vol. 12, No. 1, p. 68, 1940.

CHAPTER VIII

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

impeding 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

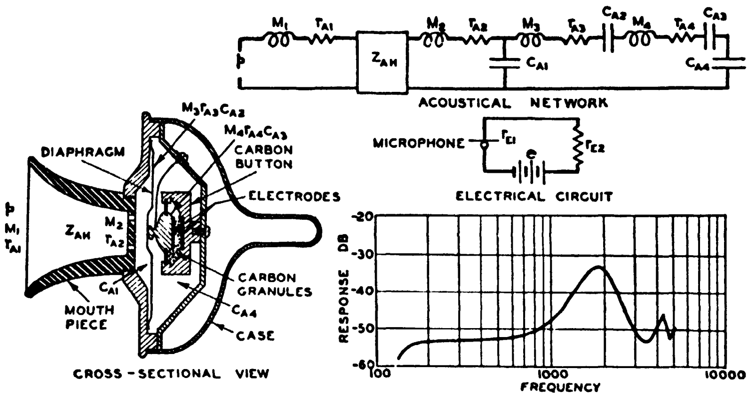

FIG. 8.1. Cross-sectional view, the electrical circuit and the acoustical network of a single-button carbon microphone. In the electrical circuit: r_{E1} = the electrical resistance of the carbon element, r_{E2} = the electrical resistance of the load, and e = the polarizing voltage of the battery. In 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.

is a change in the electrical resistance between the diaphragm and the carbon cup. For small displacements the change in resistance is proportional to 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,

$$
\begin{aligned}
i &= \frac{e}{r_{E0}}\left(1 - \frac{hx}{r_{E0}} \sin \omega t + \frac{h^2x^2}{r_{E0}^2} \sin^2 \omega t \ldots\right) \\
&= \frac{e}{r_{E0}}\left(1 - \frac{hx}{r_{E0}} \sin \omega t + \frac{h^2x^2}{2r_{E0}^2} - \frac{h^2x^2}{2r_{E0}^2} \cos 2\omega t \ldots\right) \qquad 8.2
\end{aligned}
$$

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. 5.23). 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.

A new 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 in-

[1] Jones, W. C., *Jour. A.I.E.E.*, Vol. 57, No. 10, p. 559, 1939.

cludes 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

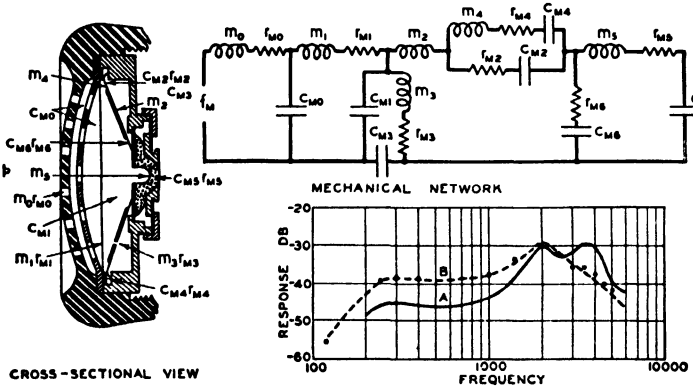

FIG. 8.2. Cross-sectional view and the mechanical network of an improved single-button carbon microphone. The electrical circuit is the same as that of Fig. 8.1. In the mechanical network: m_O and r_{MO} = the mass and mechanical resistance of the holes in the outer grill. C_{MO} = compliance of the 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} = 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.

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. 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 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.

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.

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 [2] by means of a system consisting of a stretched diaphragm and two carbon buttons as shown in Fig. 8.3. The performance of the system may be obtained from a consideration of the mechanical network of the vibrating system. The 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

[2] Jones, W. C., *Bell Syst. Tech. Jour.*, Vol. 10, No. 1, p. 46, 1931.

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

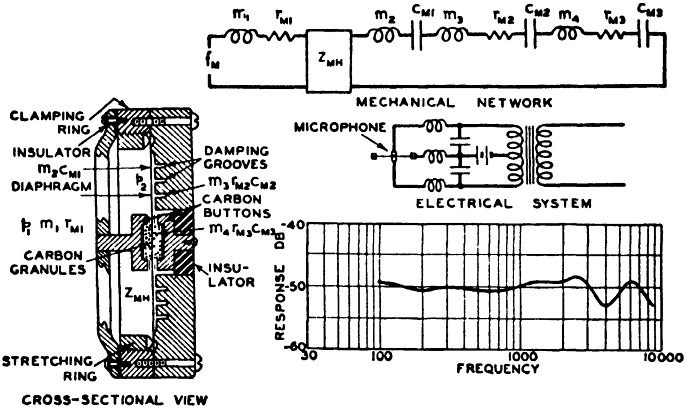

FIG. 8.3. Cross-sectional view, the electrical circuit and the mechanical circuit of a double-button, stretched diaphragm, carbon microphone. In the mechanical circuit: m_1 and r_{M1} = the mass and mechanical resistance of the air load. z_{MH} = the mechanical quadripole representing the cylindrical 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.

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. 8.3.

The electrical circuit diagram for this microphone is shown in Fig. 8.3. 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

$$i_2 - i_1 = \frac{2e}{r_{E0}}\left(\frac{hx}{r_{E0}} \sin \omega t + \frac{h^3x^3}{r_{E0}{}^3} \sin^3 \omega t \ldots\right)$$

$$= \frac{2e}{r_{E0}}\left(\frac{hx \sin \omega t}{r_{E0}} + \frac{3}{4}\frac{h^3x^3}{r_{E0}{}^3} \sin \omega t - \frac{h^3x^3}{4r_{E0}{}^3} \sin 3\omega t \ldots\right) \qquad 8.5$$

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.

B. *Condenser Microphone.* — A condenser microphone is a microphone which depends for its operation on variations in electrical capacitance. The typical condenser microphone[3] consists of a thin stretched plate separated from a parallel rigid plate (Fig. 8.4). The electrical system of this microphone is shown in Fig. 8.4.

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,

[3] Wente, E. C., *Phys. Rev.*, Vol. 10, No. 1, p. 39, 1917.

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.

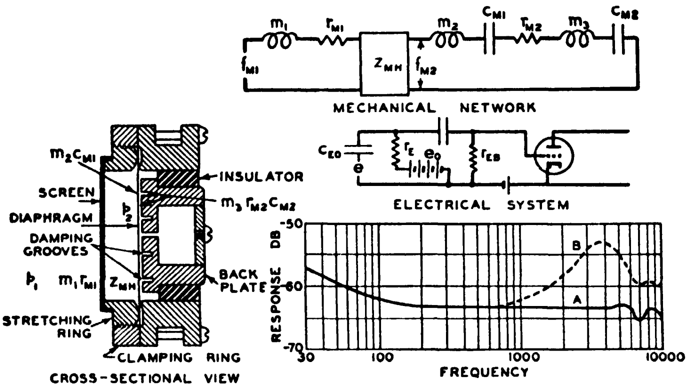

FIG. 8.4. Cross-sectional view, electrical system and mechanical circuit of a condenser microphone. In the electrical system: e_O = the polarizing voltage, r_E = the polarizing electrical resistance, r_{EB} = the bias electrical resistance, C_{EO} = the electrical capacitance of the microphone. In the 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.

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

$$(C_{E0} + C_{E1}\sin\omega t) r_E \frac{di}{dt} + (1 + r_E C_{E1}\omega\cos\omega t)i - e_0 C_{E1}\omega\cos\omega t = 0 \qquad 8.8$$

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[4] 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.

The mechanical network of the mechanical system of the condenser microphone is shown in Fig. 8.4. 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[5,6] 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

[4] Wente, E. C., *Phys. Rev.*, Vol. 19, No. 5, p. 498, 1922.
[5] Crandall, I. B., *Phys. Rev.*, Vol. 11, No. 6, p. 449, 1918.
[6] Crandall, "Vibrating Systems and Sound," D. Van Nostrand Company, New York, N. Y., 1926.

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

$$x = \frac{f_{M2}}{\left[r_{M2} + j\omega(m_2 + m_3) + \frac{1}{j\omega}\left(\frac{1}{C_{M1}} + \frac{1}{C_{M2}}\right)\right] j\omega} \qquad 8.12$$

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 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 constant sound pressure on the diaphragm and for constant free wave sound pressure are shown in Fig. 8.4.

C. *Piezoelectric (Crystal) Microphones.*[7,8,9] — A piezoelectric microphone is a microphone which depends upon the generation of an electromotive force by the deformation of a crystal having piezoelectric properties. The voltage generated[10] 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 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. 8.5, 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. 8.5. 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 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. 8.5. 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.

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[7] Sawyer, C. B., *Proc. Inst. Rad. Eng.*, Vol. 19, No. 11, p. 2020, 1931.
[8] Williams, A. L., *Jour. Soc. Mot. Pic. Eng.*, Vol. 18, No. 4, p. 196, 1934.
[9] Nicolson, U.S. Patent 1,495,429.
[10] Olson, " Dynamical Analogies," D. Van Nostrand Company, New York, N. Y., 1943.

The internal voltage, e, developed by the crystal is

$$e = Kx \tag{8.13}$$

where K = constant of the crystal, and
x = effective amplitude of the deformation of the crystal by an applied force.

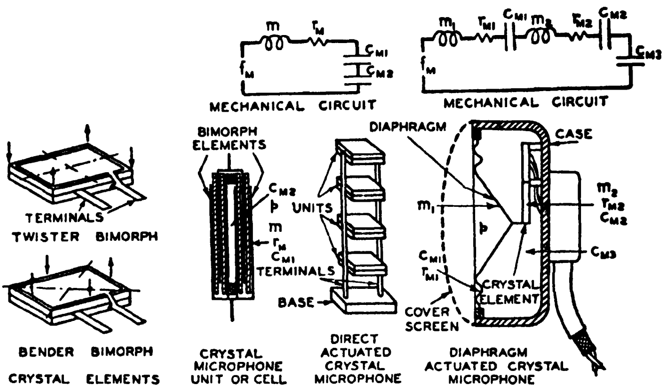

FIG. 8.5. Crystal elements and sound cells. A direct actuated crystal microphone. In the mechanical circuit: m, r_M and C_{M1} = the mass, mechanical resistance and compliance of the crystal. C_{M2} = the compliance of one half of the air chamber. f_M = the driving force. $f_M = pA$. A = the effective 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.

From the mechanical circuit of Fig. 8.5, the amplitude, in centimeters, is

$$x = \frac{f_M}{\left(r_M + j\omega m + \dfrac{1}{j\omega C_{M1}} + \dfrac{1}{j\omega C_{M2}}\right) j\omega} \tag{8.14}$$

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. 8.5, 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 showin in Fig. 8.5. The response frequency characteristic for constant sound pressure on the diaphragm may be obtained from the mechanical circuit of Fig. 8.5 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 wave length. This effect accentuates the response in the high-frequency range.

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).*[11] — A cross-

[11] Wente and Thuras, *Jour. Acous. Soc. Amer.*, Vol. 3, No. 1, p. 44, 1931.

sectional view of a moving coil microphone is shown in Fig. 8.6. The motion of the diaphragm is transferred to a coil located in a magnetic field.

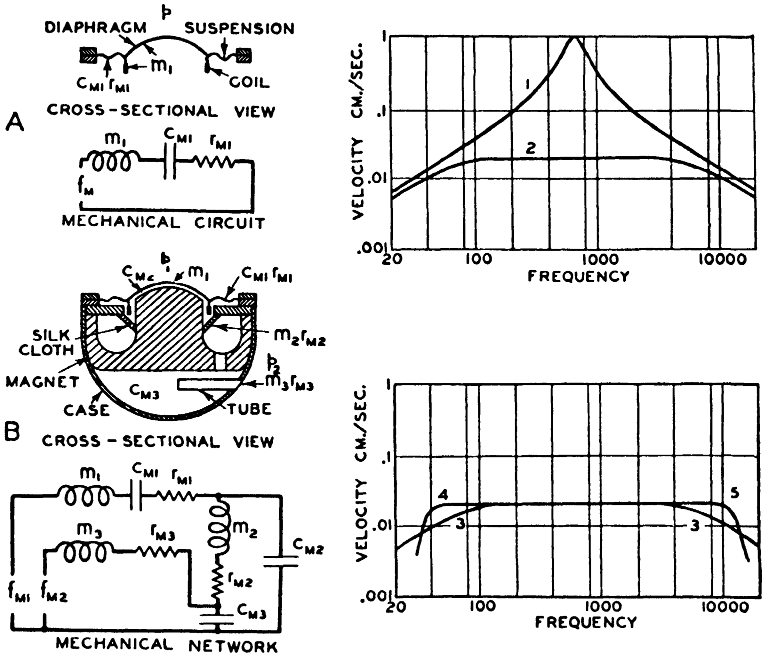

FIG. 8.6. *A*. Cross-sectional view and mechanical circuit of a diaphragm, coil and suspension. In the mechanical circuit: m_1, r_{M1} and C_{M1} = the mass, mechanical resistance and compliance of the vibrating system. f_M = the driving force. $f_M = pA$. A = the area of the diaphragm. p = the sound 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.

The mechanical circuit of the mechanical system consisting of the diaphragm coil and suspension system is shown in Fig. 8.6*A*.

The velocity, in centimeters per second, of the voice coil is given by

$$\dot{x} = \frac{f_M}{r_{M1} + j\omega m_1 + \dfrac{1}{j\omega C_{M1}}} \tag{8.15}$$

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

$$e = Bl\dot{x} \tag{8.16}$$

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. 8.6*A* 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. 8.6*A* shows some falling off in velocity at the high and low frequencies. This can be corrected by the use of some additional elements (Fig. 8.6*B*). 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 microphones 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. 6.17 and Fig. 6.45).

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, of diameter equal to the spherical case directly above the diaphragm and spaced one-fourth inch. Employing this expedient, a nondirectional characteristic[12] is obtained over the response frequency range.

2. *Inductor Microphone*[13] (*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. 8.7. The diaphragm, $r_{M1}C_{M1}m_1$, of this microphone is " V " shaped with a straight conductor located in the bottom of the " V." The mechanical network of this microphone 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[14,15] 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. 8.8. These elements may take various forms as, for example, the pipe is usually coiled in the form of a labyrinth (see Fig. 8.29).

The acoustical network[16] of the pressure ribbon microphone is shown in Fig. 8.8.

The inertance and acoustical capacitance of the ribbon are designated by M_R and C_{AR}.

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[12] Marshall and Romanow, *Bell Syst. Tech. Jour.*, Vol. 15, No. 3, p. 405, 1936.
[13] Olson, H. F., U.S. Patent 2,106,224.
[14] Olson, H. F., U.S. Patent 2,102,736.
[15] Olson, H. F., *Jour. Soc. Mot. Pic. Eng.*, Vol. 27, No. 3, p. 284, 1936.
[16] Olson, H. F., *Broadcast News*, No. 30, p. 3, May, 1939.

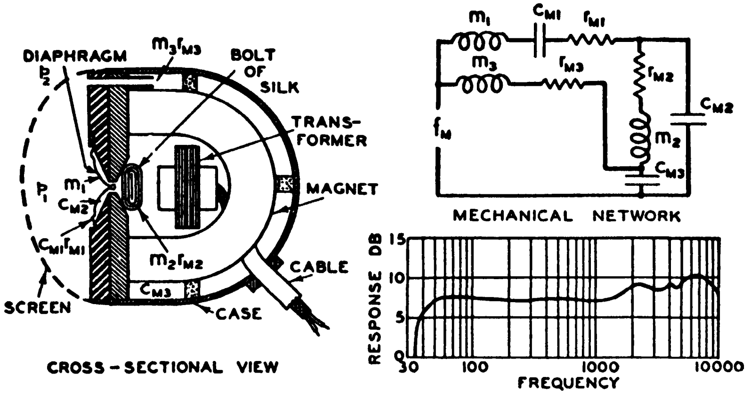

Fig. 8.7. Cross-sectional view and mechanical network of an inductor microphone. In the mechanical network: m_1, r_{M1} and C_{M1} = the mass, mechanical resistance and compliance of the diaphragm and conductor. m_2 and r_{M2} = the mass and mechanical resistance of the bolt of silk. C_{M2} = the compliance 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. $f_{M1} = Ap_1$ and $f_{M2} = Ap_2$. 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.

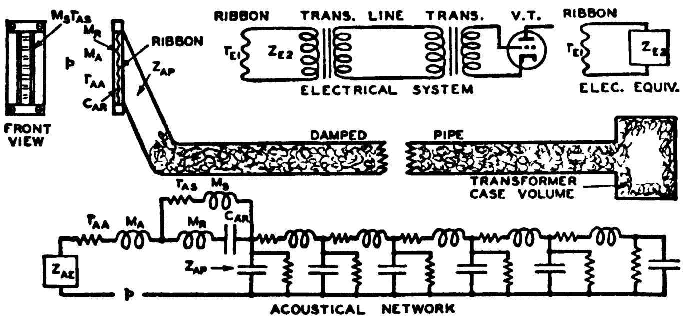

Fig. 8.8. Schematic view, electrical system and its equivalent and acoustical network of a pressure ribbon microphone. In the electrical circuit: r_{E1} = the electrical resistance of the ribbon. z_{E2} = the external electrical impedance load presented to the ribbon. In the acoustical network: M_A and 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 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)

$$p = \frac{dS}{4\pi a} j\rho\omega u_{\max}\epsilon^{j\omega t}\epsilon^{-jka} \tag{8.17}$$

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

$$p = \frac{j\omega\rho}{4\pi} u_{\max}\epsilon^{j\omega t}\int\int \frac{dS}{a_1}\epsilon^{-jka_1} \tag{8.18}$$

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 integration must be performed and then the resulting pressure integrated over the surface of the ribbon.

The total force is

$$f_{MA} = \frac{j\omega\rho u_{\max}\epsilon^{j\omega t}}{4\pi}\int\int dS'\int\int \frac{dS}{a_1}\epsilon^{jka_1} \tag{8.19}$$

where dS' = surface element at 1.

The acoustical impedance due to the air load is

$$z_{AA} = r_{AA} + jx_{AA} = \frac{f_{MA}}{A^2 u_{\max}\epsilon^{j\omega t}} \tag{8.20}$$

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),

$$z_{AS} = r_{AS} + j\omega M_S \tag{8.21}$$

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

$$r_{AP} = \frac{42}{A_P} \tag{8.22}$$

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

$$z_{AE} = \frac{(Bl)^2}{A_R{}^2 z_{ET}} \tag{8.23}$$

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. 8.9.

The volume current of the ribbon, in cubic centimeters per second, is given by

$$U = \frac{p}{r_{AP} + r_{AA} + jx_{AR} + jx_{AA} - jx_{AP}} \tag{8.24}$$

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. 8.9. The velocity of the ribbon, in centimeters per second, is

$$\dot{x} = \frac{U}{A_R} \tag{8.25}$$

The voltage, in abvolts, generated in the ribbon is given by

$$e = Bl\dot{x} \tag{8.26}$$

where B = flux density, in gausses, and
l = length of the ribbon, in centimeters.

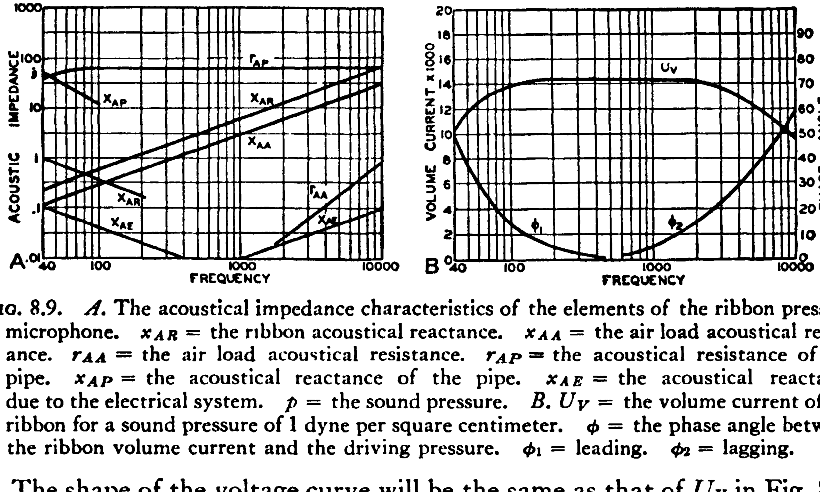

FIG. 8.9. *A*. The acoustical impedance characteristics of the elements of the ribbon pressure microphone. x_{AR} = the ribbon acoustical reactance. x_{AA} = the air load acoustical reactance. r_{AA} = the air load acoustical resistance. r_{AP} = the acoustical resistance of the pipe. x_{AP} = the acoustical 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.

The shape of the voltage curve will be the same as that of U_V in Fig. 8.9. 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.

E. *Electronic Microphone.*[17] — 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. 8.10. The voltage output of an electronic transducer is given by

$$e = Kx_3 \tag{8.27}$$

where K = constant of the system, and
x_3 = amplitude of the element.

[17] Olson, H. F., *Jour. Acous. Soc. Amer.* Vol. 19, No. 2, p. 307, 1947.

The output of the electronic microphone may be computed from the mechanical network of Fig. 8.10.

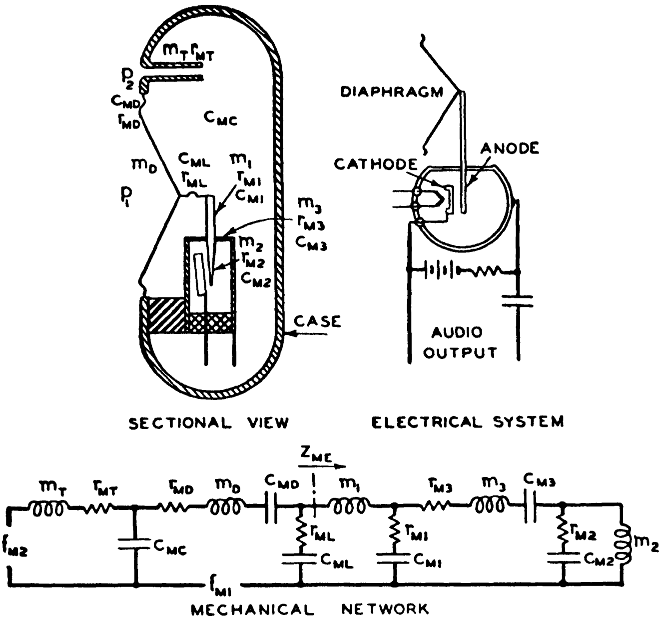

FIG. 8.10. Sectional view, electrical system, and mechanical network of an electronic microphone. In the mechanical network: z_{ME} = the mechanical impedance of the electronic transducer. m_1, r_{M1} and C_{M1} = the effective mass, mechanical resistance and compliance of the outer portion of the bar. 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. 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 opening. The electrical system shows the wiring diagram for a diode type electronic transducer.

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

$$f_{M1} = p_1A_D \tag{8.28}$$

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

$$f_{M2} = p_2 A_D \qquad 8.29$$

where A_D = area of the diaphragm, in square centimeters, and
p_2 = sound pressure at the port, in dynes per square centimeter.

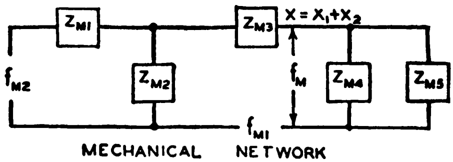

Fig. 8.11. Mechanical network of Fig. 8.10 in terms of the following:

$$z_{M1} = r_{MT} + j\omega m_T$$
$$z_{M2} = \frac{1}{j\omega C_{MC}}$$
$$z_{M3} = r_{MD} + j\omega m_D + \frac{1}{j\omega C_{MD}}$$
$$z_{M4} = r_{ML} + \frac{1}{j\omega C_{ML}}$$
$$z_{M5} = z_{ME}$$

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.

The response may be obtained from a consideration of the mechanical network. The mechanical network of Fig. 8.10 may be reduced to the mechanical network of Fig. 8.11 in which

$$z_{M1} = r_{MT} + j\omega m_T \qquad 8.30$$

$$z_{M2} = \frac{1}{j\omega C_{MC}} \qquad 8.31$$

$$z_{M3} = r_{MD} + j\omega m_D + \frac{1}{j\omega C_{MD}} \qquad 8.32$$

$$z_{M4} = r_{ML} + \frac{1}{j\omega C_{ML}} \qquad 8.33$$

$$z_{M5} = z_{ME} \qquad 8.34$$

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

$$z_{ME} = z_{M5} = \frac{z_{M6}(z_{M7}+z_{M8})(z_{M9}+z_{M10})+z_{M9}z_{M10}(z_{M6}+z_{M7})+z_{M7}z_{M8}(z_{M9}+z_{M10})}{(z_{M7}+z_{M8})(z_{M9}+z_{M10})+z_{M9}z_{M10}} \qquad 8.35$$

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

$$x_1 = \frac{f_{M1}}{j\omega\left(z_{M3} + \dfrac{z_{M1}z_{M2}}{z_{M1}+z_{M2}} + \dfrac{z_{M4}z_{M5}}{z_{M4}+z_{M5}}\right)} \qquad 8.36$$

The amplitude, in centimeters, due to the driving force f_{M2} is

$$x_2 = \frac{-f_{M2}z_{M2}(z_{M4}+z_{M5})}{j\omega[z_{M1}(z_{M2}+z_{M3})(z_{M4}+z_{M5})+z_{M4}z_{M5}(z_{M1}+z_{M2})+z_{M2}z_{M3}(z_{M4}+z_{M5})]} \qquad 8.37$$

The amplitude, in centimeters, of m_2 is

$$x_3 = \frac{f_M z_{M7} z_{M9}}{j\omega[z_{M6}(z_{M7}+z_{M8})(z_{M9}+z_{M10})+z_{M9}z_{M10}(z_{M6}+z_{M7})+z_{M7}z_{M8}(z_{M9}+z_{M10})]} \qquad 8.38$$

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 8.38 and the constants of the system. The voltage output can be obtained from the amplitude and equation 8.27.

The electrical connections for the electronic microphone are shown in Fig. 8.10.

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 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.*[18,19,20,21] — The response of a pressure gradient microphone, as the name implies, is a function of the difference 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 8.12 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

[18] Olson, H. F., *Jour. Soc. Mot. Pic. Eng.*, Vol. 16, No. 6, p. 695, 1931.
[19] Olson, H. F., *Jour. Acous. Soc. Amer.*, Vol. 3, No. 1, p. 56, 1931.
[20] Olson, H. F., *Proc. Inst. Rad. Eng.*, Vol. 21, No. 5, p. 655, 1933.
[21] Massa, F., *Jour. Acous. Soc. Amer.*, Vol. 10, No. 3, p. 173, 1939

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.

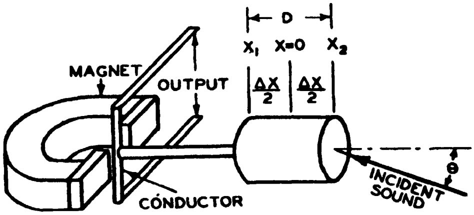

FIG. 8.12. Pressure gradient microphone.

Assume a plane sound wave, from equation 1.22, the pressure, in dynes per square centimeter, at $x = 0$ may be written

$$p = kc\rho A \sin (kct)$$
$$p = p_m \sin kct \qquad 8.39$$

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

$$p_1 = p_m \sin k\left(ct + \frac{\Delta x}{2}\cos\theta\right) \qquad 8.40$$

The pressure at the other end of the cylinder $x_2 = \Delta x/2$ is

$$p_2 = p_m \sin k\left(ct - \frac{\Delta x}{2}\cos\theta\right) \qquad 8.41$$

The difference in pressure between the two ends of the cylinder is

$$\Delta p = p_1 - p_2 = 2p_m \cos (kct) \sin\left(\frac{k\Delta x}{2}\cos\theta\right) \qquad 8.42$$

The driving force, in dynes, available for driving the cylinder along the x axis is

$$f_M = S\Delta p = 2Sp_m \cos (kct) \sin \left(\frac{k\Delta x}{2} \cos \theta\right) \qquad 8.43$$

where S = area of the end of the cylinder, in square centimeters.

If Δx is small compared to the wavelength the driving force is

$$f_M = S \frac{2\pi f}{c} p_m \Delta x \cos \theta \cos kct \qquad 8.44$$

A comparison of equations 8.39 and 8.44 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

$$\dot{x} = \frac{f_M}{j\omega m} = \frac{Sp_m}{jcm} \Delta x \cos \theta \cos kct = \frac{Sp_m}{cm} \Delta x \cos \theta \sin kct \qquad 8.45$$

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

$$\dot{x} = \frac{2Sp_m}{m\omega} \sin (kct) \sin \left(\frac{k\Delta x}{2} \cos \theta\right) \qquad 8.46$$

$$\dot{x} = \frac{2Sp_m}{m\omega} \sin (kct) \sin \left(\frac{kD}{2} \cos \theta\right) \qquad 8.47$$

where D = distance between the two ends of the cylinder.

The voltage output, in abvolts, of the conductor is

$$e = Bl\dot{x} \qquad 8.48$$

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 8.47 and 8.48 is shown in Fig. 8.13.

The directional characteristics of a pressure gradient system of the type shown in Fig. 8.12 and computed from equation 8.47 are shown in Fig. 8.14. 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

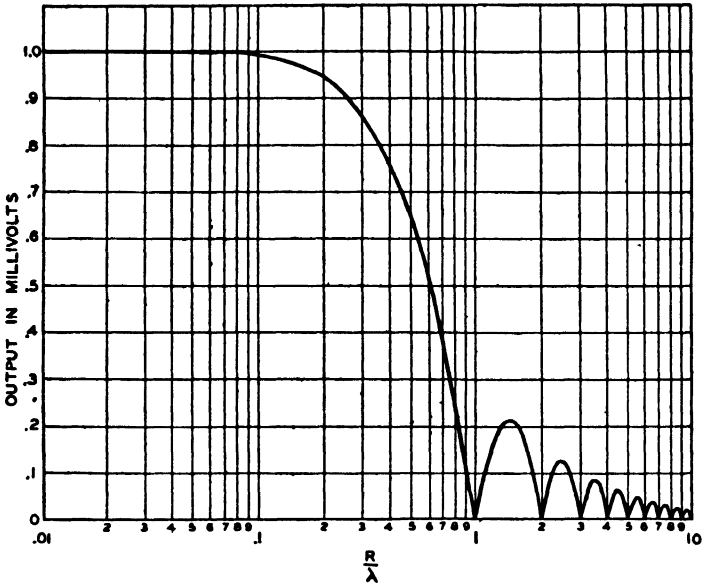

FIG. 8.13. Computed open-circuit voltage response frequency characteristic of a pressure gradient, mass controlled, electrodynamic microphone.

case of the baffle type ribbon microphone, the directional characteristics 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

$$p = \frac{kc\rho A}{r} \sin k(ct - r) \qquad 8.49$$

Let the distance on the axis of the cylinder between the source and points

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

$$\Delta p = kc\rho A \left[\frac{2r \cos k(ct - r) \sin \left(\frac{kD}{2}\right) + 2D \sin k(ct - r) \cos \left(\frac{kD}{2}\right)}{r^2 - \left(\frac{D}{2}\right)^2} \right] \qquad 8.50$$

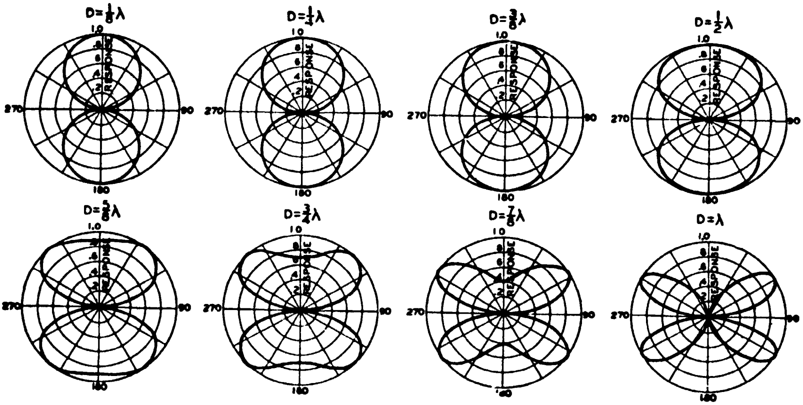

Fig. 8.14. Directional characteristics of a pressure gradient microphone as a function of the dimensions and the wavelength. The polar graph depicts the output, in volts, as a function of the angle, in degrees. The maximum response is arbitrarily chosen as unity.

If D is small compared to r and kD is small compared to unity, equation 8.50 becomes approximately

$$\Delta p = kc\rho AD \left[\frac{kr \cos k(ct - r) + \sin k(ct - r)}{r^2} \right] \qquad 8.51$$

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. 8.31*A*.

B. *Velocity Microphone.*[22,23,24] — Free ribbon microphones are used for all types of sound collection. Essentially these microphones consist of a

[22] Olson, H. F., *Jour. Soc. Mot. Pic. Eng.*, Vol. 16, No. 6, p. 695, 1931.
[23] Olson, H. F., *Jour. Acous. Soc. Amer.*, Vol. 3, No. 1, p. 56, 1931.
[24] Olson, H. F., *Proc. Inst. Rad. Eng.*, Vol. 21, No. 5, p. 655, 1933.

loosely stretched ribbon suspended in the air gap between two pole pieces (Fig. 8.15). 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

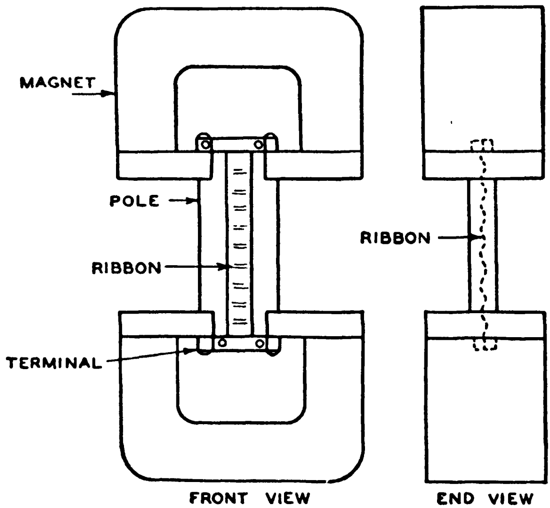

FIG. 8.15. The essential elements of a velocity microphone.

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 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 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.[25] 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

$$\frac{p_\theta}{p} = 1 + \frac{\cos\theta}{\sqrt{1-\sin^2\theta}}\left[1 - \epsilon^{jkR}\sum_{u=0}^{\infty}\epsilon_u j^u \mu^u J_u\,(kR\sin\theta)\right] \qquad 8.52$$

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.

The pressure at the center on the front and back of a circular plate for normal incidence $\theta = 0°$ or $180°$, from equation 8.52, is

$$\left|\frac{p_0}{p}\right| = \sqrt{5 - 4\cos kR} \qquad 8.53$$

$$\left|\frac{p_{180}}{p}\right| = 1 \qquad 8.54$$

The pressure frequency characteristic on the front and back of a circular baffle for normal incidence computed from equations 8.53 and 8.54 is shown in Fig. 8.16. It will be seen that the pressure at the front rises to a value 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. 8.17. 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. 8.17 represent an average of these determinations. It will be seen that theory and experiment are in fairly

[25] Sivian and O'Neil, *Jour. Acous. Soc. Amer.*, Vol. 3, No. 4, p. 483, 1932.

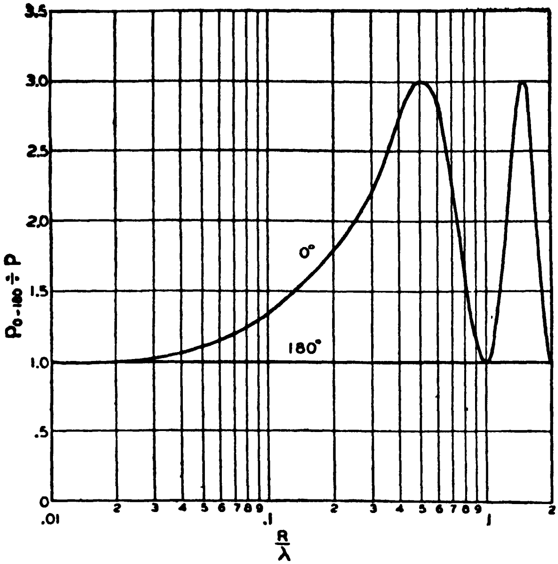

FIG. 8.16. Computed pressure frequency characteristic, at the center, on the front and the back of a circular baffle for normal incidence of the impinging sound wave.

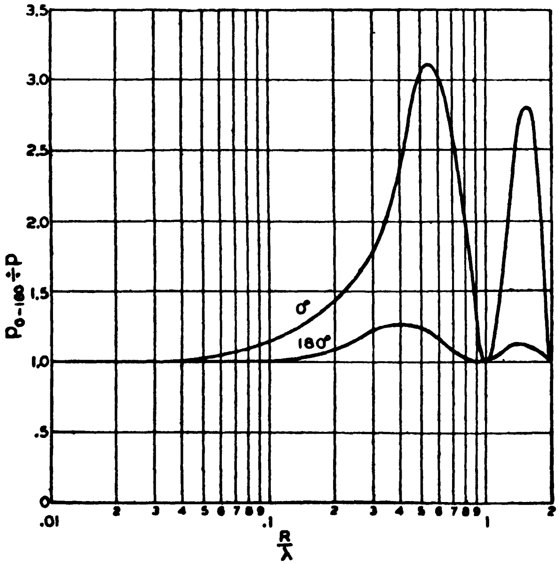

FIG. 8.17. Measured pressure frequency characteristic at the center, on the front and the back of a circular baffle for normal incidence of the impinging sound wave.

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 8.52 are shown in Fig. 8.18. A point in the plane wave cor-

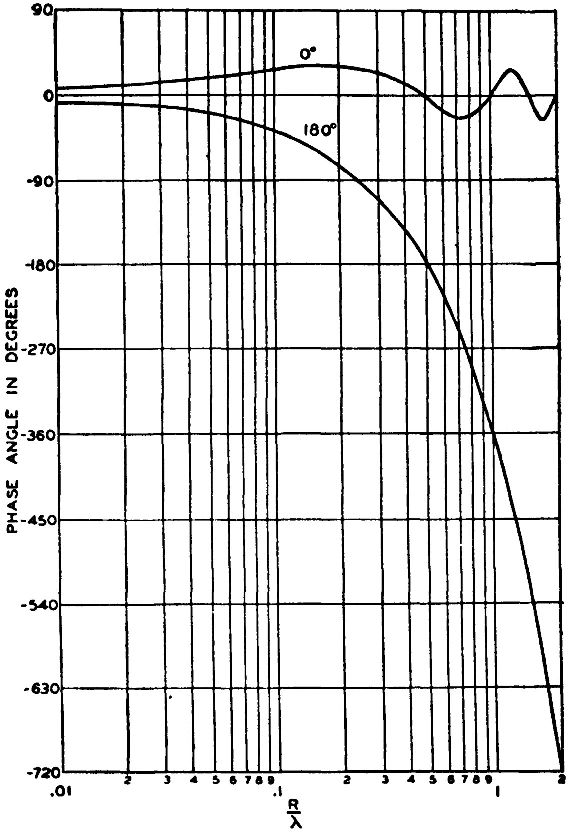

FIG. 8.18. Computed phase frequency characteristic at the center, on the front and the back of a circular baffle for normal incidence of the impinging sound wave.

responding 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 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 8.52 may be used to compute the difference in pressure between the two sides of a relatively small ribbon located in a large baffle (Fig. 8.19). The difference in pressure between the two sides of the ribbon in a circular baffle, Fig. 8.19, is

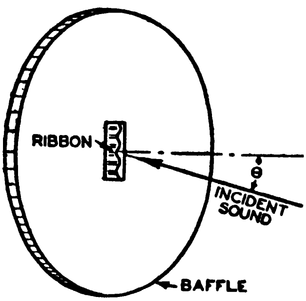

FIG. 8.19. A velocity microphone with a large circular baffle.

$$\Delta p = p_\theta - p_{\theta+180} \qquad 8.55$$

where p_θ and $p_{\theta+180}$ may be obtained from equation 8.52. The acoustical impedance of the ribbon, Fig. 8.19, is given by

$$z_{AR} = j\omega M_R - \frac{j}{\omega C_{AR}} \qquad 8.56$$

where M_R = inertance of the ribbon, and

C_{AR} = acoustical capacitance of the ribbon.

From equation 8.19 the total force of the air load upon the ribbon is

$$f_{MA} = \frac{j\omega\rho u_{\max}\epsilon^{j\omega t}}{4\pi} \iint dS' \iint \frac{dS}{a_1}\epsilon^{jka_1} \qquad 8.57$$

The above integration extends over both sides of the ribbon and cognizance must be taken of the 180° difference in phase between the front and back when integrating between the two surfaces. The integration of equation 8.57 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

$$z_{AA} = r_{AA} + jx_{AA} = \frac{f_{MA}}{A_R{}^2 u_{\max}\epsilon^{j\omega t}} \qquad 8.58$$

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. 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 rectance term in equation 8.56 may be neglected. Under these conditions the acoustical impedance of the vibrating system is

$$z_{AT} = j\omega M_R + j\omega M_{AA} \qquad 8.59$$

where M_{AA} = inertance of the air load.

The velocity, in centimeters per second, of the ribbon is

$$\dot{x} = \frac{\Delta p}{A_R z_{AT}} \qquad 8.60$$

where A_R = area of the ribbon, in square centimeters.

The voltage output in abvolts is

$$e = Bl\dot{x} \qquad 8.61$$

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. 8.19, computed from equation 8.61 is shown in Fig. 8.20.

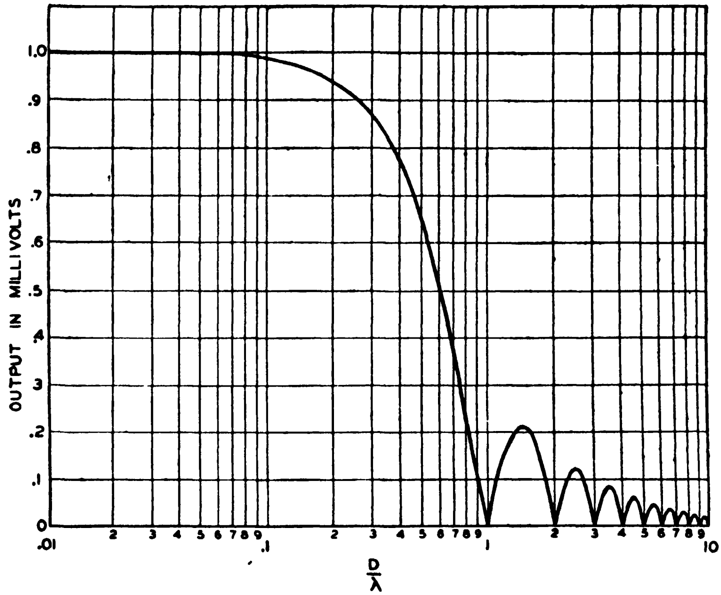

FIG. 8.20. Computed open-circuit voltage response frequency characteristic of a mass controlled, electrodynamic ribbon located in a large circular baffle.

The experimental response of a ribbon microphone with a circular baffle is shown in Fig. 8.21. The agreement between the measured response and the computed response is quite good. There is some deviation between

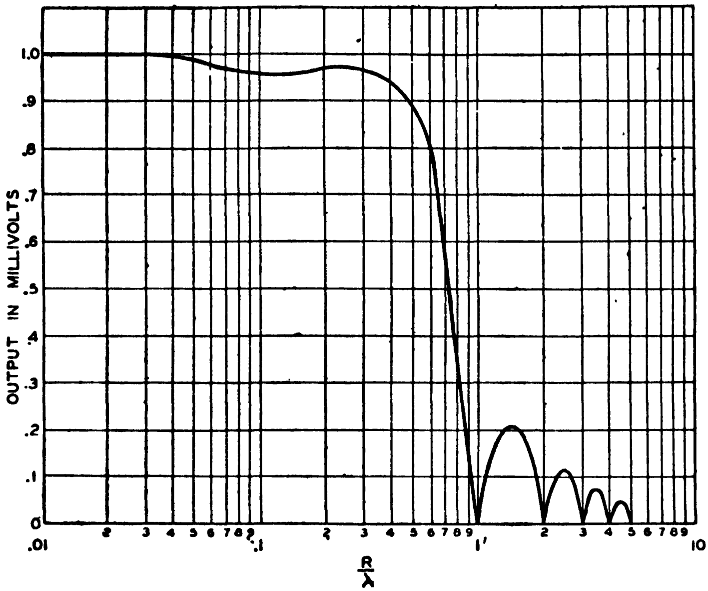

FIG. 8.21. Measured open-circuit voltage response frequency characteristic of a mass controlled, electrodynamic ribbon located in a large circular baffle.

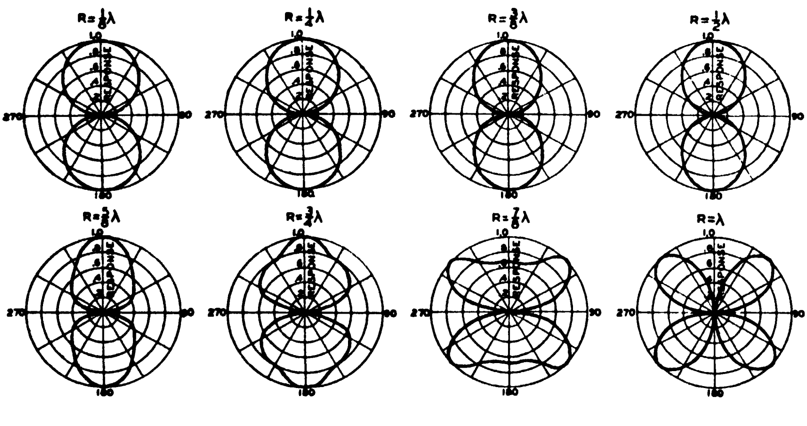

FIG. 8.22. Measured directional characteristics of a ribbon microphone with a large circular baffle (see Fig. 8.19) as a function of the radius of the baffle and the wavelength. The polar graph depicts the output, in volts, as a function of the angle, in degrees. The maximum response is arbitrarily chosen as unity.

$R/\lambda = .5$ and $R/\lambda = .8$. There is also some discrepancy in this region between computed and measured pressures (Figs. 8.16 and 8.17). 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. 8.13 and 8.20.

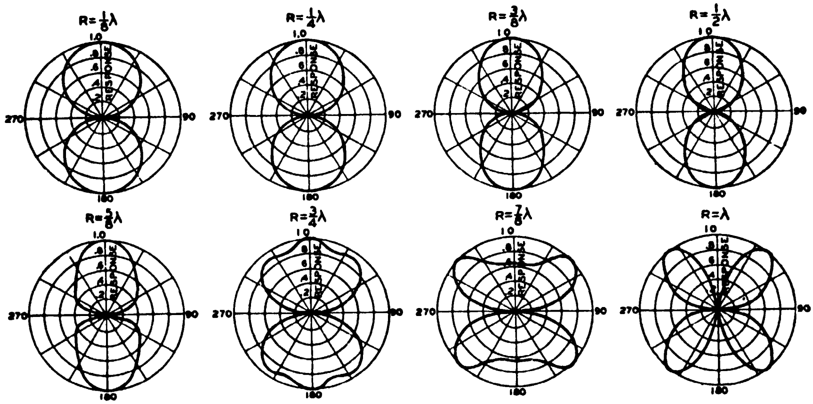

FIG. 8.23. Computed directional characteristics of a ribbon microphone with a large circular baffle (see Fig. 8.19) as a function of the radius of the baffle and the wavelength. The polar graph depicts the output, in volts, as a function of the angle, in degrees. The maximum response is arbitrarily chosen as unity.

The measured directional characteristic of the ribbon microphone with a circular baffle is shown in Fig. 8.22. 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 8.52, 8.55, 8.60 and 8.61 are shown in Fig. 8.22. It will be seen that the agreement with the experimental results of Fig. 8.23 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. 8.16 and 8.17, and for the response, Figs. 8.20 and 8.21. The theoretical directional characteristic for a doublet, Fig. 8.14, 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 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 8.55, and the particle velocity in a plane wave, for a ribbon microphone with a circular baffle, is shown in Fig. 8.24. It will be seen that this force leads the particle velocity

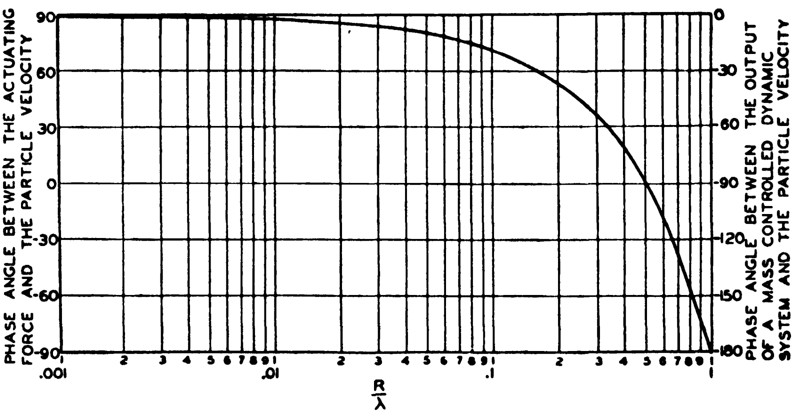

FIG. 8.24. The phase angle, in degrees, between the actuating force and the particle velocity for a mass controlled ribbon with a circular baffle as a function of R/λ. The phase angle between the voltage output of a mass controlled, electrodynamic ribbon located in a magnetic field as a function of R/λ.

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. 8.24. 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.

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

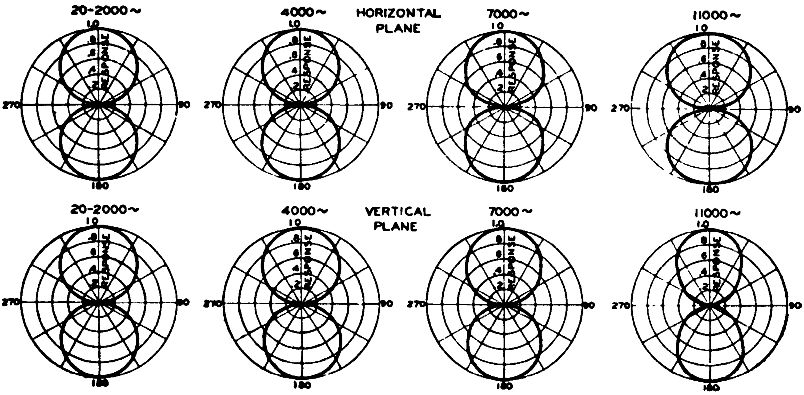

Fig. 8.25. The directional characteristic of the velocity microphone shown in Fig. 8.15. The polar graph depicts the output, in volts, as a function of the angle, in degrees. The maximum response is arbitrarily chosen as unity.

is shown in Fig. 8.15. It will be seen that the effective baffle is irregular in shape. The directional characteristics of the microphone of Fig. 8.15 are shown in Fig. 8.25. 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. 8.31*A*.

The response of the baffle type velocity microphone may be obtained from the acoustical network of Fig. 8.26 and the acoustical impedance characteristics of the acoustical elements of the system of Fig. 8.27. The ribbon is 2.2 inches in length and .2 inch in width. The flux density is 9000 gausses. The computed voltages are indicated by the dots in Fig.

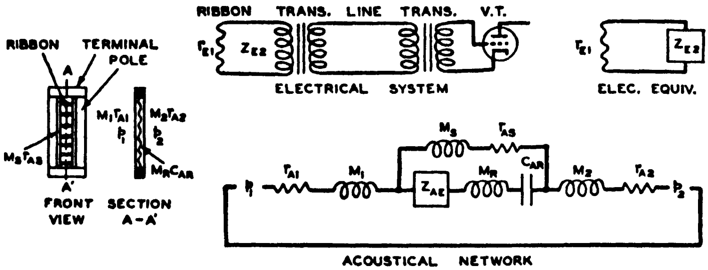

FIG. 8.26. Front and sectional views of the vibrating system of a velocity microphone. In the electrical circuit: r_{E1} = the electrical resistance of the ribbon. z_{E2} = the electrical impedance of the load upon the ribbon due to the transformers and vacuum tube. In the acoustical network: r_{A1}, M_{A1}, r_{A2} and M_{A2} = 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 pressures at the front and back of the ribbon.

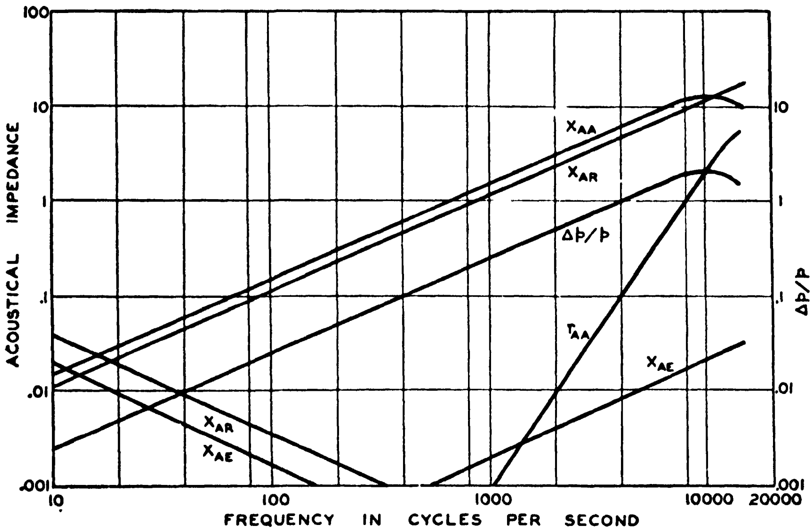

FIG. 8.27. The acoustical impedance characteristics of the velocity microphone. x_{AR} = the positive and negative acoustical reactances due to the inertance and the acoustical capacitance of the ribbon. x_{AA} = the acoustical reactance due to the air load. r_{AA} = the acoustical resistance due to 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.

8.28. The experimental response frequency characteristic of the microphone of Fig. 8.15 is shown in Fig. 8.28. 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. 8.28 depict the open-circuit

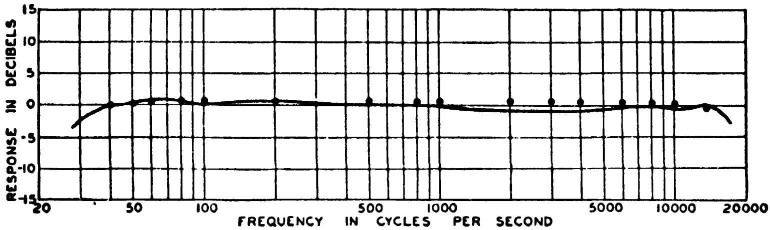

FIG. 8.28. Open-circuit voltage response frequency characteristics of a velocity microphone at the terminals of the 250-ohm output. 0 db = 600 microvolts per dyne per square centimeter. When the impedance is stepped up to 50,000 ohms for the input to the grid of a vacuum tube, the voltage is 8.4 millivolts per dyne per square centimeter. The dots represent the computed response and the solid line the measured response.

voltage from the terminals of the 250-ohm transformer of Fig. 8.26. 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. 8.26 is 23 db above the voltage in the line. This is .0085 volt per dyne per square centimeter at the grid of the vacuum tube.

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 [26,27,28] consists of a bidirectional microphone and a nondirectional microphone. A unidirectional microphone consisting of a ribbon velocity element (see Sec. 8.3*B* and Fig. 8.15) and a ribbon pressure element (see Sec. 8.2*D*3 and Fig. 8.8) is shown in Fig. 8.29. The damped

[26] Olson, H. F., *Jour. Acous. Soc. Amer.*, Vol. 3, No. 3, p. 315, 1932.
[27] Weinberger, Olson and Massa, *Jour. Acous. Soc. Amer.*, Vol. 5, No. 2, p. 139, 1933.
[28] Olson, H. F., *Jour. Soc. Mot. Pic. Eng.*, Vol. 27, No. 3, p. 284, 1936.

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

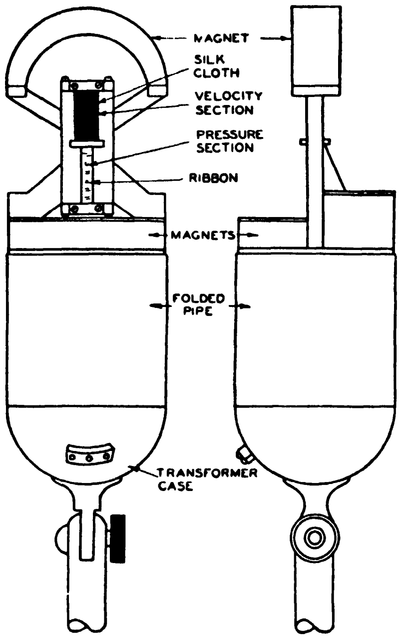

FIG. 8.29. Unidirectional microphone with the screen removed. Ribbon type pressure and velocity elements.

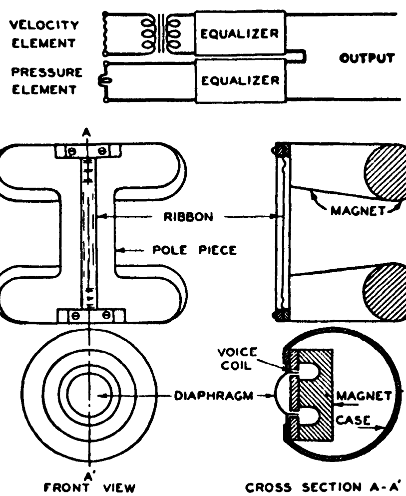

FIG. 8.30. Unidirectional microphone consisting of a ribbon type velocity element and a dynamic type pressure element.

pressure in the sound wave at the low frequencies (see Sec. 8.2*D*3 and Fig. 8.9). 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.

A unidirectional microphone [29] consisting of a ribbon velocity element and a dynamic pressure element is shown in Fig. 8.30. Equalizers are used to correct the amplitude and phase of the dynamic element to conform with the velocity element.

[29] Marshall and Harry, *Jour. Acous. Soc. Amer.*, Vol. 12, No. 4, p. 481, 1941.

1. *The Response of the Unidirectional Microphone as a Function of the Distance and the Frequency.*[30] — 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

$$e_{ND} = \frac{R_1}{r} \sin \omega t \qquad 8.62$$

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.

The voltage output of the bidirectional velocity microphone as a function of the distance and the wavelength λ, in centimeters, is

$$e_{BD} = R_2 \left(\frac{1}{r} \sin \omega t - \frac{\lambda}{2\pi r^2} \cos \omega t \right) \cos \theta \qquad 8.63$$

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

$$\text{Response Ratio} = \sqrt{\frac{\left(\frac{1}{r} + \frac{\cos \theta}{r}\right)^2 + \left(\frac{\lambda \cos \theta}{2\pi r^2}\right)^2}{\left(\frac{2}{r}\right)^2}} \qquad 8.64$$

This ratio for θ = 0, 30°, 60°, 90°, 120°, 150° and 180° for 1, 2 and 5 feet is shown in Fig. 8.31. The same ratio for a conventional velocity micro-

[30] Olson, H. F., *Broadcast News*, No. 30, p. 3, May, 1939.

phone for 1, 2 and 5 feet is shown in Fig. 8.31. It will be seen that the accentuation in the unidirectional microphone is smaller than in the case of the velocity microphone.

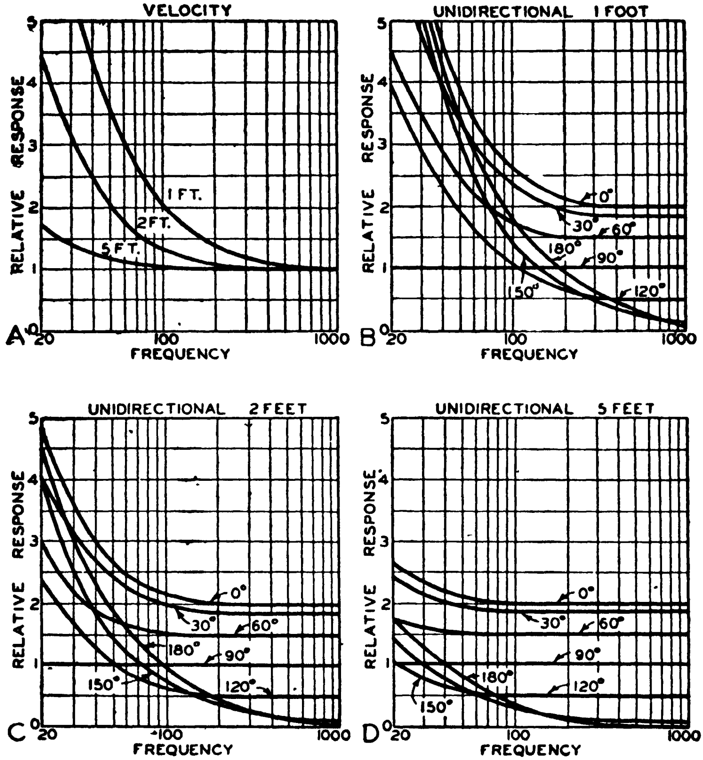

Fig. 8.31. *A*. The relative voltage output of a velocity (or pressure gradient) microphone as compared to a nondirectional pressure microphone for distances of 1, 2 and 5 feet. *B*, *C*, *D*, the relative voltage output of a unidirectional microphone as compared to a nondirectional pressure microphone for distances of 1, 2 and 5 feet and for various angles of the incident sound.

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.*[31] — 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 non-

[31] Olson, H. F., *Broadcast News*, No. 30, p. 3, May, 1939.

directional 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

$$e_{UD} = R_1 + R_2 \cos \theta \qquad 8.65$$

where R_1 = voltage output of the nondirectional microphone, and
R_2 = voltage output of the bidirectional unit for $\theta = 0$.

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 8.106, is

$$\text{Directional efficiency} = \frac{2\pi \int_0 (R_1 + R_2 \cos \theta)^2 \sin \theta \, d\theta}{4\pi (R_1 + R_2)^2}$$

$$= \frac{1}{6} \frac{(R_1 + R_2)^3 - (R_1 - R_2)^3}{(R_1 + R_2)^2 R_2} \qquad 8.66$$

For the standard velocity microphone $R_1 = 0$, $R_2 = 1$ and the ratio is $\frac{1}{3}$. For the cardioid unidirectional $R_1 = 1$ and $R_2 = 1$ 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. 8.32. The data in Fig. 8.32 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. 8.33 by means of polar diagrams. Fig. 8.33 shows that the energy response of the bidirectional microphone 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.*[32] — The

[32] Olson, H. F., *Broadcast News*, No. 30, p. 3, May, 1939.

preceding discussions have assumed that the phase angle between the outputs of the two units did not change with frequency. There are two

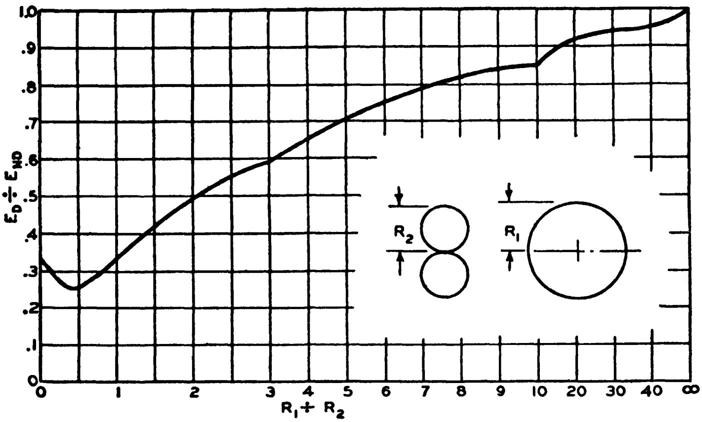

FIG. 8.32. The ratio of energy response to random sounds of a directional microphone consisting of a bidirectional and a nondirectional unit as a function of the ratio of the outputs of the elements, as compared to the nondirectional microphone. E_{ND} = energy response of a nondirectional microphone. E_D = energy response of a directional microphone. R_1 = voltage output of the nondirectional unit. R_2 = voltage output of the directional unit.

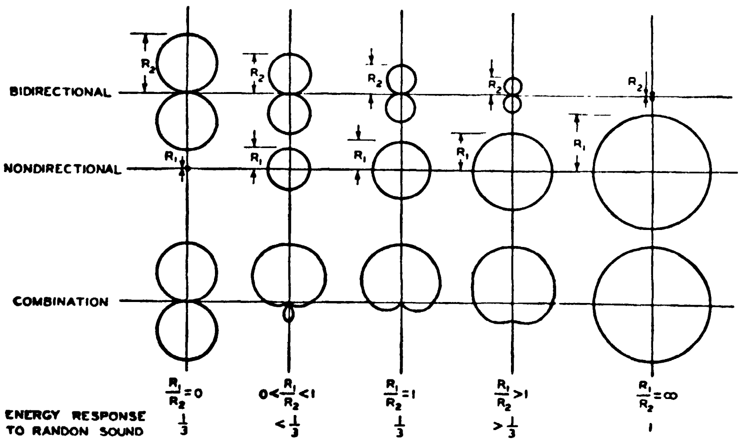

FIG. 8.33. Directional diagrams of various combinations of bidirectional and nondirectional microphones and the energy response to random sounds.

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

$$e = e_0\sqrt{(\cos\theta + \cos\phi)^2 + (\sin\phi)^2} \qquad 8.67$$

The efficiency of the energy response of the above system to that of a nondirectional microphone is

Directional efficiency =

$$\frac{2\pi e_0^2\int_0^\pi [(\cos^2\theta + 2\cos\theta\cos\phi + \cos^2\phi) + \sin^2\phi]\sin\theta\, d\theta}{16\pi e_0^2} \qquad 8.68$$

Directional efficiency =

$$\frac{2\pi e_0^2\int_0^\pi [\cos^2\theta + 2\cos\theta\cos\phi + 1]\sin\theta\, d\theta}{16\pi e_0^2} = \frac{1}{3} \qquad 8.69$$

The efficiency is the same as in the case of no phase angle shift.

If the units are separated by a finite distance d, then there will be a phase difference between the units which is

$$\phi = \frac{d}{\lambda} 360\cos\theta \qquad 8.70$$

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 8.67 the output is

$$e = e_0\sqrt{[\cos\theta + \cos(K\cos\theta)]^2 + [\sin(K\cos\theta)]^2} \qquad 8.71$$

The efficiency of the energy response of the above system to a nondirectional system is given by

Directional efficiency =

$$\frac{2\pi e_0^2\int_0^\pi \{[\cos\theta + \cos(K\cos\theta)]^2 + [\sin(K\cos\theta]^2\}\sin\theta\, d\theta}{16\pi e_0^2} = \frac{1}{3} \qquad 8.72$$

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. 8.3*B*. The phase shift in a pressure ribbon microphone has been considered in Sec. 8.2*D*3. 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.* — Unidirectional microphones consisting of the combination of a nondirectional and a bidirectional microphones have been described in Sec. 8.4*A*. 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 electroacoustic 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. 8.34. The open ends of the pipes are separated by a distance D. A bend of 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

$$\Delta p = 2p_0 \sin\left(\frac{d - D}{\lambda}\pi + \frac{D\pi}{\lambda}\cos\theta\right) \qquad 8.73$$

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.

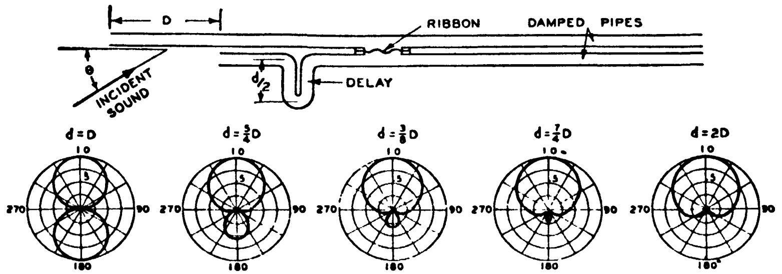

Fig. 8.34. A directional microphone employing a phase shifting system. The polar graphs show the directional characteristics for various ratios of d/D. The polar graph depicts the output, in volts, as a function of the angle, in degrees. The maximum response is arbitrarily chosen as unity.

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.

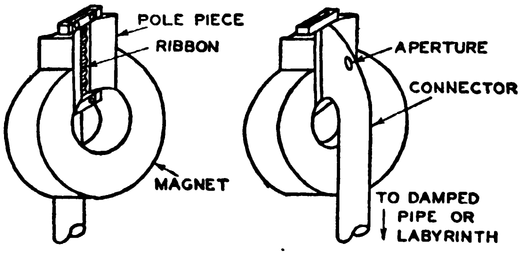

Fig. 8.35. The elements of a single ribbon polydirectional microphone.

A series of directional characteristics for various ratios of D to d is shown in Fig. 8.34.

2. *Polydirectional Microphone.* — The single element polydirectional microphone [33] is shown in Fig. 8.35. The ribbon is located in the air gap

[33] Olson, H. F., *Proc. Inst. Rad. Eng.*, Vol. 32, No. 2, p. 77, 1944.

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.

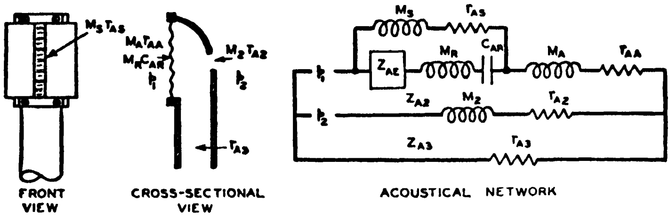

FIG. 8.36. Front view, cross-sectional view and the acoustical network of a polydirectional microphone. In the acoustical network: M_R and C_{AR} = the inertance and acoustical capacitance of the ribbon. M_A and r_{AA} = the inertance and acoustical resistance of the air load on the front of the ribbon. 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. p_1 = the sound pressure at the front of the ribbon. p_2 = the sound pressure at the back of the connector.

The action of this microphone can be obtained from Fig. 8.36 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

$$p_1 = p_{01}\epsilon^{j(\omega t+\phi_1)} \qquad 8.74$$

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.

The sound pressure acting on the aperture in the labyrinth connector may be written

$$p_2 = p_{02}\epsilon^{j(\omega t+\phi_2)} \qquad 8.75$$

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.

The reference point for the phase may be changed so that

$$p_1 = p_{01}\epsilon^{j(\omega t)} \qquad 8.76$$

and

$$p_2 = p_{02}\epsilon^{j(\omega t + \phi_3)} \tag{8.77}$$

The phase angle ϕ_3 is a function of the angle of the incident sound as follows:

$$\phi_3 = \phi \cos \theta \tag{8.78}$$

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

$$\dot{X}_1 = \frac{p_1(z_{A2} + z_{A3})}{z_{A1}z_{A2} + z_{A1}z_{A3} + z_{A2}z_{A3}} \tag{8.79}$$

where $z_{A1} = r_{AA} + j\omega M_A +$

$$\frac{r_{AS} - \omega^2 r_{AS} M_R C_{AR} + j\omega M_S - j\omega^3 M_S M_R C_{AR} - \omega M_S C_{AR} z_{AE} + j\omega C_{AR} r_{AS} z_{AE}}{1 - \omega^2 C_{AR}(M_R + M_S) + j\omega r_{AS} C_{AR} + j\omega C_{AR} z_{AE}}$$

$z_{A2} = r_{A2} + j\omega M_2$

$z_{A3} = r_{A3}$

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

$$\dot{X}_2 = \frac{p_2(z_{A3})}{z_{A1}z_{A2} + z_{A1}z_{A3} + z_{A2}z_{A3}} \qquad 8.80$$

The resultant volume current, in cubic centimeters per second, $\dot{X}_R$, of the the ribbon is the difference between equations 8.79 and 8.80,

$$\dot{X}_R = \dot{X}_1 - \dot{X}_2 \qquad 8.81$$

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. 8.36).

In Fig. 8.37*A* 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 8.79, 8.80, and 8.81 the volume current of the ribbon is

$$\dot{X}_R = \dot{X}_1 - \dot{X}_2 = \frac{(p_1 - p_2)}{z_{A1}} \qquad 8.82$$

If the amplitudes of p_1 and p_2 are equal, then

$$\dot{X}_R = \frac{(p_{01} - p_{01}\epsilon^{j\phi\cos\theta})\epsilon^{j\omega t}}{z_{A1}} \qquad 8.83$$

If the angle ϕ is small

$$\dot{X}_R = \frac{p_1\phi}{z_{A1}}\cos\theta = \frac{\Delta p\cos\theta}{z_{A1}} \qquad 8.84$$

where $\Delta p = p_1\phi$ the difference in pressure between the two sides of the ribbon. Equation 8.84 will be recognized as that of the velocity microphone. The directional characteristic is bidirectional.

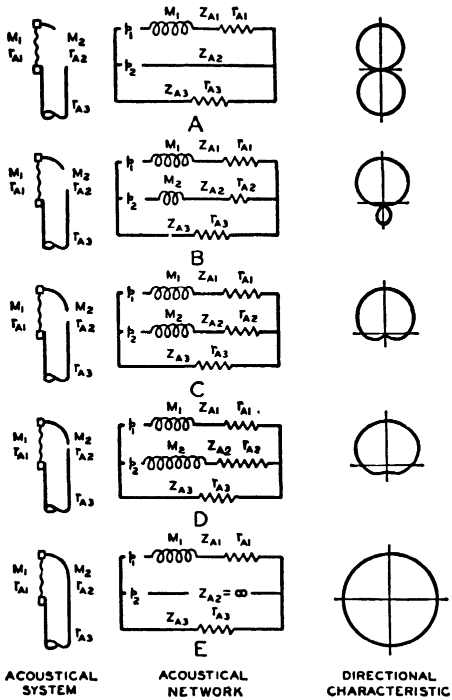

FIG. 8.37. The acoustical system, the acoustical network and the directional characteristics of the polydirectional microphone for various values of the aperture M_2, r_{A3}.

In Fig. 8.37*E* 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 8.79, 8.80 and 8.81 the volume current of the ribbon is given by

$$\dot{X}_R = \dot{X}_1 - \dot{X}_2 = \dot{X}_1 = \frac{p_1}{z_{A1} + z_{A3}} \qquad 8.85$$

Equation 8.85 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. 8.37*A* and the nondirectional characteristic Fig. 8.37*E*, as depicted by Fig. 8.37, parts *B*, *C*, and *D*. The directional characteristic of Fig. 8.37*C* is given by

$$e = R + R \cos \theta \tag{8.86}$$

This is a cardioid characteristic which is obtained in the two-element unidirectional microphone by making the output of the bidirectional element equal to the nondirectional unit. The directional characteristic of Fig. 8.37*B* is given by

$$e = \frac{R}{2} + \frac{3R}{2} \cos \theta \tag{8.87}$$

For a wider directional pickup angle the characteristic of Fig. 8.37*D* may be more desirable. This characteristic is given by

$$e = \frac{8R}{7} + \frac{6R}{7} \cos \theta \tag{8.88}$$

The energy response to random sounds as compared to that of a nondirectional microphone is $\frac{1}{3}$ for the bidirectional characteristic, Fig. 8.37*A*, and the cardioid characteristic, Fig. 8.37*C*. The energy response for the characteristic of Fig. 8.37*B* 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. 8.37*A*, to the cardioid characteristic of Fig. 8.37*C*. The energy response of the characteristic of Fig. 8.37*D* given by equation 8.88 is 0.39. The energy response varies from $\frac{1}{3}$ to 1 in going from the cardioid characteristic of Fig. 8.37*C* to the nondirectional characteristic of Fig. 8.37*E*. The general expression[34] for the directional characteristics obtainable with this microphone is

$$e = R_1 + R_2 \cos \theta \tag{8.89}$$

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. 8.32.

3. *Uniphase Dynamic Microphone.*—The uniphase dynamic microphone[35, 36] is a unidirectional microphone employing a diaphragm-voice

[34] 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.

[35] Bauer, B. B., *Jour. Acous. Soc. Amer.*, Vol. 13, No. 1, p. 41, 1941.

[36] Bauer, B. B., *Electronics*, Vol. 15, No. 1, p. 31, 1942.

coil transducer unit and a phase shifting acoustical network to obtain unidirectional characteristics. Schematic views and the acoustical network of the unidyne microphone are shown in Fig. 8.38. 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, $M_2 r_{A2}$. 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

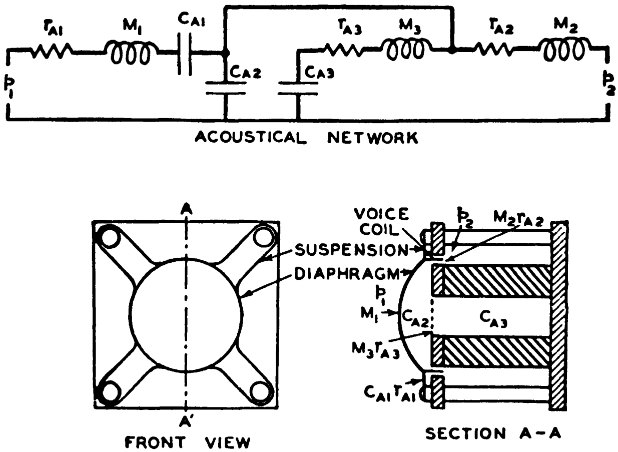

FIG. 8.38. Front view, sectional view and the acoustical network of a unidyne unidirectional microphone. In the acoustic circuit: M_1, r_{A1} and C_{A1} = the inertance, acoustical resistance and acoustical capacitance of the diaphragm and suspension system. M_2 and r_{A2} = the inertance and acoustical 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.

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 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. 8.32.

4. *Dipole Microphone.*[37] — A dipole microphone is a microphone in which the response is a function of the sound pressure between two distinct points. A schematic view of the acoustical system is shown in Fig. 8.39. 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 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.

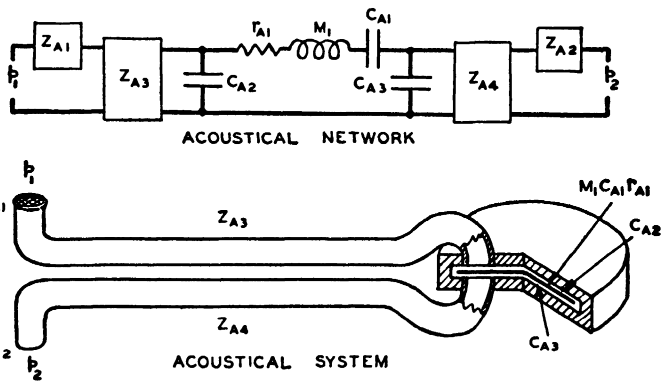

FIG. 8.39. Schematic view and acoustical network of the dipole microphone. In the acoustical circuit: z_{A1} and z_{A2} = the acoustical impedances of the silk cloth terminating the ends of the pipes. z_{A3} and z_{A4} = the acoustical quadripoles representing the cylindrical pipes. C_{A2} and C_{A3} = 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 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. 5.21.

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 out-

[37] Olney, Slaymaker and Meeker, *Jour. Acous. Soc. Amer.*, Vol. 16, No. 3, p. 172, 1945.

put will be independent of the frequency. It also possesses the anti-noise characteristics of a close talking, first-order, gradient microphone.

5. *Differential Microphone.*[38] *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. 8.40. The transducer is a carbon element. The performance of the system may be determined from a consideration of the acoustical network of Fig. 8.40. The differential

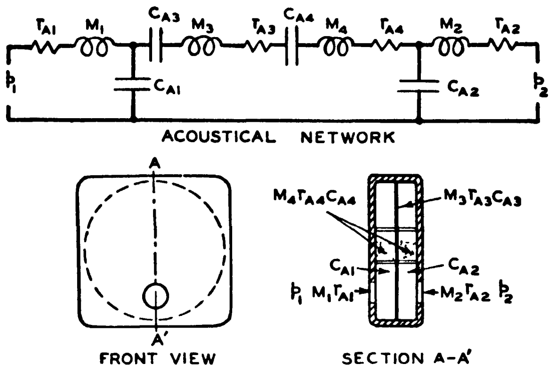

FIG. 8.40. Front view, sectional view and acoustical network of the differential microphone. In the acoustical network: M_1, r_{A1} and M_2, r_{A2} = the inertances and acoustical resistances of the two holes in the case. C_{A1} and C_{A2} = the acoustical capacitances of the air chambers on the two 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.

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.

8.5. Higher Order Gradient Microphones.[39, 40] — 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

[38] Ellithorn and Wiggins, *Proc. Inst. Rad. Eng.*, Vol. 34, No. 2, p. 84P, 1946.
[39] Olson, H. F., U. S. Patent 2,301,744.
[40] Olson, H. F., *Jour. Acous. Soc. Amer.*, Vol. 17, No. 3, p. 192, 1946.

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 h'gher order gradient microphones.

A. *Second-Order Gradient Microphones.*[41,42] — 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

$$\Delta(\Delta p) = p_M D_1 D_2 \left[\frac{-k^2 r^2 \sin k(c - r) + 2kr \cos k(ct - r) + 2 \sin k(ct - r)}{r^3} \right] \cos^2 \theta \qquad 8.90$$

where D_1 = distance between the points in the pair 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. 8.41.

The acoustical network of the acoustical system of one of the units in the second-order microphone is shown in Fig. 8.41. 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 anti-noise characteristics of a close talking, second-order, gradient microphone. The directional characteristics of the second-order gradient microphone, as equation 8.90 shows, are bidirectional and proportional to the square of the frequency.

B. *Gradient Microphones of Any Order.*[43] — The general expression for the actuating pressure for a microphone of any order n for any two points separated by a distance δr is

$$\delta n_p = \frac{\partial n_p}{\partial r^n} \delta r^n = \frac{\partial}{\partial r^n} \left(-j \frac{p_M}{r} \epsilon^{jk(ct-r)}\right) (\delta r \cos \theta)^n \qquad 8.91$$

Equation 8.91 shows that the pressure available for driving the microphone is proportional to the nth power of the frequency. The directional

[41] Olson, H. F., U. S. Patent 2,301,744.
[42] Olson, H. F., *Jour. Acous. Soc. Amer.*, Vol. 17, No. 3, p. 192, 1946.
[43] Olson, H. F., *Jour. Acous. Soc. Amer.*, Vol. 17, No. 3, p. 192, 1946.

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. 8.42.

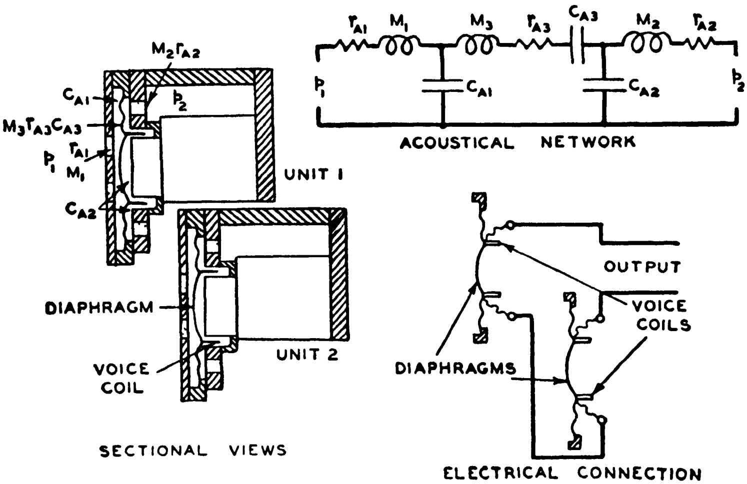

FIG. 8.41. Sectional view, electrical connection and acoustical network of a second-order gradient microphone. The acoustical network applies to a single unit. The electrical connection shows the two units connected in opposition. In the acoustical network: r_{A1} and M_1 = the acoustical resistance 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.

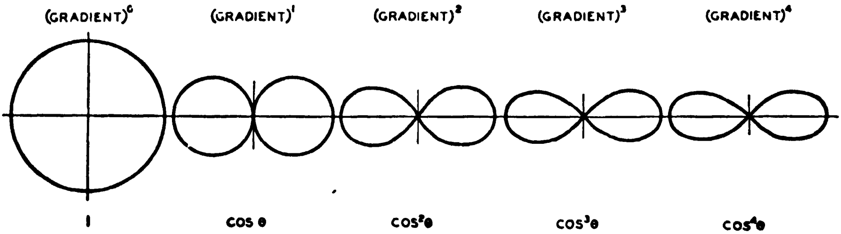

FIG. 8.42. The directional characteristics of gradient microphones of order zero, one, two, three and four.

C. *Noise Discrimination of Gradient Microphones.*[44] — Gradient microphones of order one and higher are directional. Therefore, these micro-

[44] 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 8.106 as follows,

$$\text{Directional efficiency} = \frac{2\pi \int_0^\pi R_n^2 \cos^{2n} \sin \theta \, d\theta}{4\pi R_0^2} \tag{8.92}$$

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.

If the sensitivity of the gradient microphone of order zero is the same as that of a gradient microphone or order n, equation 8.92 becomes

$$\text{Directional efficiency} = \frac{1}{2n + 1} \tag{8.93}$$

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. 8.43. The response frequency characteristics of all three are assumed to be independent of the frequency for a plane sound wave. Referring to Fig. 8.43 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 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. 8.43. 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

¾-inch distance from the small sound source, the response frequency characteristics for distant sounds will be as shown in Fig. 8.44. 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.

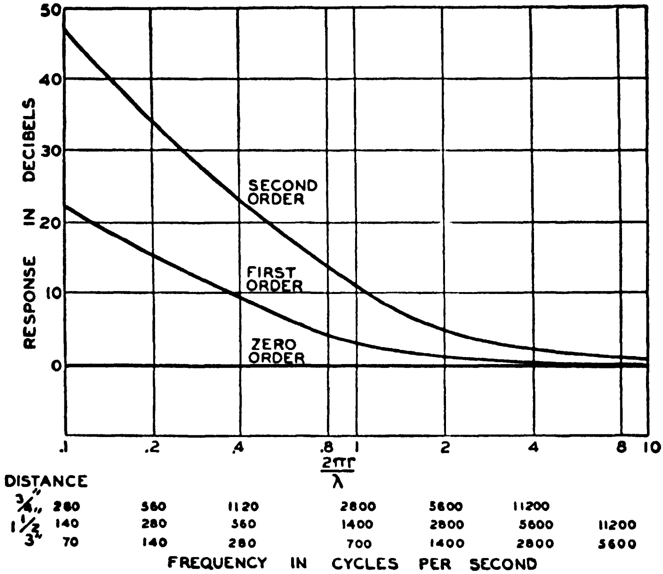

FIG. 8.43. Response of zero-, first- and second-order gradient microphones to a small source as a function of $\frac{2\pi r}{\lambda}$ where r = distance and λ = wavelength. The response frequency characteristics of all three are assumed to be independent of the frequency for a plane wave, that is, $\frac{2\pi r}{\lambda} = \infty$. The frequency scales below the graph apply to three distances, namely: 3, 1½ and ¾ inches.

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 ¾-inch distance, to distant sound originating in random directions is shown in Fig. 8.45. First-order gradient antinoise microphones have been described in Secs. 8.4*B*4 and 8.4*B*5. The characteristics for a first-order gradient microphone apply to these microphones. A second-order gradient micro-

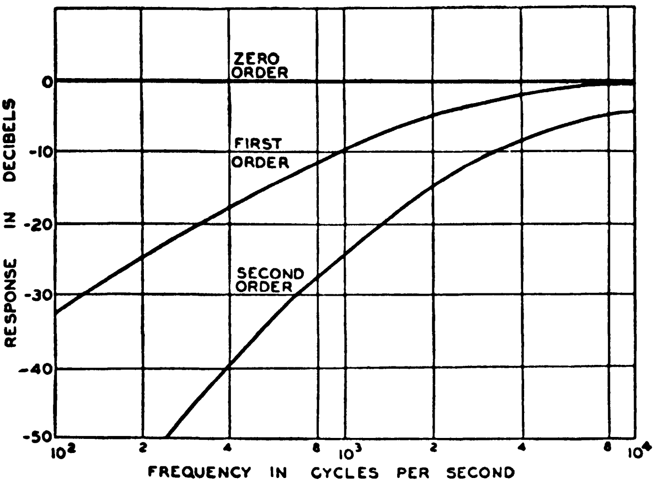

Fig. 8.44. Response frequency characteristics of zero-, first- and second-order gradient microphones to a plane wave. The microphones are compensated so that the responses of all three are the same and independent of the frequency when operating at a distance of $\frac{3}{4}$ inch from a small sound source.

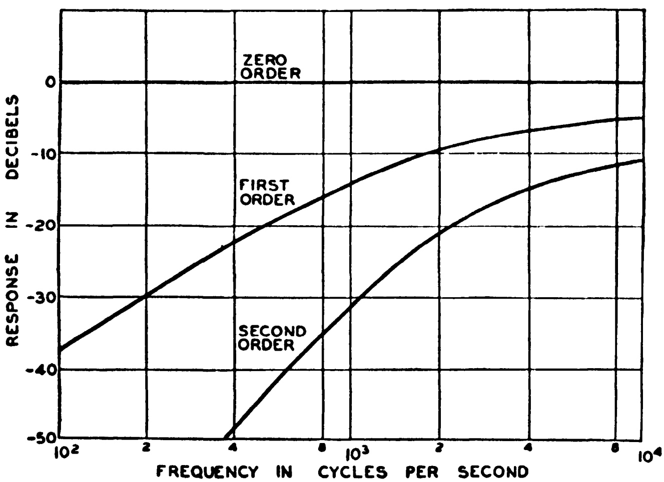

Fig. 8.45. Response frequency characteristics of zero-, first- and second-order gradient microphones to random sounds originating at a distance. The microphones are compensated so that the responses of all three are the same and independent of the frequency when operating at a distance of $\frac{3}{4}$ inch from a small sound source.

phone has been described in Sec. 8.5*A*. 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 witstand without pain.

D. *Higher Order Unidirectional Gradient Microphones.*[45] — 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 in Sec. 8.4. It is the purpose of this section to consider higher order combination gradient microphones with unidirectional characteristics.

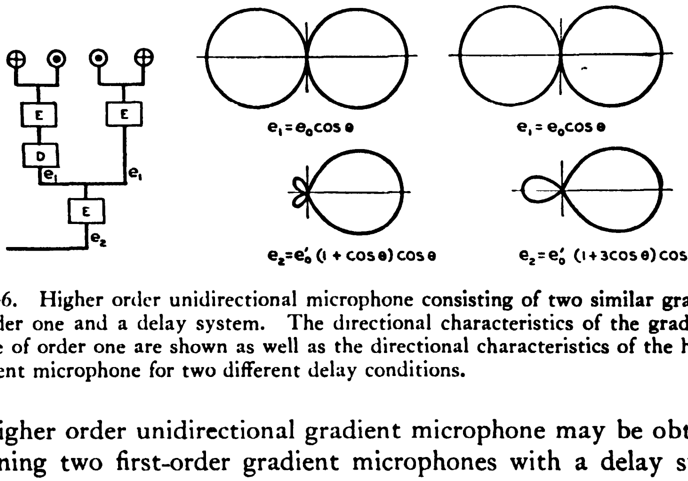

FIG. 8.46. Higher order unidirectional microphone consisting of two similar gradient units of order one and a delay system. The directional characteristics of the gradient microphone of order one are shown as well as the directional characteristics of the higher order gradient microphone for two different delay conditions.

A higher order unidirectional gradient microphone may be obtained by combining two first-order gradient microphones with a delay system as shown in Fig. 8.46. The voltage output of this system is given by

$$e_2 = e_0(D_2 + D_1 \cos \theta) \cos \theta \qquad 8.94$$

where e_0 = reference voltage output,
D_1 = distance between the first-order gradient elements, and
D_2 = path length of the delay.

Equation 8.94 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

[45] Olson. H. F., *Jour. Acous. Soc. Amer.*, Vol. 17, No. 3, p. 192, 1946.

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. 8.46.

The system of Fig. 8.47 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. 8.47.

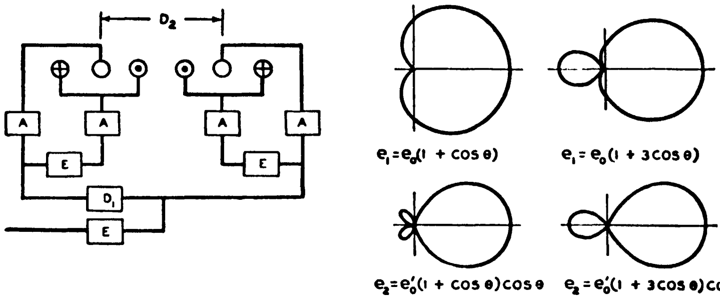

FIG. 8.47. Higher order unidirectional microphone consisting of two unidirectional elements described in Sec. 8.4 and a delay system. The directional characteristics of two different unidirectional units are shown and the combination higher order gradient for $D_2 > 0$ and $D_1 = 0$.

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 have 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.*[46, 47, 48, 49, 50] — 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. 8.48). 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 sharp directional characteristics. If this condition is satisfied at the low frequencies the size of the reflector becomes prohibitive to be used with facility.

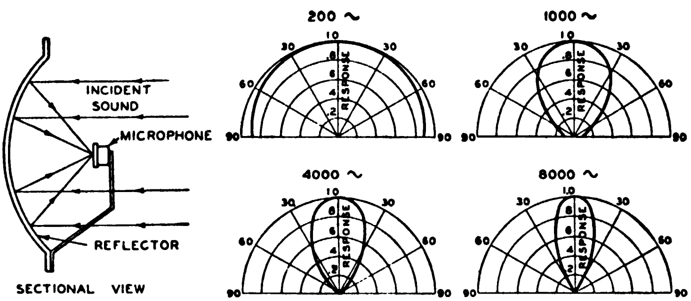

FIG. 8.48. Cross-sectional view of a parabolic reflector for a microphone. The polar graphs show the directional characteristics. The polar graph depicts the pressure, in dynes, at the microphone as a function of the angle, in degrees. The maximum response is arbitrarily chosen as unity.

A cross-sectional view of a parabolic reflector and a pressure microphone located at the focus is shown in Fig. 8.48. 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.

[46] Olson and Wolff, *Jour. Acous. Soc. Amer.*, Vol. 1, No. 2, p. 173, 1930.
[47] Hanson, O. B., *Jour. Acous. Soc. Amer.*, Vol. 3, No. 1, Part 1, p. 9, 1931.
[48] Hanson, O. B., *Jour. Acous. Soc. Amer.*, Vol. 3, No. 1, Part 1, p. 81, 1931.
[49] Dreher, Carl, *Jour. Soc. Mot. Pic. Eng.*, Vol. 16, No. 1, p. 29, 1932.
[50] Olson and Wolff, *Jour. Acous. Soc. Amer.*, Vol. 1, No. 3, p. 410, 1930.

The directional characteristics of a parabolic reflector 3 feet in diameter, used with a pressure microphone, are shown in Fig. 8.48. It will be seen 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. *Line Microphones.*[51,52,53] — 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 terminated in an acoustical resistance. Under these conditions the output of the pipes can be added vectorially.

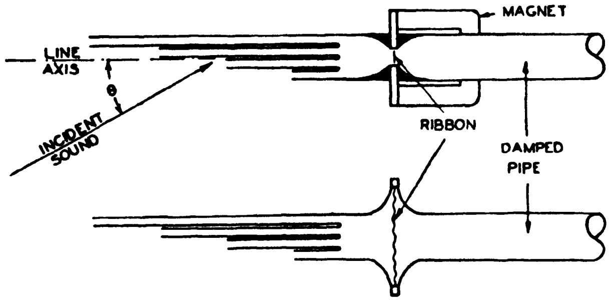

FIG. 8.49. Line microphone. Useful directivity on the line axis. This microphone consists of a large number of small pipes arranged in a line with the distance from the opening of each pipe to the common junction decreasing in equal steps. The system is terminated in a ribbon element and an acoustical resistance.

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. 8.49). 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

[51] Olson, H. F., *Jour. Inst. Rad. Eng.*, Vol. 27, No. 7, p. 438, 1939.
[52] Mason and Marshall, *Jour. Acous. Soc. Amer.*, Vol. 10, No. 3, p. 206, 1939.
[53] Olson, H. F., *Broadcast News*, No. 28, p. 32, July, 1938.

the common junction of the microphone may be expressed as

$$p_n = B_n \cos 2\pi\left(ft - \frac{x_n - x_n \cos\theta}{\lambda}\right) + jB_n \sin 2\pi\left(ft - \frac{x_n - x_n\cos\theta}{\lambda}\right) \qquad 8.95$$

$$p_n = B_n\epsilon^{2\pi jft}\epsilon^{2\pi j(x_n - x_n\cos\theta)/\lambda} \qquad 8.96$$

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 unifrom 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

$$R_\theta = \frac{1}{B_n l}\left|\int_{-l/2}^{l/2} B_n\epsilon^{2\pi j[ft + (x - x\cos\theta)/\lambda]}\,dx\right| \qquad 8.97$$

The absolute value of the term on the right is given by

$$R_\theta = \frac{1}{l}\left|\int_{-l/2}^{l/2}\epsilon^{2\pi j(x - x\cos\theta)/\lambda}\,dx\right| \qquad 8.98$$

$$R_\theta = \frac{\sin\frac{\pi}{\lambda}(l - l\cos\theta)}{\frac{\pi}{\lambda}(l - l\cos\theta)} \qquad 8.99$$

The directional characteristics of the microphone of Fig. 8.49 for various ratios of length of the line to the wavelength are shown in Fig. 8.50. 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.

2. *Line Microphone: Useful Directivity on the Line Axis. Line with Progressive Delay.* — As in the case of Fig. 8.49 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. 8.51).

$$R_\theta = \frac{1}{B_n l}\left| \int_{-l/2}^{l/2} B_n \epsilon^{2\pi j[(x - x\cos\theta/\lambda) + d/\lambda]}\, dx \right| \qquad 8.100$$

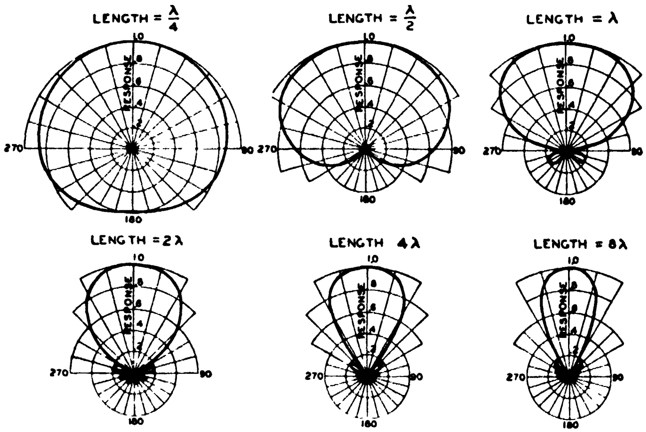

FIG. 8.50. The directional characteristics of the microphone shown in Fig. 8.49 as a function of the ratio of the length of the line to the wavelength. The polar graph depicts the output, in volts, as a function of the angle, in degrees.

where d is the path length of the delay introduced for the point furthest removed from the common junction.

$$R_\theta = \frac{\sin \frac{\pi}{\lambda}(l - l\cos\theta + d)}{\frac{\pi}{\lambda}(l - l\cos\theta + d)} \qquad 8.101$$

The directional characteristic of the microphone of Fig. 8.51 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. 8.52. Comparing Fig. 8.52 with Fig. 8.50, 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. 8.51 arranged so that the ribbon element measures

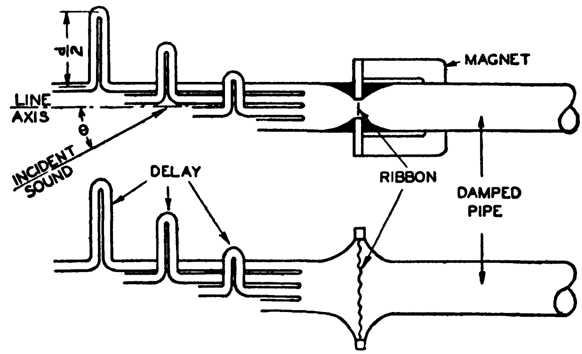

Fig. 8.51. Line microphone. Useful directivity on the line axis. This microphone differs from Fig. 8.49 in that a delay is inserted in each small pipe. The amount of delay is proportional to the distance from the pipe opening to the pickup point nearest the common junction.

the difference in the pressures generated in the two lines (Fig. 8.53). 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.

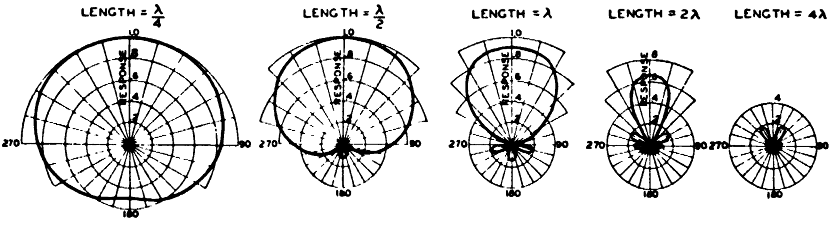

Fig. 8.52. The directional characteristics of the microphone shown in Fig. 8.51 for a time delay of one quarter the length of the line as a function of the ratio of the length of the lines to the wavelength. The polar graph depicts the output, in volts, as a function of the angle, in degrees.

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

$$f_M = A \cos (2\pi ft) \sin \left(\frac{\pi D \cos \theta}{\lambda}\right) \qquad 8.102$$

where A = constant, including the pressure of the impinging sound wave and dimensions of the microphone.

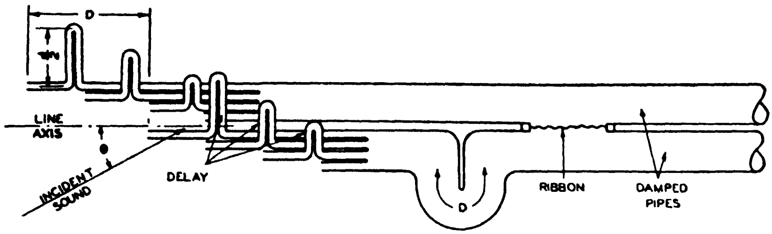

FIG. 8.53. Line microphone. Useful directivity on the line axis. This microphone consists of two lines of the type shown in Fig. 8.51 displayed by a distance D along the axis. In the line nearest the ribbon element a bend is inserted which introduces a path length D. The ribbon element measures the difference in pressure in the two lines.

If D is small compared to the wavelength, equation 8.102 becomes

$$f_M = A \frac{\pi D}{\lambda} \cos (2\pi ft) \cos \theta \qquad 8.103$$

Equation 8.103 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

$$\dot{x} = \frac{A}{j2\pi fm_r}\left(\frac{\pi D}{\lambda}\right) \cos (2\pi ft) \cos \theta$$

$$= \frac{A}{2\pi m_r}\left(\frac{\pi D}{c}\right) \sin 2\pi ft \cos \theta \qquad 8.104$$

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. 8.53 are given by equation 8.101. The directional characteristics of the microphone, shown in Fig. 8.53, for D small compared to the wavelength are the product of equations 8.101 and 8.104. The directional characteristics may be written as

$$R_\theta = \frac{\sin \frac{\pi}{\lambda}(l - l\cos\theta + d)}{\frac{\pi}{\lambda}(l - l\cos\theta + d)}\cos\theta \qquad 8.105$$

The directional characteristics of the microphone shown in Fig. 8.53 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. 8.54. A measure

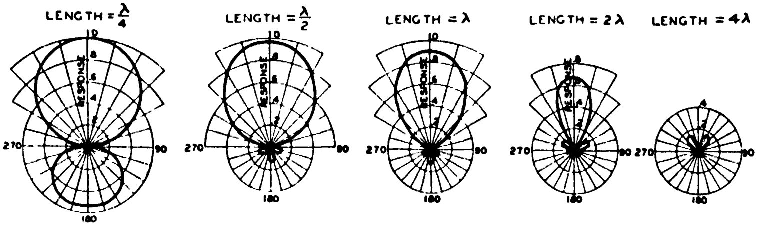

FIG. 8.54. The directional characteristics of the microphone shown in Fig. 8.53 for a time delay equivalent to one quarter the length of the line as a function of the ratio of the length of the line to the wavelength. The polar graph depicts the output, in volts, as a function of the angle, in degrees.

of the value of a line with progressive delay and a pressure gradient element for improving the directivity may be obtained by comparing Fig. 8.54 with Fig. 8.50. 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. 8.49.

4. *Ultradirectional Microphone.*[54] — 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

[54] Olson, H. F., *Jour. Inst. Rad. Eng.*, Vol. 27, No. 7, p. 438, 1939.

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 reenforcing. However, the directional char-

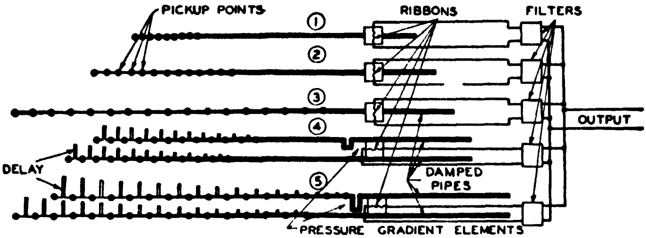

FIG. 8.55. Ultradirectional microphone consisting of five units. Units 1, 2 and 3 are of the type shown in Fig. 8.49. Units 4 and 5 are of the type shown in Fig. 8.53. An electrical filter system is used to allocate the output of the units to their respective ranges.

acteristics 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.

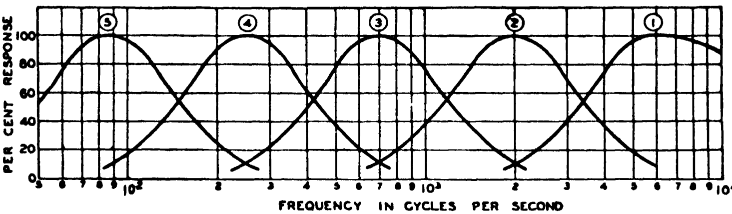

FIG. 8.56. Voltage response frequency characteristics of the units and electrical filter system shown in Fig. 8.55.

The ultradirectional microphone shown schematically in Fig. 8.55 consists of five units. Units 1, 2 and 3 are of the type shown in Fig. 8.49. Units 4 and 5 are of the type shown in Fig. 8.53. 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. 8.56. Fig. 8.57 illustrates the principles used in obtaining uniform directional characteristics. Fig. 8.57*A* is the directional characteristic of line 3 at 700 cycles. Fig. 857*B* shows the directional characteristics of lines 2 and 3 at 950 cycles. The resultant of these characteristics is also shown in Fig. 8.57*B*. The same is shown in Fig. 8.57*C* for 1250 cycles. In Figs. 8.57*B* and 8.57*C* the directional characteristic of line 2 is broader than Fig. 8.57*A* while the characteristic of line 3 is narrower. The resultant of lines 2 and 3 is a directional characteristic very close to Fig.

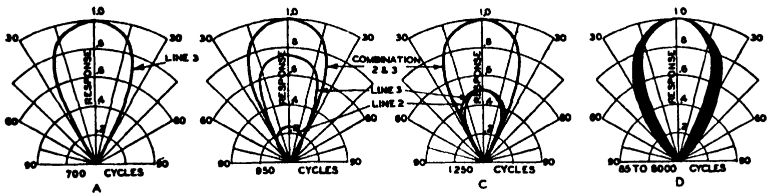

FIG. 8.57. *A*. The directional characteristic of line 3 of Fig. 8.55 at 700 cycles. *B*. The directional characteristics of lines 2 and 3 and the resultant at 950 cycles. *C*. The directional characteristics of lines 2 and 3 and the resultant at 1250 cycles. *D*. The directional characteristics of the microphone shown in Fig. 8.53 for the range from 85 to 8000 cycles fall within the shaded area.

8.57*A*. The directional characteristics of the microphone shown in Fig. 8.57 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. 8.57*D*. Considering that this microphone has a frequency range of $6\frac{1}{2}$ octaves, it is a remarkably uniform directional characteristic.

8.7. Directional Efficiency of a Sound Collecting System.[55, 56, 57, 58, 59] — 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

$$\text{Directional efficiency} = \frac{1}{4\pi}\int_0^{4\pi} f^2(\psi)\, d\Omega_\psi \qquad 8.106$$

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.

[55] Olson, H. F., *Jour. Soc. Mot. Pic. Eng.*, Vol. 16, No. 6, p. 695, 1931.
[56] Olson, H. F., *Jour. Acous. Soc. Amer.*, Vol. 3, No. 1, p. 56, 1931.
[57] Olson, H. F., *Proc. Inst. Rad. Eng.*, Vol. 21, No. 5, p. 655, 1933.
[58] American Standards Association Sectional Committee Z-24, Report on Calibration of Microphones, *Jour. Acous. Soc. Amer.*, Vol. 7, No. 4, p. 300, 1936.
[59] Baumzweiger, B., *Jour. Acous. Soc. Amer.*, Vol. 11, No. 4, p. 447, 1940.

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. 8.58. 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.

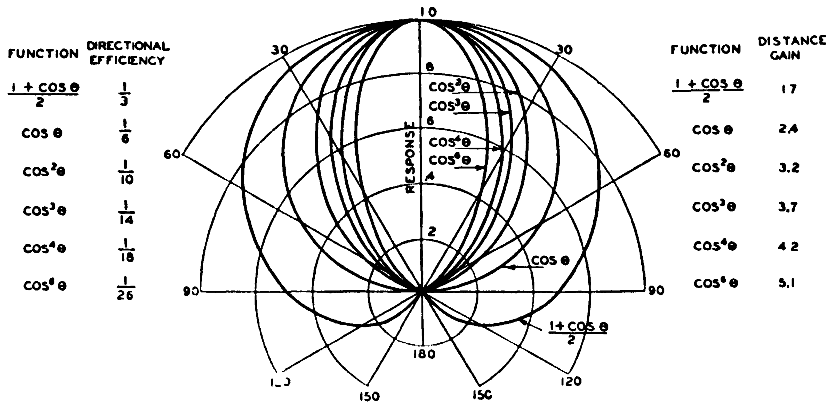

FIG. 8.58. The directional efficiency of microphones having directional characteristics which are various cosine functions. The ratio of energy response of a directional microphone to the energy response of a nondirectional microphone for sounds originating in random directions is termed directional 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. 8.58, 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 directivity index.

8.8. Throat Microphone.[60, 61, 62] — 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

[60] Shawn, J., *Communications*, Vol. 23, No. 1, p. 11, 1943

[61] Martin, D., *Jour. Acous. Soc. Amer.*, Vol. 19, No. 1, p. 43, 1947.

[62] Greibach and Pacent, *Elec. Eng.*, Vol. 65, No. 4, p. 187, 1946.

are shown in Fig. 8.59. Since the acoustical impedance of the flesh of the 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.

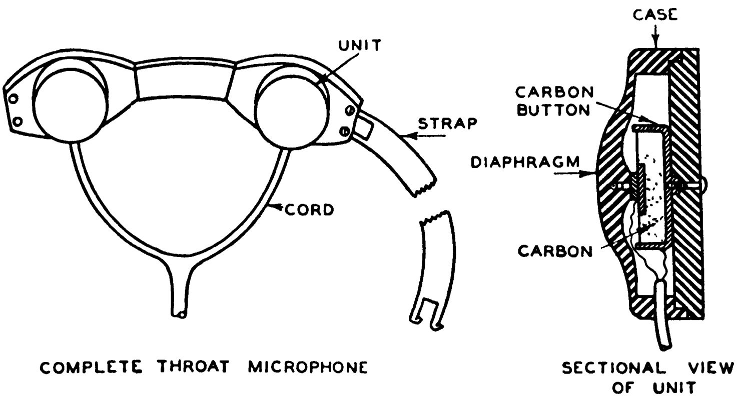

FIG. 8.59. Complete throat microphone and sectional view of the carbon type throat microphone unit.

Therefore, the high-frequency response must be accentuated to obtain intelligible speech. The units shown in Fig. 8.59 are of the carbon type. Other types of transducers as, for example, the magnetic type are also used in the throat microphone.

8.9. Lapel and Boom Microphones.[63, 64] — 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 same applications a microphone mounted upon a small light boom supported in a variety of ways has also been used. The principal purpose of the lapel 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

[63] Olson and Carlisle, *Proc. Inst. Rad. Eng.*, Vol. 22, No. 12, p. 1354, 1934.
[64] Jones and Bell, *Jour. Soc. Mot. Pic. Eng.*, Vol. 19, No. 3, p. 219, 1932.

general design of lapel and boom microphones is the same as the conventional microphones described in this chapter except that the size is smaller.

8.10. Wind Excitation and Screening of Microphones. — There are three possible sources of excitation which a microphone is subject to 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

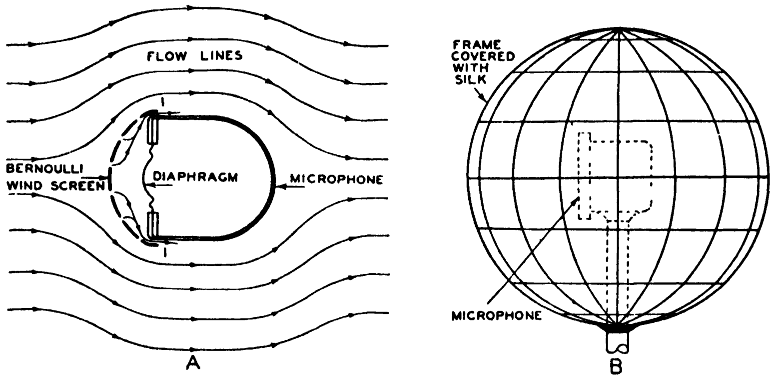

FIG. 8.60. Wind screens for microphones. *A*. Bernoulli wind screen applied to a dynamic microphone. *B*. Wind screen consisting of a wire frame covered with sheer silk.

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 steamlining 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. 8.60*B*). 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 microphone. The Bernoulli[65] wind screen is shown in Fig. 8.60*A*. The wind pulses travel through the screen and exert a pressure on the diaphragm.

[65] Phelps, W. D., *RCA Review*, Vol. 3, No. 2, p. 203, 1938.

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.

8.11. Nonlinear Distortion in Microphones. — The sources of distortion in microphones are, in general, the same as in the case of loud speakers. 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 a 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. 10.2*C*) in a velocity microphone for sound pressures up to 1000 dynes per square centimeter is less than $\frac{1}{3}$ of a 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 made in carbon materials in recent years and the distortion, although still high, has been materially reduced.

8.12. Transient Response of Microphones. — The subject of transient response of vibrating systems, together with applications to loud speakers has been considered in Sec. 6.14. The measurement of transient response of loud speakers will be considered in Sec. 10.3*G*. 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. 8.26 may be reduced to the simplified acoustical circuit of Fig. 8.61*A*, provided the elements, M_A and M_R, the inertances due to the mass of the 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,

$$\Delta p = j\omega p' \qquad 8.107$$

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 8.107 shows that the acoustical circuit 8.61*B* is equivalent to acoustical circuit 8.61*A*. From acoustical circuit 8.61*B* the volume current is

$$U = \frac{j\omega p'}{j\omega M_R + j\omega M_A} = \frac{p'}{M_R + M_A} \tag{8.108}$$

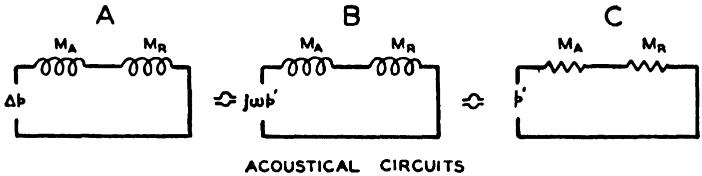

FIG. 8.61. Acoustical circuits of a velocity microphone. M_A and M_R = the inertance due to the air load and the ribbon mass. $\Delta p = j\omega p'$. Δp = the difference in pressure between the two sides of the ribbon. $\omega = 2\pi f$. f = the frequency. p' = a sound pressure proportional to the free field sound pressure. Under the conditions depicted *A* is equivalent to *B*. *B* is equivalent to *C*.

Equation 8.106 shows that the acoustical circuit 8.61*B* may be reduced to acoustical circuit 8.61*C*. Since, in acoustical circuit 8.61*C*, an acoustical resistance is driven by a constant sound pressure, the response to transients is perfect. This has been substantiated by actual tests[66] in which it is possible to obtain square waves from the output of a velocity microphone actuated by a loud speaker with a very smooth, wide-range response frequency characteristics. In multi-resonant systems with nonuniform response frequency characteristics it is impossible to obtain any semblance of a square wave from a loud-speaker microphone combination.

8.13. 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. 8.62, follow: The ambient 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.

[66] Olson and Preston, *RCA Review*, Vol. 7, No. 2, p. 155, 1946.

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

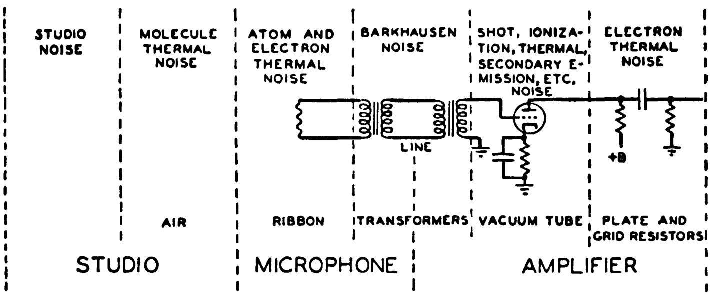

FIG. 8.62. Sources of noise in a sound pickup system.

of room noise is shown in Fig. 12.29. 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 pressure, p, in dynes per square centimeter, in the frequency interval between f_1 and f_2 may be obtained from the equation

$$\bar{p} = \sqrt{\int_{f_1}^{f_2} p_f^2 \, df} = \sqrt{\int_{f}^{f_2} 4kTr_A \, df} \qquad 8.109$$

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. 8.2*D*1 and Fig. 8.6.

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. 8.27.

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 8.109 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

$$\bar{f}_M = \sqrt{\int_{f_1}^{f_2} f_{Mf}^2 \, df} = \sqrt{\int_{f_1}^{f_2} 4kTr_M \, df} \qquad 8.110$$

where $f_{Mf}^2 \, df$ = 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.[67,68] The voltage, e, in abvolts, due to the thermal agitation of the electrons in a conductor is given by

$$e = \sqrt{4kT\,(f_2 - f_1)\,r_E} \qquad 8.111$$

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.

[67] Johnson, J. B., *Phys. Rev.*, Vol. 32, No. 1, p. 97, 1928.
[68] Nyquist, H., *Phys. Rev.*, Vol. 32, No. 1, p. 110, 1928.

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[69] 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 8.111 in Sec. 8.13*D*.

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. 8.3*B*). 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.

The above data show that the noises from all sources are comparable

[69] Terman, "Radio Engineers Handbook," McGraw-Hill Book Company, New York, N. Y., 1943.

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.

CHAPTER IX

MISCELLANEOUS TRANSDUCERS

9.1. Introduction. — Interest in the science of sound reproduction has been stimulated during the past two 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 loud speakers 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 loud speakers 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. *Bipolar 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.

The force,[1] in dynes, upon the diaphragm when an alternating current flows in the coils is

$$f_M = \frac{M^2}{4\pi R_1^2 A} + \frac{2MNi_{\max} \sin \omega t}{R_1 R_2 A} + \frac{2\pi N^2 i^2_{\max}}{R_2^2 A} - \frac{2\pi N^2 i^2_{\max} \cos 2\omega t}{R_2^2 A} \qquad 9.1$$

[1] Olson, "Dynamical Analogies," D. Van Nostrand Company, New York, N. Y., 1943.

where A = effective area of one pole, in square centimeters,
N = number of turns per coil,
R_1 = reluctance of the permanent field circuit, in gilberts per maxwell,
R_2 = reluctance of the alternating magnetic circuit, in gilberts per maxwell,
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.

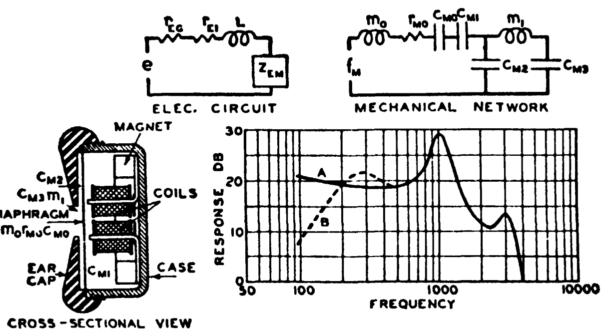

FIG. 9.1. Cross-sectional view, mechanical network, electrical circuit and response frequency characteristics of a bipolar telephone receiver. In the mechanical network: f_M = the mechanical driving force, m_0, r_{M0} and C_{M0} = the mass, mechanical resistance and compliance of the diaphragm. C_{M1} = 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.

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 ear cap. In all hard ear caps 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 ear cap 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 in actual practice. The artificial ear (see Sec. 10.4B) introduces a leak which corresponds to the leak between the ear and the ear cap. 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).

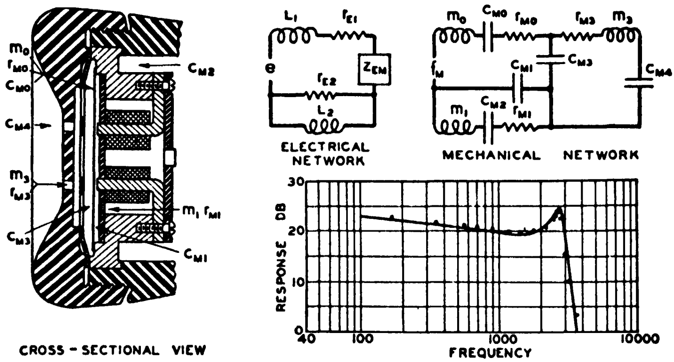

FIG. 9.2. Cross-sectional view, mechanical network, electrical circuit and response frequency characteristics of an improved bipolar telephone receiver. In the mechanical network: f_M = the mechanical driving force. m_0, r_{M0} and C_{M0} = the mass, mechanical resistance and compliance of the diaphragm. 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.

A new 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 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. 9.2. The mass of the diaphragm is represented by m_0. The compliance and

[2] Jones, W. C., *Jour. A.I.E.E.*, Vol. 57, No. 10, p. 559, 1939.

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 ear cap form the mechanical resistance r_{M3} and the mass m_3. The compliance C_{M3} is due to the cavity between the ear cap and the diaphragm. 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 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. 9.2. 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 changes in the constants of the component parts. Comparing the response of Figs. 9.1 and 9.2 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.

B. *Crystal Telephone Receiver.* — A crystal telephone receiver[4] consists of a light diaphragm connected to a Rochelle salt crystal (Fig. 9.3). 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. 9.3 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.

[3] Olson, " Dynamical Analogies," D. Van Nostrand Company, New York, N. Y., 1943.

[4] 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. 9.3.

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 artifical ear is indicated by *A*, Fig. 9.3.

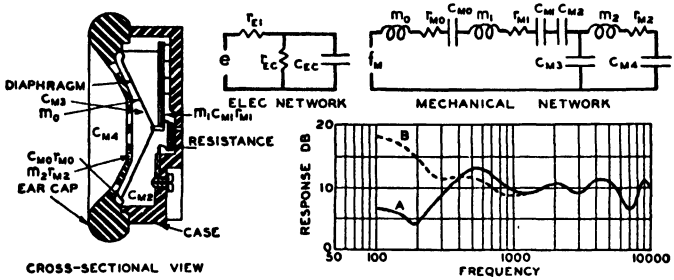

FIG. 9.3. Cross-sectional view, mechanical network, electrical network and response frequency characteristics of a crystal telephone receiver. In the mechanical network: f_M = the mechanical driving force. m_0, r_{M0} and C_{M0} = the mass, mechanical resistance and compliance of the diaphragm. m_1, 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[5] 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. 9.4. The mechanical network of the mechanical system is also shown in Fig. 9.4.

The electrical impedance,[6] in abohms, due to the mechanical system is given by

$$z_{EM} = \frac{(Bl)^2}{z_M} \qquad 9.2$$

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.

[5] Wente and Thuras, *Jour. Acous. Soc. Amer.*, Vol. III, No. 1, p. 44, 1932.
[6] Olson, "Dynamical Analogies," D. Van Nostrand Company, New York, N. Y., 1943.

In dynamic telephone receivers the flux density is relatively low and z_{EM} is small compared to r_{E1} and may be neglected.

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.

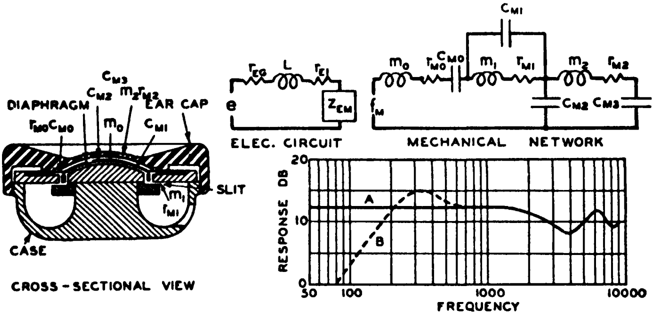

FIG. 9.4. Cross-sectional view, mechanical network, electrical circuit and response frequency characteristics of a dynamic telephone receiver. In the mechanical network: f_M = the mechanical driving force. m_0 = the mass of the diaphragm. r_{M0} and C_{M0} = the mechanical resistance and compliance of 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.

The pressure response frequency characteristic feeding a plain cavity is indicated by *A*, Fig. 9.4. The response measured on an artificial ear indicated by *B*, Fig. 9.4, shows that the response at the low frequencies is reduced due to the leak.

D. *Inductor Telephone Receiver.* — An inductor telephone receiver [7,8] 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

[7] Olson and Massa, *Jour. Acous. Soc. Amer.*, Vol. 6, No. 4, p. 240, 1935.
[8] Olson, H. F., *Jour. Soc. Mot. Pic. Eng.*, Vol. 27, No. 5, p. 537, 1936.

shown in Fig. 9.5. The acoustical network compensates for the leak between the ear and the ear cap. The effect of the leak between the ear and the ear cap upon the response of a telephone receiver has been outlined in the preceding sections. Obviously, from a practical standpoint the performance of a telephone receiver should be independent of the leak between the ear and the ear cap. In order to design the vibrating system

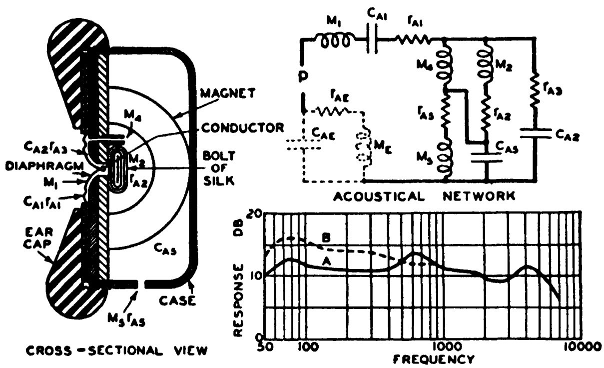

FIG. 9.5. Cross-sectional view, acoustical network and response frequency characteristic of an inductor telephone receiver. In acoustical network: M_1 = the inertance of the diaphragm and conductor. C_{A1} and r_{A1} = the acoustical capacitance and acoustical resistance of the suspension system. M_2 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 between 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, $p = \frac{f_M}{S}$. f_M = the mechanical driving force. S = the area of the diaphragm. The graph shows the pressure response frequency characteristics. A. Receiver feeding an artificial ear. B. Receiver feeding a plain cavity.

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 ear cap must be considered as a part of the vibrating system. The acoustical impedance characteristic, looking through the aperture of the ear cap of a telephone receiver, is shown in Fig. 10.24, 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. 9.5. 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 is the same as in the case of the dynamic telephone receiver. 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. 9.5. 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. 9.5. 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 ear cap.

A dynamic telephone receiver[9] 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. 9.5. 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. Phonographs. — A phonograph is a system for the reproduction of 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

[9] Anderson, L. J., *Jour. Soc. Mot. Pic. Eng.*, Vol. 37, No. 3, p. 319, 1941.

high quality reproductions as, for example, in transcriptions for broadcasting. The systems used in the recording and processing of phonograph records will be considered in Sec. 11.4G. It is the purpose of the sections which follow to consider phonograph recorders, a mechanical phonograph, phonograph pickups and distortion in phonograph reproduction.

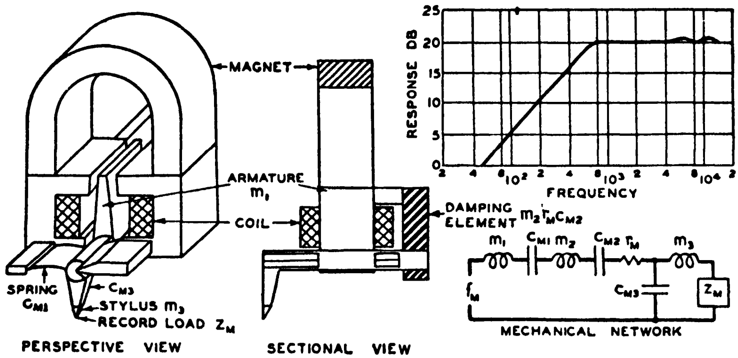

FIG. 9.6. Perspective and sectional views, mechanical network and velocity response frequency characteristic of a lateral type phonograph recorder. In the mechanical network: f_M = the mechanical driving force. m_1 = the mass of the armature. C_{M1} = the compliance of the restoring spring. m_2, r_M 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. The graph depicts the velocity response frequency characteristic of the recorder.

A. *Recording Systems.* — 1. *Lateral Recorder.* — In the lateral type of recording the undulations are cut in a directional parallel to the surface of the record and perpendicular to the groove. Perspective and sectional views and the mechanical network of a lateral type magnetic phonograph recorder[10] are shown in Fig. 9.6. The force applied to the armature in a magnetic driving system is 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.

2. *Vertical Recorder.* — In the vertical type of recording the undulations are cut in a directional perpendicular to the surface of the record. A sectional view, the mechanical circuit and electrical system of a feedback

[10] Hasbrouck, H. J., *Jour. Soc. Mot. Pic. Eng.*, Vol. 32, No. 3, p. 246, 1939.

type of a vertical type phonograph recorder[11] are shown in Fig. 9.7. 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. 9.7. 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. 9.7*B*. The use of a feedback in conjunction with a simple

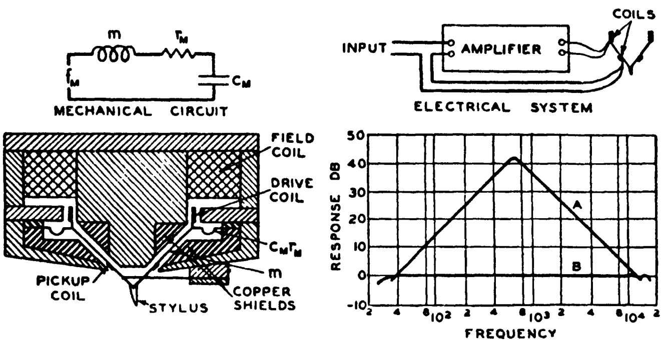

FIG. 9.7. Sectional view, mechanical circuit, electrical system and velocity response frequency characteristic of a feedback vertical type phonograph recorder. In the mechanical circuit: f_M = the mechanical driving force. m, r_M and C_M = the mass mechanical resistance and compliance of the vibrating system. In the graph: *A* = the velocity response frequency characteristic without feedback. *B* = the velocity response frequency characteristic with feedback.

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.

3. *Recording Characteristics.* — The velocity response frequency of a typical standard frequency record used in obtaining the response frequency characteristics of phonograph pickups and mechanical phonographs is shown in Fig. 9.8. 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.

[11] Vieth and Wiebusch, *Jour. Soc. Mot. Pic. Eng.*, Vol. 30, No. 1, p. 96, 1938.

In radio transcription recording, the orthacoustic[12] type of recording characteristic is employed. The orthacoustic velocity frequency characteristic for constant voltage input to the microphone amplifier is shown in

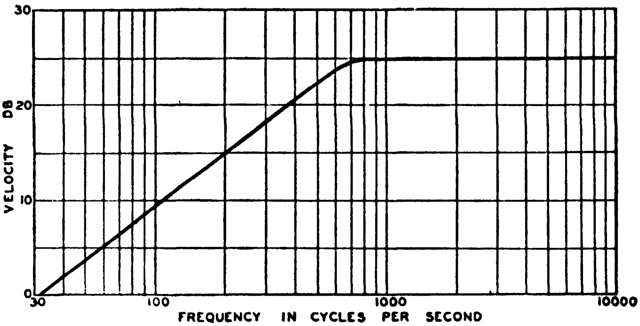

FIG. 9.8. Typical velocity response frequency characteristic of a standard frequency phonograph record for constant voltage input to the equalized recording amplifier.[9]

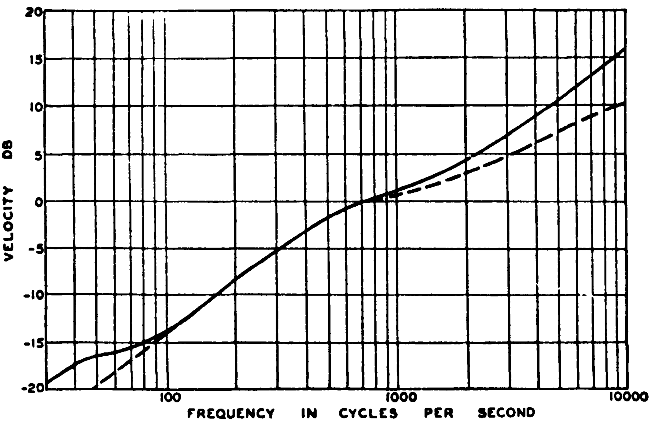

FIG. 9.9. Solid line curve: Velocity response frequency characteristic of an orthacoustic phonograph record for constant voltage input, at the microphone input terminals, to the equalized recording amplifier. Dashed line curve: Velocity response frequency characteristic of a commercial phonograph record for constant voltage input, at the microphone input terminals, to the equalized recording amplifier.

Fig. 9.9. This characteristic is essentially a constant amplitude frequency characteristic. In reproduction of the record, an inverse response frequency characteristic is used to obtain a uniform overall response frequency

[12] Recording and Reproducing Standards, *Proc. Inst. Rad. Eng.*, Vol. 30, No. 8, p. 355, 1942.

characteristic. The use of this type of response frequency characteristic reduces ground noise and distortion.

In recording of commercial phonograph records, some high-frequency accentuation is used but not to the extent employed in the orthacoustic characteristic. The velocity response frequency characteristic for constant voltage input to the microphone amplifier as employed in the recording of commercial phonograph records is shown in Fig. 9.9. In the reproduction of commerical phonograph records, an inverse response frequency characteristic is employed to obtain a uniform overall response frequency characteristic. The use of high-frequency accentuation, as shown in Fig. 9.9, reduces record ground noise and distortion.

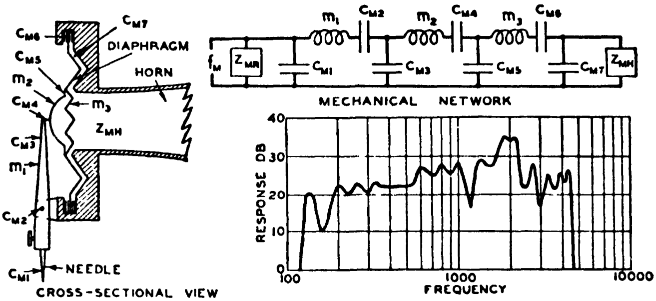

FIG. 9.10. Cross-sectional view, mechanical network and response frequency characteristic of a mechanical phonograph. In the mechanical network: z_{MR} = the mechanical impedance of the record. C_{M1}, C_{M2}, C_{M3}, C_{M4}, C_{M5}, C_{M6} and C_{M7} = the compliances of the needle, the needle holder, 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. 9.8. The graph shows the pressure response frequency characteristic of a console type mechanical phonograph using a record having a characteristic of Fig. 9.8.

B. *Mechanical Phonograph.*[13] — 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. A cross-sectional view and the mechanical network of a mechanical phonograph are shown in Fig. 9.10. The system consists essentially of a diaphragm coupled to a needle driven by a phonograph record. To improve the radiation efficiency, the diaphragm is coupled to a horn. The record mechanical impedance is usually large compared to the mechanical im-

[13] Maxfield and Harrison, *Bell Syst. Tech. Jour.*, Vol. 5, No. 3, p. 493, 1926.

pedance 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. 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 impedance of the load.

The response frequency characteristic of a mechanical phonograph of the console type is shown in Fig. 9.10.

C. *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.

1. *Crystal Pickup.* — A crystal pickup[14] 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 commerical phonographs is shown in Fig. 9.11. The needle, driven by the record, is coupled to the crystal. The elements of the system and the mechanical network are shown in Fig. 9.11. The displacement of the crystal can be determined from the mechanical network of the mechanical system and the velocity of the generator obtained from Fig. 9.8. The voltage output of the crystal is proportional to the displacement. Therefore, the open circuit voltage at the low frequencies is accentuated as shown by the response characteristic A, Fig. 9.11. 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. The voltage response frequency characteristic of a pickup with a resistance shunting the crystal is indicated by B, Fig. 9.11. The high-frequency response of the crystal shows a cutoff around 7000 cycles. This cutoff can be made any value up to 15,000 cycles by a suitable choice of constants.

Referring to the mechanical network of Fig. 9.11 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

[14] Williams, A. L., *Jour. Soc. Mot. Pic. Eng.*, Vol. 32, No. 5, p. 552, 1939.

becomes larger the vibration velocity of the record will be correspondingly greater. Vibration of the record produces radiation of sound into the air. 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 loud speaker and produces distortion.

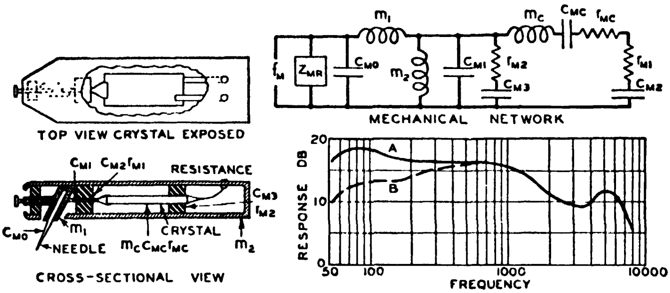

FIG. 9.11. Cross-sectional view, mechanical network and response frequency characteristics of a crystal pickup. In the mechanical network: z_{MR} = the mechanical impedance of the record. C_{M0} = the compliance of the needle. m_1 = the mass of the needle holder. C_{M1} = the compliance of the shaft. C_{M2} and C_{M3} = the compliances of the crystal supports. r_{M1} and r_{M2} = the mechanical resistances of the crystal supports. 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 having the characteristic of Fig. 9.8. The graph depicts the voltage response frequency characteristic with the record characteristic of Fig. 9.8. *A*. Open-circuit voltage response frequency characteristic. *B*. Voltage response frequency characteristic with 500,000 ohms shunting the crystal.

To overcome this, a low noise crystal pickup[15] has been developed. The essential elements and mechanical network of a low noise crystal pickup are shown in Fig. 9.12. 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. 9.12 is about 20 db lower than the replaceable needle pickup of Fig. 9.11. The voltage response frequency characteristic of the low noise crystal pickup is shown in Fig. 9.12.

2. *Magnetic Pickup*. — A magnetic pickup[16,17] is a phonograph pickup whose electrical output is generated in a coil or conductor in a magnetic field or circuit. A typical magnetic pickup is shown in Fig. 9.13. The

[15] Burt, A. D., *Electronics*, Vol. 16, No. 1, p. 90, 1943.
[16] Kellogg, E. W., *Jour. A.I.E.E.*, Vol. 46, No. 10, p. 1041, 1927.
[17] Hasbrouck, H. J., *Proc. I.R.E.*, Vol. 27, No. 3, p. 184, 1939.

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

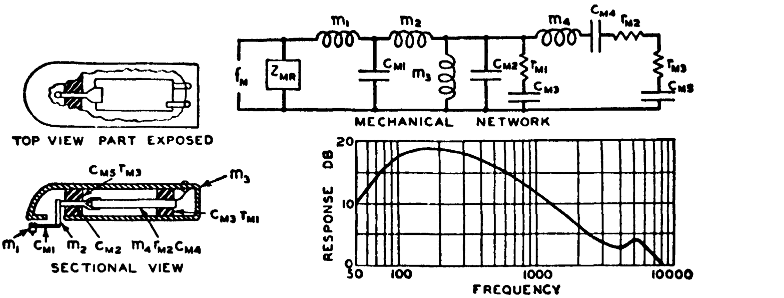

FIG. 9.12. Cross-sectional views, mechanical network and response frequency characteristic of a low noise crystal pickup. In the mechanical network: z_{MR} = the mechanical impedance of the record. m_1 = the mass of the stylus and holder. C_{M1} = the compliance of the stylus arm. m_2 = the mass of the 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 having the characteristic of Fig. 9.8. The graph depicts the voltage response frequency characteristic with the record characteristic of Fig. 9.8.

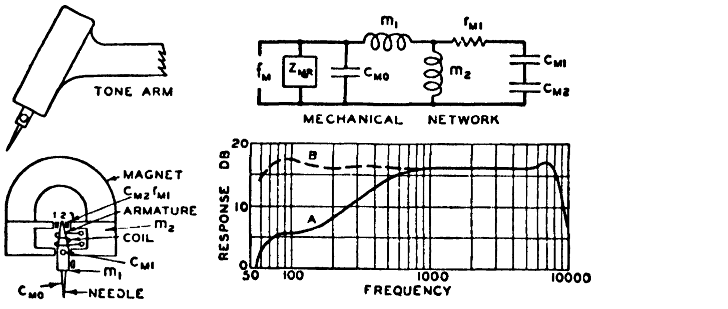

FIG. 9.13. Front and side views, mechanical network and response frequency characteristics of a magnetic pickup. In the mechanical network: z_{MR} = the mechanical impedance of the record. C_{M0} = the compliance of the needle. m_1 = the mass of the needle holder and armature. C_{M1} = the compliance 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 having the characteristic of Fig. 9.8. The graph depicts the voltage response frequency characteristic with the record characteristic of Fig. 9.8. *A*. Open-circuit voltage response frequency characteristic. *B*. Voltage response frequency characteristic of an equalized pickup..

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.

The voltage,[18] in abvolts, generated in the coil is

$$e = N\frac{d\phi}{dt} = \frac{NMA}{a^2}\dot{x} \qquad 9.4$$

where N = number of turns in the coil,
A = area of pole piece, in square centimeters,
ϕ = flux through armature, in maxwells,
M = magnetomotive force of the permanent magnet, in gilberts,
a = spacing between the armature and pole pieces, in centimeters, and
$\dot{x}$ = velocity of the armature, in centimeters per second.

Equation 9.4 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. 9.13. Damping, represented by the compliance C_{M2} and the mechanical resistance r_{M1}, is furnished by a suitable material such as viscoloid. The voltage response frequency characteristic with record characteristic shown in Fig. 9.8 is shown in Fig. 9.13. The output below 1000 cycles is similar to that of the record. Some compensation in the amplifier must be provided to compensate for this drooping characteristic. The peak at the low frequencies is due to the resonance of the total mass of the pickup and tone arm m_2 with the compliances C_{M1} and C_{M2}. Below this resonant frequency the mechanical mass reactance due to m_2 becomes small compared to the mechanical impedance of the compliance elements and the output falls off rapidly with decrease in frequency.

3. *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. 9.14 shows a cross-sectional view and mechanical network of a dynamic pickup[19] 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.

Another form of dynamic pickup shown in Fig. 9.15, 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

[18] Olson, "Dynamical Analogies," D. Van Nostrand Company, New York, N. Y., 1943.
[19] Frederick, H. A., *Jour. Soc. Mot. Pic. Eng.*, Vol. 18, No. 2, p. 141, 1932.

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 differ for the two types of

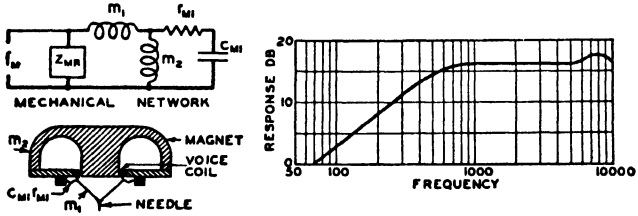

FIG. 9.14. Cross-sectional view, mechanical network and response frequency characteristic of a vertical dynamic pickup. In the mechanical network: z_{MR} = the mechanical impedance of the record. m_1 = the mass of the stylus and voice coil. C_{M1} and r_{M1} = the compliance and mechanical resistance of the suspension system. m_2 = the mass of the pickup and tone arm. f_M = the force generated by a velocity generator having the characteristic of Fig. 9.8. The graph depicts the voltage response frequency characteristic with the record characteristic of Fig. 9.8.

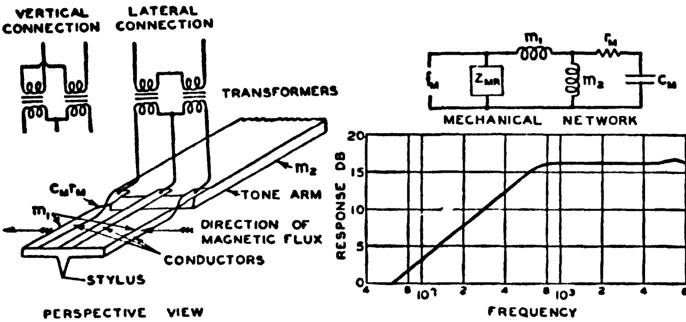

FIG. 9.15. Perspective view, electrical connection arrangement mechanical network and response frequency characteristic of a combination vertical or lateral dynamic pickup. In the mechanical network: z_{MR} = the mechanical impedance of the record. m_1, r_M and C_M = the mass, mechanical resistance and 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 having the characteristic of Fig. 9.8. The graph depicts the voltage frequency characteristic with the record characteristics of Fig. 9.8. Note: Either vertical or lateral type phonograph records can be reproduced by merely changing the transformer output connections.

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 voltage response frequency characteristic of the combination lateral and vertical pickup is shown in Fig. 9.15. This pickup is designed for wide-range reproduction of transcription phonograph records.

4. *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.

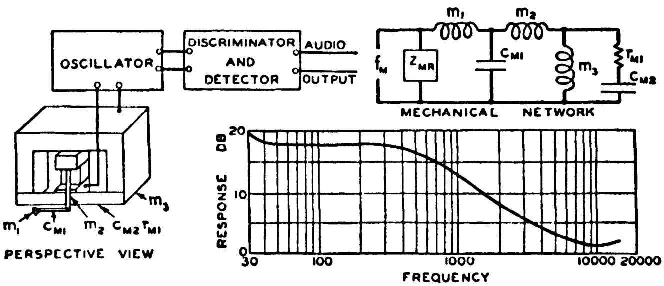

FIG. 9.16. Perspective view, electrical system, mechanical network and response frequency characteristic of a frequency modulation pickup. In the mechanical network: z_{MR} = the mechanical impedance of the record. m_1 = the mass of a stylus and holder. C_{M1} = the compliance of the stylus arm. m_2 = 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 having the characteristic of Fig. 9.8. The graph depicts the voltage response frequency characteristic with the record characteristic of Fig. 9.8.

A perspective view, electrical diagram, mechanical network and response frequency characteristic of a frequency modulation pickup[20] is shown in Fig. 9.16. 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 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 that a lateral displacement of the stylus will produce a change in the spacing between the ribbon and insulated back 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 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.

[20] Beers and Sinnett, *Proc. Inst. Rad. Eng.*, Vol. 31, No. 4, p. 138, 1943.

The output of the detector corresponds to amplitude of the stylus. A typical voltage response frequency characteristic of a wide-range frequency modulation pickup is shown in Fig. 9.16.

5. *Electronic Pickup.*[21] — 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 is shown in

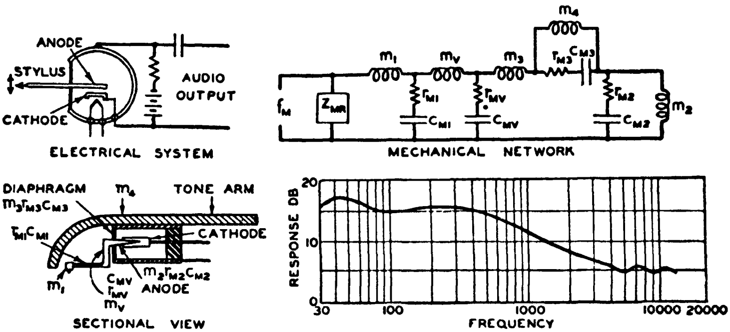

FIG. 9.17. Sectional view, electrical system, mechanical network and response frequency characteristic of an electronic pickup. In the mechanical network: z_{MR} = the mechanical impedance of the record. m_1 = the mass of the stylus. r_{M1} and C_{M1} = the mechanical resistance and compliance of the stylus arm. m_V, r_{MV} and C_{MV} = the mass, mechanical resistance and compliance of the vertical 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 having the characteristic of Fig. 9.8. The graph depicts the voltage response frequency characteristic with the record characteristic of Fig. 9.8. The electrical system shows the wiring diagram for a diode type electronic transducer.

Fig. 9.17. 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. Therefore, the response charactertistic of Fig. 9.17 is similar to the crystal and frequency modulation pickups. One of the advantages of the electronic phonograph pickup is that the mechanical impedance can be made exceedingly small. The voltage response characteristic shown in Fig. 9.17 was obtained on a pickup designed for the reproduction of wide frequency range transcription

[21] Olson, *Jour. Acous. Soc. Amer.*, Vol. 19, No. 2 , p. 307, 1947.

phonograph records. The system can also be designed for response over a limited frequency, for reproduction of commercial phonograph records, by an appropriate choice of constants.

6. *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[22] is shown in Fig. 9.18.

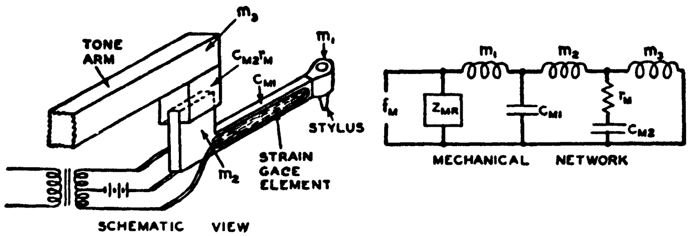

FIG. 9.18. Perspective view and mechanical network of a variable resistance pickup. In the mechanical network: z_{MR} = the mechanical impedance of the record. m_1 = the mass of the stylus. C_{M1} = the compliance of the stylus arm. m_2 = the mass at the base of the stylus arm. C_{M2} and r_M = the compliance 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 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.

D. *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 impedance to the needle. As a consequence, the vibrating

[22] Bachman, W. S., *Elec. Eng.*, Vol. 65, No. 3, p. 159, 1946.

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.[23]

Another source of nonlinear distortion is due to a deviation in tracking,[24,25] 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 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

TABLE 10.1. TONE ARM DESIGN DATA

The length of the tone arm is the distance from the stylus to the pivot. The overhang is the minimum distance between the center of the turntable and the stylus when the tone arm is swung past the center of the record. The offset angle is the angle between the vertical plane containing the vibration axis of the pickup and the line joining the stylus and the tone arm pivot. The data below are for 12-inch records with a minimum groove radius of 1.75 inches and a maximum groove radius of 5.75 inches.

Tone Arm Length in Inches	Overhang in Inches	Offset Angle in Degrees
6.5	.64	28
7.0	.60	26
7.5	.56	24
8.0	.52	22
9.0	.47	20
10.0	.42	18
12.0	.35	15
15.0	.28	12
20.0	.20	9

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. For a tracking error of 15° with a straight tone arm, the distortion is approximately 4 per cent. However, by the use of a tone arm with angular offset and overhang the tracking error can be reduced to ±5°. With this tracking error the distortion is negligible compared to other distortions in the system. Table 10.1 contains the angular offset and the overhang for various tone arm lengths for minimum tracking error for 12-inch records.

[23] Begun and Lynch, *Jour. Acous. Soc. Amer.*, Vol. 13, No. 3, p. 284, 1942.
[24] Olney, Benj., *Electronics*, Vol. 10, No. 1, p. 19, 1937.
[25] Bauer, B. B., *Electronics*, Vol. 18, No. 3, p. 110, 1945.

Another source of nonlinear distortion [26, 27, 28] is due to the finite size of the stylus or needle point. The curve traced by the center of the needle sliding in a sinusoidal groove is not sinusoidal. This distortion may be reduced by reducing the size of the needle point. It is interesting to note that this distortion is greater in vertical type records than in the lateral type. The push-pull effect of the lateral record tends to reduce the second harmonic distortion.

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 [29] introduces nonlinear distortion and a loss in the high-frequency response.

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." 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,[30] in the absence of any signal, is one of the factors which limits the volume range and the frequency range of 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 it is usually necessary to limit the high-frequency range. A method for 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.

E. *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 and the walls of the record. The force, in dynes, developed by the interaction of the pickup stylus and the record is given by

$$f_M = j\frac{z_{MR}z_{MP}x\omega}{z_{MR} + z_{MP}} \qquad 9.5$$

[26] DiToro, M. J., *Jour. Soc. Mot. Pic. Eng.*, Vol. 29, No. 5, p. 493, 1938.
[27] Pierce and Hunt, *Jour. Acous. Soc. Amer.*, Vol. 10, No. 1, p. 14, 1938.
[28] Sepmeyer, L. W., *Jour. Acous. Soc. Amer.*, Vol. 13, No. 3, p. 276, 1942.
[29] Bauer, B. B., *Jour. Acous. Soc. Amer.*, Vol. 16, No. 4, p. 246, 1945.
[30] Reid, J. D., *Jour. Acous. Soc. Amer.*, Vol. 13, No. 3, p. 274, 1942.

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 mechanical ohms.

Equation 9.5 illustrates the importance of a pickup with a small mechanical 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 loud speaker. 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 9.5 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. 10.5*D*.

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[31] is shown in Fig. 9.19. The crystal is a Rochelle salt bimorph type. With

[31] Bauer, B. B., *Jour. Acous. Soc. Amer.*, Vol. 11, No. 3, p. 303, 1940.

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 sys-

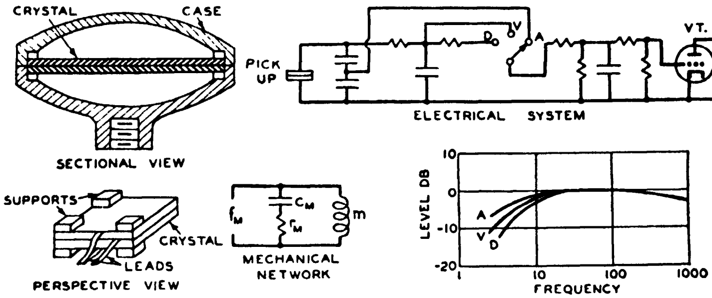

FIG. 9.19. Sectional view, perspective view of the crystal mounting arrangement, mechanical network, electrical connection and response frequency characteristic of a vibration pickup. In the mechanical network: m = the mass of the crystal. r_M and C_M = the mechanical resistance and compliance of the crystal and supports. 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, and D, displacement.

tem is shown in Fig. 9.19. 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

$$\dot{x} = \frac{f_M}{j\omega m}(1 - \omega^2 m C_M) \tag{9.6}$$

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

$$\dot{x} \sim \frac{f_M}{j\omega m} \tag{9.7}$$

The acceleration is given by

$$\ddot{x} = j\dot{x}\omega = \frac{f_M}{m} \tag{9.8}$$

The displacement is given by

$$x = \frac{\dot{x}}{j\omega} = -\frac{f_M}{\omega^2 m} \qquad 9.9$$

Below the resonant frequency the force is proportional to and in phase with the acceleration. The voltage output of the unit then corresponds 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. 9.19 to obtain velocity and displacement.

The response frequency characteristics of the vibration pickup and the electrical system are shown in Fig. 9.19. 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.

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 power 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. 9.20. Cross-sectional views, mechanical and electrical networks of the microphone and receiver are shown in Fig. 9.20. In order to obtain a tolerable sound level at the receiver the overall 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. 9.21. 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

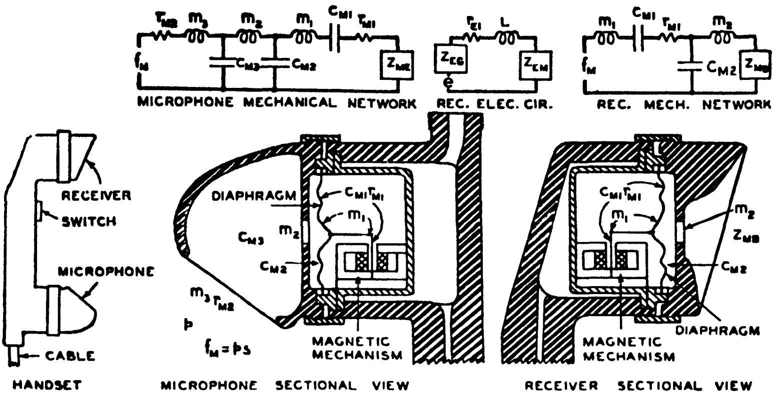

FIG. 9.20. The sound powered handset, sectional views of the microphone and receiver, mechanical networks of the microphone and receiver and electrical circuit of the receiver. In the microphone mechanical network: z_{ME} = the mechanical impedance due to the electrical circuit. m_1, r_{M1} and C_{M1} = 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.

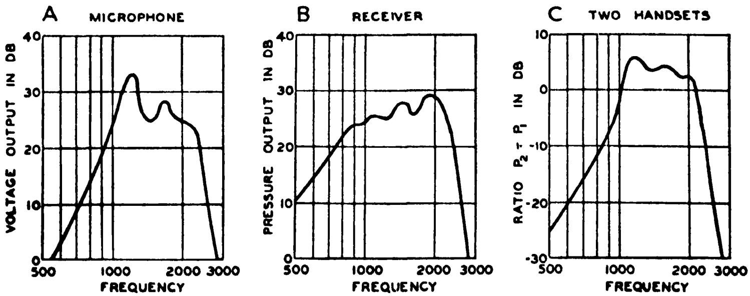

FIG. 9.21. Response frequency characteristics of a sound powered telephone. *A*. Voltage response frequency characteristic of the microphone. *B*. Voltage response frequency characteristic of the receiver. *C*. Overall pressure ratio response frequency characteristic of two handsets, one used as a transmitter and the other as a receiver.

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[32] consists of the combination of a microphone, an amplifier and a horn loud speaker (Fig. 9.22). The microphone and horn loud speaker form a single unit. In use, the operator speaks into the microphone. The voice is reinforced by the amplifier and loud speaker. The resulting power output is many times that of the unaided voice or the voice and an acoustical megaphone.

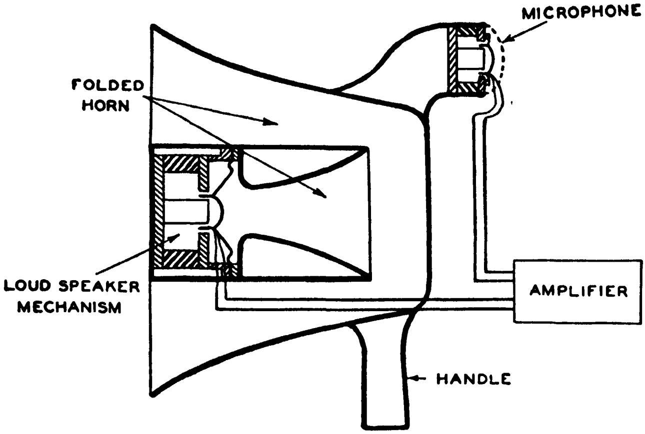

Fig. 9.22. Sectional view depicting the elements and electrical connections of an electrical megaphone.

The only theoretical limitation to the amount of renforcing is the production of continuous oscillations due to regenerative feedback from the loud speaker 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. 2.14). By placing the microphone at the rear of the horn and 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 loud speaker is a conventional, light-weight horn loud speaker. Two types of amplifiers have been used — namely, a portable battery operated amplifier and a semi-portable a-c line operated amplifier.

[32] Sanial, A. J., *Communications*, Vol. 25, No. 7, p. 33, 1945.

√ **9.7. Magnetic Wire Sound Reproducing System.**— Magnetic recording on steel wire and tape has been done for almost fifty years. The telegraphone was invented by Poulsen [33] in 1898. Since that time there has been a periodic revival of magnetic recording. Within the last few years interest has been revived due to improvement in the recording medium and the recording and reproducing means.[34, 35, 36, 37, 38, 38'] The elements of a complete magnetic wire sound reproducing system are described in Sec. 11.4*H*. It is the purpose of this section to describe a magnetic wire reproducer.

The recording or reproducing head located midway between the two spools is designed to use longitudinal magnetization [39] (Fig. 9.23). The

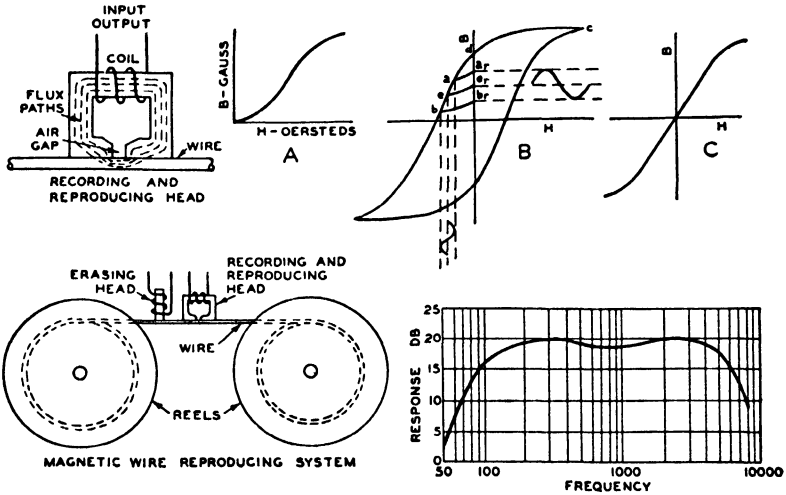

FIG. 9.23. The elements of a magnetic wire reproducing system. *A*. The B-H characteristic of an iron or steel wire. *B*. Magnetic characteristics using d-c polarizing. *C*. Magnetic characteristics using high frequency superimposed upon the signal. The response frequency characteristic depicts the overall response as a function of the frequency.

magnetizing force, during recording, is directed longitudinally along the wire, parallel to the direction of motion. It may be mentioned in passing that with tape any of three methods[40] may be used: longitudinal, magnetic

[33] Poulsen, V., *Ann. der Phys.*, Vol. 3, p. 754, 1900.
[34] Hickman, C. N., *Bell Syst. Tech Jour.*, Vol. 16, No. 2, p. 165, 1937.
[35] Begun, S. J., *Jour. Soc. Mot. Pic. Eng.*, Vol. 28, No. 5, p. 464, 1937.
[36] Begun, S. J., *Proc. Inst. Rad. Eng.*, Vol. 29, No. 8, p. 423, 1941.
[37] Toomin and Wildfeuer, *Proc. Inst. Rad. Eng.*, Vol. 32, No. 11, p. 664, 1944.
[38] Selby, M. C., *Electronics*, Vol. 17, No. 5, p. 133, 1944.
[38'] Begun, S. J., *Jour. Soc. Mot. Pic. Eng.*, Vol. 48, No. 1, p 1., 1947.
[39] Pugsley, D. W., *Electronic Industries*, Vol. 3, No. 1, p. 116, 1944.
[40] Wooldridge, D. E., *Elec. Eng.*, Vol. 65, No. 6, p. 343, 1946.

force parallel to the direction of motion; transverse, magnetic force, magnetic force perpendicular to the direction of motion and parallel to the face of the tape; and perpendicular, magnetic force perpendicular to both the direction of motion and the face of the tape. In the sectional view of the head and wire, the coil is energized with audio-frequency current from an amplifier. A complete magnetic recording system is described in Sec. 11.4*H* and depicted in Fig. 11.37. The current in the coil sets up a magnetic flux as shown in Fig. 9.23. The flux paths pass through the wire instead of the air gap because the reluctance of the steel wire is much smaller than the air gap. The section of the wire in the air gap is magnetized longitudinally. It will be seen that it is important that the wire be kept in close contact with the head as it passes the air gap in order to keep the reluctance of the magnetic path as low as possible.

The same head may be used for reproducing. If a magnetized wire is pulled past the head, the flux in the head will vary in accordance with the magnetization in the wire. A voltage will be generated in the coil due to variation of the flux.

An erasing coil is located ahead of the recording head. This erases any previous record just prior to recording. The erasing is accomplished by energizing the erasing coil with a current of 30,000 cycles.

The B–H characteristic of steel or iron depicted in Fig. 9.23*A* is not a straight line. Therefore, recording in the manner indicated above would produce distortion due to the nonlinear magnetic charactersitics of the steel wire.

The magnetic nonlinearity may be overcome by saturating the wire with a direct current field to the point c in Fig. 9.23*B*. When the field is removed the residual flux returns to point d. Then a reverse bias field is applied which lowers the flux to point e. After these operations the recording is made about the point e. All these operations are carried out in a single head. After the wire has passed the recording head the flux in the wire is a_r, e_r and b_r instead of a, e and b. If the characteristic between a and b is linear there will be no little distortion due to the counter magnetization when the head is removed from the wire.

In another method the magnetic nonlinearity[40'] is overcome by using a high-frequency current of about 30,000 cycles. The same source is used as the erasing coil. An appropriate high-frequency current is mixed with the audio signal current and both applied to the recording head at the same time. The retained B plotted against H employing the high-fre-

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[40'] Carpenter, G. W., and Carlson, W. L., U. S. Patent 1,640,881., quency bias is shown in Fig. 9.23*C*. This is a linear characteristic over a wide range. Under these conditions, recordings with low distortion can be made.

The response frequency characteristic, employing a wire .004 inch in diameter moving at 6 feet per second is shown in Fig. 9.23.

9.8. Motion Picture Film Sound Reproducing System.[41] — The elements of a complete motion film reproducing system are described in Sec. 11.4*E*. The sound track on 35 millimeter film occupies a space about .1 inch wide just inside the sprocket holes as shown in Fig. 9.24. 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. 9.24

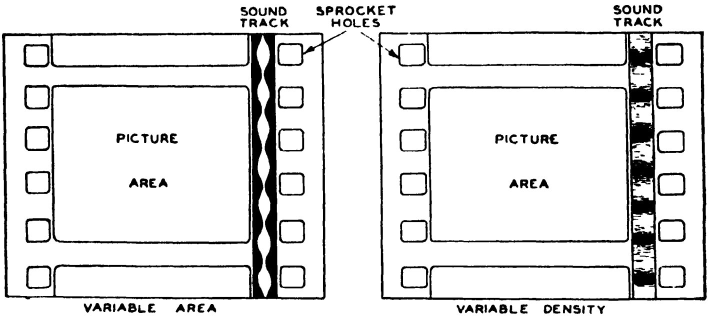

FIG. 9.24. The position of the picture and sound track in 35-millimeter sound motion picture film. Two types of sound track are shown — namely, variable area and variable density.

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. In addition to the conventional variable density sound track shown in Fig. 9.24, there are other types as, for example, squeeze, class A pushpull and class B pushpull variable density sound tracks. It is the purpose of this section to describe the elements for recording the sound on the film and the elements for reproducing a sound motion picture film record.

A. *Recording System.* — 1. *Variable area.*[42, 43, 44, 45] In the variable area

[41] Albin, F., Clark, L. E., Hill, A. P., Hilhard, J. K., Kimball, Harry, Lambert, Kenneth, and M'ller, Wesley, "Motion Picture Sound Engineering," D. Van Nostrand Company, New York, N. Y., 1938.
[42] Hardy, A. C., *Trans Soc. Mot. Pic. Eng.*, Vol. 11, No. 31, p. 475, 1927.
[43] Dimmick, G. L., *Jour. Soc. Mot. Pic. Eng.*, Vol. 15, No. 4, p. 428, 1930.
[44] Kellogg, E. W., *Jour. Soc. Mot. Pic. Eng.*, Vol. 25, No. 3, p. 203, 1935.
[45] Sachtleben, I. T., *Jour. Soc. Mot. Pic. Eng.*, Vol. 25, No. 2, p. 175, 1935.

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. 9.25. The triangular aperture is uniformly illuminated by means of a lamp and lens system. The image of the tri-

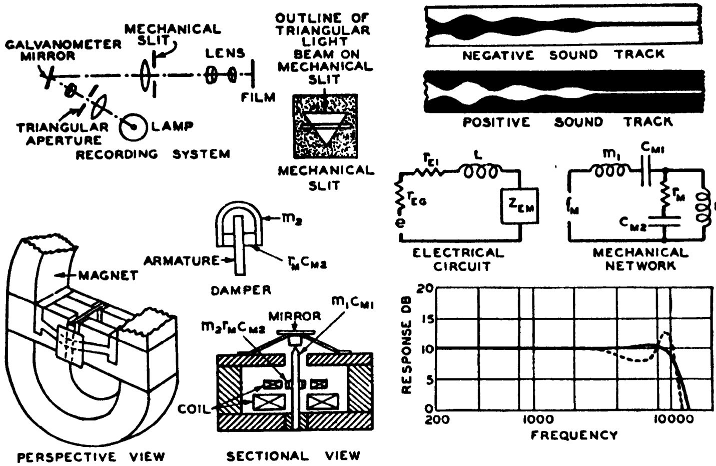

FIG. 9.25. The elements of a variable area sound motion picture film recording system. The negative and positive sound tracks. Perspective and sectional views, mechanical network and electrical circuit of the galvanometer. In the mechanical network: f_M = the mechanical driving force. m_1 and C_{M1} = 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.

angular aperture is reflected by the galvanometer mirror focused on the mechanical slit. The mechanical slit in turn is focused on the film. The galvanometer mirror swings about an axis parallel to the plane of the paper. The triangular light image on the mechanical slit moves up and down on the mechanical slit. The result is that the width of the exposed portion of the negative sound track corresponds to the rotational vibrations of the galvanometer. In the positive record the width of the unexposed portion corresponds to the signal.

The amount of ground noise produced is proportional to the exposed portion of the positive sound track. For this reason it is desirable to make the unexposed portion[46,47,48] of the record just wide enough to accommodate the modulation. This is accomplished by applying a bias signal to the gavlanometer. In the absence of a signal a very narrow exposed portion 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 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. 9.25. 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. 9.25. It will be seen that the rotational amplitude is uniform with respect to frequency to about 10,000 cycles.

2. *Variable Density.*[49] — 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. 9.26. The ribbons of the light value 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.

[46] Kreuzer, B., *Jour. Soc. Mot. Pic. Eng.*, Vol. 16, No. 6, p. 671, 1931.
[47] Dimmick, G. L., *Jour. Soc. Mot. Pic. Eng.*, Vol. 29, No. 3, p. 258, 1937.
[48] Kellogg, E. W., *Jour. Soc. Mot. Pic. Eng.*, Vol. 36, No. 2, p. 137, 1941.
[49] MacKensie, D., *Trans. S.M.P.E.*, Vol. 12, No. 35, p. 730, 1928.

Ground noise reduction[50, 50'] 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. 9.26. 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

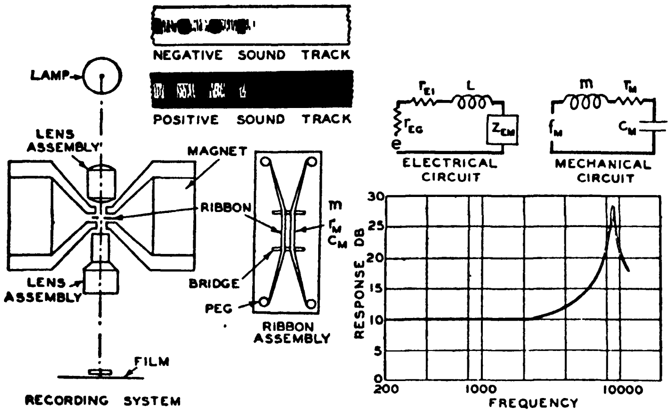

FIG. 9.26. The elements of a variable density sound motion film recording system. The negative and positive sound tracks. Sectional and ribbon assembly views, mechanical circuit and electrical circuit of the light value. In the mechanical circuit: f_M = the mechanical driving force. m, r_M and C_M = the mass, mechanical resistance and compliance of the ribbons. In the electrical circuit: z_{EM} = the motional electrical impedance. L and r_{EI} 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 value.

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. 9.26.

B. *Reproducing System.* — The elements of a motion picture film sound reproducing system are shown in Fig. 9.27. 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 usually about .00075 inch.

[50] Silent and Frayne, *Jour. Soc. Mot. Pic. Eng.*, Vol. 18, No. 5, p. 551, 1932.
[50'] Scoville and Bell, *Jour. Soc. Mot. Pic. Eng.*, Vol. 38, No. 2, p. 125, 1942.

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

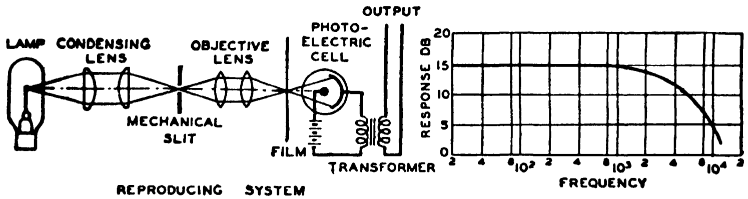

FIG. 9.27. The elements of a motion picture film sound reproducing system and the voltage response frequency characteristic with a constant amplitude film.

voltage output response frequency characteristic of a typical motion picture film sound reproducing system using a constant amplitude film is shown in Fig. 9.27. 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.

9.9. 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 simplest hearing aid[51] consists of a carbon microphone, a battery, an attenuator and a telephone receiver (Fig. 9.28*A*). This hearing aid will give satisfactory service where the hearing loss is about 20 db.

The hearing aid shown in Fig. 9.28*B* consists of a carbon microphone, a carbon amplifier, an attenuator, a battery and a telephone receiver. This hearing aid will give satisfactory service where the hearing loss is about 40 db.

The quality of the carbon type hearing aids is usually very poor, due to the frequency and the amplitude distortion produced by the carbon microphone and amplifier.

[51] Tuffnell, W. L., *Bell Labs. Record*, Vol. 18, No. 1, p. 8, 1939.

During the past few years hearing aids[52,53,54,55,56,57,58,59] employing vacuum tube amplifiers have almost completely replaced 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

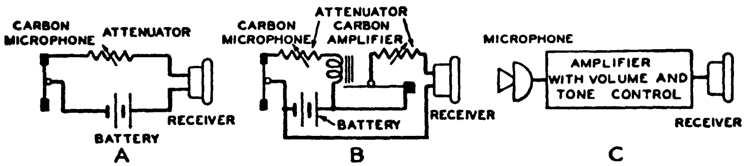

FIG. 9.28. Hearing aids. *A*. Simple carbon microphone hearing aid. *B*. Carbon microphone hearing aid with a mechanical carbon amplifier. *C*. Schematic diagram of a vacuum tube hearing aid.

arrangement of the components of a vacuum tube hearing aid is shown in Fig. 9.28*C*. The microphone used in hearing aids today is a diaphragm crystal type similar to that described in Sec. 8.2*C*2 and depicted in Fig. 8.5. 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. 9.29. A moulded plug fits the ear cavity and holds the receiver in place. Under these conditions the leak at the ear is very small. Therefore, good response is obtained at the low frequencies. The action of the system is essentially the same as that of a bipolar telephone receiver considered in Sec. 9.2*A* 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

[52] Ramanow, F. F., *Jour. Acous. Soc. Amer.*, Vol. 13, No. 3, p. 295, 1942.
[53] Sabine, P. E., *Jour Acous. Soc. Amer.*, Vol. 16, No. 1, p. 38, 1944.
[54] Carlisle and Mundel, *Jour. Acous. Soc. Amer.*, Vol. 16, No. 1, p. 45, 1944.
[55] Grossman and Molloy, *Jour. Acous. Soc. Amer.*, Vol. 16, No. 1, p. 52, 1944.
[56] Hanson, W. W., *Jour. Acous. Soc. Amer.*, Vol. 16, No. 1, p. 60, 1944.
[57] LeBel, C. J., *Jour. Acous. Soc. Amer.*, Vol. 16, No. 1, p. 63, 1944.
[58] Watson, N. A., *Jour. Acous. Soc. Amer.*, Vol. 16, No. 3, p. 194, 1945.
[59] Strommen, E., *Jour. Acous. Soc. Amer.*, Vol. 15, No. 4, p. 211, 1944.

inner ear by means of a bone conduction receiver.[60] 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. 9.30. By means of the multiple resonant system it is possible to deliver a large force to the relatively high mechanical impedance,

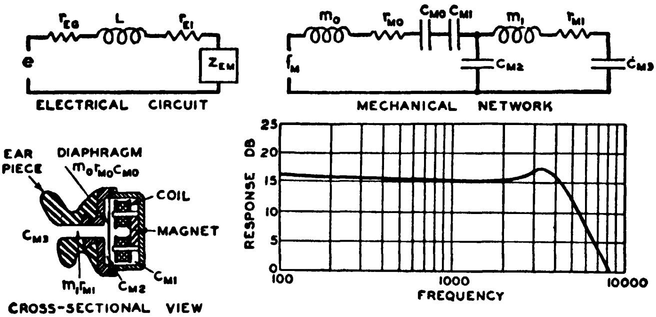

FIG. 9.29. Cross-sectional view, mechanical network, electrical circuit and response frequency characteristic of an insert type telephone receiver. In the mechanical network: f_M = the mechanical driving force. m_0, r_{M0} and C_{M0} = the mass, mechanical resistance and compliance of the diaphragm. 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.

z_{ME}, of the mastoid bone. The response frequency characteristic is quite good considering the difficult conditions under which the bone conduction receiver operates.

9.10. Electrical Musical Instruments.[61] — 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.

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 loud speaker.

[60] Hawley, M. S., *Bell Labs. Record*, Vol. 18, No. 1, p. 12, 1939.

[61] For a comprehensive paper on "Electronic Music and Instruments," Miessner, B. F., *Proc. I.R.E.*, Vol. 24, No. 11, p. 1427, 1936.

Electric pianos[62] 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 loud speakers.

One type of electric organ[63] consists of a number of small alternators

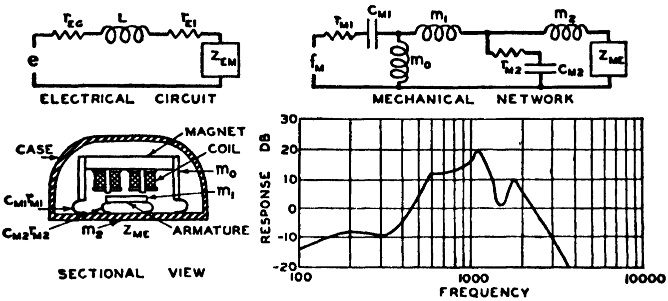

FIG. 9.30. Cross-sectional view, mechanical network, electrical circuit and response frequency characteristic. In the mechanical network: m_O = the mass of the coil and magnetic structure. m_1 = the mass of the armature. C_{M1} and r_{M1} = compliance and the mechanical resistance connecting the coil and magnetic structure to the case. C_{M2} and r_{M2} = the compliance and mechanical resistance connecting the armature to the case. m_2 = 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.

(one for each note), a keying and mixing system for adjusting the quality, an amplifier and a loud speaker. Another electric organ[64] employs wind driven reeds. The vibrations of the reeds are converted into the corresponding electrical variations, amplified and reconverted into sound by means of loud speakers.

A versatile electronic musical instrument[65,66] in which the wave shape and harmonic content may be varied over wide limits derives the needed frequencies from twelve high-frequency oscillators followed by cascade

[62] Miessner, B. F., *Proc. I.R.E.*, Vol. 24, No. 11, p. 1427, 1936.
[63] Hammond, L., U. S. Patent 1,956,350.
[64] Hoschke, U. S. Patent 2,015,014.
[65] Hammond, L., *Science*, Vol. 89, p. 6, Feb. 10, 1939.
[66] Hammond, L., *Electronics*, Vol. 12, No. 11, p. 16, 1939.

frequency dividers in which the frequency is divided in each stage. The overtone structure of each note is made a function of the input level by the use of an over-biased nonlinear amplifier. Practically any musical instrument such as the organ, piano, guitar, violin, trombone, etc., can be simulated by this instrument.

Electric carillons[67] consisting of tuned coiled vibrators, magnetoelectric translators, amplifiers and reproducers possess qualities which are quite similar to the conventional carillons.

9.11. Sirens.[68] — 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[69] 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.

9.12. Compressed Air Loud Speaker.— The compressed air loud speaker consists of an electrically actuated valve which interrupts or modulates an air stream. Thus, the output consists of a series of puffs, the envelope of which corresponds to electrical impulses which actuate the valve. Horns are usually coupled to the system to improve the efficiency. In these systems the sound power output may be several times the electrical input to the valve. As a consequence, it is possible to obtain very large acoustical outputs with relatively small electrical inputs (small power amplifiers).

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[67] Curtiss, A. N., U. S. Patent 2,026,342.

[68] Wood, " A Textbook of Sound," The Macmillan Company., New York, N. Y.

[69] Jones, Clark, *Jour. Acous. Soc. Amer.*, Vol. 18, No. 2, p. 371, 1946.

9.13. Seismic Detectors.[70] — The variation of the velocity of sound in the various strata comprising the earth's crust forms the basis of geophysical 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 confirmation 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.

9.14. Stethoscopes.[71,72,73] — The ordinary acoustical stetnoscope 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,

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[70] Silverman, Daniel., *Jour. A.I.E.E.*, Vol. 58, No. 11, 455, 1939.
[71] Rappaport and Sprague, *Amer. Heart Jour.*, Vol. 21, p. 257, 1941.
[72] Frederick and Dodge, *Bell Syst. Tech. Jour.*, Vol. 3, No. 4, p. 531, 1924.
[73] Singer, C., *Electronics*, Vol. 38, No. 6, p. 66, 1939.

that makes it possible to diagnose normal or abnormal conditions by ausculation.

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. 9.31 shows the frequency bands of some of the most common sounds. The fundamentals are shown as dark areas and the harmonics or overtones as

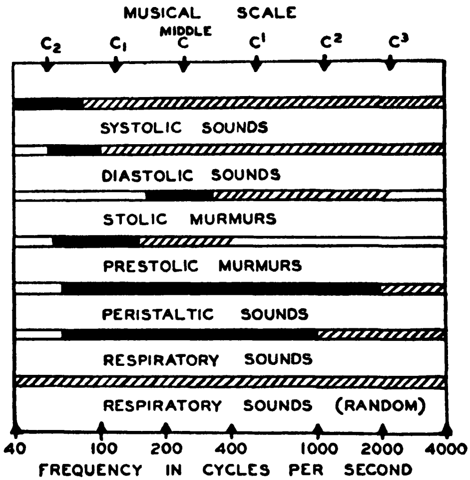

FIG. 9.31. The frequency ranges of sounds generated in the body. The frequency ranges of the fundamental frequencies are shown as solid lines. The frequency ranges of the harmonics and overtones are shown as cross-hatched lines.

cross-hatched areas. The fundamental of the systolic sound ranges from 40 to 80 cycles. There 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. 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 true fundamental. The components of these sounds are scattered over the entire audible spectrum.

From Fig. 9.31 it is quite evident that, in order to obtain the maximum intelligence from the stethscope 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. 9.32. 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 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 ranges 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[74,75] is shown in Fig. 9.32C. The

[74] Olson, H. F., *Electronics*, Vol. 16, No. 8, p. 184, 1943.
[75] Olson, H. F., U. S. Patents 2,363,686 and 2,389,868.

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

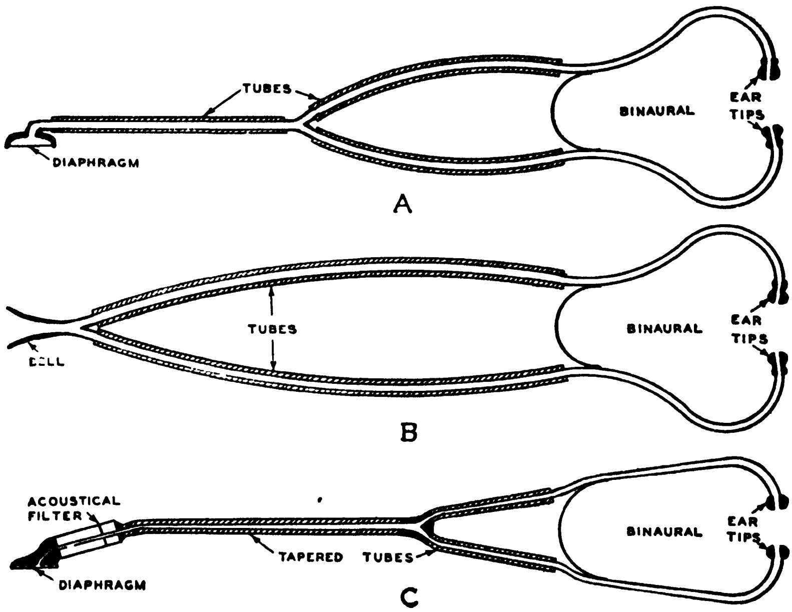

FIG. 9.32. Sectional views of stethoscopes. *A*. Diaphragm type. *B*. Open bell type. *C*. Wide range selective type.

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 murmers, 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 characteristics 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 acoustical stethoscope This is due to the fact that most of these noises in the acoustical stethoscope fall below the threshold of hearing.

9.15. Ear Defenders.[76]—Ear defender is a term used to designate a device which introduces attentuation of sound between a point outside the head and the ear drum. 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 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 attentuation below 500 cycles is quite small. In order to obtain high attentuation 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. 9.33. A successful insert type ear defender must be made of suitable material combined with a shape which will provide

[76] Watson and Knudsen, *Jour. Acous. Soc. Amer.*, Vol. 15, No. 3, p. 153, 1944.

adequate attentuation, comfort, easy insertion and positive retention. 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

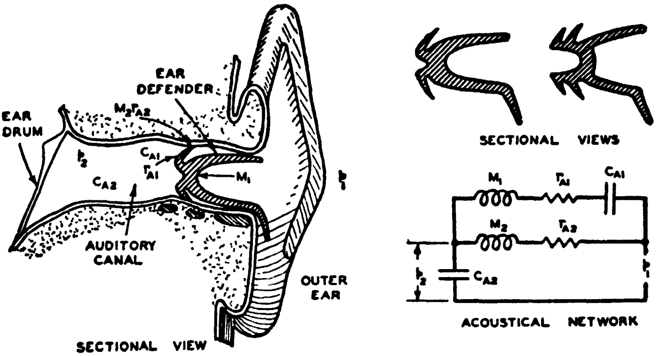

FIG. 9.33. Sectional view of an ear defender in the ear canal and the acoustical network of the system. In the acoustical network: M_1 = the inertance due to the mass of the ear defender. C_{A1} and r_{A1} = the effective acoustical capacitance and acoustical resistance of the ear defender with respect 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.

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. 9.33). 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.

CHAPTER X

MEASUREMENTS

10.1. Introduction. — The rapid progress made in acoustics during the past two decades has resulted in a corresponding advance in acoustical measurements. 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, loud speakers and telephone receivers together with fundamental acoustical measurements.

10.2. Calibration of Microphones.[1,2] — 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 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

[1] 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.

[2] Standards of the Institute of Radio Engineers, 1933.

e/p is usually expressed in decibels with respect to some arbitrary reference level. The pressure upon the diaphragm may be generated by a pistonphone, thermophone or an electrostatic actuator.

(a) *Pistonphone.*[3,4,5,6] — A schematic arrangement of a pistonphone for use in calibrating a pressure type microphone having a diaphragm of high acoustical impedance is shown in Fig. 10.1*A*. The small piston is driven

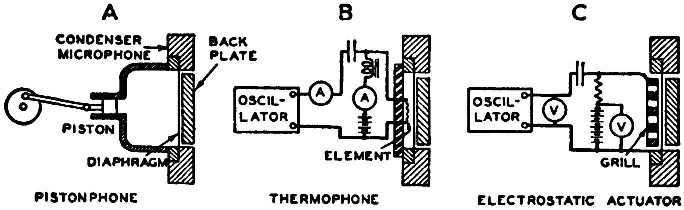

FIG. 10.1. Apparatus for obtaining the pressure frequency characteristic of a condenser type microphone. The pistonphone and thermophone may be used for other types of pressure microphones.

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.

[3] Wente, E. C., *Phys. Rev.*, Vol. 10, No. 1, p. 39, 1917.
[4] Wente, E. C., *Phys. Rev.*, Vol. 19, No. 4, p. 333, 1922.
[5] Kaye, G. W. C., *Jour. Acous. Soc. Amer.*, Vol. 7, No. 3, p. 174, 1936.
[6] Glover and Baumzweiger, *Jour. Acous. Soc. Amer.*, Vol. 10, No. 3, p. 200, 1939.

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.

(b) *Thermophone.*[7,8,9] — The thermophone consists of one or more strips of thin gold leaf mounted upon terminal blocks (Fig. 10.1*B*). 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

$$p = \frac{.96 S i_0 i r_E}{\omega m C V A \alpha D^{1/2}} \tag{10.2}$$

where $D = \left(1 - \dfrac{4KS^2}{\omega CVA}\right)^2 + \left(1 + \dfrac{4S}{VA\alpha} + \dfrac{4KS\alpha}{\omega C} + \dfrac{4KS^2}{\omega CVA}\right)^2$

$$A = \frac{T_a}{T_S}\frac{\gamma}{\gamma - 1} - 1$$

$$m = \frac{(\gamma - 1)T_S}{\gamma p_0}$$

$$\alpha = \sqrt{\frac{\omega C_p \rho}{2K}}$$

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,

[7] Arnold and Crandall, *Phys. Rev.*, Vol. 10, No. 1, p. 22, 1917.

[8] Wente, E. C., *Phys. Rev.*, Vol. 19, No. 4, p. 333, 1922.

[9] Ballantine, S., *Jour. Acous. Soc. Amer.*, Vol. III, No. 3, p. 319, 1932.

r_E = total electrical resistance of the strip, in ohms,
T_S = mean temperature of the strip, in degrees Kelvin,
T_α = 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.

The determination of the ratio e/p is carried out in the same manner as the pistonphone.

(c) *Electrostatic Actuator.*[10] — The electrostatic actuator consists of an auxiliary electrode in the form of a grill mounted in front of the microphone diaphragm, Fig. 10.1*C*. The actuator is perforated so that it does not 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,[11] in dynes per square centimeter of the grill, assuming no tufting of the electrostatic lines, is

$$p = \frac{8.85 e_0 e}{d^2} \times 10^{-7} \tag{10.3}$$

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

[10] Ballantine, S., *Jour. Acous. Soc. Amer.*, Vol. 3, No. 3, p. 319, 1932.
[11] Olson, "Dynamical Analogies," D. Van Nostrand Company, New York, N. Y., 1943.

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.

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 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.*[12,13,14,15] — 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[16] developed the formula for the turning moment of the disk as

$$M = \frac{4}{3} \rho a^3 u^2 \sin 2\theta \qquad 10.4$$

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

$$\phi = \frac{M}{S} \qquad 10.5$$

where S = moment of torsion of the suspension, in dyne centimeter.

The moment of torsion of the suspension is given by

$$S = \frac{I}{T^2} [4\pi^2 + (\log_e \gamma)^2] \qquad 10.6$$

[12] Rayleigh, *Phil. Mag.*, Vol. 14, p. 186, 1882.
[13] Ballantine, *Phys. Rev.*, Vol. 32, No. 6, p. 988, 1928.
[14] Olson and Goldman, *Electronics*, Vol. 4, No. 9, p. 106, 1931.
[15] Sivian, L. J., *Bell Syst. Tech. Jour.*, Vol. 10, No. 1, p. 96, 1931.
[16] Koenig, *Ann. d. Physik*, Vol. 43, p. 43, 1891.

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 10.4, 10.5 and 10.6 it is possible to determine the particle velocity u in the sound wave.

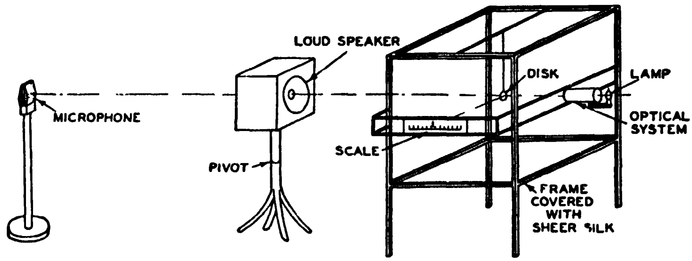

FIG. 10.2. Arrangement of apparatus for obtaining the free-field response of a microphone by means of a Rayleigh disk.

The arrangement of a Rayleigh disk for field calibrations of microphones is shown in Fig. 10.2. The source of sound is a small direct radiator loud speaker, with the back completely enclosed, placed halfway between the disk and the microphone. A small loud speaker 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. 1.5*D* and Fig. 1.3). From the geometry of the system of Fig. 10.2 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.[17] Ballantine [18] established reciprocity theorems for mechanoacoustic, electromechano, and electromechanoacoustic systems. Ballantine also carried out a generalized discussion to show that a microphone may be calibrated in terms of electrical standards by the use of the extended reciprocity relations. Later other

[17] Rayleigh, "Theory of Sound," The Macmillan Company, Vol. I., p. 145.
[18] Ballantine, S., *Proc. Inst. Rad. Engrs.*, Vol. 17, No. 6, p. 929, 1929.

investigators[19,20,21] 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,

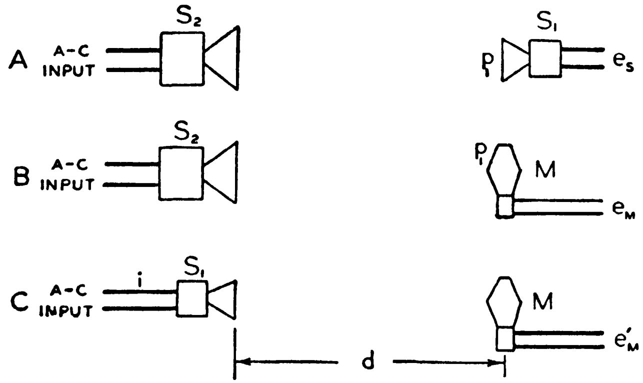

FIG. 10.3. The three experiments of the reciprocity procedure for obtaining the free-field calibration of a microphone. *A*. The open-circuit voltage, e_S, of the reversible microphone loud speaker, S_1, when used as a microphone and actuated by a sound pressure, p_1. *B*. The open-circuit voltage, e_M, 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 loud speaker, S_1, used as a loud speaker with a current input, i, and a spacial separation, d.

to be calibrated, a reversible microphone loud speaker S_1, and a loud speaker S_2. For the reversible microphone loud speaker it is convenient to used a small back-enclosed loud speaker.

The first and second experiments are shown schematically in Fig. 10.3*A* and Fig. 10.3*B*. An alternating current is fed to the loud speaker 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 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

$$p_1 = \frac{e_S}{K_S} = \frac{e_M}{K_M} \qquad . \qquad 10.7$$

[19] Cook, R. K., *Jour. of Research*, Nat. Bur. of Standards, Vol. 25, No. 5, p. 489, 1940.

[20] McLean, W. R., *Jour. Acous. Soc. Amer.*, Vol. 12, No. 1, p. 140. 1940.

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

The voltage output,[22] in abvolts, of the microphone loud speaker S_1 used as a microphone is

$$e_S = Bl\dot{x}_1 \qquad 10.8$$

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.

The velocity, in centimeters per second, of the vibrating system of S_1 as a microphone is

$$\dot{x}_1 = \frac{p_1 A}{z_M} \qquad 10.9$$

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.

From equations 10.7, 10.8 and 10.9

$$\frac{e_S}{p_1} = \frac{BlA}{z_M} = K_S \qquad 10.10$$

The third experiment is shown in Fig. 10.3*C*. The velocity,[23] in centimeters per second, of the diaphragm and voice coil of S_1 for a current i, in abamperes, in the voice coil is

$$\dot{x} = \frac{Bli}{z_M} \qquad 10.11$$

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

$$p = \frac{\rho ckA\dot{x}}{4\pi d} \qquad 10.12$$

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

[22] Olson, "Dynamical Analogies," D. Van Nostrand Company, New York, N. Y., 1943.
[23] Olson, "Dynamical Analogies," D. Van Nostrand Company, New York, N. Y., 1943.

$k = 2\pi/\lambda$,
λ = wavelength, in centimeters, and
c = velocity of sound.

From equations 10.11 and 10.12,

$$p = \frac{\rho ckABli}{4\pi dz_M} \qquad 10.13$$

From equations 10.10 and 10.13,

$$p = \frac{\rho ckiK_S}{4\pi d} = \frac{r_A iK_S}{2d\lambda} \qquad 10.14$$

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

$$p = \frac{e'_M}{K_M} \qquad 10.15$$

From equations 10.14 and 10.15,

$$\frac{e'_M}{K_M} = \frac{r_A iK_S}{2d\lambda} \qquad 10.16$$

From equation 10.7,

$$K_M = \frac{e_M}{e_S} K_S \qquad 10.17$$

When K_S is eliminated from equations 10.16 and 10.17, the response of the microphone M, in abvolts per dyne per square centimeter, is

$$K_M = \sqrt{\frac{2d\lambda e_M e'_M}{r_A i e_S}} \qquad 10.18$$

where e_S, e_M, e'_M and i are obtained from the experiments of Fig. 10.3. 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. 10.3*A*4.

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. *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 [24] which simulate actual operating conditions. The artificial voice consisting of a small loud-speaker unit mounted in the head of a manikin, as shown in Fig. 10.4, provides a means for obtaining the response frequency characteristics of close talking microphones. Resonances in the tube connecting the loud-speaker unit and the mouth are eliminated

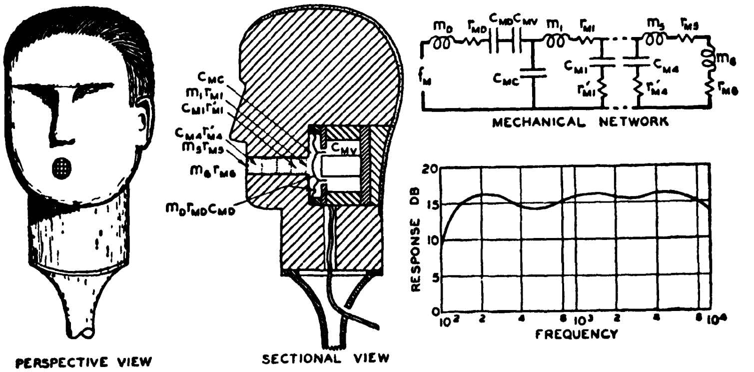

FIG. 10.4. Perspective view, sectional view, mechanical network and response frequency characteristic of an artificial voice. In the mechanical network: m_D = the mass of the diaphragm and suspension of the small loud-speaker unit. r_{MD} and C_{MD} = the mechanical resistance and compliance of the suspension system of the small loud-speaker unit. C_{MC} = the compliance of the air chamber behind the diaphragm. $m_1, r_M \ldots m_5, r_{M5}$ = the masses 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.

by the introduction of series and shunt mechanical resistances. The response frequency characteristic shown in Fig. 10.4 can be obtained by a suitable choice of constants of the mechanical system.

4. *Artificial Throat.* — Throat microphones have been described in Sec. 8.8. Throat microphones are actuated by sound waves transmitted through the throat. An artificial throat [25] for testing throat microphones consists of a mass controlled system driven by a voice located in magnetic field. Specifically the voice coil is coupled to a massive platform. The

[24] Inglis, Gray and Jenkins, *Bell Syst. Tech. Jour.*, Vol. 11, No. 2, p. 293, 1932.
[25] Greibach, E. H., *Elec. Eng.*, Vol. 65, No. 4, p. 184, 1946.

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 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 with respect to direction. A polar diagram showing the output variation of the microphone with direction is usually employed.

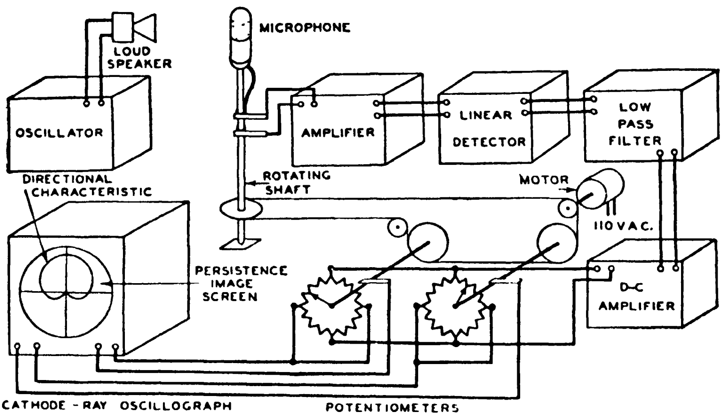

FIG. 10.5. Schematic arrangement of the apparatus employing a cathode-ray tube with a long persistence screen as a polar directional characteristic indicator and recorder.

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 loud speaker. The apparatus of Fig. 10.5 is arranged to obtain the directional characteristic of the microphone. The directional characteristic of the loud speaker may be obtained by placing the loud speaker 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.

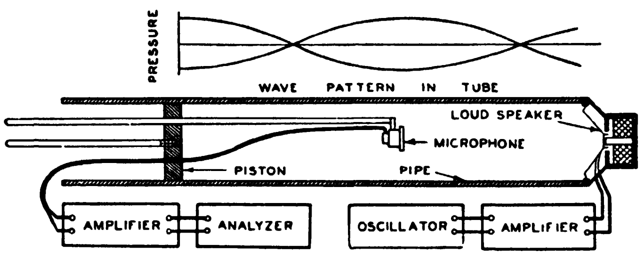

Fig. 10.6. Arrangement of apparatus for measuring the nonlinear distortion generated by a microphone. (After Phelps.)

C. *Nonlinear Distortion Characteristic.* — The harmonic distortion tests are intended to show the spurious harmonics generated by the microphone 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[26] shown in Fig. 10.6 provides a simple means of obtaining a sound wave free from 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 loud speaker. 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. 10.3*C*). For the third harmonic the microphone is

[26] Phelps, W. D., *Jour. Acous. Soc. Amer.*, Vol. 11, No. 2, p. 219, 1939.

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. 8.3*B*), 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 used as the reference standard. The standard microphone and the micro-

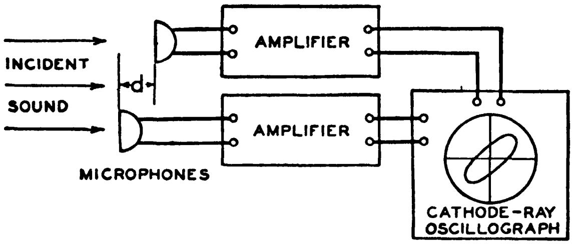

FIG. 10.7. Schematic arrangement of apparatus for measuring the phase characteristic of a microphone.

phone to be tested may be placed side by side in a plane progressive wave in free space, Fig. 10.7. 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

$$\phi = \frac{d}{\lambda} 360° \qquad 10.19$$

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. 10.3*G* and 8.12.

10.3. Testing of Loud Speakers.[27, 28] — Many different measurements are required to determine the performance of a loud speaker. The most important characteristics which depict the performance of a loud speaker 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 loud speaker 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

$$\text{Absolute response} = \frac{p}{\dfrac{e}{\sqrt{z_E}}} = \frac{p\sqrt{z_E}}{e} \qquad 10.20$$

where p = measured sound pressure, in dynes per square centimeter,

e = effective voltage applied to the voice coil, in volts, and

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

[27] American Standards Association, Loud Speaker Testing, C. 16.4, 1942.

[28] *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.

$$\text{Absolute response} = 20 \log_{10} \frac{\dfrac{p}{e/\sqrt{z_E}}}{\dfrac{1}{1/\sqrt{1}}} = 20 \log_{10} \frac{e/\sqrt{z_E}}{p} \qquad 10.21$$

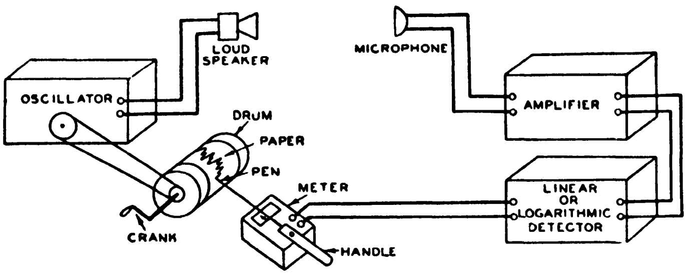

FIG. 10.8. Schematic arrangement of the apparatus for manually recording the sound pressure frequency characteristic of a sound source. (After Wolff and Ringel.)

The apparatus and methods employed for obtaining the response frequency characteristics of loud speakers 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[29] is shown in Fig. 10.8.

This method yields a response frequency curve on semilogarithmic paper in about 3 minutes. Rotation of a condenser governing the beat frequency 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.

[29] Wolff and Ringel, *Proc. I.R.E.*, Vol. 15, No. 5, p. 363, 1927.

A linear or logarithmic detector [30, 31] may be employed. In the former, the deflection of the meter is proportional to the sound pressure. In the 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

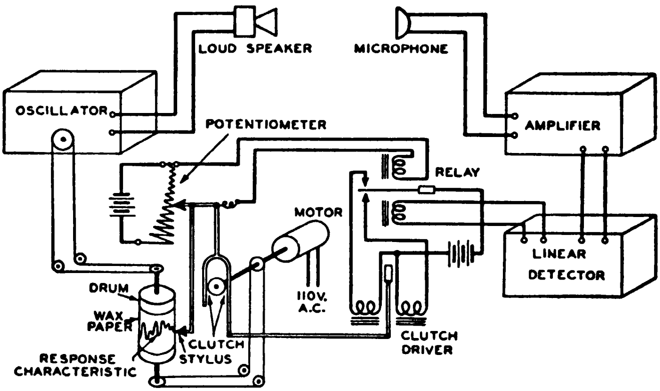

FIG. 10.9. Schematic arrangement of the apparatus used in a high-speed level recorder for automatically recording a sound pressure frequency characteristic. (After Wente, Bedell, and Swartzel.)

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 [32] 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. 10.9 shows how a pressure characteristic can be made with the sound level recorder. A dark colored tape coated 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 loud speaker 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.

[30] Ballantine, S., *Jour. Acous Soc. Amer.*, Vol. 5, No. 1, p. 10, 1933.
[31] Hackley, R. A., *Broadcast News*, No. 28, p. 20, July, 1938.
[32] Wente, Bedell and Swartzel, *Jour. Acous. Soc. Amer.*, Vol. 6, No. 3, p. 121, 1935.

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 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[33] 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 cathode-ray tube,[34, 35] with a long persistence screen, may be used as a response indicator and recorder, Fig. 10.10. A motor drives the beat frequency oscillator and a potentiometer. The potentiometer varies the 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 loud speaker. 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 time required to trace a response frequency characteristic of a loud speaker is about 30 seconds. The apparatus is very useful for development work because the motor sweeps through the range again and

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[33] Clark, W. R., *A.I.E.E. Trans.*, Vol. 59, p. 957, 1940.
[34] Hackley, R. A., *Broadcast News*, No. 28, p. 20, July, 1938.
[35] Sherman, J. B., *Proc. I. R. E.*, Vol. 26, No. 16, p. 700, 1938.

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.

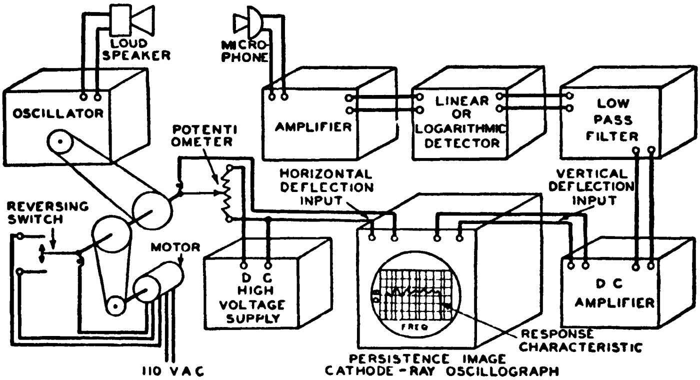

FIG. 10.10. Schematic arrangement of the apparatus employing a cathode-ray tube with a long persistence screen as a pressure response frequency indicator and recorder. (After Hackley.)

A system[36] for measuring the response of a loud speaker employing a thermal noise generator is shown in Fig. 10.11. A diode may be used as a source of thermal noise. The output is amplified, filtered and fed to a loud speaker. The frequency distribution of the energy fed to the loud speaker is shown in Fig. 10.11*A*. The output of the loud speaker 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. 10.11*B*. The band width of the filter should be independent of the frequency. The position of the filter pass band 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 loud speaker is shown in Fig. 10.11*C*.

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.*[37] —The microphone

[36] Olney, B., *Jour. Acous. Soc. Amer.*, Vol. 13, No.1, p. 79, 1942.
[37] Standards on Electroacoustics, Institute of Radio Engineers, 1938.

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.

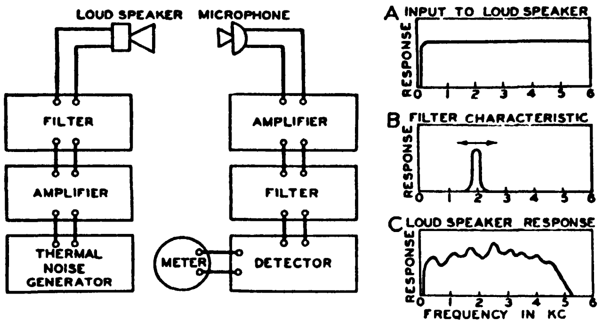

FIG. 10.11. Schematic arrangement of the apparatus employing a thermal noise generator and a band pass filter for obtaining the response frequency characteristic of a loud speaker. *A*. Input to the loud speaker. *B*. The response frequency characteristic of the band pass filter. *C*. Response frequency characteristic of the loud speaker.

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. 10.12. This arrangement has the feature that it does not require an absolute calibration of the measuring system.

Referring to Fig. 10.12, the absolute response is given by

$$\text{Absolute response, in decibels,} = 20 \log_{10} \frac{p}{e} \sqrt{z_E}$$

$$= [A - B - C - D] + 10 \log_{10} z_E \qquad 10.22$$

where A = output of measuring system, in decibels, with the microphone picking up sound from the loud speaker 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 loud speaker, in ohms.

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.

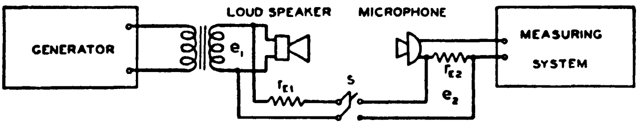

FIG. 10.12. Schematic arrangement for obtaining the factor p/e in the formula for absolute response of a loud speaker.

4. *Free-Field Sound Room.*[38, 39, 40] — 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:

$$E_R/E_D = 16\pi D^2(1 - a)/aS \qquad 10.23$$

where E_R = energy density of reflected sound, in ergs per cubic centimeter,
E_D = energy density of the direct sound, in ergs per cubic centimeter,
D = distance from the source to the observation point, in centimeters,

[38] 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).

[39] Olson, H. F., *Jour. Acous. Soc. Amer.*, Vol. 15, No. 2, p. 96, 1943.

[40] Beranek, L., *Jour. Acous. Soc. Amer.*, Vol. 18, No. 1, p. 140, 1946.

S = area of absorbing material, in square centimeters,
V = volume of room, in cubic centimeters, and
a = absorption coefficient (see Sec. 11.2A).

An examination of equation 10.23 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 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. 10.13. 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. 10.13. One-inch Ozite is spaced 1 foot from the walls, ceiling and floor. One-inch Ozite

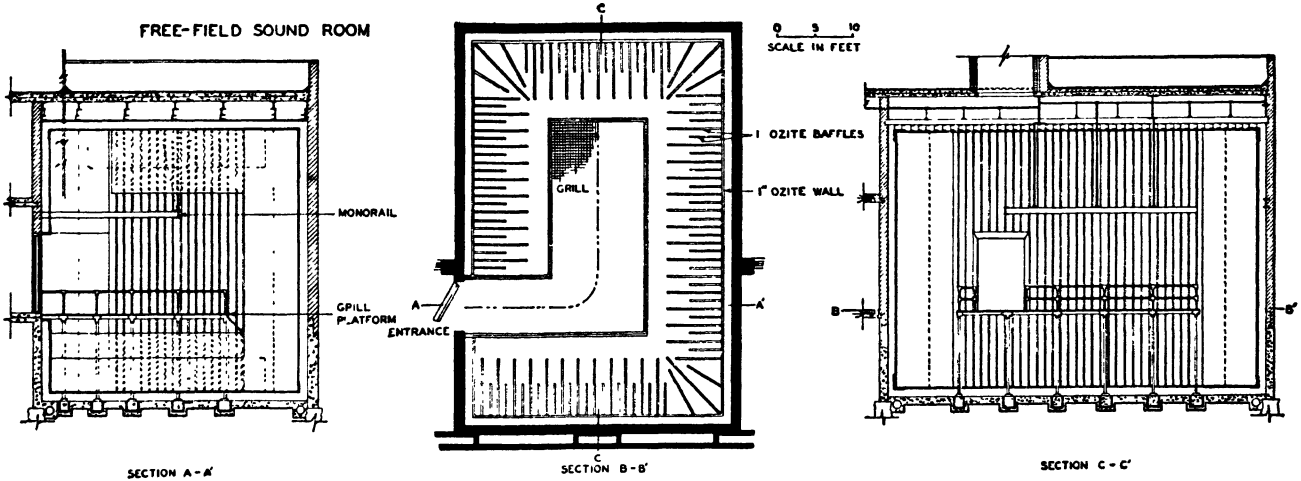

FIG. 10.13. End elevation, plan and side elevation of a free-field sound room.

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 conicides with the first floor level which makes it readily accessible to the adjoining laboratory.

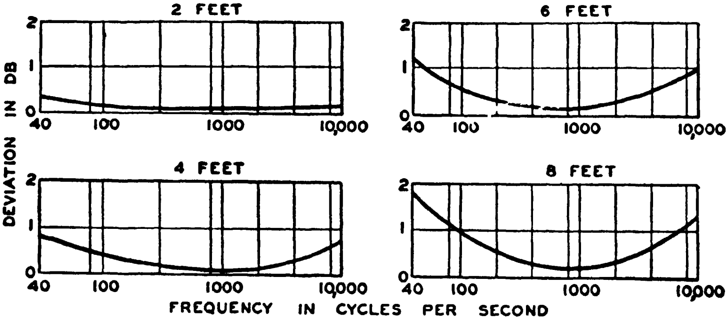

FIG. 10.14. Deviation of the pressure from an inverse distance characteristic for various distances from a sound source in the free-field sound room.

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 loud speaker. The maximum deviation in pressure from an inverse distance characteristic for various frequencies is shown in Fig. 10.14. 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 will be treated, which will make 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 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. 10.15.

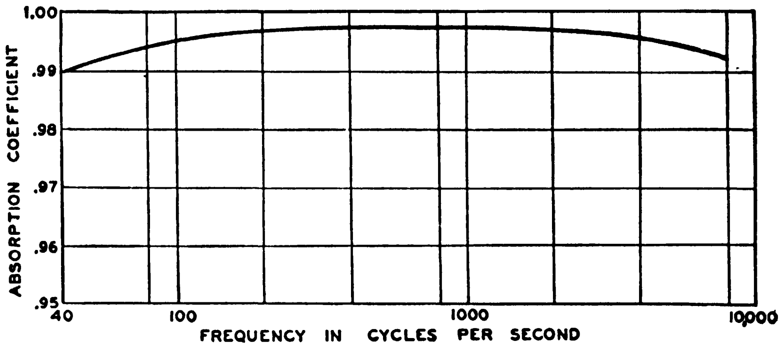

FIG. 10.15. Absorption coefficient frequency characteristic of free-field sound room.

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 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 ±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 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.

5. *Outdoor Response.* — If a free-field sound room is not available, free-field conditions may be obtained outdoors by locating the microphone and loud speaker 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 loud speaker 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 loud speakers is carrried 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 loud speaker 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 loud speaker, 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 loud speaker response. A check response frequency measurement taken close to the loud speaker with no warble should be made to determine if there are any abrupt variations in its response.

7. *Arrangement of Loud Speakers for Test.* — In obtaining response frequency characteristic of loud speakers, the systems may be divided into two classes — namely, direct radiator, loud-speaker units designed to operate in some additional structure and complete systems such as direct radiator mechanisms mounted in cabinets and horn loud speakers.

In the test of direct radiator, loud-speaker 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 2½ 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 loud-speaker systems such as direct radiator mechanisms mounted in cabinets and horn loud speakers are tested in the same manner as in the case of direct radiator, loud-speaker units, but without the use of additional baffles.

8. *Living Room Measurements.*[41] — The performance of a radio receiver in a living room will be discussed in Sec. 11.20. The characteristics shown in Fig. 11.19 were obtained with the cathode-ray response measuring system described in Sec. 10.3*A*2. However, any of the systems described in Sec. 10.3*A*2 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 loud speaker in a theater will be discussed in Sec. 11.2*E*. The characteristics for the various parts of the theater may be obtained with any equipment described in Sec. 10.3*A*2. 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 loud speaker 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 backseat 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. At high speeds, wind, road rumble and engine noises are quite high and mask 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 loud speaker 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. 10.3*A*2.

[41] Wheeler and Whitman, *Proc. Inst. Rad. Eng.*, Vol. 23, No. 6, p. 610, 1935.

B. *Directional Characteristic.* — The directional characteristic of a loud speaker 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 loud speaker 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 loud speaker 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. 10.2*B* and depicted in Fig. 10.5. The same system may be used to obtain the directional pattern of a loud speaker. In this case the loud speaker and microphone are interchanged, that is, the microphone is fixed and the loud speaker rotated.

C. *Nonlinear Distortion Characteristic.* — The nonlinear distortion characteristic of a loud speaker 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 loud speaker.

The apparatus and circuit in schematic form for measuring the distortion produced by a loud speaker are shown in Fig. 10.16. 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 loud speaker. 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 loud speaker 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. 10.2*C*.

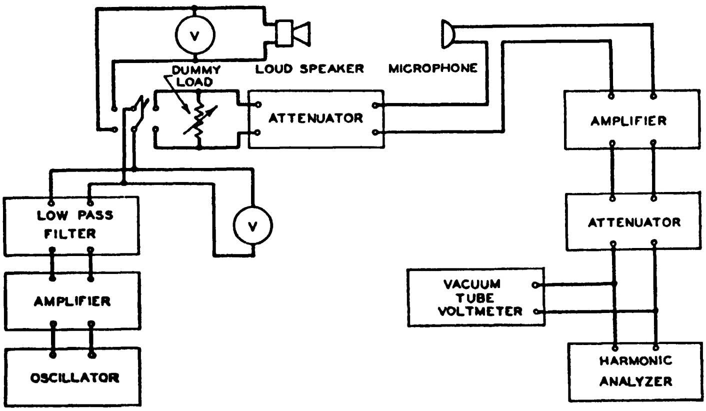

FIG. 10.16. Schematic arrangement of the apparatus for measuring the nonlinear distortion of a loud speaker.

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.

Two systems, which are in common use as harmonic analyzers, are shown in Fig. 10.17.

The heterodyne analyzer [42] is shown schematically in Fig. 10.17*A*. The 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

[42] Arguimbau, L. B., *General Radio Experimenter*, Vol. 8, p. 1, June, July, 1933.

band is passed through the selective tuned amplifier, detected and then measured on a meter.

The balance bridge for measuring the total distortion is shown schematically in Fig. 10.17*B*. 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 to be tested are adjusted by means of suitable networks so that none of the fundamental remains. The remainder is the total harmonic generated by the system under test. This is measured by means of a root-mean-square meter.

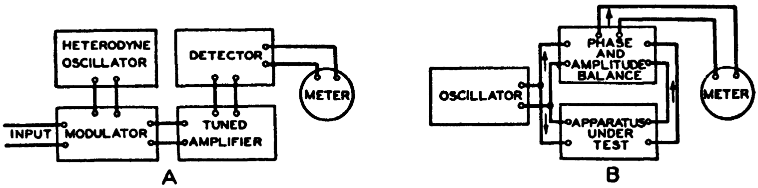

FIG. 10.17. Harmonic analyzers. *A*. Heterodyne analyzer. *B*. Balance bridge analyzer.

The above procedure for obtaining the harmonic content requires considerable time and sometimes a qualitative indication requiring only a few minutes has been found to be very useful.

Fig. 10.18 shows a cathode-ray oscillograph used to indicate the extent to which harmonic distortion is introduced by a loud speaker. 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 loud speaker. 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 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 loud speaker can be estimated from a slight, moderate or very marked change in the wave form.

A schematic diagram of the apparatus for the intermodulation method[43]

[43] Hilliard, J. K., *Proc. Inst. Rad. Eng.*, Vol. 29, No. 12, p. 614, 1941.

of measuring nonlinear distortion is shown in Fig. 10.19. Two tones are impressed upon the loud speaker 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 apppa-ratus under test is shown in Fig. 10.20*A*. The output of the microphone is shown in Fig. 10.20*B*. This output is fed to an 800-cycle high-pass electrical

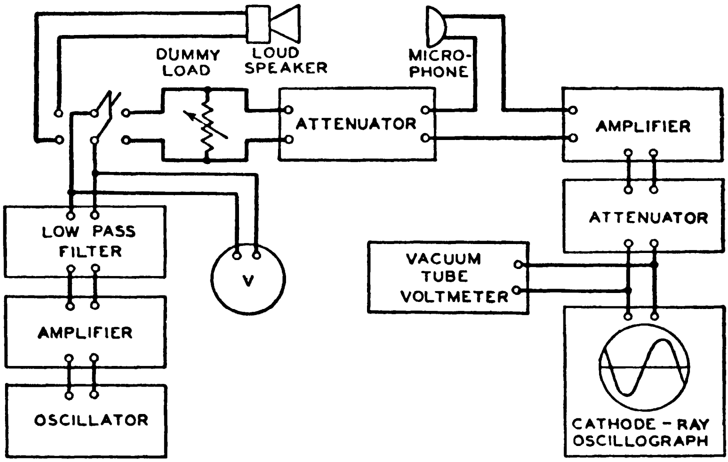

FIG. 10.18. Schematic arrangement of the apparatus employing a cathode-ray tube for indicating the nonlinear distortion of a loud speaker.

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. 10.20*C*. Beyond the electrical filters the signal is amplified and impressed upon a full-wave detector. The output of the detector is shown in Fig. 10.20*D*. 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. 10.20*E*. 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.

D. *Efficiency Frequency Characteristic.*[44] — The efficiency of a loud speaker 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

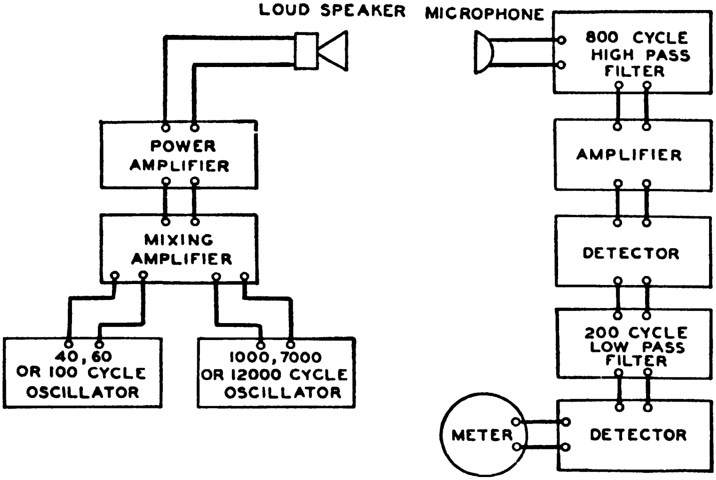

FIG. 10.19. Schematic arrangement of the apparatus for measuring the nonlinear distortion of a loud speaker employing the intermodulation method. (After Hilliard.)

pressure. The plot of efficiency, in per cent, versus frequency, in cycles, is termed the efficiency frequency characteristic.

The measurement of efficiency of a loud speaker 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 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

[44] Standards on Electroacoustics, Institute of Radio Engineers, 1938.

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

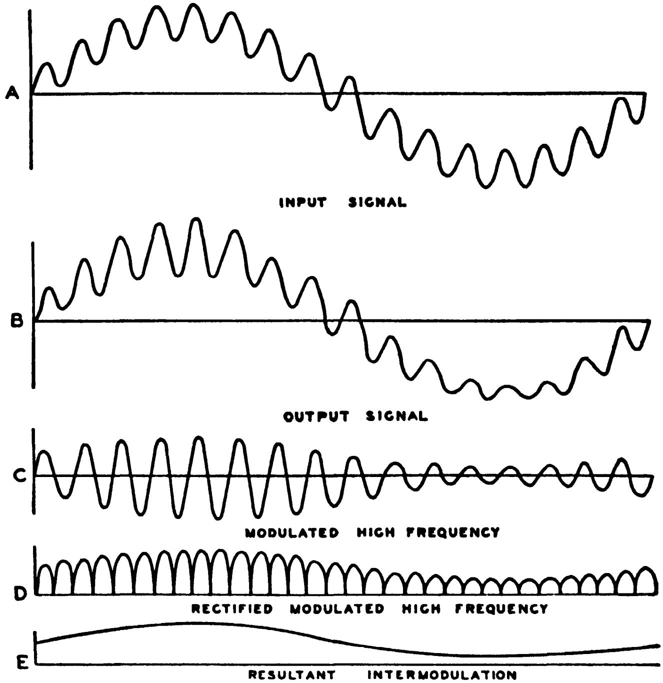

FIG. 10.20. Typical wave shapes in the various stages of the intermodulation system of distortion measurement. *A*. The input signal to the loud speaker. *B*. The output signal of the loud speaker. *C*. The modulated high-frequency output of the band pass filter. *D*. The rectified modulated high frequency of the detector. *E*. The resultant intermodulation output of the copper oxide rectifier.

displacement 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

$$P_A = \frac{1}{\rho c} \iint p^2 \, dS \times 10^{-7} \qquad 10.24$$

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

$$\mu = \frac{P_A}{P_E} \times 100 \qquad 10.25$$

where P_A = total acoustical output, in watts, and
P_E = electrical input, in watts.

As previously mentioned, the loud speaker 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 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 the 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 loud-speaker 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 loud speaker 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 loud speaker is shown in Fig. 10.21. 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 10.24. 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 loud speaker. If the pattern is unsymmetrical, the loud speaker 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 loud-speaker efficiency by indirect means. The most common method is to measure the electrical impedance under various conditions of diaphragm loading.

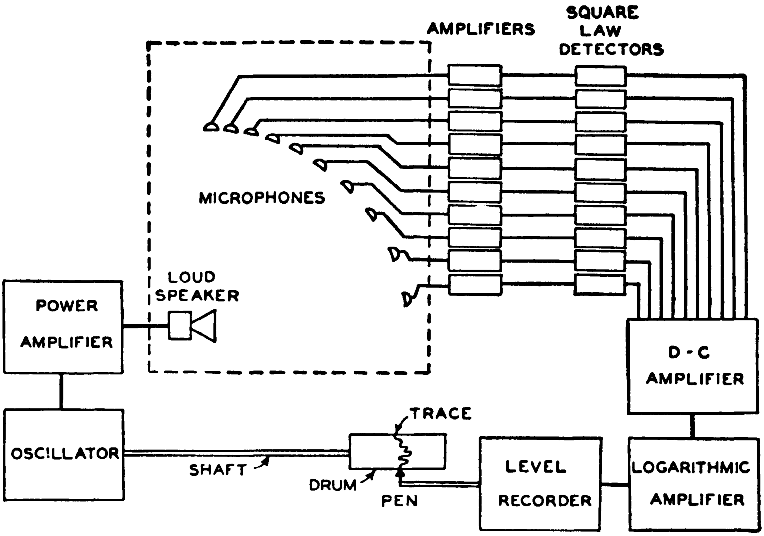

FIG. 10.21. Schematic arrangement of the apparatus for obtaining the total sound power output frequency characteristic of a loud speaker.

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[45] 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

$$\mu = \frac{r_{EM}}{r_{EN}} \times 100 \qquad 10.26$$

[45] Kennelly and Pierce, *Proc. A. A. A. S.*, Vol. 48, No. 6, 1912.

where $r_{EM} = r_{EN} - r_{ED}$ motional electrical resistance, in ohms,
r_{EN} = resistive component of the electrical impedance with the system in the normal state, in ohms, and
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 diaphragm 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 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 loud speaker 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. 10.2*D*, Fig. 10.7. A reference sound may be set up by a separate loud speaker, in which the phase shift is small, and picked up by one microphone. A reference voltage source may be substituted for the reference microphone. The sound from the loud speaker to be tested may be picked up on the other microphone. The phase difference may be determined as outlined in Sec. 10.2*D*. 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. 7.4*B*).

F. *Electrical Impedance Frequency Characteristic.* — The electrical impedance characteristic of a loud speaker 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.

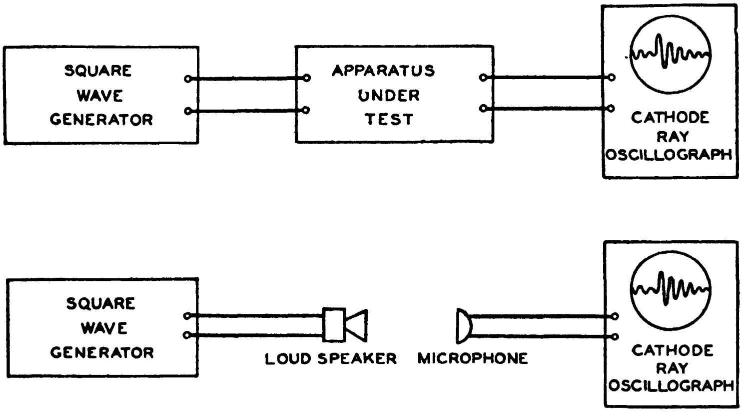

FIG. 10.22. Schematic arrangement of the apparatus employing a square wave generator and a cathode-ray tube for indicating the transient response characteristics of acoustical apparatus such as microphones or loud speakers.

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. 6.15).

The apparatus for investigating the transient response of an audio system is shown schematically in Fig. 10.22. The output of a square wave generator is fed to the apparatus to be tested. The output of the apparatus under test 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[46] offer a simple and rapid method of including both phase shift and amplitude response in a single test.

[46] Kallmann, Spencer and Singer, *Proc. Inst. Rad. Eng.*, Vol. 33, No. 3, p. 169, 1945.

H. *Subjective Measurements.* — In many cases the apparatus for making all the objective tests outlined in the preceding sections is not always 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 loud speakers are compared with a reference loud-speaker system, is widely used. Or, in other words, this test determines the lumped effects of the following factors: loudness, frequency range, tone balance, special distribution, quality and hangover. The electrical input is usually broadcasting program material such as speech or music. The inputs should be adjusted until the reference and test loud speakers are judged to have equal loudness. The required attenuation of the electrical input determines the relative loudness efficiency of the loud speaker under test. The loud speakers should occupy positions which are symmetrical with respect to the room boundaries and the observer. The loud speakers should be sufficiently separated that interaction is negligible. A number of observers and a variety of program material should be used in order to insure statistical significance.

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 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 [47, 48] of telephone receivers is shown in Fig. 10.23. A free progressive sound wave is established by means of a loud speaker driven by an oscillator and power amplifier. The test should be conducted in a free-field sound room (see Sec. 10.3*A*4). With the receivers removed the observer listens to the sound produced by the loud speaker as shown in Fig. 10.23*A*. Next the observer places the receivers on his ears and the output of the oscillator and amplifier is transferred from the loud speaker to the headphones by throwing the switch and reducing the power by means of a suitable attenuator as shown in Fig. 10.23*B*. 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 loud speaker at the distance of the observer is measured by means of a cali-

[47] Olson and Massa, *Jour. Acous. Soc. Amer.*, Vol. 6, No. 4, p. 250, 1935.
[48] Olson, H. F., *Jour. Soc. Mot. Pic. Eng.*, Vol. 27, No. 5, p. 537, 1936.

brated microphone, amplifier and meter combination as shown in Fig. 10.23*C*. This procedure is repeated at several frequencies. The absolute response of the telephone receiver is given by equation 10.20, where p

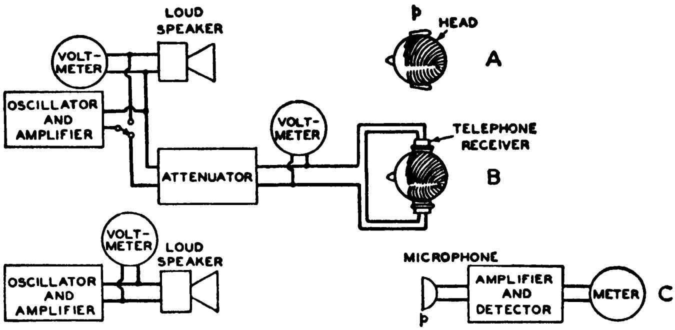

FIG. 10.23. Schematic arrangement of the apparatus for obtaining the subjective response frequency characteristic of telephone receivers. In *A*, the observer listens to the sound produced by the loud speaker. In *B*, the observer adjusts the attenuator until the sound level produced by the telephone receivers appears to be equal to the sound level produced by the loud speaker. In *C*, a sound level meter is used to measure the sound pressure produced by the loud speaker.

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.*[49, 50, 51, 52] — The acoustical impedance frequency characteristic looking through the ear cap of a telephone receiver as normally worn has been investigated by Inglis, Gray and Jenkins. This is shown in Fig. 10.24. An artificial ear and the acoustical network which yields approximately the same acoustical impedance characteristic are shown in Fig. 10.24. 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 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 loud speakers may be performed upon telephone

[49] Inglis, Gray and Jenkins, *Bell Syst. Tech. Jour.*, Vol. 11, No. 2, p. 293, 1932.
[50] Olson and Massa, *Jour. Acous. Soc. Amer.*, Vol. 6, No. 2, p. 250, 1935.
[51] Olson, H. F., *Jour. S. M. P. E.*, Vol. 27, No. 5, p. 537, 1936.
[52] Romanow, F. F., *Jour. Acous. Soc. Amer.*, Vol. 13, No. 3, p. 294, 1942.

receivers by employing the artificial ear. The same apparatus as described in Sec. 10.3*A*2 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. 10.24 may be used for this purpose by closing the slits.

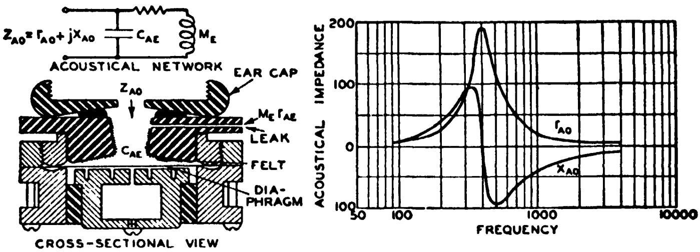

FIG. 10.24. A cross-sectional view of an artificial ear employing a standard condenser microphone and the acoustical network of the acoustical system. The graph shows the resistive, r_{A0}, and reactive, x_{A0}, components of the acoustical impedance, looking into the aperture of the ear cap, as a function of the frequency. C_{AE} = acoustical capacitance of the cavity. M_E and r_{AE} = inertance and acoustical resistance of the leak.

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 [53, 54] 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. 9.3*C*3).

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 [55, 56, 56'] as shown in the schematic diagram of Fig. 10.25. 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

[53] Hawley, M. S., *Bell Labs. Rec.*, Vol. 18, No. 3, p. 73, 1939.
[54] Romanow, F. F., *Jour. Acous. Soc. Amer.*, Vol. 13, No. 3, p. 294, 1942.
[55] Buchman and Meyers, *ENT*, Vol. 7, p. 147, 1930.
[56] King, D. R., *Electronics*, Vol. 14, No. 5, p. 47, 1941.
[56'] Bauer, B. B., *Jour. Acous. Soc. Amer.*, Vol. 18, No. 2, p. 387, 1946.

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. 10.25*A* can be measured accurately at this distance by employing

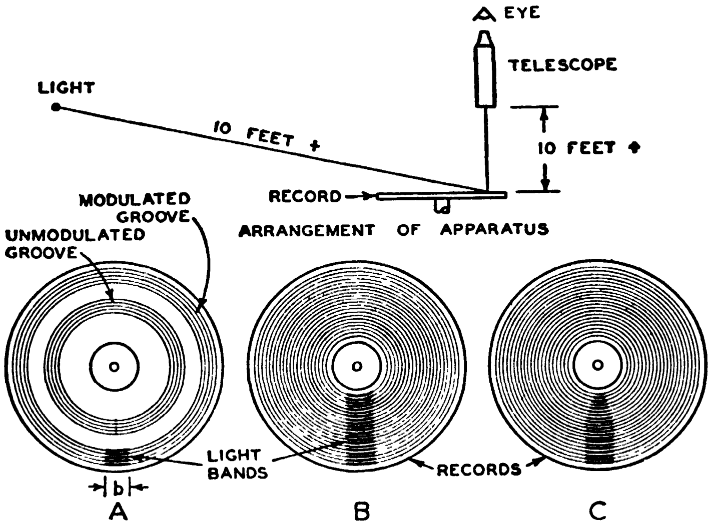

FIG. 10.25. Schematic arrangement of the apparatus for the measurement of the response of a phonograph record by the optical method. The light patterns for various types of modulation are shown in *A*, *B* and *C*. *A* shows a record with a modulated and unmodulated groove. *B* shows a record with constant velocity and any frequency. *C* shows a record cut with sine wave modulation having the response frequency characteristic of Fig. 9.8. The frequency increases as a logarithmic function from the inside to the outside. Note; the complementary pattern at 180° is not shown.

a telescope. Under these conditions, the amplitude of the modulation, in inches, is given by

$$x_p = \frac{bu}{2f} \qquad 10.27$$

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.

The peak velocity, in inches per second, is given by

$$\dot{x}_p = \pi bu \qquad 10.28$$

The rms velocity, in inches per second, is given by

$$\dot{x} = \frac{\pi}{\sqrt{2}} bu \qquad 10.29$$

The image shown in Fig. 10.25*B* 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 is constant. The image in Fig. 10.25*C* depicts a frequency record cut with a sine wave modulation having the velocity response frequency characteristic as shown in Fig. 9.8.

B. *Testing of Phonograph Pickups.* — Phonograph pickups are generally tested by employing a standard frequency record. The velocity response frequency characteristic of a standard frequency record is shown in Fig. 9.8. The velocity response frequency characteristic of the record may be determined by the optical method outlined in Sec. 10.5*A*. 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. 10.3*A*2. Using this arrangement a continuous response frequency characteristic may be obtained.

Nonlinear distortion characteristic of a phonograph pickup and record combination may be obtained by employing apparatus described in Sec. 10.3*C*.

C. *Testing of 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. 9.8. 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. 10.3*A*2. The horn or diaphragm of the mechanical phonograph is treated as a loud speaker and the test conducted as outlined in Sec. 10.3*A*.

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. 9.3*E*). 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 loud speaker, 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 pickup 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 [57] 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.[58] — In the reproduction of sound by film, wire, 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 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.[59,60,61,62] — 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. 10.26. The loud speakers are connected to two pipes, one of which is variable in

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[57] Burt, A. D., *Electronics*, Vol. 16, No. 1, p. 90, 1943.

[58] Kellogg and Morgan, *Jour. Acous. Soc. Amer.*, Vol. 7, No. 4, p. 271, 1936.

[59] Stewart, G. W., *Phys. Rev.*, 28, Vol. No. 5, p. 1038, 1926.

[60] Stewart and Lindsay, " Acoustics," D. Van Nostrand Company, New York N. Y., 1930.

[61] Flanders, P. B., *Bell Syst. Tech. Jour.*, Vol. 11, No. 3, p. 402, 1932.

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

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.

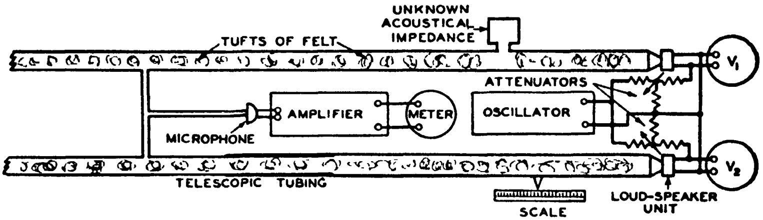

FIG. 10.26. Schematic arrangement of an acoustical impedance bridge for measuring acoustical impedance. (After Stewart.)

With the branch closed the voltage across the two loud-speaker 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

$$\frac{2z_{AU}}{2z_{AU} + r_A} = \frac{p_0}{p_0'} (\cos \theta + j \sin \theta) \qquad 10.30$$

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

$$\frac{p_0}{p_0'} = \frac{e_1 e_2'}{e_2 e_1'} \qquad 10.31$$

where e_1 and e_2 = the voltages applied to the loud speaker without the branch, and
e_1' and e_2' = the voltages with the branch attached.

The phase angle θ in radians is given by

$$\theta = \frac{2\pi d}{\lambda} \tag{10.32}$$

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 loud-speaker 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 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.

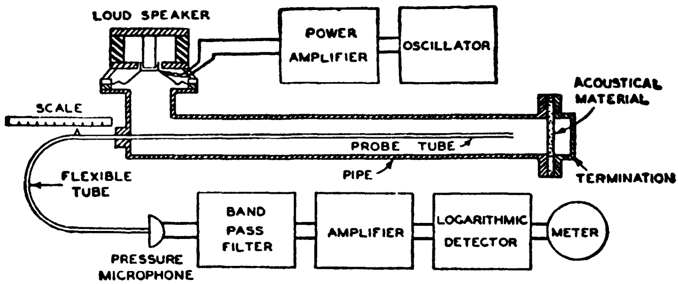

FIG. 10.27. Schematic arrangement of the apparatus for measuring the acoustical impedance by means of the standing wave pattern in a tube terminated by the unknown acoustical impedance.

The arrangement of the apparatus [63, 64, 65, 66] for the measurement of acoustical impedance by the tube method is shown in Fig. 10.27. 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.

[63] Taylor, H. O., *Phys. Rev.*, Vol. 2, No. 4, p. 270, 1913.
[64] Hall, W. M., *Jour. Acous. Soc. Amer.*, Vol. 11, No. 1, p. 140, 1939.
[65] Bolt and Brown., *Jour. Acous. Soc. Amer.*, Vol. 12, No. 1, p. 31, 1940.
[66] Sabine, H. J., *Jour. Acous. Soc. Amer.*, Vol. 14, No. 2, p. 143, 1942.

The sound absorption coefficient for normal incidence is given by

$$\alpha_n = 1 - \left(\frac{\log_{10}^{-1}(L/20) - 1}{\log_{10}^{-1}(L/20) + 1}\right)^2 \qquad 10.33$$

$$= 1 - K^2$$

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.

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

$$r_{A1} = \rho c \frac{1 - K^2}{1 + K^2 + 2K \cos\left(\frac{2\pi D_1}{D_2}\right)} \qquad 10.34$$

The acoustical reactance, in acoustical ohms per unit area, is given by

$$x_{A1} = \rho c \frac{2K \sin\left(\frac{2\pi D_1}{D_2}\right)}{1 + K^2 + 2K \cos\left(\frac{2\pi D_1}{D_2}\right)} \qquad 10.35$$

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 10.33.

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,

$$z_{A1} = r_{A1} + jx_{A1} \qquad 10.36$$

10.8. Mechanical Impedance Bridge. — 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[67] 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

$$C_M = \frac{4l^3}{\mathcal{Q}ba^3} \qquad 10.37$$

where l = length of the reed, in centimeters,
b = width of the reed, in centimeters,
a = thickness of the reed, in centimeters, and
$\mathcal{Q}$ = Young's modulus, in dynes per square centimeter, see Table 1.1.

The mechanical circuit of a 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. 10.28. 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 microphoné. 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

[67] Wiggins, A. M., *Jour. Acous. Soc. Amer.*, Vol. 15, No. 1, p. 50, 1943.

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.

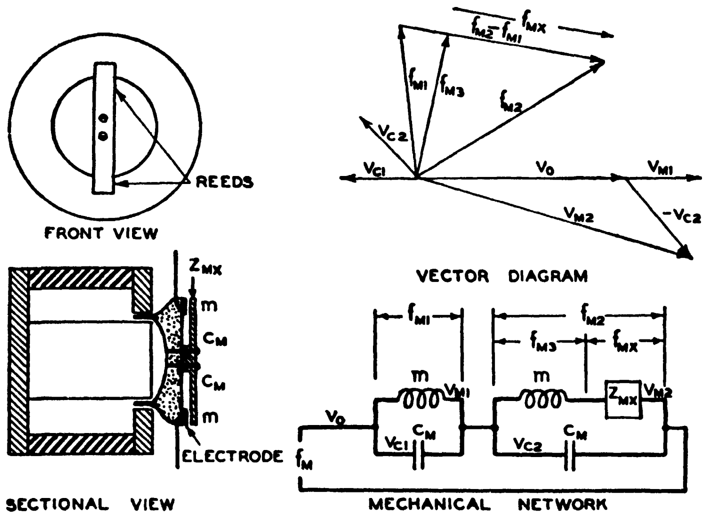

Fig. 10.28. Front and sectional views, mechanical network and vector diagram of a mechanical impedance bridge. In the mechanical network: m = the effective mass of the reed. C_M = the compliance of the reed. z_{MX} = the mechanical impedance being measured. f_M = the total driving force. f_{M1} = the 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 themagnitudes and phases of the forces and velocities for a typical condition.

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 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 25 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 frequency at which the measurement is to be made is generally acceptable. The proximity to the natural frequency at which a measurement can accurately 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 impressed 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

$$|z_{MX}| = \frac{f_{M2} - f_{M1}}{f_{M2}}\left(x_{MM} - \frac{x_{MM}^2}{x_{MC}}\right) \qquad 10.38$$

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 10.37.

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.

10.9. Measurement of Porosity.— Porosity is [68, 69, 70, 71, 72, 73] a relevant mechanical property of a porous material. The porosity of a substance 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. 10.29. 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

[68] Rettinger, M., *Jour. Acous. Soc. Amer.*, Vol. 6, No. 3, p. 188, 1935.
[69] Gemant, A., *Jour. App. Phys.*, Vol. 12, No. 10, p. 725, 1941.
[70] Morse, Bolt and Brown, *Jour. Acous. Soc. Amer.*, Vol. 12, No. 2, p. 217, 1940.
[71] Brown and Bolt, *Jour. Acous. Soc. Amer.*, Vol. 13, No. 4, p. 337, 1942.
[72] Beranek, L., *Jour. Acous. Soc. Amer.*, Vol. 13, No. 3, p. 248, 1942.
[73] Morse and Bolt, *Rev. Mod. Phys.*, Vol. 16, No. 2, p. 69, 1944.

h of the water column on the two sides of the U tube measured. Then the 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 centi-

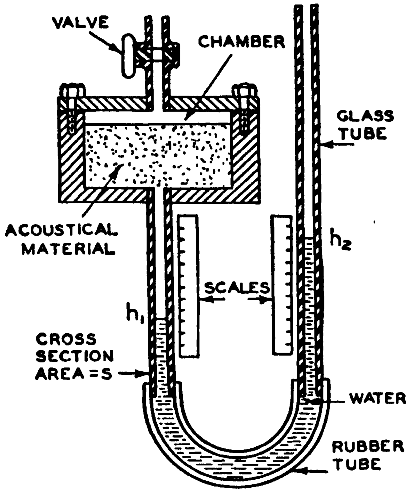

FIG. 10.29. Schematic arrangement of the apparatus for the measurement of porosity of a porous material.

meters, 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

$$P = \frac{p_0 \Delta V_0}{V_t \Delta p_0} + 1 - \frac{V}{V_t} \qquad 10.39$$

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.[74, 75, 76, 77, 78, 79]

The d-c acoustical resistance, r_{ADC}, in acoustical ohms, may be obtained from the volume current and the pressure as follows:

$$r_{ADC} = \frac{p}{U} \tag{10.40}$$

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.

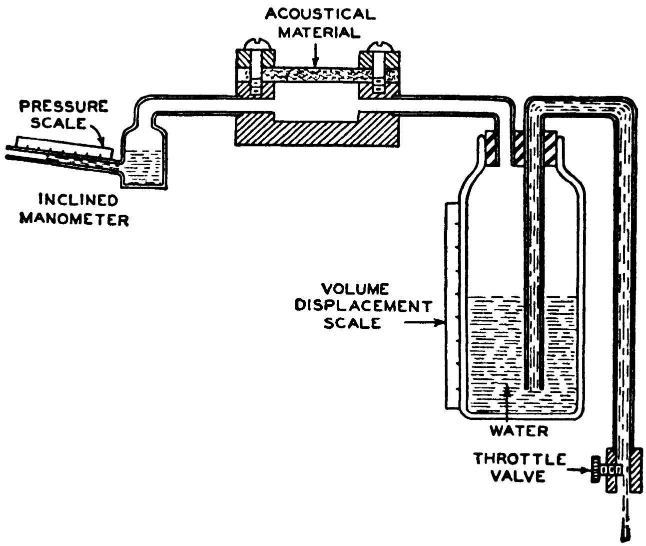

FIG. 10.30. Schematic arrangement of the apparatus for the measurement of flow resistance or d-c acoustical resistance of an acoustical material.

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. 10.30. The difference in pressure between the two sides is

[74] Rettinger, M., *Jour. Acous. Soc. Amer.*, Vol. 6, No. 3, p. 188, 1935.
[75] Gemant, A., *Jour. App. Phys.*, Vol. 12, No. 10, p. 725, 1941.
[76] Morse, Bolt and Brown, *Jour. Acous. Soc. Amer.*, Vol. 12, No. 2, p. 217, 1940.
[77] Brown and Bolt, *Jour. Acous. Soc. Amer.*, Vol. 13, No. 4, p. 337, 1942.
[78] Beranek, L., *Jour. Acous. Soc. Amer.*, Vol. 13, No. 3, p. 248, 1942.
[79] Morse and Bolt, *Rev. Mod. Phys.*, Vol. 16, No. 2, p. 69, 1944.

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.

The d-c acoustical resistance or flow resistance is usually specified as the acoustical resistance per unit cube as follows:

$$r_{ADC1} = \frac{pA}{Ud} \tag{10.41}$$

where A = area of the material, in square centimeters, and
d = thickness of the material, in centimeters.

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.

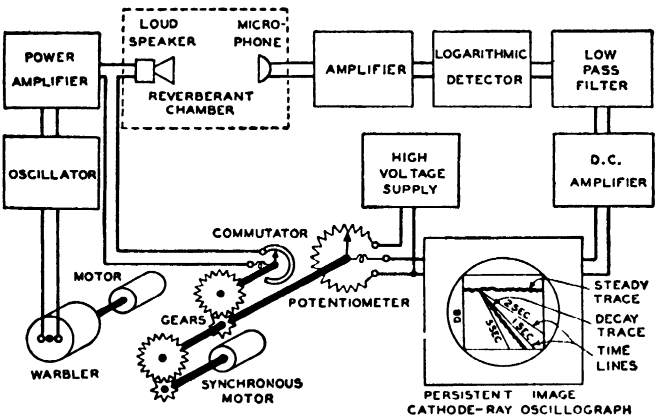

Fig. 10.31. Schematic arrangement of the apparatus for obtaining the reverberation time of a reverberation chamber or room employing a cathode-ray oscillograph with persistent image screen.

Many systems[80] 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.

[80] Olson and Massa, " Applied Acoustics," 2nd Ed., P. Blakiston's Son and Co., Philadelphia, 1939.

The high-speed level recorder, Fig. 10.9, and the high-speed level indicator, Fig. 10.10, have been found to be useful means for measuring reverberation time because the trace of the entire decay of the sounds may be examined.

A schematic diagram depicting the apparatus for measuring the reverberation time is shown in Fig. 10.31. 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 loud speaker. 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 a smoother decay trace. As a further aid in smoothing the decay response multiple loud speakers and microphones may be used.

10.12. Measurement of Absorption Coefficient.[81, 82, 83, 84] — 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.

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 20,0000 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. 11.2*A*.

[81] Watson, F. R., " Acoustics of Buildings," John Wiley and Sons, New York, N. Y., 1923.

[82] Bagenal and Wood, " Planning for Good Acoustics," Methuen, 1931.

[83] Knudsen, V. O., " Architectural Acoustics," John Wiley and Sons, New York, N. Y., 1932.

[84] Sabine, P. E.," Acoustics and Architecture," McGraw-Hill Book Company, New York, N. Y., 1932.

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. 11.2*A*, 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.

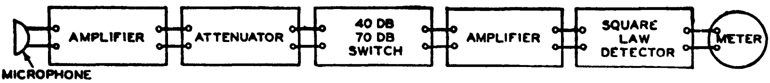

FIG. 10.32. Schematic arrangement of the components of a noise meter.

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 impossible, at the present time, to build a noise meter which will show the

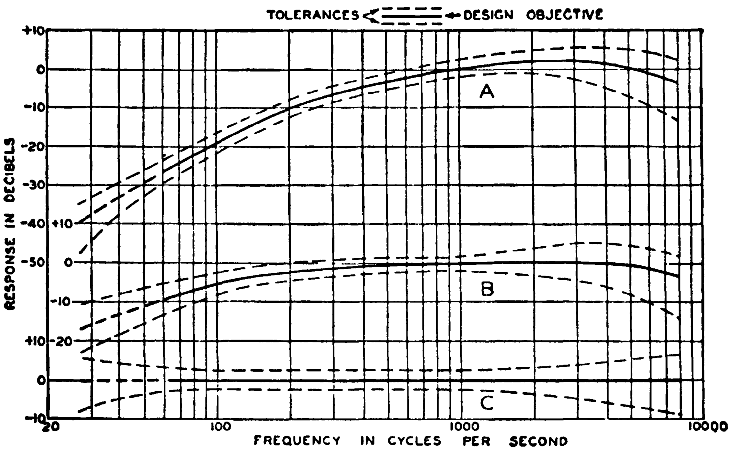

FIG. 10.33. Recommended characteristics for sound level meters. (American Standards Association.)

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. 10.32. 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 overall frequency response of an ideal noise meter should be the reciprocal of the ear response frequency characteristics. This would make the noise meter unduly complicated. The response frequency characteristics recommended for the noise meters by the American Standards Association are[85,86,87] shown in Fig. 10.33. 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 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. 10.14 and 11.2R).

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 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 calibration remain correct within a decibel, i.e., $\pm\frac{1}{2}$ db.

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[85] Amer. Tent. Standards for Sound Level Meters Z-24.3, American Standards Association, New York, N. Y., 1936; or *Jour. Acous. Soc. Amer.*, Vol. 8, No. 2, 1936.

[86] American Standard for Noise Measurement Z-24.2, 1942, or *Jour. Acous. Soc. Amer.*, Vol. 14, No. 1, p. 102, 1942. Also American Standards Association, Z-24.2, 1942.

[87] American Standard for Sound Level Meters for Measurement of Noise and other Sounds, Z-24.3, American Standards Association, New York, N. Y., 1944. Also American Standards Association, Z-24.3, 1944.

10.14. 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 definition and equations relating to transmission loss and reduction factor, see Sec. 11.2*R*. 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. 10.3*A*2) 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. 11.2*R*.

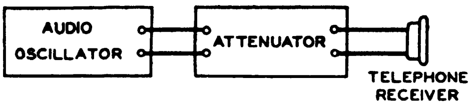

FIG. 10.34. Schematic arrangement of the components of an audiometer.

10.15. Audiometry.[88, 89] — The acuity of hearing is measured by an audiometer. The audiometer consists of an audio oscillator for generating pure tones, an attenuator calibrated in decibels and a telephone receiver (Fig. 10.34). 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.

[88] Fletcher, H., " Speech and Hearing," D. Van Nostrand Company, New York, N. Y., 1929.

[89] Proposed Specifications for Audiometers for General Diagnostic Purposes, *Jour. Acous. Soc. Amer.*, Vol. 9, No. 1, p. 72, 1937.

10.16. Articulation Measurements.[90, 91] — 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 accuracy and ease with which the articulated sounds of speech are recognized.

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.

10.17. Testing of Hearing Aids.[92] — Apparatus for testing the overall performance of a hearing aid is depicted in Fig. 10.35. A sound wave is produced by the oscillator, amplifier and loud-speaker 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 by means of an artificial ear, amplifier and meter. Any of the response measuring systems described in Sec. 10.3*A*2 may be employed instead of the point-by-point system depicted in Fig. 10.35. The artificial

[90] Fletcher, H., " Speech and Hearing," D. Van Nostrand Company, New York, N. Y., 1929.

[91] Fletcher and Steinberg, *Jour. Acous. Soc. Amer.*, Vol. 1, No. 2, Part 2, p. 1, 1930.

[92] Tentative Code for Measurements of Performance of Hearing Aids, *Jour. Acous. Soc. Amer.*, Vol. 17, No. 2, p. 44, 1945.

ear consists of a cylindrical tube, 1.80 centimeters in length and .305 centimeters 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. 10.35*A* has been obtained, the sound

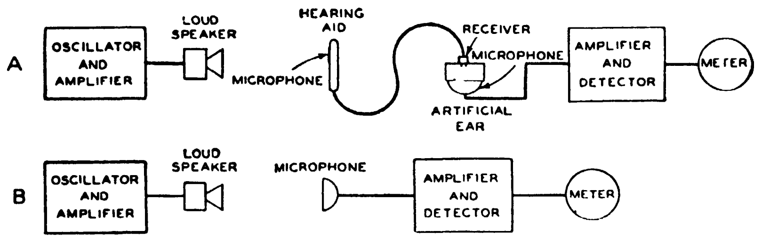

FIG. 10.35. Schematic arrangement of the apparatus for obtaining the response frequency characteristic of a hearing aid.

pressure response frequency characteristic in free space is obtained by placing the microphone of the system of Fig. 10.35*B* 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.

CHAPTER XI

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, 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] — 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 many 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[7] for the decay of the sound in a room is

$$E = E_0 \epsilon^{-cAt/4V} \qquad 11.1$$

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.

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[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] 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

$$T = .050\,\frac{V}{A} \qquad 11.2$$

where T = time, in seconds,
V = volume, in cubic feet, and
A = total absorption, in sabins.

Later work[8] has shown that equation 11.2 is unsatisfactory for large rooms or rooms with very large absorption. The equation developed by Eyring is

$$T = \frac{.05V}{-\,S\log_e(1 - a_{av})} \qquad 11.3$$

where V = volume, in cubic feet,
S = total area, in square feet, and
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

[8] 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	Thickness (in)	Mounting	Frequency 128	256	512	1024	2048	4096	Author
			Coefficient						
Corkoustic	1¼	2	.11	.34	.67	.47	.57	.53	A.M.A
Cushiontone	⅞	2	.17	.58	.70	.90	.76	.71	"
Koustex	1	2	.15	.27	.75	.99	.80	.87	"
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	"
Absorbex F	1	2	11	22	73	.72	77	.75	"
Q-T Ductliner	1	4	29	.41	.78	89	.88	.78	"
Macoustic Plaster	½	2	.32	.24	.53	.81	.68	.67	"
Reverbolite Plaster	½	2	.29	.30	.40	.49	.54	.60	"
Sabinite Plaster	½	2	.26	.16	.32	.70	.73	.72	"
Draperies Hung Straight, in 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" x 4" — 16" O.C.	¼		.22	.21	.28	.31	.44	.55	V.K.

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" x 2" furring 12" O.C.
3. Attached to metal supports applied to 1" x 2" wood furring.
4. Laid on 24 ga. sheet iron, nailed to 1" x 2" wood furring 24" O.C.

TABLE 11.1. (*Continued*)

Material	Thick-ness (in.)	Mount-ing	Frequency 128	256	512	1024	2048	4096	Author
			Coefficient						
Masonite, on 2" x 4" — 16" O.C.	$\frac{1}{2}$		.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	.C8	08	.11	W.S.
Brick Wall, Unpainted			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, Motar	2		.15	.21	.43	.37	.39	.51	B.R.
Plaster, Lime on Wood Lath on Wood Studs, Rough Finish	$\frac{1}{2}$		.039	.056	.061	.089	.054	.070	P.S.
Plaster, Gypsum on Wood Lath on Wood Studs, Rough Finish	$\frac{1}{2}$		023	039	039	052	037	.035	P.S.
Ozite	$\frac{3}{4}$		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" x 2" furring 12" O.C.
3. Attached to metal supports applied to 1" x 2" wood furring.
4. Laid on 24 ga. sheet iron, nailed to 1" x 2" wood furring 24" O.C.

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. 10.3*A*2).

B. *Mechanism of Sound Absorption by Acoustical Materials.*[9,10,11,12,13,14] —

[9] Rettinger, M., *Jour. Acous. Soc. Amer.*, Vol. 6. No. 3, p. 188, 1935.
[10] Gemant, A., *Jour. App. Phys.*, Vol. 12, No. 10, p. 725, 1941.
[11] Morse, Bolt and Brown, *Jour. Acous. Soc. Amer.*, Vol. 12, No. 2, p. 217, 1940.
[12] Brown and Bolt, *Jour. Acous. Soc. Amer.*, Vol. 13, No. 4, p. 337, 1942.
[13] Beranek, L., *Jour. Acous. Soc. Amer.*, Vol. 13, No. 3, p. 248, 1942.
[14] Morse and Bolt, *Rev. Mod. Phys.*, Vol. 16, No. 2, p. 69, 1944.

The mechanism of sound absorption may be illustrated by means of the acoustical impedance concept. This phase of sound absorption of 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

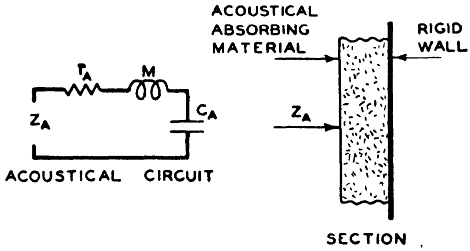

FIG. 11.1. Sectional view of sound absorbing material and the acoustical circuit. In the acoustical circuit: z_A = the acoustical impedance of the material per unit area. r_A, M and C_A = the acoustical resistance, inertance and acoustical capacitance of the material.

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

$$r_{A1} = \frac{r_{A1D}d}{3} \tag{11.5}$$

The inertance per unit area is

$$M_1 = \frac{dm\rho}{3} \tag{11.6}$$

The acoustical capacitance per unit area is

$$C_{A1} = \frac{Pd}{\rho c^2} \tag{11.7}$$

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 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 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,

$$\alpha = 1 - \left|\frac{z_{A1} - \rho c}{z_{A1} + \rho c}\right|^2 \qquad 11.8$$

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.

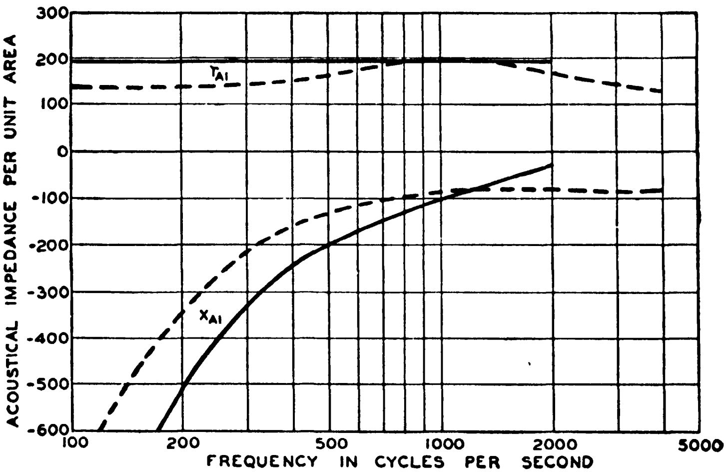

FIG. 11.2. The acoustical resistance, x_{A1}, and acoustical reactance, r_{A1}, characteristics of Permacoustic 1 inch in thickness and backed by a rigid wall. Dashed lines: experimentally measured characteristics. Solid lines: computed characteristics. The ordinates represent the acoustical impedance per square centimeter.

Equation 11.8 may be written

$$\alpha = \frac{4r_{A1}\rho c}{(r_{A1} + \rho c)^2 + x_{A1}^2} \qquad 11.9$$

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

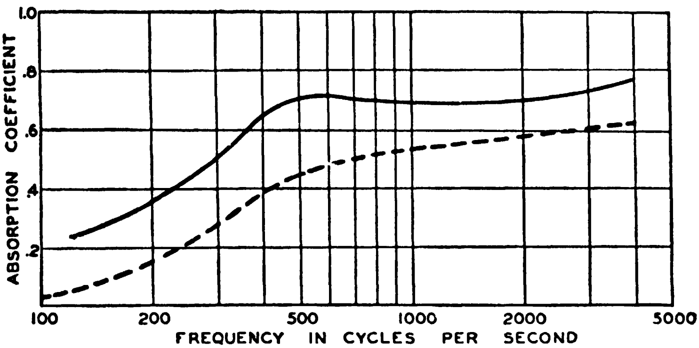

FIG. 11.3. The absorption coefficient characteristics of Permacoustic 1 inch in thickness and backed by a rigid wall. Solid line: reverberation chamber measurements. Dashed line: computed from acoustical impedance characteristic of Fig. 11.2.

in reverberation chambers is larger [15] than the values obtained under actual conditions.

C. *Functional Sound Absorbers.*[16] — 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 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 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. 11.4*A* shows that the maximum

[15] Morris, Nixon and Parkinson, *Jour. Acous. Soc. Amer.*, Vol. 9, No. 3, p. 234, 1938.
[16] Olson, H. F., *RCA Review*, Vol. 7, No. 4, p. 503, 1946.

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.

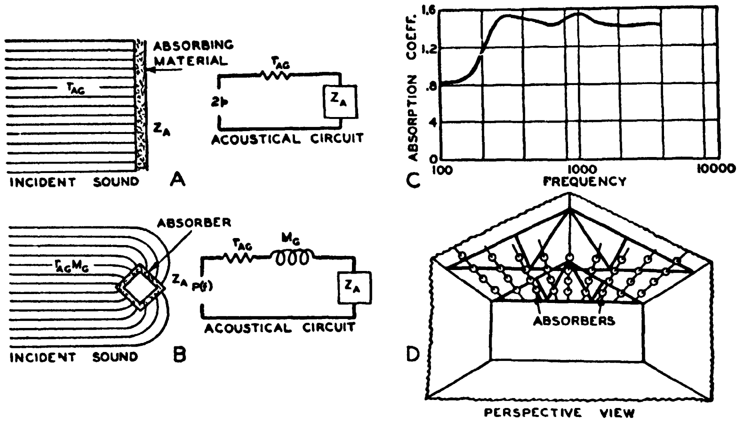

FIG. 11.4. *A*. Conventional sound absorbing material. In the acoustical circuit: r_{AG} = the characteristic acoustical impedance of air for a plane wave. z_A = the acoustical impedance of the sound absorbing material. p = the sound pressure in free space. *B*. Functional sound absorber. In the acoustical 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.

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. 11.4*B*. 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. 11.4*C*. It will be seen that the absorption per unit area is about two times that of conventional materials. The shell type functional 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. 10.4*D*. 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.

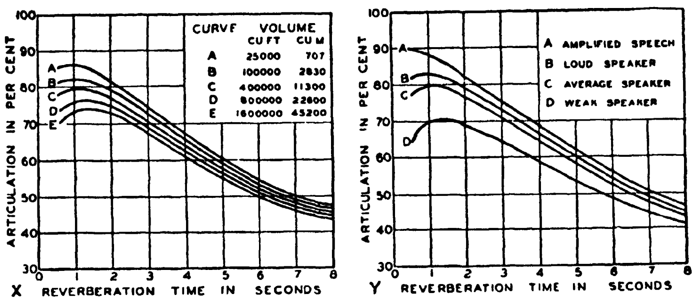

FIG. 11.5. *X*. The articulation of a speaker in auditoriums of various volumes. *Y* = the articulation of amplified speech, loud, average, and a weak speaker in an auditorium of 400,000 cubic feet. (After Knudsen.)

D. *Articulation and Reverberation Time.* — The articulation[17] (see Sec. 10.16) of unamplified speech in auditoriums of various sizes as a function of the reverberation time is shown in Fig. 11.5. 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. 11.5. The obvious solution is the use of sound rein-

[17] Knudsen, V. O., *Jour. Acous. Soc. Amer.*, Vol. 9, No. 3, p. 175, 1938.

forcing equipment. The articulation for a weak, average and loud talker without amplification as compared to amplified speech is shown in Fig. 11.5. By proper selection and placement of the loud speakers the articulation characteristic may be made considerably higher.

E. *Sound Motion Picture Reproducing Systems.*[18] — The resultant sound energy density at the position of the auditor in a theater depends upon the response and the directional characteristics of the loud speaker 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 loud speaker 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. 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 loud speaker.

An elevation view of a reproducer in a theater is shown in Fig. 11.6. The two extreme points to be supplied are indicated as A and E. If the loud speaker 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 loud speaker furnishes a solution of the problem.

The directional characteristics of the loud speaker are shown in Fig. 11.6. 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 loud speaker is adjusted until the axis of the characteristic passes through the point E. Then the position of the loud speaker is adjusted until the angle θ is 40°. The distribution over the distance under consideration is shown in Fig. 11.6. 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.

[18] Olson and Massa, *Jour. Soc. Mot. Pict. Eng.*, Vol. 23, No. 2, p. 63, 1934.

From the response frequency characteristic of the loud speaker the pressure at any distance r centimeters on the axis may be obtained from the following equation.

$$p = p_0 \frac{x_0}{r} \tag{11.10}$$

where p_0 = pressure, in dynes per square centimeter, obtained at a distance x_0, in centimeters.

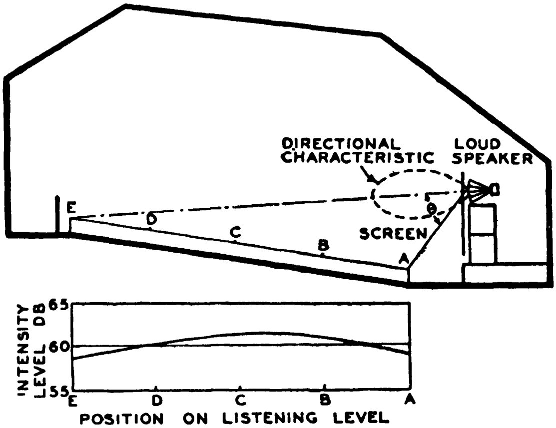

FIG. 11.6. Arrangement of the loud speaker for a sound motion picture reproducing system in a theater. The graph shows the intensity level at points along the listening level.

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 at this frequency. The direct radiation from the loud speaker can then be obtained for any point in the space.

The energy density, ergs per cubic centimeter, due to direct radiation from the loud speaker is

$$E_D = \frac{p_0^2 x_0^2 R_\theta^2}{r^2 \rho c^2} \tag{11.11}$$

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 loud speaker, 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 loud speakers for obtaining a uniform energy distribution of the direct sound. The energy density of reflected sound, as shown by equation 11.12, is independent of the observation point. As a consequence, by employing directional loud speakers, 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. 11.7. 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 loud speakers are usually placed behind a perforated screen upon which the picture is projected. Fig. 11.7. 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 5.12). 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

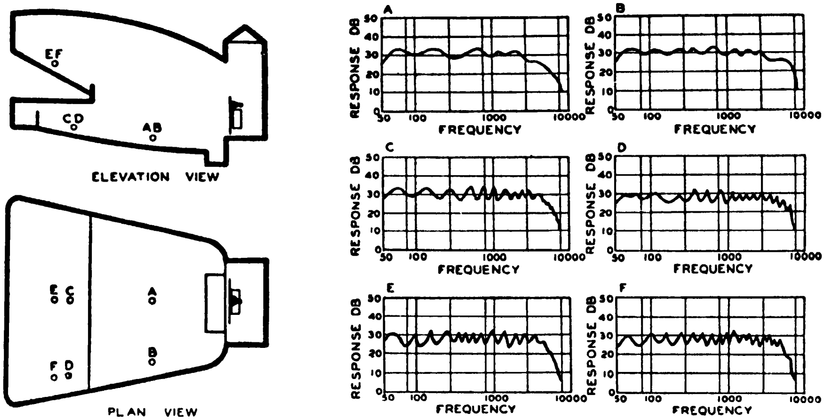

FIG. 11.7. A cross-sectional elevation and plan view of a theater equipped with a loud speaker for the reproduction of sound motion pictures. The graphs show the response frequency characteristics in various parts of the theater.

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 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.

F. *Sound Reinforcing Systems.*[19] — A large theater equipped with a sound reinforcing system is shown in Fig. 11.8. 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 loud

[19] Olson, H. F., *RCA Review*, Vol. 1, No. 1, p. 44, 1936.

speakers 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 loud speakers. Usually

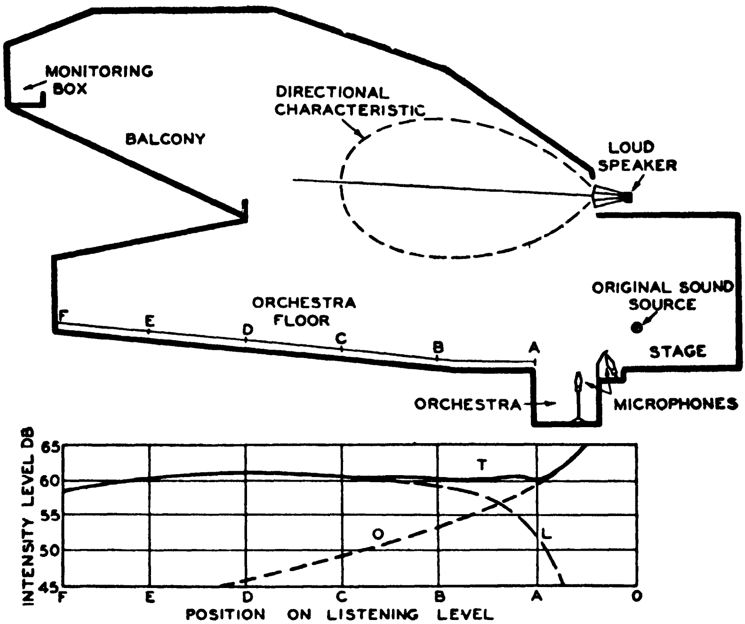

Fig. 11.8. Arrangement of the components of a sound reinforcing system in a theater. The graph shows the intensity level due to direct sound at the points indicated on the orchestra floor. Curve O is the intensity due to the original source of sound. Curve L is the intensity level due to the loud speaker. Curve T is the resultant intensity level.

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 loud speakers 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. 11.8. 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

$$E_{D1} = \frac{P_{D1}}{4\pi r^2 c} \tag{11.14}$$

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.

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. 11.8. 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 combination of the original source and the loud speakers, 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 loud speaker, from equation 11.11, is

$$E_{D2} = \frac{p_0^2 x_0^2 R_\theta^2}{r^2 \rho c^2} \tag{11.15}$$

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 $\theta = O$,
ρ = density of air, in grams per cubic centimeter, and
c = velocity of sound, in centimeters per second.

The problem is to select a loud speaker 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. 11.8, due to the direct sound from a loud speaker 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 loud speaker 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. 11.8 shows that the total intensity level characteristic remains uniform when the output of the loud speakers, 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,
S = area of absorbing materials, in square centimeters,
V = volume of the theater, in cubic centimeters,
t = time, in seconds,
c = velocity of sound, in centimeters per second,
P_{D1} = power output of the original sound, in ergs per second, and
P_{D2} = power output of the loud speaker, 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 microphones for collecting the sounds are usually concealed in the footlight trough.[20] By employing directional loud speakers, as outlined above, the sound level at the microphones due to the loud speakers is low and thereby reduces the possibility of oscillations due to acoustic feedback or regeneration in the reproducing system. In large theaters, having an 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 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. 11.9, 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. 11.9.

[20] Olson, H. F., *RCA Review*, Vol. 1, No. 1, p. 49, 1936.

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

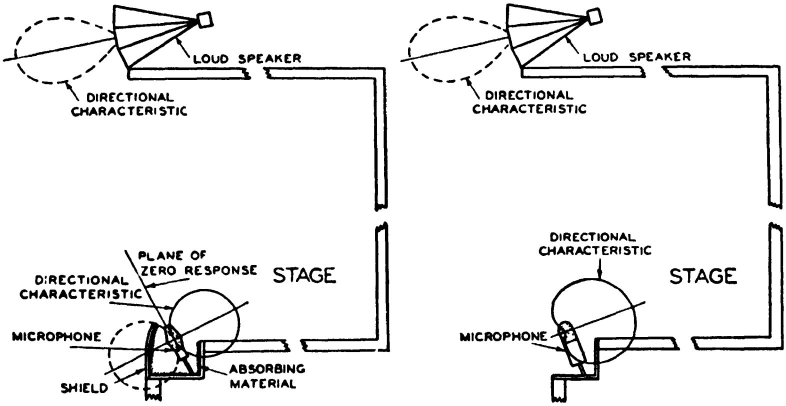

FIG. 11.9. Arrangements depicting the use of directional loud speakers and microphones for reducing feedback between the loud speaker and the microphone. The arrangement on the left employs a velocity microphone. A shield is used to reduce sound pickup from the orchestra and audience. The arrangement 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.

stage sound to that received from the orchestra when there is an orchestral accompaniment as well as the overall intensity of the augmented sound. The monitoring box is usually located in the rear balcony, the position which is the most susceptible to the augmented sound and one which also furnishes a good view of the action.

G. *Theater Acoustics.*[21] — Many theaters have major acoustical defects which cause echos and objectionable concentrations due to focusing of the reflected sound. These reflections may be more important than the reverberation time.

[21] Standardization Committee, *Jour. Soc. Mot. Pic. Eng.*, Vol. 36, No. 3, p. 267, 1941.

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 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 echos and sound concentrations. These, as well as other defects, 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. 11.10 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.

H. *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 frequency of 1000 cycles, is shown in the lower graph of Fig. 11.11. The reverberation time for other frequencies can be obtained by multiplying by

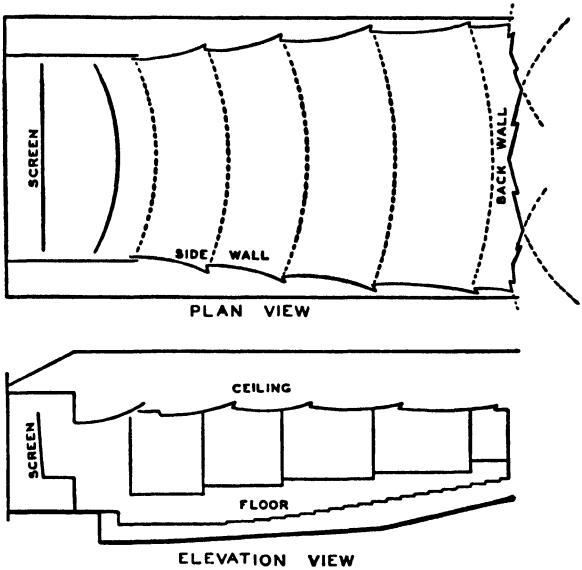

FIG. 11.10. Plan and elevation of a theater designed for good acoustics.

the factor K, obtained from the upper graph of Fig. 11.11. 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 12.12).

I. *Power Requirements for Reproducing Systems.*[22] — The power requirement 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. 11.12 shows the acoustical power required, as a function of the volume, in auditoriums to produce a sound level of 80 db. In large auditoriums where the orchestra is also reinforced the power available should be greater. For example, to render full artistic appeal, the system should be capable of producing a level up to 100 db. This means a power of 100 times that shown in Fig. 11.12. Systems for producing this sound

[22] Olson, H. F., *RCA Review*, Vol. 1, No. 1, p. 49, 1936.

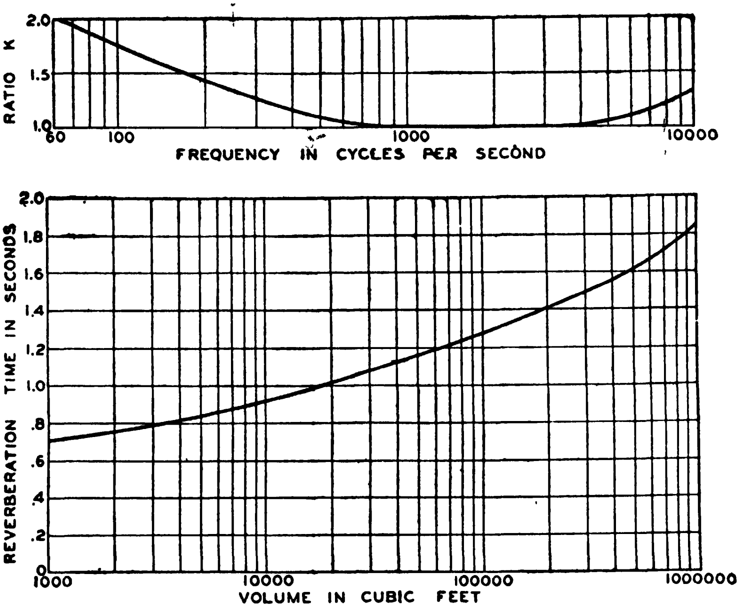

FIG. 11.11. Lower graph shows the optimum reverberation time for a theater as a function of the volume for 1000 cycles. Upper graph shows the relation between the reverberation time and the frequency, that is, the reverberation time at other frequencies is obtained by multiplying by K.

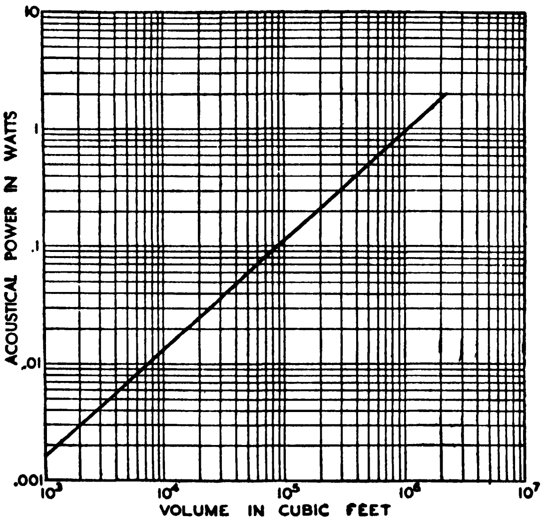

FIG. 11.12. Acoustic power required to produce an intensity level of 80 db as a function of the volume of the auditorium.

level without distortion usually require special amplifiers and loud speakers.

J. *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 range. This volume range, of course, depends upon the noise level at the listening point. The tolerable level of the noises generated in any reproducing system depends upon the noise level at the reproducing point.

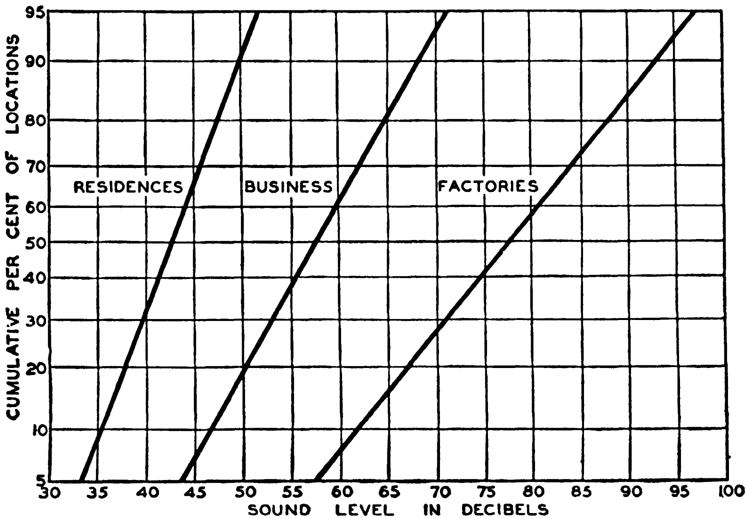

FIG. 11.13. Room noise in residence, business and factory locations. (After Seacord.)

The noise level [23] of residences, business offices and factories is shown in Fig. 11.13. 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 [24] is 25 db. With an audience the average noise level is 42 db. It may rise to 48 db and go down to as 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.

[23] Seacord, D. F., *Jour. Acous. Soc. Amer.*, Vol. 12, No. 1, p. 183, 1940.
[24] Muller, W. A., *Jour. Soc. Mot. Pict. Eng.*, Vol. 35, No. 1, p. 49, 1940.

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. 10.33.

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
Theater	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

K. *Public Address Systems.*[25] — The term public address system ordinarily refers to a sound reproducing apparatus for use in addressing large assemblages. There are innumerable specific applications of sound reproducing apparatus for this purpose. The problems in all these situations are 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.

[25] Olson, H. F., *RCA Review*, Vol. 1, No. 1, p. 49, 1936.

A large stadium equipped with a public address system is illustrated by the left portion of Fig. 11.14. Due to the size and configuration of the audience area it is practically impossible to obtain satisfactory sound level and coverage with a single loud speaker. Consequently, the loud speakers 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. 11.14 shows how uniform sound distribution is obtained in the vertical plane by means of the directional characteristics.

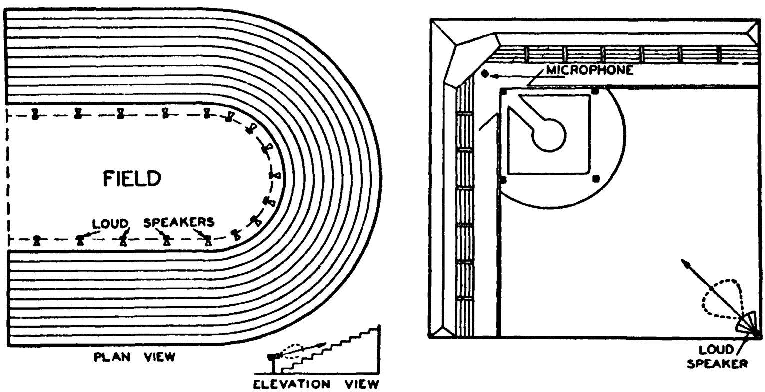

Fig. 11.14. Two arrangements of sound systems for addressing assemblages in large grandstands. For the stadium on the left a large number of loud speakers are used, each loud speaker covering a small portion of the total area. For the ball park on the right, a single loud speaker cluster is used to supply the entire grandstand.

A baseball field equipped with a public address system is illustrated by the right portion of Fig. 11.14. As contrasted to the stadium, here a single loud-speaker station is used to supply the entire audience area. The distance between the loud speakers 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 loud speaker should correspond to the vertical angle subtended by the audience at the loud speaker. Since the distance of those nearest the loud speaker to those farthest removed (that is, considering the vertical angle only) is very nearly the same, the sound intensity from the loud speaker 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 loud speaker should correspond to the angle subtended by the stands at the loud speaker. Since 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 loud-speaker 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 quite 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 loud speakers 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 loud speakers 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 loud speakers 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. 11.15. The system on the left employs a single loud-speaker station located either above or below the stage as shown in Fig. 11.15. If the stage is quite low the logical position for the loud speakers is at the top of the stage. The same procedure for obtaining uniform sound coverage and adequate intensity level of the sound from the loud speakers as used in the preceding considerations is applicable in this case. If the stage is very high the separation between the loud speakers 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 loud speaker under the stage as shown in Fig. 11.15. The system depicted on the right employs a large number of loud speakers, each one supplying a small portion of the audience. The directional characteristics of the loud speakers should be selected so that each individual area is adequately supplied. Cognizance must be taken of the energy supplied from adjacent loud speakers.

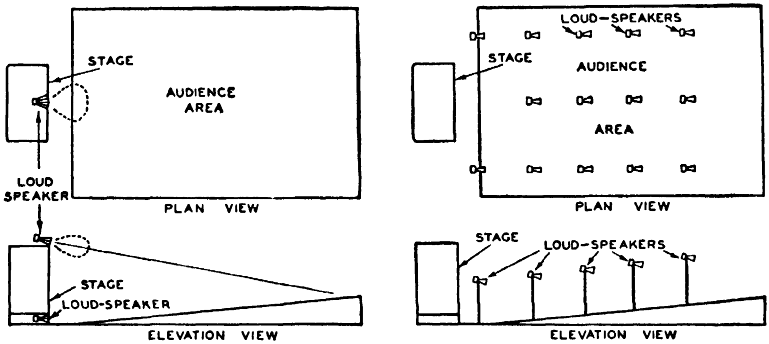

FIG. 11.15. Two arrangements of sound reinforcing systems for an outdoor theater. The arrangement on the left employs a single loud speaker located above the stage and having suitable directional characteristics to produce a uniform intensity level over the audience area. An alternate arrangement employs a loud speaker located below the stage floor. The arrangement on the right employs a number of loud speakers, each covering a small portion of the audience.

There are certain advantages in each system. In the case of the single loud-speaker 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 loud speaker, and by directing the loud speakers downward, the sound intensity level outside the audience area will be considerably lower than in the case of the single loud-speaker station and usually eliminates any annoyance difficulties. The short sound projection distance is another advantage of the multiple loud-speaker system.

The above typical examples of outdoor public address and sound reinforcing systems illustrate the principal factors involved in this field of sound reproduction.

L. *Orchestra and Stage Shell.* — When orchestra and stage productions are conducted in outdoor theaters it is desirable to provide a shell to augment 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

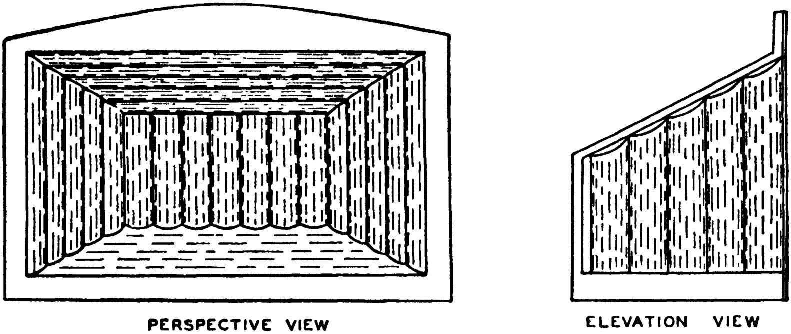

FIG. 11.16. Perspective and elevation views of an orchestra shell with polycylindrical surfaces.

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. 11.16. 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.

M. *General Announce and Paging Systems.*[26] — General announce systems are useful in factories, warehouses, railroad stations, airport terminals. etc. A typical installation is depicted on the left portion of Fig. 11.17, 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 response of the system. Furthermore, the cost of the amplifiers

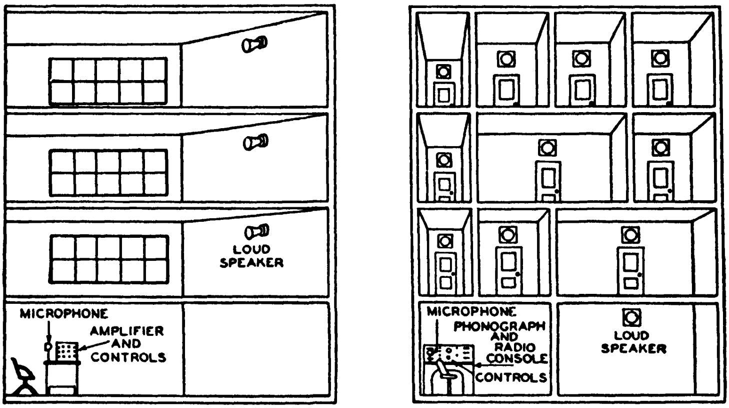

Fig. 11.17. Two uses of call, general announce, and sound distributing systems. On the left, a high efficiency horn loud speaker is used to obtain a high sound level over a large floor area as in a factory or warehouse. On the right, small, direct radiator loud speakers are used to supply the small rooms at a relatively low level as in paging, announcing, and centralized radio installations used in hospitals, hotels, or schools.

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 loud speakers 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

[26] 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. 11.12, it will be seen that the power requirements for the loud speakers will be small. To blend with the furnishings of the room, it is desirable to mount the loud speakers flush with the wall surface. Therefore, for these applications, a direct radiator loud speaker of the permanent magnet dynamic type is most suitable. In this connection it should be mentioned that these loud speakers 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 loud speakers.

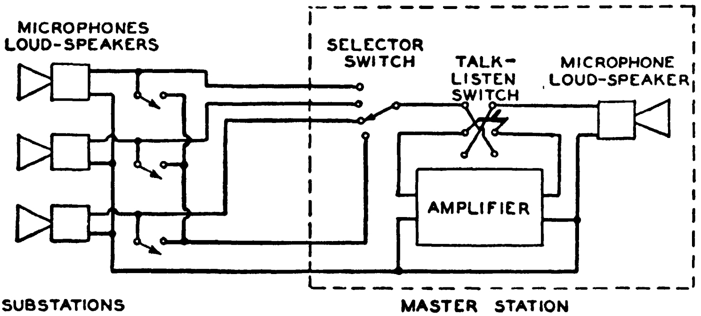

FIG. 11.18. A schematic arrangement of the elements of a simple intercommunicating system.

For large rooms requiring large acoustical outputs it is more economical to use a high efficiency loud speaker 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. 11.17 and requiring a large number of units, it is more economical to use relatively inefficient low-cost loud speakers and correspondingly larger amplifiers.

N. *Intercommunicating Systems.* — Intercommunicating systems are loud speaking telephones for use in communicating between two rooms. The more elaborate systems are similar to the general announce system described in Sec. 11.2*M* 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, loud speaker, and a talk-listen switch. The remote unit consists of a microphone, loud speaker, and talk-listen switch. In the simplest systems the loud speaker with suitable electrical compensation is also used as a microphone as shown in Fig. 11.18. 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.

O. *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 loud speakers so that the

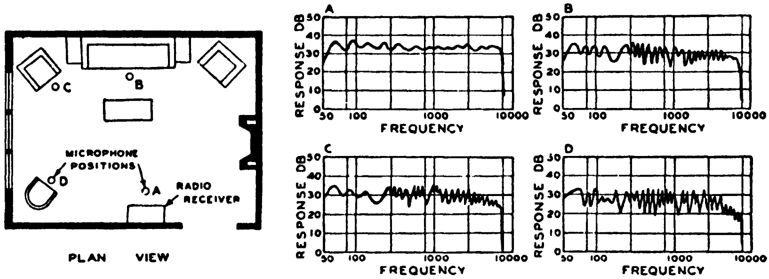

FIG. 11.19. A plan view of a living room with a radio receiver. The graphs show the response frequency characteristics for various positions in the room.

direct sound is the same in all parts of the auditorium. It is not practical to arrange the loud speakers 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. 11.19. 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.[27]

P. *Radio Receiver Operating in an Automobile.* — The loud speaker in an automobile is usually placed in one of the following three positions: in the header (above the windshield), in the instrument panel, and on the fire wall or dash. The header position gives somewhat better distribution of high-frequency response in the back seat than the other two positions. However, the low-frequency response of a loud speaker mounted in the header is usually attenuated due to the small volume behind the loud speaker. The dash or fire wall position gives good distribution of high-frequency response in the front seat but not as good distribution in the back seat. The low-frequency response in this position can be made very good by employing a large loud-speaker case or by venting the back of the case into the engine compartment. Sometimes a combination of a low-frequency dash loud speaker and a high-frequency header or instrument panel loud speaker is employed. At the present time the favored position for the loud speaker appears to be in the instrument panel because in this location the radio receiver, loud speaker, 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

[27] Wolff and Cornell, *Electronics*, Vol. 6, No. 2, p. 50, 1933.

to enclose the back of the loud speaker. Therefore, the response may be maintained in the low-frequency range. In order to reduce annoyance from hiss generated in the receiver due to the relatively weak signals delivered by the antenna, it is customary to attenuate the response above 4500 cycles. The low-frequency response in reproduction is usually masked at the higher speeds by wind noise and road rumble (also see Sec. 10.3*A*10).

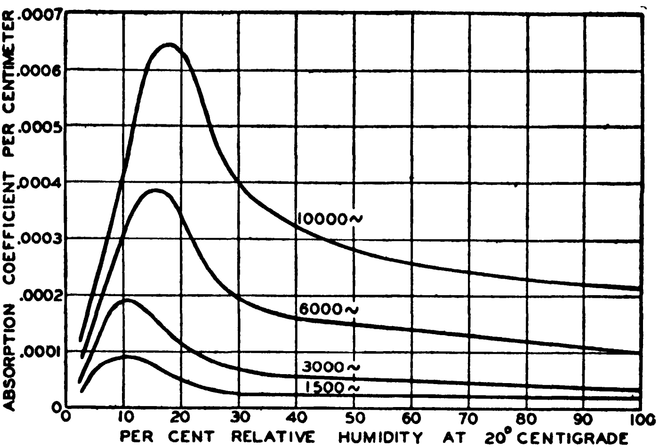

FIG. 11.20. Curves showing the absorption of a plane sound wave in passing through air, at 20° Centigrade, for different frequencies, as a function of the relative humidity. The intensity after a plane wave has travelled a distance x centimeters is $I_0\epsilon^{-mx}$, where I_0 is the intensity at $x = 0$ and m is the coefficient given by the above graph. (After Knudsen.)

Q. *Absorption of Sound in Passing Through Air.* — The absorption[28] of a plane progressive sound wave in passing through air may be several times 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 per centimeter for 1500, 3000, 6000 and 10,000 cycles as a function of the humidity is shown in Fig. 11.20.

[28] Knudsen, V. D., *Jour. Acous. Soc. Amer.*, Vol. 6, No. 4, p. 199, 1935.

R. *Sound Transmission Through Partitions.*[29,30,31,32,33,34] — 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

$$\text{T.L.} = 10 \log_{10} \frac{I_i}{I_t} = 10 \log_{10} \frac{1}{\tau} \qquad 11.17$$

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

$$\text{R.F.} = \text{T.L.} + 10 \log_{10} \frac{A}{S} \qquad 11.18$$

where A = total absorption in the receiving room, in sabins, and
S = area of the partition, in square feet.

[29] Rayleigh, "Theory of Sound," Macmillan Co., London.
[30] Eckhardt and Chrisler, Bureau of Standards, Paper No. 526.
[31] Knudsen, "Architectural Acoustics," John Wiley and Sons, New York, N. Y., 1932.
[32] Sabine, "Acoustics and Architecture," McGraw-Hill Book Co., New York, N. Y., 1932.
[33] Watson, "Acoustics of Buildings," John Wiley and Sons, New York, N. Y., 1923.
[34] Morrical, K. C., *Jour. Acous. Soc. Amer.*, Vol. XI, No. 2, p. 211, 1939.

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.

TABLE 11.3. NOISE REDUCTION THROUGH VARIOUS STRUCTURES

Material	Weight in Lbs. per Sq. Ft.	Transmission Loss in DB					Av. T.L. in DB	Average τ	Author
		Frequency							
		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, 1/8"...........	8 2		31	27	37	44	32	.00063	B.S.
Plywood, 1/4"....... ..	.73		21	21	25	26	21	008	B.S.
Celotex, Standard 1/4"..	.30		14	15	18	24	15	.032	B.S.
Celotex, Standard 1/2"..	.66		22	17	23	27	20	.01	B.S.
Hair Felt, 1"	.75	4 9	4.6	6 0	7 1	6.7	5	.32	P.E.S.
Hair Felt, 4"........		7.5	12.5	15	19	19	14	.04	P.E.S.
Wood Studs, Wood Lath, Lime Plaster.	18	27	29	38	47	43	43	.00005	P.E.S.
Tile, 2" Gypsum . ..	20	25	34	44	51	63	48	.000016	P.E.S.
Tile Clay 6" x 12" x 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.
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 1/4"		25	27	31	36	31	35	.00032	P.E.S.
Window Glass, Plate 1/4"	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.

The abbreviations in the above table are as follows: B.S., Bureau of Standards; P.E.S., P. E. Sabine.

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.*[35] — 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 bound-

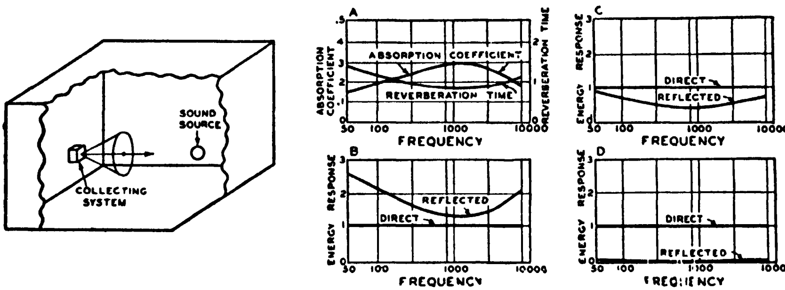

FIG. 11.21. Sound collecting system in a studio. Graph *A* shows the reverberation time and the absorption coefficient of the boundaries of a typical studio. Graphs *B*, *C* and *D* show the energy response for the direct and reflected sounds for various microphones as follows: *B*. Nondirectional microphone. *C*. Bidirectional velocity or unidirectional microphone. *D*. Ultradirectional microphone.

aries 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. 11.21, 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

$$e = \mathcal{Q} p f_1(\psi) \tag{11.19}$$

[35] Olson, H. F., *Proc. Inst. Rad. Eng.*, Vol. 21, No. 5, p. 655, 1933.

where e = voltage output of the microphone, in volts,
p = sound pressure, in dynes per square centimeter,
$\mathcal{Q}$ = sensitivity constant of the microphone, and
ψ = angle between incident pencil of sound and the reference axis of the microphone.

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

$$E_R = \frac{4E_0\Omega}{caS4\pi}\left[1 - \epsilon^{(cS\,[\log_\epsilon (1-a)]\,t)/4V}\right](1 - a) \qquad 11.21$$

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. 11.21*A*. 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. 11.21*B*, 11.21*C* and 11.21*D*). The generally reflected sound picked up by a nondirectional microphone is shown in Fig. 11.21*B*. The generally reflected sound picked up by a velocity or unidirectional microphone in which $\Omega = 4\pi/3$ is shown in Fig. 11.21*C*. The generally reflected sound picked up by an ultradirectional microphone in which $\Omega = \pi/10$ is shown in Fig. 11.21*D*. The effectiveness of a directional sound collecting system in overcoming reverberation and undesirable sounds is graphically depicted in Fig. 11.21.

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 [36] of a velocity microphone and a number of sound sources is shown in Fig. 11.22*A*. 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.

The same procedure [37] for balancing the instruments of an orchestra

[36] Olson, H. F., *Proc. Inst. Rad. Eng.*, Vol. 21, No. 5, p. 655, 1933.
[37] Olson, H. F., *Jour. Soc. Mot. Pic. Eng.*, Vol. 27, No. 3, p. 284, 1936.

may be used in connection with a unidrectional microphone (Fig. 11.22*B*). 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.

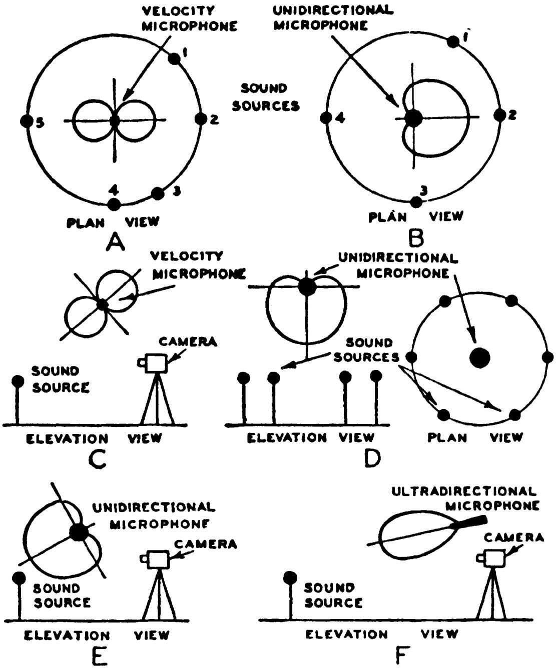

FIG. 11.22. Examples illustrating the use of directional microphones.

The velocity microphone is used in recording music and other sound in sound motion pictures. In some instances noises produced by the camera and devices are objectionable and must be reduced. The directional characteristics of the velocity microphone are useful in overcoming objectionable noises.[38] 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. 11.22*C*.

In certain types of recording [39] it is desirable to place the microphone at the center of action directed downward and collect sounds over an angle of

[38] Olson, H. F., *Jour. Soc. Mot. Pic. Eng.*, Vol. 16, No. 6, p. 695, 1931.
[39] Olson, H. F., *Jour. Soc. Mot. Pic. Eng.*, Vol. 27, No. 3, p. 284, 1936.

360° with respect to the microphone. (Fig. 11.22*D* illustrates the use of a unidirectional microphone for this application.)

The unidirectional microphone is almost universally employed for speech pickup in sound motion pictures and television (Fig. 11.22*E*). 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[40] provides a high order of discrimination against reverberation and other undesirable sounds (see Sec. 8.6*B*). 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. 11.22*F*). 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. 11.9.

B. *Broadcasting Studios.*[41,42,43,44,45,46,47,48] — 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.

[40] Olson, H. F., *Proc. Inst. Rad. Eng.*, Vol. 27, No. 7, p. 438, 1939.
[41] Morris and Nixon, *Jour. Acous. Soc. Amer.*, Vol. 8, No. 2, p. 81, 1936.
[42] Chinn and Bradley, *Proc. I. R. E.*, Vol. 27, No. 7, p. 421, 1939.
[43] Potwin and Maxfield, *Jour. Acous. Soc. Amer.*, Vol. 11, No. 1, Part 1, p. 48, 1939.
[44] Nixon, G. M., *RCA Review*, Vol. 6, No. 3, p. 259, 1942.
[45] Volkmann, J. E., *Jour. Acous. Soc. Amer.*, Vol. 13, No. 3, p. 234, 1942.
[46] Boner, C. P., *Jour. Acous. Soc. Amer.*, Vol. 13, No. 3, p. 244, 1942.
[47] Content and Green, *Proc. Inst. Rad. Eng.*, Vol. 32, No. 2, p. 72, 1944.
[48] Nygren, A., *FM and Television*, Vol. 6, No. 5, p. 25, 1946.

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

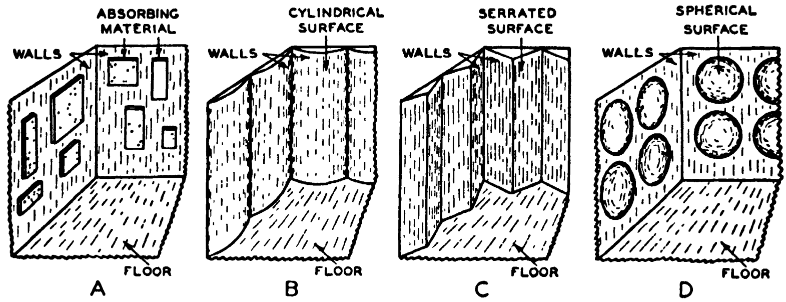

FIG. 11.23. Wall structures for diffusing the reflected sound. *A*. Absorbing material located in spots. *B*. Cylindrical surfaces. *C*. Serrated surfaces. *D*. Spherical surfaces.

value as possible. This can, of course, be done by making the walls very 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. 11.23. In Fig. 11.23*A*, the absorbing material is distributed in discrete spots[49] on the wall surface. This distribution of material breaks up the reflected wave front and thereby produces a diffuse sound field. In Fig. 11.23*B*, *C* and *D*; polycylindrical,[50] serrated[51] and spherical surfaces[52] are employed to produce a diffuse sound field. These surfaces have been used for walls and ceilings in broadcast studios. The polycylindrical and spherical surfaces increase the wavefront of the reflected sound. The convex surfaces also reduce the interference effect between direct and reflected sounds. The treatments shown

[49] Potwin and Maxfield, *Jour. Acous. Soc. Amer.*, Vol. 11, No. 1, Part 1, p. 48, 1939.
[50] Volkmann, J. E., *Jour. Acous. Soc. Amer.*, Vol. 13, No. 3, p. 234, 1942.
[51] Nixon, G. M., *RCA Review*, Vol. 6, No. 3, p. 259, 1942.
[52] Nygren, A., *FM and Television*, Vol. 6, No. 5, p. 25, 1946.

in Fig. 11.23 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 Fig. 11.23 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 parallelepipeds.

Various expedients, in addition to the wall structures of Fig. 11.23, 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[53] of the studio is important in distributing the characteristic resonant frequencies uniformly over the frequency range. The graph of Fig. 11.24 shows the ratio of the dimensions for small, medium 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. 11.24. 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. 11.24. 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. 11.24.

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. 11.25. The wall and ceiling surfaces of the stage are arranged to provide 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

[53] Volkmann, J. E., *Jour. Acous. Soc. Amer.*, Vol. 13, No. 3, p. 234, 1942.

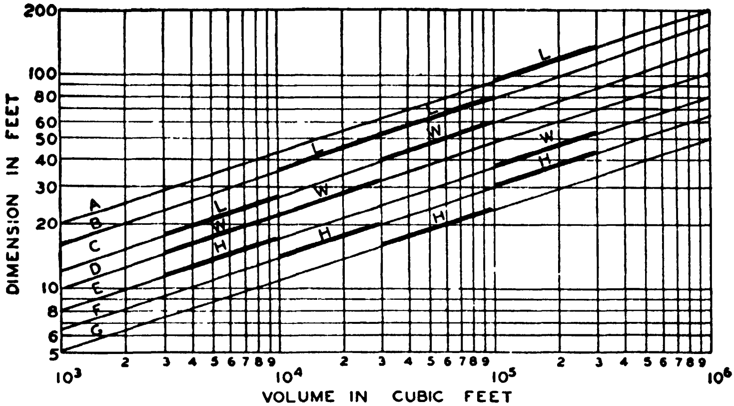

FIG. 11.24. Preferred studio dimensions. In the graph, H = height, W = width, L = length. Small rooms, H:W:L = 1:1.25.1.6 = E:D:C. Average shape rooms, H:W:L: = 1:1.6:2.5 = F:D:B. Low ceiling rooms, H:W:L:= 1:2.5:3.2 = G:C:B. Long rooms, H:W:L = 1:1.25:3.2 = F:E:A. (After Volkmann.)

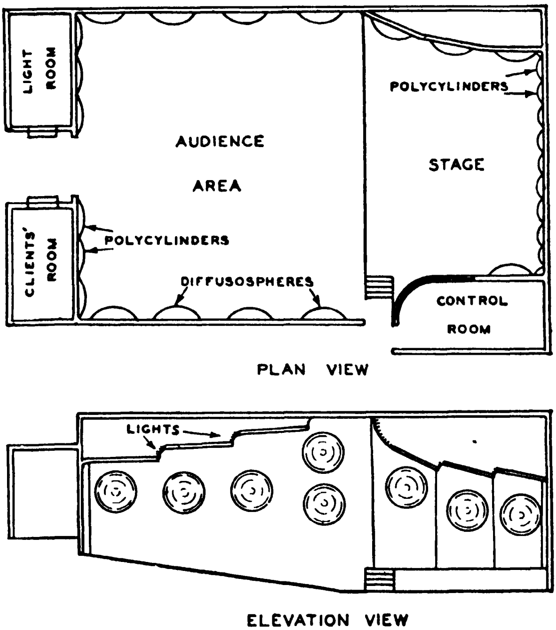

FIG. 11.25. Plan and elevation views of an auditorium type broadcasting studio. (After Nixon and Nygren.)

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, 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. *Scoring and Recording Studios.*[54, 55, 56, 57, 58, 59] — 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 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. 11.26. 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 $\frac{1}{4}$-inch plywood. One fourth of the convex surface is made absorbent as shown in Fig. 11.26. 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

[54] Rettinger, M., *Proc. I. R. E.*, Vol. 28, No. 7, p. 296, 1940.
[55] Volkmann, J. E., *Jour. Acous. Soc. Amer.*, Vol. 13, No. 3, p. 234, 1942.
[56] Rettinger, M., *Jour. Soc. Mot. Pic. Eng.*, Vol. 39, No. 3, p. 186, 1942.
[57] Livadary and Rettinger, *Jour. Soc. Mot. Pic. Eng.*, Vol. 42, No. 6, p. 361, 1944.
[58] Slyfield, C. O., *Jour. Soc. Mot. Pic. Eng.*, Vol. 42, No. 6, p. 367, 1944.
[59] Ryder, L. L., *Jour. Soc. Mot. Pic. Eng.*, Vol. 42, No. 6, p. 369, 1944.

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 $\frac{1}{2}$ inch thick and plywood $\frac{3}{8}$ inch thick were applied to the studs between the stripping to produce a series of horizontal rockwool, fireboard 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

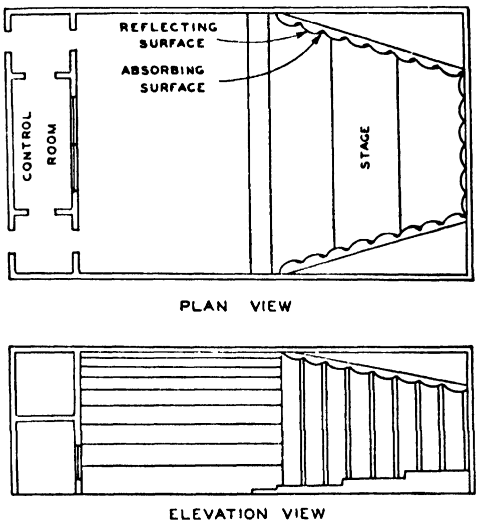

Fig. 11.26. Plan and elevation views of a scoring and recording studio. (After Rettinger.)

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.

D. *Vocal Studios.*[60, 61] — In sound motion pictures when scoring an

[60] Mounce, Portman and Rettinger, *Jour. Soc. Mot. Pic. Eng.*, Vol. 42, No. 6, p. 375, 1944.
[61] Ryder, L. L., *Jour. Soc. Mot. Pic. Eng.*, Vol. 42, No. 6, p. 379, 1944.

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 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. 11.3*B*). 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. 11.2*R*.

E. *Reverberation Time of Broadcasting, Recording and Scoring Studios.*[62] — The optimum reverberation time of broadcasting, recording and scoring studios as a function of the volume of the studio, for a frequency of 1000 cycles, is shown in the lower graph of Fig. 11.27. The reverberation for other frequencies can be obtained by multiplying by the factor K, obtained from the upper graph of Fig. 11.27. The reverberation time is greater 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 12.12).

F. *Sound Stages for Motion Pictures and Television.*[63] — 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

[62] Morris and Nixon, *Jour. Acous. Soc. Amer.*, Vol. 8, No. 2, p. 81, 1936.
[63] Ringel, A. S., *Jour. Soc. Mot. Pic. Eng.*, Vol. 15, No. 3, p. 352, 1930.

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 technic 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

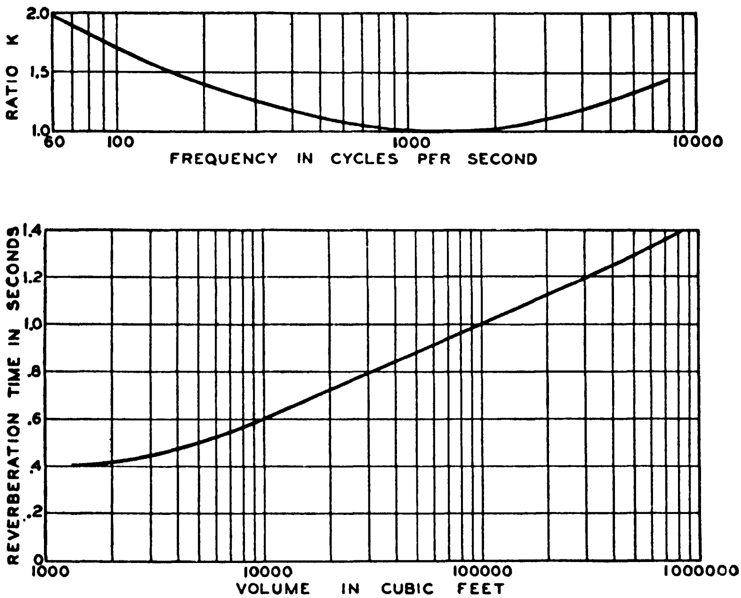

FIG. 11.27. Lower graph shows the reverberation time for a recording or broadcasting studio as a function of the volume for 1000 cycles. Upper graph shows the relation between the reverberation time and the frequency, that is, the reverberation time at other frequencies is obtained by multiplying by K. (After Morris and Nixon.)

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. 11.3*A*). 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 overall 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 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 ½-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.

G. *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.

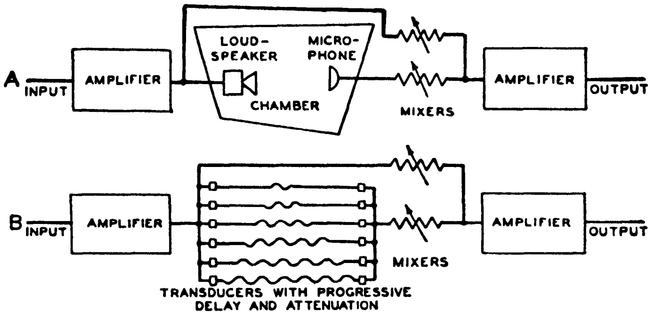

FIG. 11.28. Schematic arrangement of systems for introducing synthetic reverberation in reproduced sound. *A*. Loud speaker, microphone and reverberant chamber combination. *B*. A system consisting of transducers with progressive delay and attenuation.

Artificial reverberation may be added to a sound signal by means of the loud speaker, reverberant chamber and microphone combination shown in Fig. 11.28*A*. 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 both the reverberation time of the chambers and the coupling between the chambers. The loud speakers, 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. 11.28*B*. 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 loud speakers and microphones terminating the ends. The transducer may be a recorder and a series of pickups on a phonograph record or magnetic tape [64] or phosphor wheel.[65] The reverberation time may be varied by varying the progressive attenuation. Mixers are provided so that any ratio of the original sound to the reverberant sound may be obtained.

H. *Volume Limiters, Compressors and Expanders.*[66, 67, 68, 69] — 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. 11.29. The input signal is amplified and rectified. The rectified signal is applied to a resistance 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. 11.29, 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. 11.29, there is a gradual reduction in the gain with increase of the input. A reduction in the volume range in radio and phonograph reproduction makes it pos-

[64] Wolf, S. K., *Proc. I. R. E.*, Vol. 27, No. 7, p. 365, 1939.
[65] Goldmark and Hendricks, *Proc. I. R. E.*, Vol. 27, No. 12, p. 747, 1939.
[66] Sinnett, C. M., *Electronics*, Vol. 8, No. 11, p. 14, 1935.
[67] Norman, N. C., *Bell Labs. Record*, Vol. 13, No. 4, p. 98, 1934.
[68] Mathes and Wright, *Bell Syst. Tech. Jour.*, Vol. 13, No. 3, p. 315, 1934.
[69] Steinberg, J. C., *Jour. Acous. Soc. Amer.*, Vol. 13, No. 2, p. 107, 1941.

sible 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. 11.29, there is a gradual increase in gain with increase in output. The combination of a

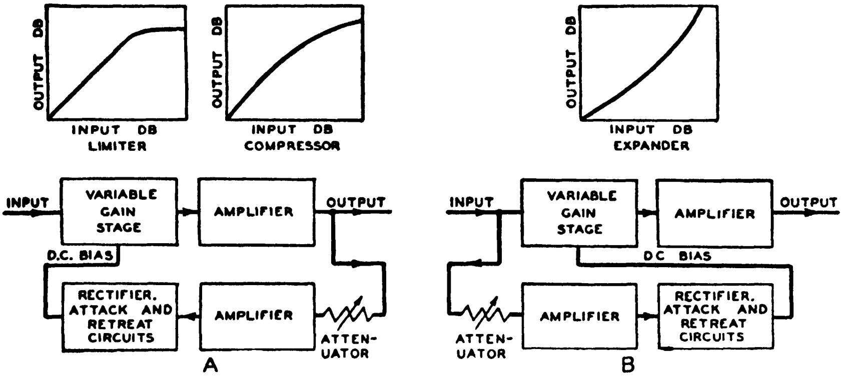

FIG. 11.29. Input versus output characteristics of limiters, compressors, and expanders. *A*. The elements of a limiter or a compressor. *B*. The elements of an expander.

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.

11.4. Complete Sound Reproducing Systems.— A. *Telephone.*[70]— The telephone is a sound reproducing system consisting of a carbon microphone (sometimes termed the transmitter), a telephone receiver, and a battery. The action of the carbon microphone has been described in Sec. 8.2*A*. The action of the telephone receiver has been described in Sec. 9.2. A schematic diagram of a telephone system is shown in Fig. 11.30. Each telephone station is connected by a line to a central office. The battery supply or other equipment is located at the central office. The function of the central office is to connect any subscriber to any other subscriber.

[70] Johnson, "Transmission Circuits for Telephonic Communication," D. Van Nostrand Company, New York, N. Y., 1924.

In large cities, there are many central or local offices, because it is not economical or practical for a central office to serve more than about 10,000 stations. The local offices are interconnected by lines. In local transmission the electrical output of the microphone 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 normal. The system in Fig. 11.30 depicts the electroacoustic elements of a telephone transmission system. In addition, equipment must be supplied for the subscriber to signal the operator and to permit the operator to send ringing current to the subscriber. The further consideration of circuits, switchboards, repeaters, manual and automatic exchanges, etc., are outside the scope of this book and the reader is referred to books on these subjects.

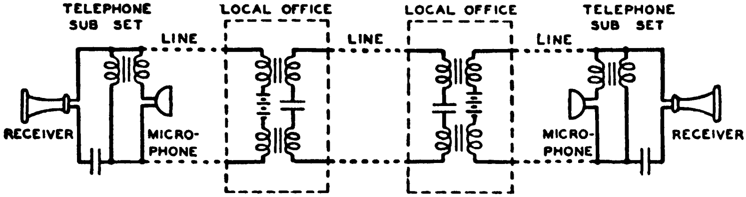

FIG. 11.30. Schematic arrangement of the apparatus in a telephone system.

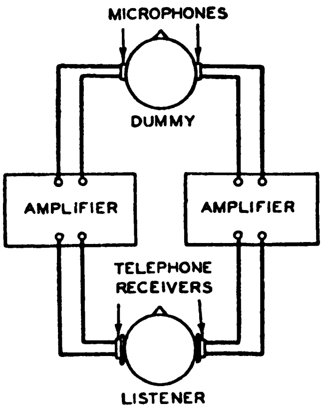

FIG. 11.31. Schematic arrangement of the apparatus for a binaural sound reproducing system.

B. *Binaural Sound Reproducing System.* — An ideal binaural sound reproducing system[71] is shown schematically in Fig. 11.31 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, M_R and M_L, 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 tele-

[71] Olson and Massa, *Jour. Soc. Mot. Pic. Eng.*, Vol. 23, No. 2, p. 63, 1934.

phone 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.

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.

There are two serious disadvantages to this ideal system in addition to the requirement for a double channel. In the first place, a set of ear phones which must be worn throughout the performance and would not

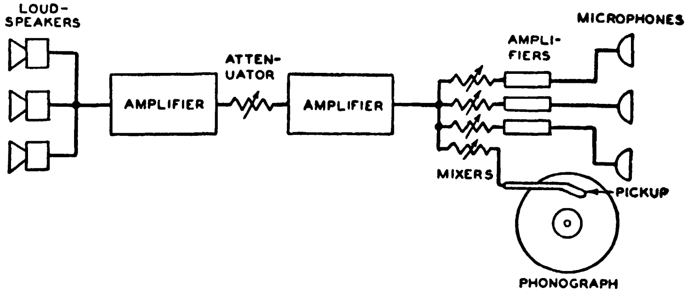

FIG. 11.32. Schematic arrangement of the apparatus for a monaural sound reproducing system.

be tolerated by most persons, is required for each member of the audience. Second, in sound motion pictures, each listener should be in the same position relative to the screen as the dummy was relative to the original set. Such a condition is obviously impossible of realization and, consequently, those members of the audience who are somewhat removed from the screen will recognize a binaural effect not in accord with their distances from the scene. It appears, therefore, that the practical limitations of the ideal system render it undesirable for commercial application.

C. *Monaural Sound Reproducing System.* — A monaural or single-channel sound reproducing system is shown in Fig. 11.32. The system consists of one or more micróphones with preamplifiers and mixers, voltage amplifier, attenuator, power amplifier and one or more loud speakers. A phonograph turntable is provided for use in reproducing phonograph records. The system shown in Fig. 11.31 is used for sound reinforcement, public address, announce, paging and numerous other applications in theaters, churches, auditoriums, outdoor theaters, mass meetings, athletic events, factories, offices, restaurants, etc.

D. *Auditory Perspective Reproducing System.*[72, 73, 74] — 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 schematically in Fig. 11.33. The sound is picked up by three microphones and amplified in separate channels, each channel feeding a separate loud speaker amplified in separate channels, each channel feeding a separate loud speaker. The three loud speakers are arranged on the stage in the same positions as the microphones on the pickup stage. One of the principal objections to the system is the number of channels required. In the case of sound motion picture reproduction three separate sound tracks would be required. Radio reproduction would require three separate transmitters and channels. Some laboratory tests have been made in which two channels are used instead of three. This arrangement is a distinct improvement over a single-channel system and appears to have commercial possibilities in both sound motion picture and radio reproduction.

In addition to the objections that arise from the need of several channels, two other conditions tend to operate against the ideality of the system. The first arises from the fact that the acoustical characteristics of the theater or auditorium are superimposed upon that of the set. Another objection arises from the necessity of requiring that the sound sources be spread far apart for the best effect. That means that the picture that is being reproduced should be spread out to cover the same area as that occupied by the sound source; otherwise, the sound will appear to come " off stage " instead of from the picture. In the case of radio reproduction it may be impossible to spread the loud speakers sufficiently to obtain good illustration of perspective in the average living room.

E. *Sound Motion Picture Reproducing System.* — A complete sound motion picture recording and reproducing system is shown in Fig. 11.34. The first element is the acoustics of the set. The factors which influence 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 re-

[72] Fletcher, H., *Jour. Soc. Mot. Pic. Eng.*, Vol. 22, No. 5, p. 314, 1934.

[73] Maxfield, Colledge and Friebus, *Jour. Soc. Mot. Pic. Eng.*, Vol. 30, No. 6, p. 666, 1938.

[74] Fletcher, H., *Jour. Acous. Soc. Amer.*, Vol. 13, No. 2, p. 89, 1941.

production. A high-pass filter is used on speech with the lower limit placed below the speech range. This latter expediency reduces low-frequency noises without impairing the speech quality. An equalizer is used to accentuate the high frequencies to compensate for the film transfer loss

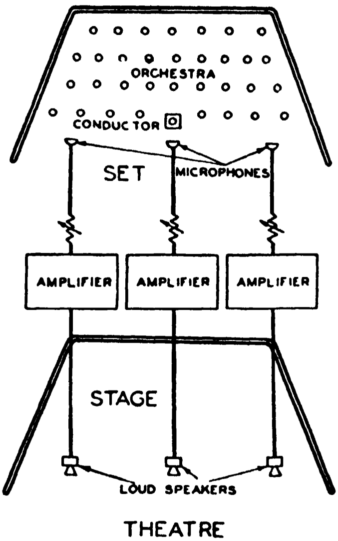

Fig. 11.33. Schematic arrangement of the apparatus for an auditory perspective reproducing system.

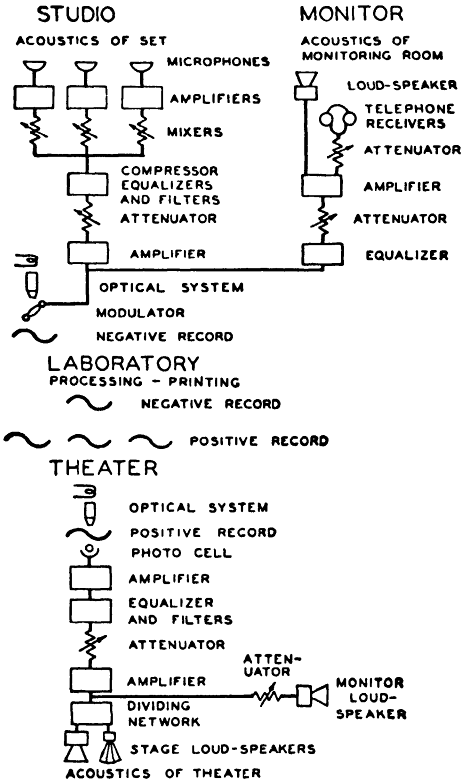

Fig. 11.34. Complete sound motion picture recording, processing and reproducing system.

at the high frequencies. A compressor is used to reduce the volume range (see Sec. 11.3*H*). The following attenuator controls the overall 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. 9.8*A*). 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 monitoring is carried out in a room a loud speaker is used. When the monitoring and mixing is carried out on the set, head phones are used for monitoring.

The negative record is sent to the laboratory and developed. Then positive records of both the sound and the picture are printed from the sound and picture negatives. These positive records are then developed and are ready for reproduction in a theater.

The variable density or variable area record is reproduced in the theater by pulling it pass 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. 9.8*B*). These are then amplified 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 loud speakers and monitoring loud speaker. The monitoring loud speakers and the attenuator are located in the projection booth. As a matter of fact, the entire system, save for the stage loud speakers, is located in the projection booth. A dividing network and a two-channel loud-speaker system are shown in Fig. 11.34. Of course, any type of suitable loud speaker described in Chapter VII may be used. The action of a sound motion picture reproducer in a theater has been discussed in Sec. 11.2*E* and will not be repeated here.

In some cases, the original record is re-recorded and additional sound is added. For example, in certain dialogue sequences it may be desirable to add incidental music to heighten the artistic effects. In this case the dialogue is recorded first. Then this record is reproduced on a system similar to the theater reproducer. The orchestra is picked up on the standard recording system. The two outputs are mixed and fed to the modulator and a new negative record is made.

F. *Radio Sound Reproducing System.* — A complete radio broadcasting and receiving system is shown in Fig. 11.35. The factors which influence the collection of sound in a broadcasting studio have been discussed in Sec. 11.3. The mixers, compressors, and attenuators are used in the same manner as in the case of the sound motion picture system. The monitoring system and mixer console 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. The output of the studio amplifier feeds the isolating amplifiers and the monitoring system. Isolating amplifiers feed the wire lines to the various transmitters. In the case

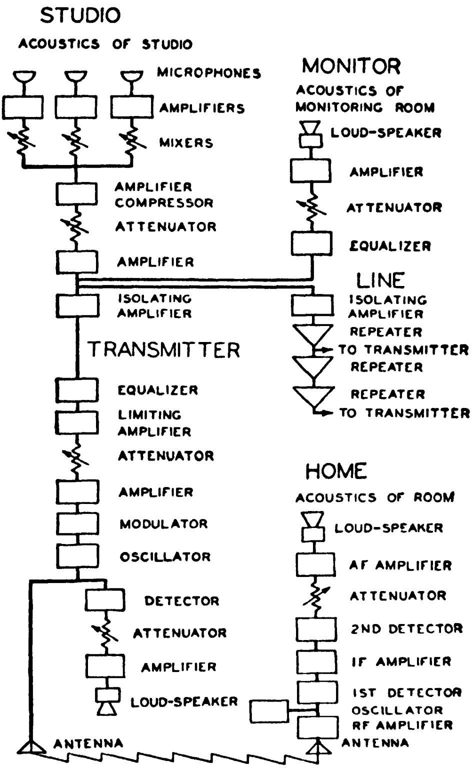

FIG. 11.35. Complete radio broadcasting and receiving system.

of the local transmitter, repeaters are not needed and the output of the isolating amplifier is fed directly to the equalizer and limiting amplifier. The equalizer compensates for the frequency discrimination in the line. The limiting amplifier is of the compressor type in which the peak volume levels are reduced. The limiting amplifier is followed by an attenuator for controlling the input to the transmitter amplifier. The amplifier is followed by the modulator and radio frequency oscillator. The output of the oscillator feeds the antenna. The monitoring system at the transmitter consists of a detector coupled to the output of the transmitter and followed by an attenuator, amplifier and loud speaker.

A very small portion of the radio frequency energy radiated by the transmitter antenna is picked up by the receiving antenna. A typical superheterodyne receiver is shown in Fig. 11.35. The output of the antenna is amplified in the radio frequency stages and combined with an intermediate frequency oscillator and fed to the first detector. The output of the first detector is amplified by the intermediate frequency amplifier and then fed to the second detector. The audio-frequency output of the detector is followed by an attenuator and power amplifier which drives the loud speaker. The action of a radio reproducer in a living room has been discussed in Sec. 11.20.

G. *Phonograph Reproducing System.* — A complete phonograph recording and reproducing system is shown in Fig. 11.36. The general studio equipment is quite similar to that of a broadcasting studio except for the equalizers and filters. An equalizer is used to attenuate the output fed to the cutter below about 800 cycles to yield approximately uniform amplitude below 800 cycles. The high-frequency response is accentuated in recording so that a corresponding attenuation in the high-frequency response in reproduction will effect a reduction in ground noise without frequency discrimination in the reproduced sound (see Sec. 9.3*A*3 and Fig. 9.9). The cutter actuated by the amplifier cuts a spiral wavy path in the revolving record corresponding to the undulations in the original sound wave striking the microphone (see Sec. 9.3*A*).

In the record plant the original wax is metalized by sputtering with gold and electroplating with copper. The resulting plating is separated from the wax, is backed by solid metal plate, and is termed the master. The master is electroplated with copper. This plating is separated from the master matrix, is backed by a solid plate, and is termed the mother. The mother in turn is electroplated with copper. This plating is separated from the mother, is backed by a solid metal plate, and is termed the stamper. By means of a hydraulic press the final records or pressings are pressed in thermoplastic material by the stamper. The spiral wavy path in the final record corresponds to that in the original soft wax record.

In reproducing, the record is turned at a constant speed by the turntable. The stylus or needle of the pickup follows the wavy spiral groove and generates voltage corresponding to the undulations in the record (see Sec. 9.3*C*). The output of the pickup is equalized to compensate for the equalization in the original recording. A low-pass filter is used to attenuate the response above the upper frequency limit of reproduction and thereby effects a reduction in the ground noise. An attenuator controls the volume of reproduction. The attenuator is followed by the amplifier which drives the loud speaker. The action of a reproducer in a living room has been considered in Sec. 11.20.

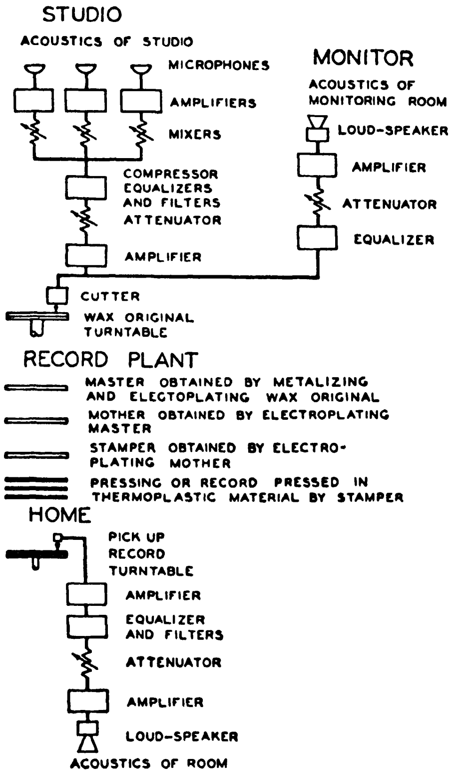

FIG. 11.36. Complete phonograph recording, record pressing and reproducing system.

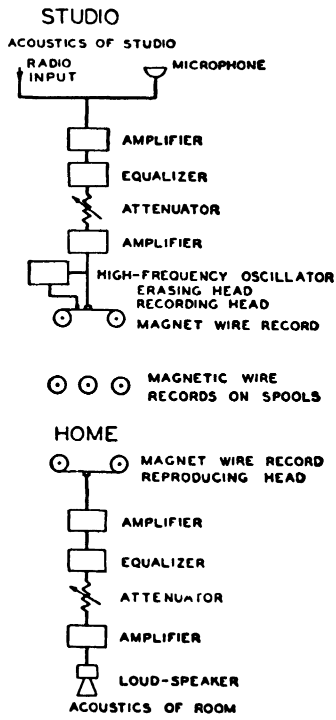

FIG. 11.37. Complete magnetic wire or tape recording, records and reproducing system.

H. *Magnetic Sound Reproducing System.* — A complete magnetic sound recording and reproducing system is shown in Fig. 11.37. The magnetic recorder system may be used to record from a studio, from the output of a radio receiver, or in the home. The elements of the magnetic recorder and reproducer have been described in Sec. 9.7. Fig. 11.37 depicts all the elements of a magnetic recording and reproducing system. The output of a microphone or a radio receiver is amplified and sent through an equalizer to compensate for the characteristics of the magnetic wire or tape. The recording head, actuated by the amplifier, magnetizes the wire or tape in a manner corresponding to the undulations in the original sound wave. The magnetic wire or tape records are stored on spools. In reproducing, the wire or tape is pulled past the recording head at the speed of the original recording. The magnetic undulations in the wire generate a voltage in the coil of the recording head which correspond to the original sound vibrations. The output of the reproducing head is equalized and amplified and fed to a loud speaker. A variable attenuator controls the volume of reproduction. The action of a reproducer in a living room has been considered in Sec. 11.20.

CHAPTER XII

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] — 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 side of the basilar membrane is the organ of Corti, which contains the

[1] Fletcher, "Speech and Hearing," D. Van Nostrand Company, New York, N. Y., 1929.

[2] Steinberg, "Acoustics"; Pender and McIlwain, "Electrical Engineers Handbook," *Communication and Electronics*, John Wiley and Sons, New York, N. Y., 1936.

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.

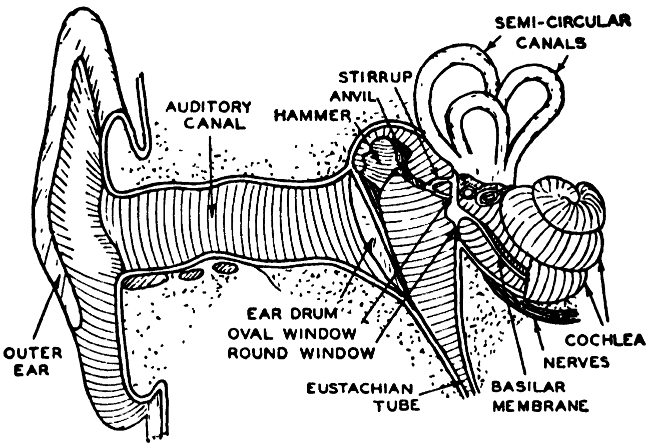

FIG. 12.1. Sectional and perspective views of the hearing mechanism.

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 ear drum to vibrate. The vibration of the ear drum 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 $\dot{X}_1$, $\dot{X}_2 \ldots \dot{X}_K$ in the branches $1, 2 \ldots K$. 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 some what greater than the actual volume which is due to the resilient walls and ear drum.

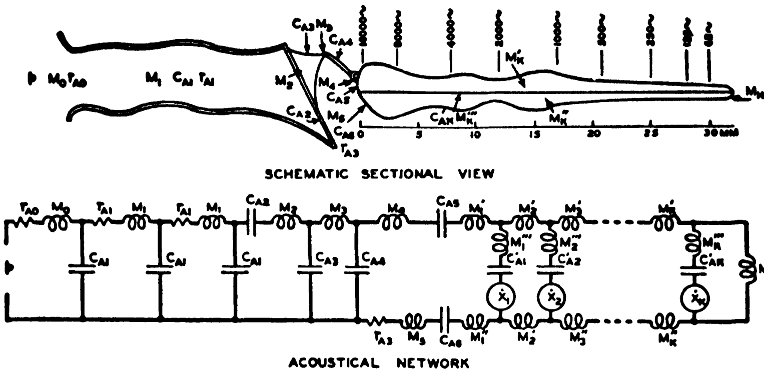

FIG. 12.2. Schematic sectional view and acoustical network of the hearing mechanism. In the acoustical network: p = the actuating sound pressure. M_0 and r_{A0} = the inertance and acoustical resistance of the air load upon the opening to the ear canal. M_1, C_{A1} and r_{A1} = the distributed inertance, acoustical capacitance and acoustical resistance of the ear canal. M_2 = the inertance of the ear drum and hammer. C_{A2} = the acoustical capacitance of the ear drum 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 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 vestibula. M''_1, M''_2, $M''_3 \ldots M''_K$ represent inertances of the liquid in the scala tympani. 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.[3, 4]—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

[3] Fletcher, "Speech and Hearing," D. Van Nostrand Company, New York, N. Y., 1929.

[4] Wegel, R. L., *Jour. Acous. Soc. Amer.*, Vol. 1, No. 3, Part 2, p. 1, 1930.

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 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

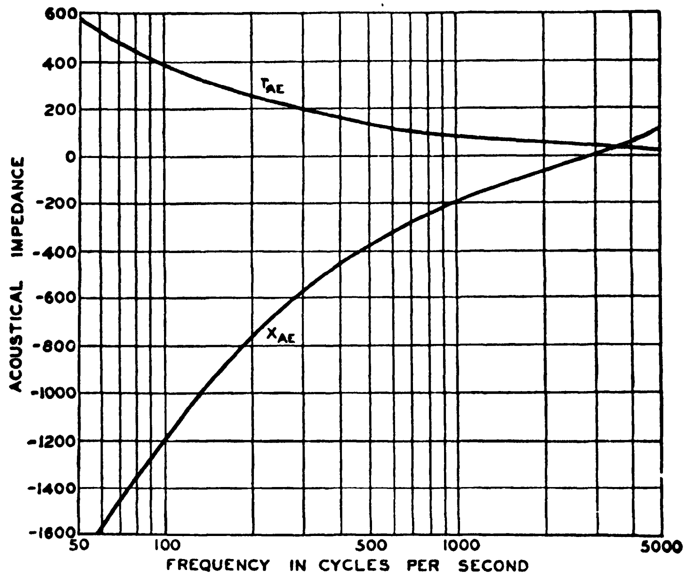

FIG. 12.3. The acoustical impedance components of acoustical resistance r_{AE} and acoustical reactance x_{AE} characteristics at the input to the ear canal. (After Steinberg.)

be replaced by a generator having an internal pressure μp_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 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 [5] by the rapid

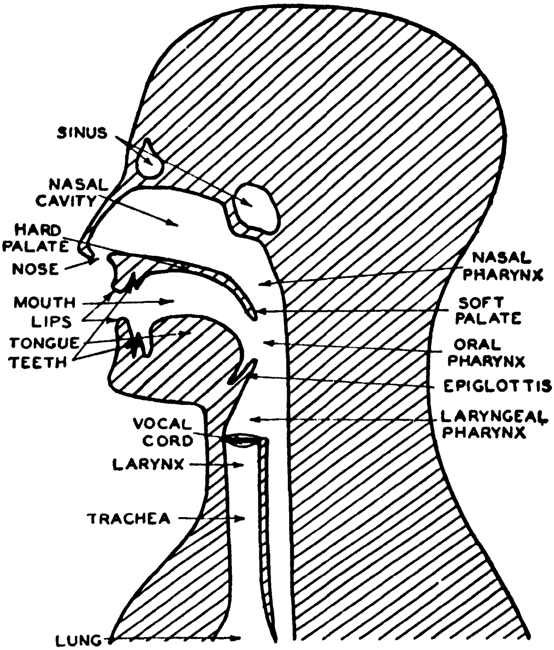

FIG. 12.4. Sectional view of the head showing the voice mechanism.

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 constants and the nonlinear element are suitable. The wave shape of a relaxation oscillator corresponds to the general wave shape of the output of the vocal cords 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. A saw-tooth wave contains the fundamental and all the harmonics. Therefore, the generator, μp_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

[5] Drew and Kellogg, *Jour. Acous. Soc. Amer.*, Vol. 12, No. 1, p. 95, 1940.

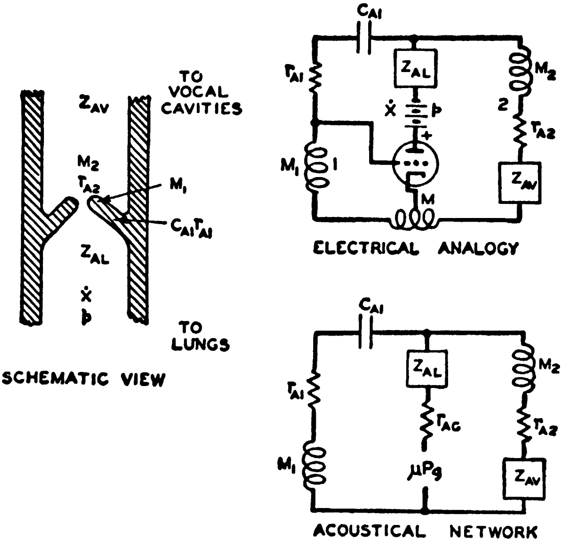

FIG. 12.5. Sectional view of a vibrating system approximating the larynx. In the electrical analogy and acoustical network: p and X = the d-c air pressure and d-c volume current supplied by the lungs. M_2 and r_{A2} = the inertance and acoustical resistance of the aperture formed by the vocal cords. 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 cords. r_{AG} = the acoustical resistance of the generator. μp_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.

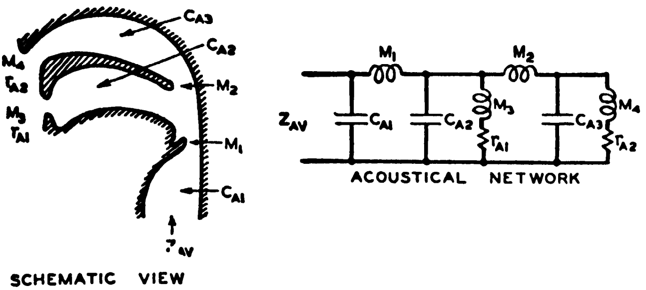

FIG. 12.6. Schematic sectional view and the acoustical network of the vocal cavities. In the acoustical circuit: z_{AV} = the input acoustical impedance to the vocal cavities. C_{A1} = the acoustical capacitance of the laryngeal pharynx. M_1 = the inertance of the narrow passage determined by the epiglottis. C_{M2} = the acoustical capacitance of the mouth cavity. M_2 = the inertance of the passage connecting the mouth and nasal cavities. M_3, M , r_{A1} and r_{A2} = the inertances and acoustical resistances of the mouth and nose openings and the air load upon these openings.

(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 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

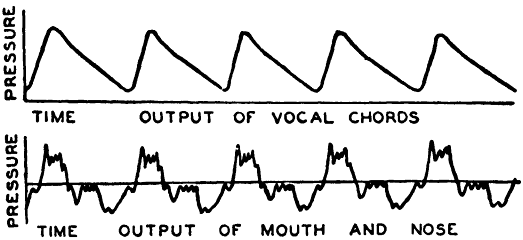

FIG. 12.7. Wave shapes of the output of the vocal cords and the mouth and nose for the vowel sound ĕ.

altered which in turn alters the output harmonic content. These changes together with a change in the fundamental frequency of the vocal cords 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 principle 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 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.

12.4. Artificial Voice Mechanisms. A. *Artificial Larynx.*[6] — A surgical operation known as a tracheotomy leaves no connection between the lungs and mouth. It is performed in an emergency to prevent the patient

[6] Riesz, R. R., *Jour. Acous. Soc. Amer.*, Vol. 1, No. 2, p. 273, 1930.

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

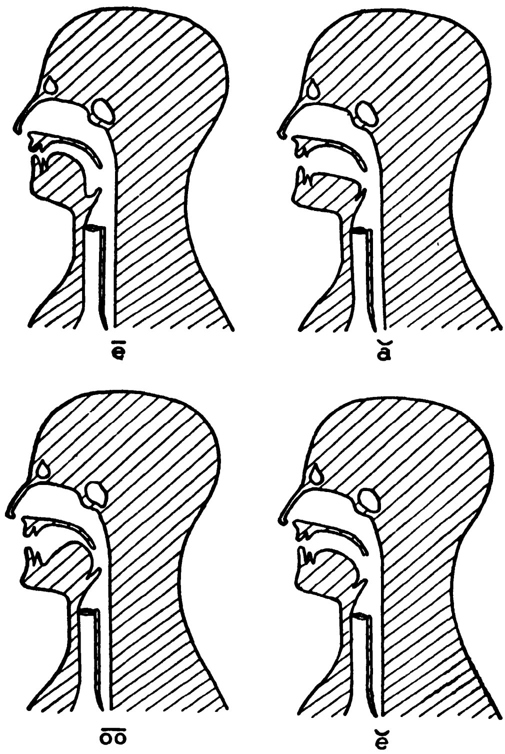

Fig. 12.8. Sectional views of the head showing the configuration of the mouth cavity for different vowel sounds.

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.*[7] — The vocal tone, as outlined in Sec. 12.3, consists of a buzzer-like tone. The breath tone is a hiss-like 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

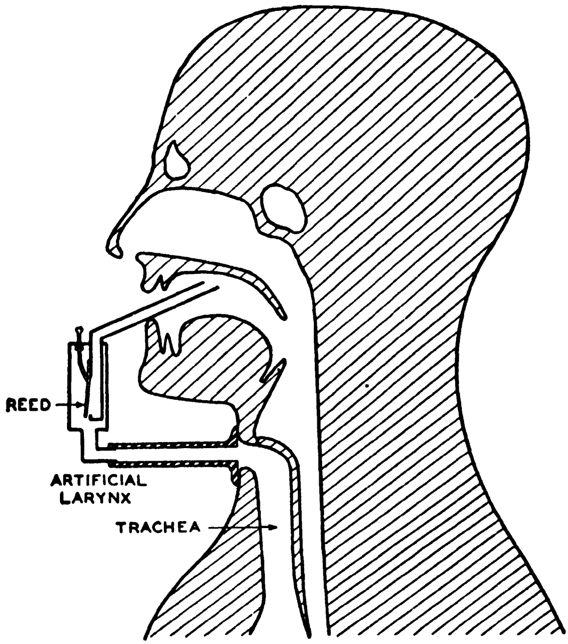

FIG. 12.9. The elements of an artificial larynx.

hiss-like 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. 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 loud speaker 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.

[7] *Bell Labs. Record*, Vol. 17, No. 6, p. 170, 1939.

C. *Vocoder.*[8] — 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 " vocoder " because it operates on the principle of coding the voice and then reconstructing the voice in accordance with the code.

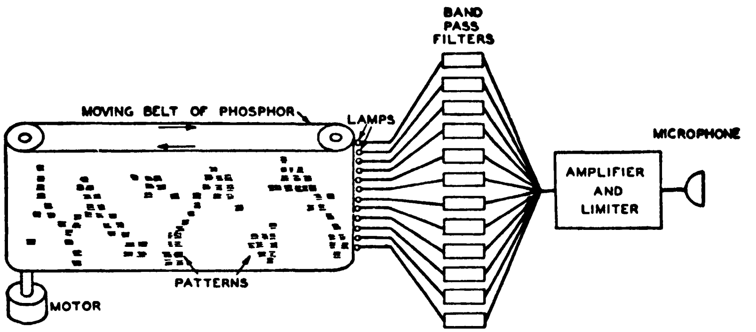

FIG. 12.10. The elements of a system for producing visible speech. (After Potter.)

12.5. Visible Speech.[9] — 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 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

[8] Dudley, H., *Jour. Acous. Soc. Amer.*, Vol. 11, No. 2, p. 169, 1939.

[9] 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, p. 1 to 89 inc., 1946.

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.

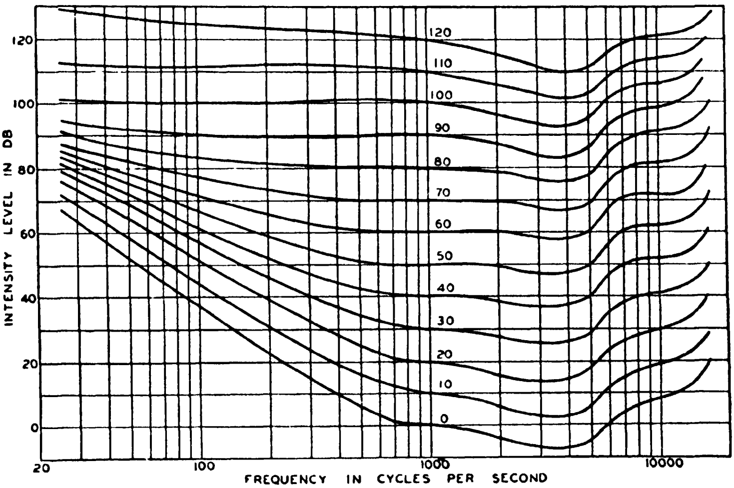

FIG. 12.11. Contour lines of equal loudness for normal ears. Numbers on curves indicate loudness level. 0 db = 10^{-16} watts per square centimeter. 0 db = 0.000204 dyne per square centimeter. (After Fletcher and Munson.)

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[10] 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

[10] Fletcher and Munson, *Jour. Acous. Soc. Amer.*, Vol. 5, No. 2, p. 82, 1933.

3000 and 4000 cycles. The sensitivity of the ear decreases above and below this frequency. The response frequency characteristics of ears are useful 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. 11.20).

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).

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 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[11] showing the relation between loudness level, in phons, and the loudness, in loudness units, is shown in Fig. 12.12.

The loudness[11] 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. 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

[11] Fletcher, H., *Jour. Acous. Soc. Amer.*, Vol. 9, No. 4, p. 275, 1938.

propagated. Pitch is that subjective quality of a 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

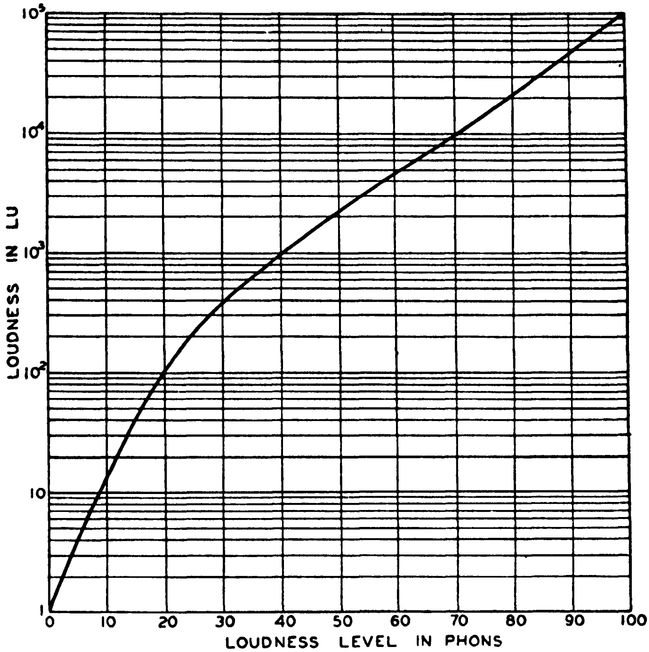

FIG. 12.12. Loudness versus loudness level. (After Fletcher.)

pitch as the intensity is increased. The change [12] in pitch with loudness is shown in Fig. 12.14.

12.9. Masking.[13, 14] — 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.

[12] Snow, W. B., *Jour. Acous. Soc. Amer.*, Vol. 8, No. 1, p. 14, 1936.
[13] Wegel and Lane, *Phys. Rev.*, Vol. 23, No. 2, p. 266, 1924.
[14] Fletcher and Munson, *Jour. Acous. Soc. Amer.*, Vol. 9, No. 1, p. 1, 1937

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

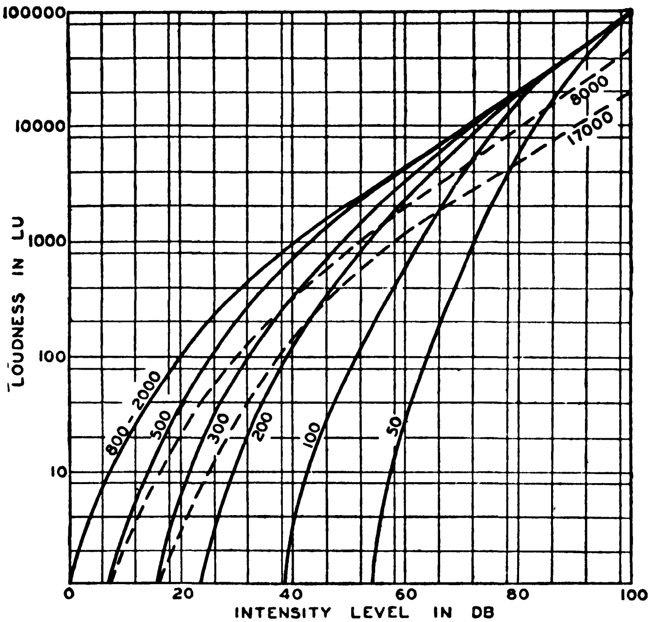

FIG. 12.13. The relation between the intensity level and the loudness of pure tones of the frequencies indicated. 0 db = 0.000204 dyne per square centimeter. (After Fletcher.)

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.

12.10. Nonlinearity of the Ear.[15] — 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.

[15] Stevens and Davis, "Hearing," John Wiley and Sons, New York, N. Y., 1939.

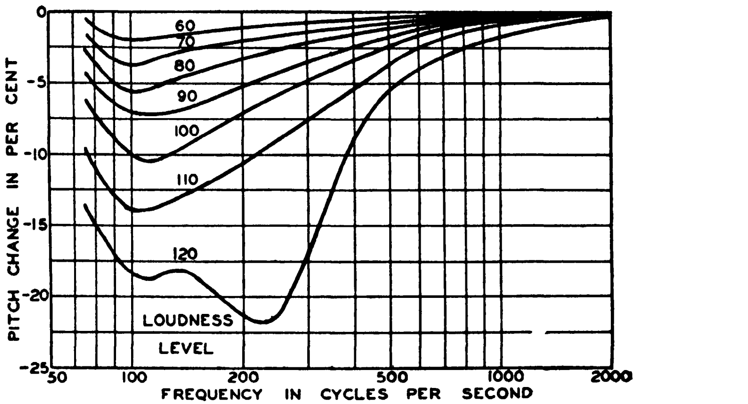

FIG. 12.14. Contours of constant loudness level. Curves show the amount by which the pitch of a pure tone of any frequency is shifted as the tone is raised in loudness level from 40 to the level of the contour. For example, a 100-cycle tone will be changed 10 per cent downward in pitch if raised from 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.)

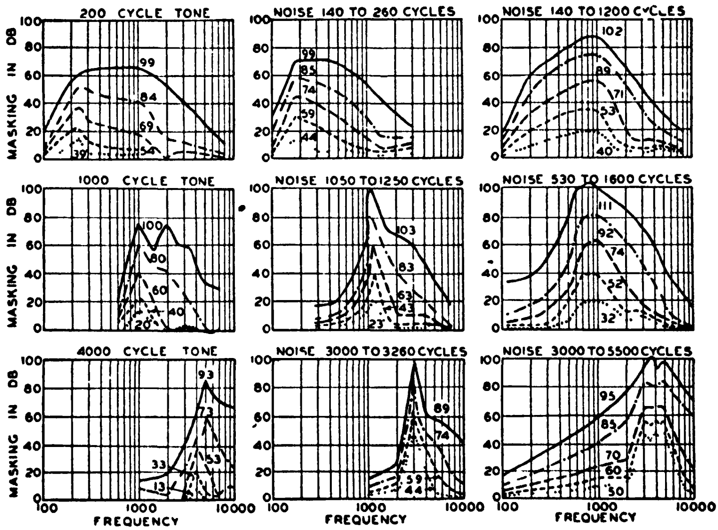

FIG. 12.15. Masking audiograms for single frequency tones, narrow bands of thermal noise and wide bands of thermal noise. The curves are labeled in db above the threshold. (After Fletcher and Munson.)

The levels above threshold of the fundamental at which the various harmonics first become detectable,[16, 17] 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.

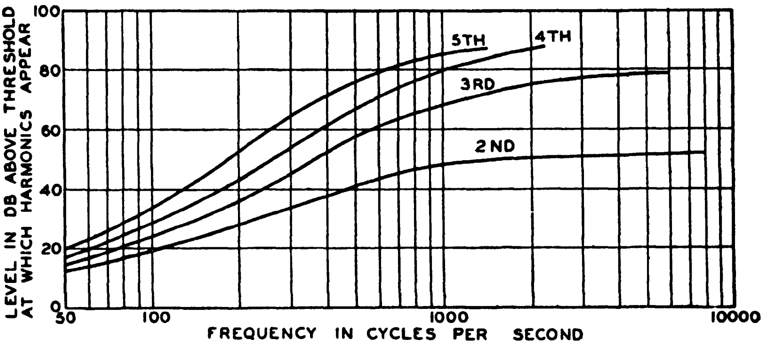

FIG. 12.16. The level above threshold at which harmonics are generated in the ear at various frequencies. (After Wegel and Lane.)

12.11. Effect of Phase Relations Among the Harmonics.[18, 19, 20] — The phase of a harmonic affects the threshold of perceptible distortion as well 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).[21, 22] — 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 loud speaker, Fig. 10.8. If the volume control is varied, the result is amplitude modulation. If the frequency control is varied, the result is frequency modu-

[16] Wegel and Lane, *Phys. Rev.*, Vol. 23, No. 2, p. 266, 1924.
[17] Fletcher, "Speech and Hearing," D. Van Nostrand Company, New York, N. Y., 1929.
[18] Chapin and Firestone, *Jour. Acous. Soc. Amer.*, Vol. 5, No. 3, p. 173, 1934.
[19] Lewis and Larsen, *Nat. Acad. Sci.*, Vol. 23, p. 415, 1937.
[20] Stevens and Davis, "Hearing," John Wiley and Sons, New York, N. Y., 1938.
[21] Seashore, C. E., "Psychology of Music," p. 33, McGraw-Hill Book Company, New York, N. Y., 1938.
[22] Stevens and Davis, "Hearing," John Wiley and Sons, New York, N. Y., 1938.

lation. If the position of the loud speaker 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.

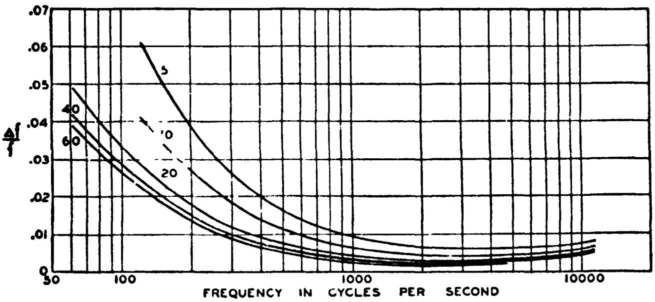

FIG. 12.17. The variation of $\Delta f/f$ with frequency for various sensation levels. Δf is the change in frequency. Sensation level is level above threshold. (After Shower and Biddulph.)

12.13. Minimum Perceptible Differences. — The minimum perceptible difference in frequency[23] 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.

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[24] which the ear can

[23] Shower and Biddulph, *Jour. Acous. Soc. Amer.*, Vol. 3, No. 2, Part 1, p. 275, 1931.

[24] Fletcher, "Speech and Hearing," D. Van Nostrand Company, New York, N. Y., 1929.

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 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.

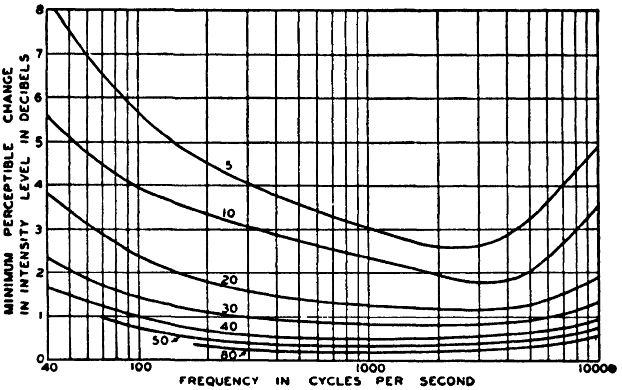

FIG. 12.18. The minimum perceptible change in intensity level of pure tones as a function of the frequency for various sensation levels. Sensation level is level above threshold. (After Fletcher, "Speech and Hearing," D. Van Nostrand Company.)

12.14. Timbre (Tone Quality).[25, 26] — 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

[25] Seashore, "Psychology of Music," McGraw-Hill Book Company, New York, N. Y., 1938.

[26] Fletcher, H., *Amer. Jour. Phys.*, Vol. 14, No. 4, p. 215, 1946.

more complex than that of loudness and pitch, because it is 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 interpetation of music.

12.16. Growth and Decay. — 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.

12.17. Auditory Localization.[27, 28, 29] — The human hearing mechanism 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. 11.31), 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.

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[27] Stewart, G. W., *Phys. Rev.*, Vol. 15, No. 5, p. 425, 1920.
[28] Stevens and Davis, "Hearing," John Wiley and Sons, New York, N. Y., 1938.
[29] Steinberg and Snow, *Bell Syst. Tech. Jour.*, Vol. 13, No. 2, p. 245, 1934.

The human hearing mechanism is also a directional collecting system. Using the system of Fig. 11.31, 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.

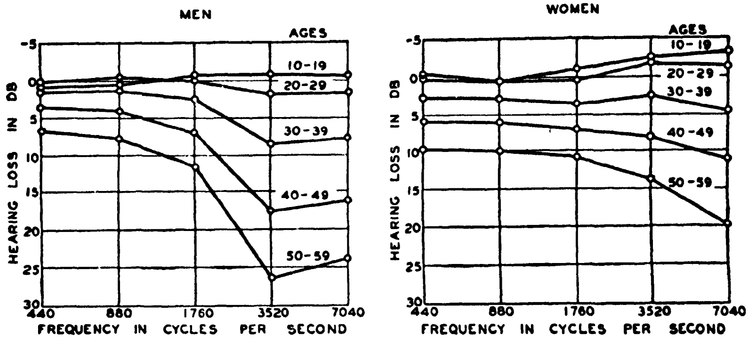

FIG. 12.19. Hearing loss frequency characteristic of men and women for different age groups. (After Steinberg, Montgomery and Gardner.)

12.18. Hearing Acuity in the United States Population.[30] — Hearing tests at New York and San Francisco World's Fairs in 1939 were conducted 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. 12.19. 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. 12.20. The upper curve is the threshold of feeling level. The lower 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

[30] Steinberg, Montgomery and Gardner, *Jour. Acous. Soc. Amer.*, Vol. 12, No. 2, p. 291, 1940.

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.

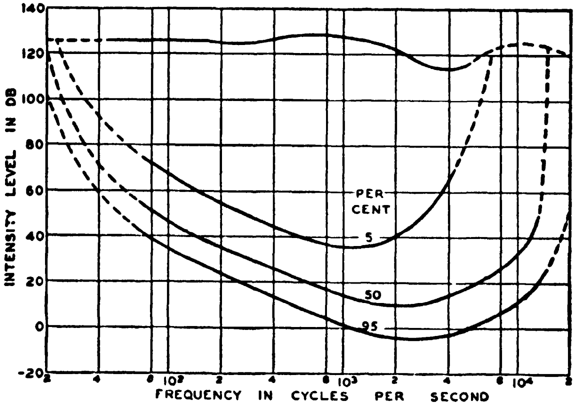

FIG. 12.20. Contours of hearing loss. (After Steinberg, Montgomery and Gardner.)

12.19. 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 [31] and volume ranges of speech and orchestral music are shown in Fig. 12.21. 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.

The frequency ranges[32] required for the reproduction of speech, musical instruments and noises without any noticeable frequency discrimination or distortion are shown in Fig. 12.22.

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[31] *Bell Labs. Record*, Vol. 12, No. 6, p. 314, 1934.
[32] Snow, W. B., *Jour. Acous. Soc. Amer.*, Vol. 3, No. 1, Part 1, p. 155, 1931.

12.20. The Effect of Frequency Discrimination upon the Articulation of Reproduced Speech. — The effect [33] of reducing the high- and low-frequency range upon speech articulation is shown in Fig. 12.23. It will be seen that a relatively high articulation can be obtained with a very narrow transmission band. However, the quality of the reproduced speech

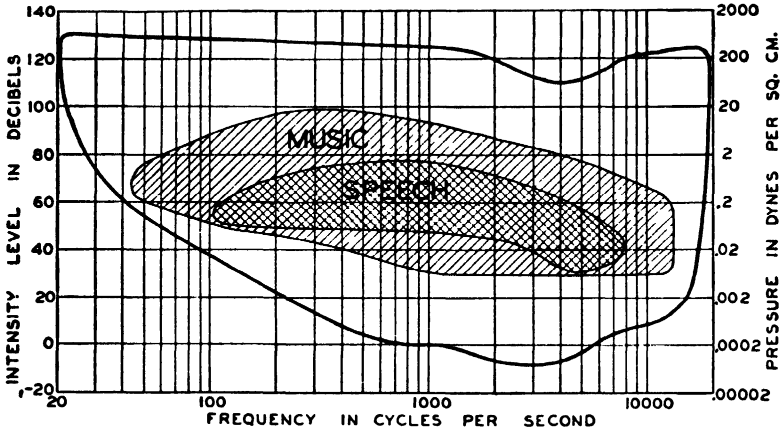

FIG. 12.21. Frequency and volume ranges of speech and music. The solid line depicts the boundaries of normal hearing, that is, the upper and lower limits of intensity and frequency. (From *Bell Laboratories Record*, June, 1934.)

is very much impaired by transmission over a narrow frequency band. A limited range may be actually superior to a wider band due to the introduction of additional noises and distortions in a wider band unless particular precautions are observed. In the case of speeches, plays and songs, a limited frequency range impairs the quality and artistic value of the reproduced sound.

12.21. The Effect of Frequency Discrimination upon the Quality of Reproduced Music. — The effect of the frequency range[34] upon the quality of reproduction of orchestral music is shown in Fig. 12.24. It will be seen that the frequency range required for no appreciable loss in quality is 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. 12.21, but will not cover the orchestral

[33] Fletcher, "Speech and Hearing," p. 280, D. Van Nostrand Company, New York, N. Y., 1929.

[34] Snow, W. B., *Jour. Acous. Soc. Amer.*, Vol. 3, No. 1, Part 1, p. 155, 1931.

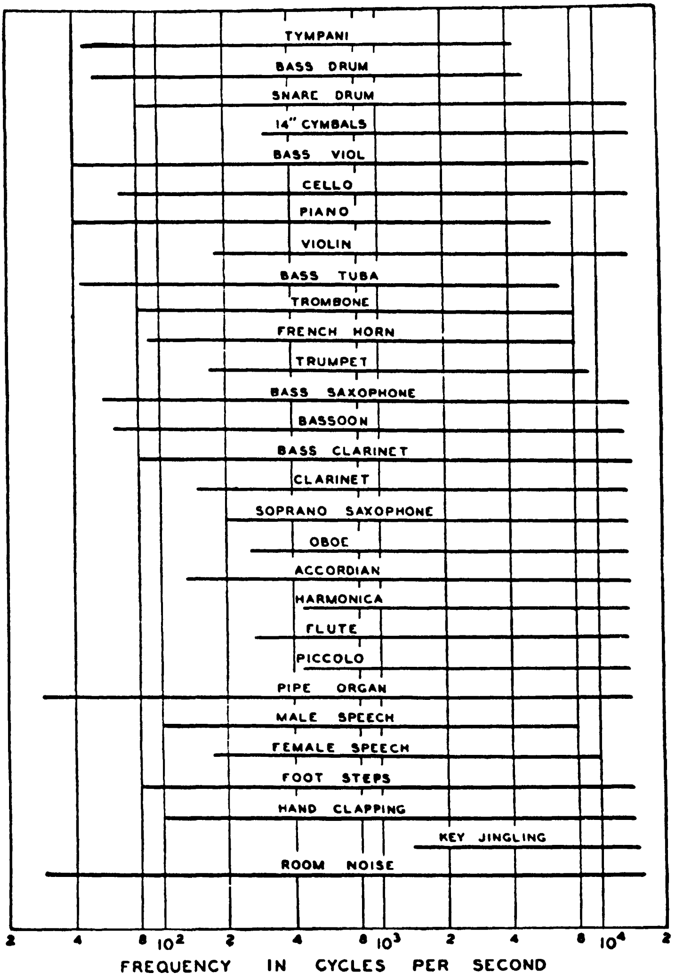

FIG. 12.22. The frequency ranges required for the reproduction of speech, musical instruments and noises without any noticeable distortion. (After Snow.)

music area. For the latter case some form of volume compression must be used.

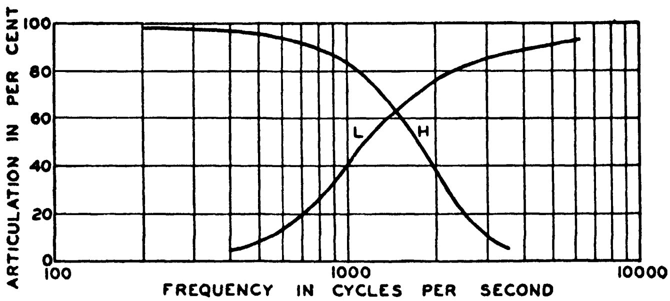

FIG. 12.23. The effect of the frequency range upon the articulation of speech. *H*. High-pass filter — all frequencies below the frequency given by the abscissa removed. *L*. low-pass filter — all frequencies above the frequency given by the abscissa removed. (After Fletcher, "Speech and Hearing," D. Van Nostrand Company.)

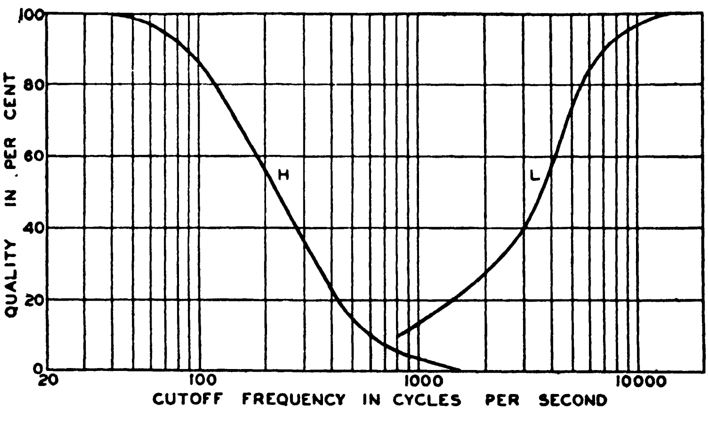

FIG. 12.24. The effect of the frequency range upon the quality of orchestra music. *H*. High-pass filter — all frequencies below the frequency given by the abscissa removed. *L*. Low-pass filter — all frequencies above the frequency given by the abscissa removed. (After Snow.)

12.22. Absolute Amplitudes and Spectra of Speech, Musical Instruments and Orchestras.[35, 36, 37, 38] — The average and peak outputs of speech

[35] Sivian, Dunn and White, *Jour. Acous. Soc. Amer.*, Vol. 2. No. 3, p. 330, 1931.
[36] Sivian, L. J., *Jour. Acous. Soc. Amer.*, Vol. 1, No. 2, Part 2, p. 1, 1930.
[37] Hathaway, J. L., *Electronics*, Vol. 12, No. 11, p. 29, 1939.
[38] Drew and Kellogg, *Jour. Acous. Soc. Amer.*, Vol. 12, No. 1, p. 95, 1940.

and musical instruments are of importance in the design of all types of reproducing equipment. For example, the average power output in-

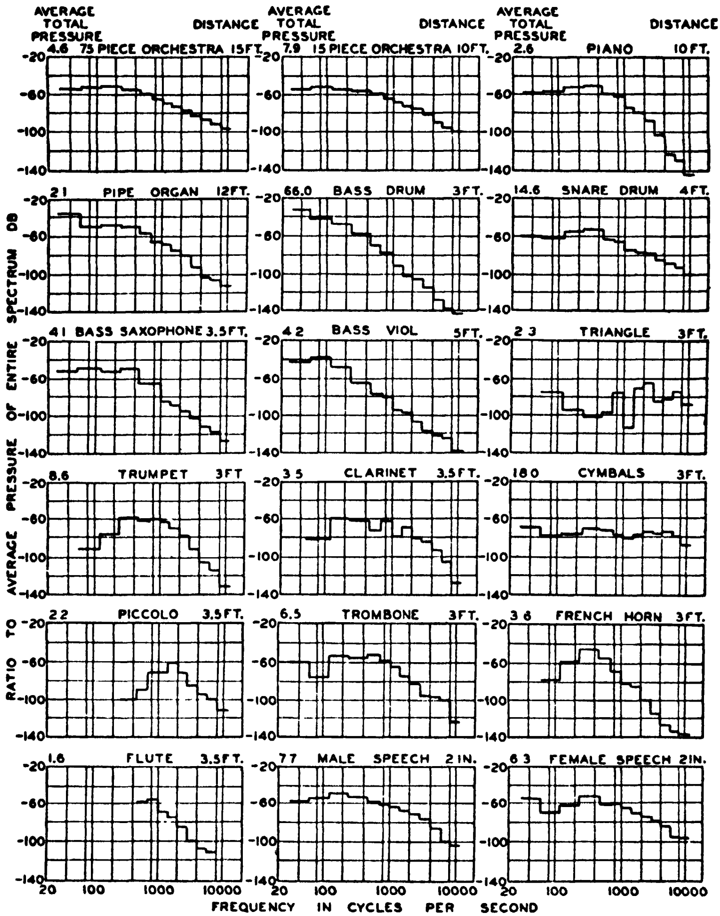

FIG. 12.25. Ratio of the average pressure per cycle to the average total pressure of the entire spectrum for speech, various musical instruments and orchestras. The distance and average total pressure, in dynes per square centimeter, are shown above each graph. (After Sivian, Dunn and White.)

volves 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

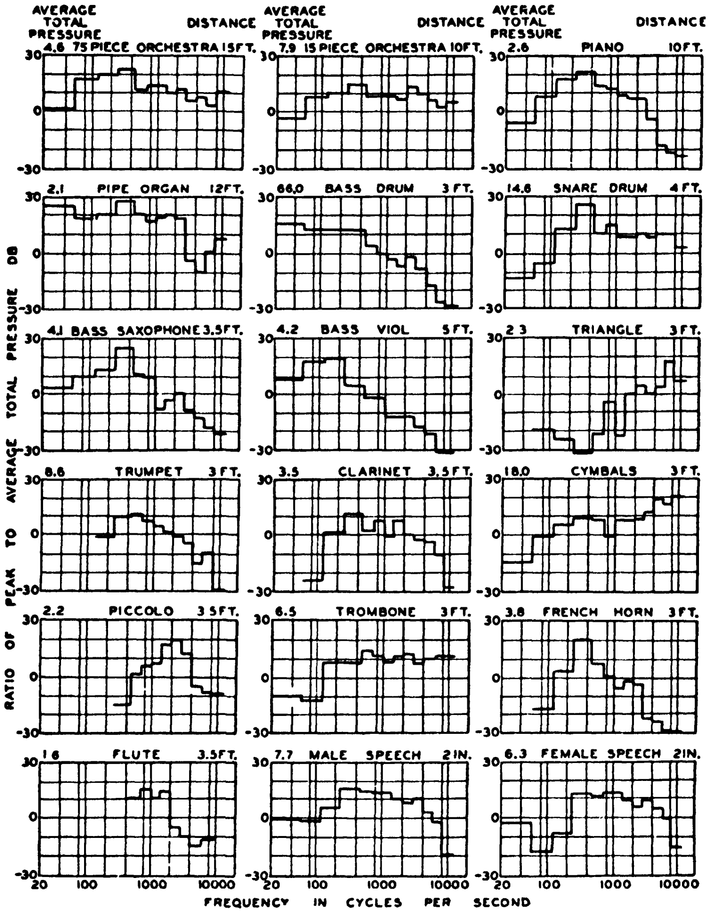

FIG. 12.26. Ratio of the peak pressure to the average total pressure of the entire spectrum for speech, various musical instruments and orchestras. The distance and average total pressure, in dynes per square centimeter, are shown above each graph. (After Sivian, Dunn and White.)

and orchestras is shown in Fig. 12.25. 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. 12.26. The peak or total power output can be computed from the pressure and the distances following the procedure as outlined in Sec. 10.3*D*.

The peak sound pressures[39] in speech for various frequency bands at a distance of 30 centimeters are shown in Fig. 12.27. The percentages on the graphs are those of intervals having peak pressures greater than

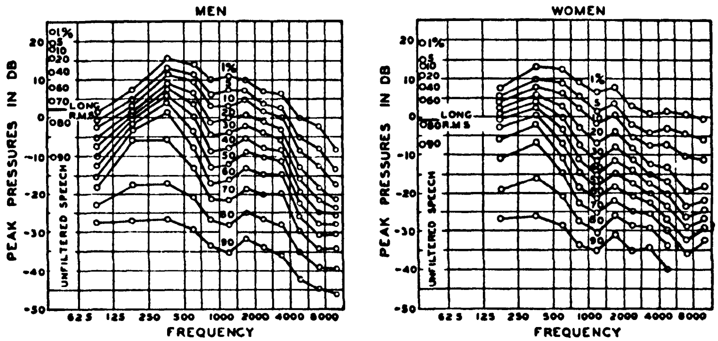

Fig. 12.27. Peak pressures in one-eighth-second intervals of conversational speech, at 30 centimeters from the mouth; composite from the voices of 6 men and 5 women. Measurements were made in the bands indicated by divisions of the frequency scale, and the percentages are those of intervals having peak pressures greater than the indicated ordinates. At the left in the graphs, peak measurements on speech as a whole are given. The r.m.s. pressure over a long-time interval is also given. 0 db = 1 dyne per square centimeter. (After Dunn and White.)

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 r.m.s. sound pressures in speech for various frequency bands at a distance of 30 centimeters are shown in Fig. 12.28.

12.23. Noise in Reproducng 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 [40] in conductors, and the small shot effect [41] and fluctuating noise in vacuum tubes are atmospherics and man-made interference. The

[39] Dunn and White, *Jour. Acous. Soc. Amer.*, Vol. 11, No. 3, p. 278, 1940.
[40] Johnson, J. B., *Phys. Rev.*, Vol. 32, No. 1, p. 97, 1928.
[41] Schottky, W., *Ann. d. Phys.*, Vol. 57, p. 541, 1918.

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.

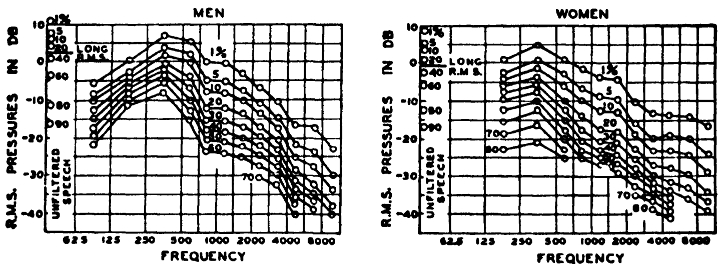

FIG. 12.28. R.m.s. pressures in one-eighth-second intervals of conversational speech, at 30 centimeters from the mouth; composite of 6 men and 5 women. Measurements were made in the bands indicated by the divisions of the frequency scale, and the percentages are those of intervals hvaing peak pressures greater than the indicated ordinates. At the left in the graphs, r.m.s. measurements on speech as a whole are given. The r m.s. pressure over a longtime interval is also given. 0 db = 1 dyne per square centimeter. (After Dunn and White.)

Hum [42] 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 nat-

[42] Standards on Radio Receivers, Institute of Radio Engineers, 1938.

ural grain of the film. In the case of film the noise is proportional to the 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.[43] 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. 8.13).

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.

12.24. Room Noise and the Reproduction of Sound.[44] — The hearing curves of Figs. 12.11 and 12.20 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[45] may be obtained from the measurements of room noise. In general, the noise spectrum is the same for all types of rooms. From Fig. 11.13, 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. 12.29. The ordinates, depicting the spectrum level, are given by

$$B = 10 \log \frac{I}{WI_0} \qquad 12.1$$

[43] Sivian and White, *Jour. Acous. Soc. Amer.*, Vol. 4, No. 4, p. 288, 1933.
[44] Fletcher, H., *Proc. I. R. E.*, Vol. 30, No. 6, p. 266, 1942.
[45] Hoth, D. F., *Jour. Acous. Soc. Amer.*, Vol. 12, No. 4, p. 499, 1941.

where I = sound intensity in a frequency band width W, and
I_0 = zero reference level of 10^{-16} watts per square centimeter.

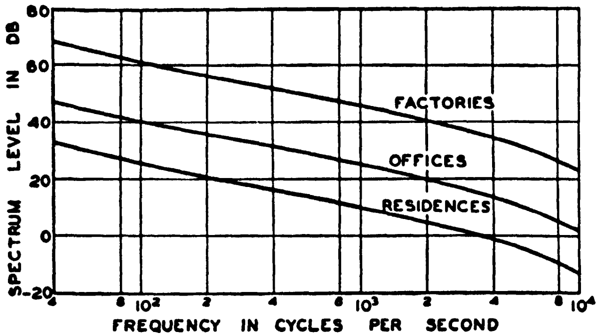

Fig. 12.29. Average noise spectrum for residences, offices and factories. 0 db = 0.000204 dyne per square centimeter. (After Fletcher and Hoth.)

In the case of thermal noise or wide-band random noise it is possible to obtain the masking [46] from the spectrum level. The masking contours

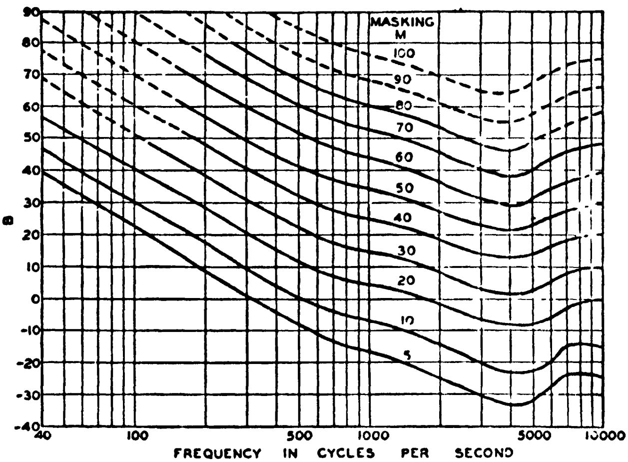

Fig. 12.30. Masking contours for wide-band noise. M is the difference between the threshold of hearing in the presence of noise and a quiet place. B is the spectrum level. (After Fletcher and Munson.)

for wide-band sounds are shown in Fig. 12.30. These masking contours have been drawn after a careful consideration of all data on the subject

[46] Fletcher and Munson, *Jour. Acous. Soc. Amer.*, Vol. 9, No. 1, p. 1, 1937.

of masking. 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. 12.29, the threshold curve, Fig. 12.11, and the masking contours of Fig. 12.30. 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. 12.31. The curves for an average business office and an average factory are also shown in Fig. 12.31.

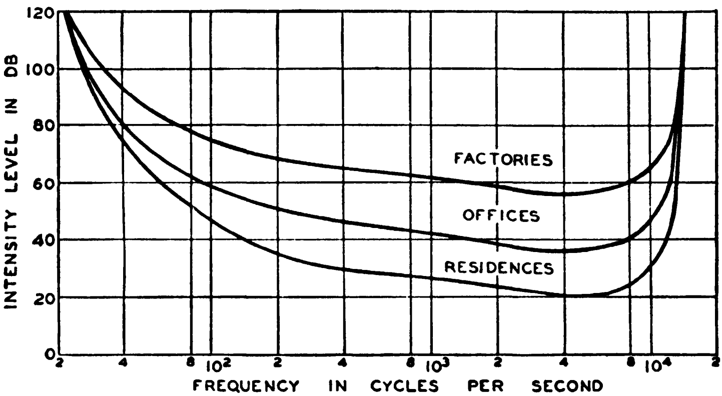

FIG. 12.31. Hearing limits for pure tones. The above characteristics are for a typical listener in typical residence, office and factory noise. (After Fletcher.)

From the masking curve of Fig. 12.31 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.

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.

12.25. 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

$$y = ap + bp^2 \qquad 12.2$$

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

$$p = p_1 + p_2 = p_0 \cos \omega_1 t + p_0 \cos \omega_2 t \qquad 12.3$$

Substituting equation 12.3 in 12.2 the resulting equation may be put in the form

$$\begin{aligned} y = {} & ap_0 \cos \omega_1 t + ap_0 \cos \omega_2 t + \tfrac{1}{2}bp_0^2 \cos 2\omega_1 t + \tfrac{1}{2}bp_0^2 \cos 2\omega_2 t \\ & + bp_0^2 \cos (\omega_1 + \omega_2)t + bp_0^2 \cos (\omega_1 - \omega_2)t + bp_0^2 \end{aligned} \qquad 12.4$$

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.

12.26. Effect of Nonlinear Distortion upon the Quality of Reproduced Speech and Music. — In an ideal reproducing system the elements are invarient with respect to the time. However, in practical systems the elements are nonlinear. These elements introduce nonlinear distortion. 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 higher 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 on the reproduction of speech and music has been determined through the system shown in

Fig. 12.32. The overall response frequency characteristic of the microphone, amplifier and loud speaker was uniform to within 2 db from 45 to

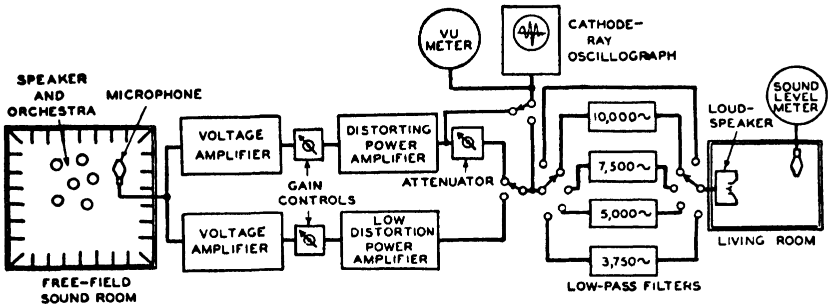

FIG. 12.32. Schematic arrangement of the apparatus for a subjective determination of the relation between nonlinear distortion and frequency range in reproduced sound.

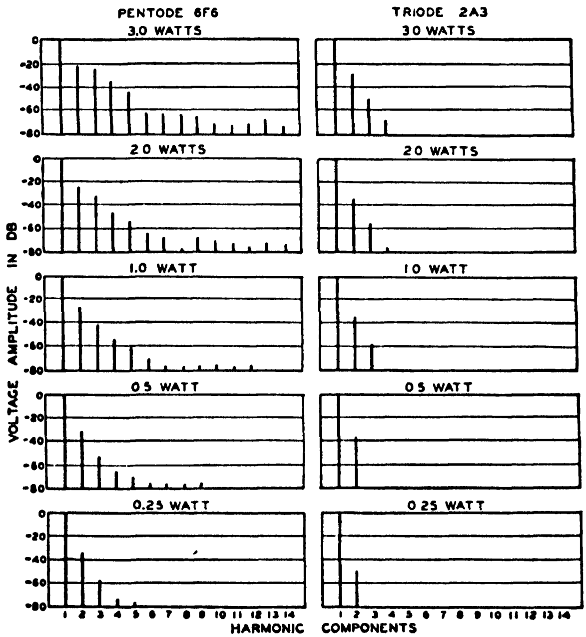

FIG. 12.33. Distortion characteristics of single-ended pentode and triode amplifiers.

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 overall 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

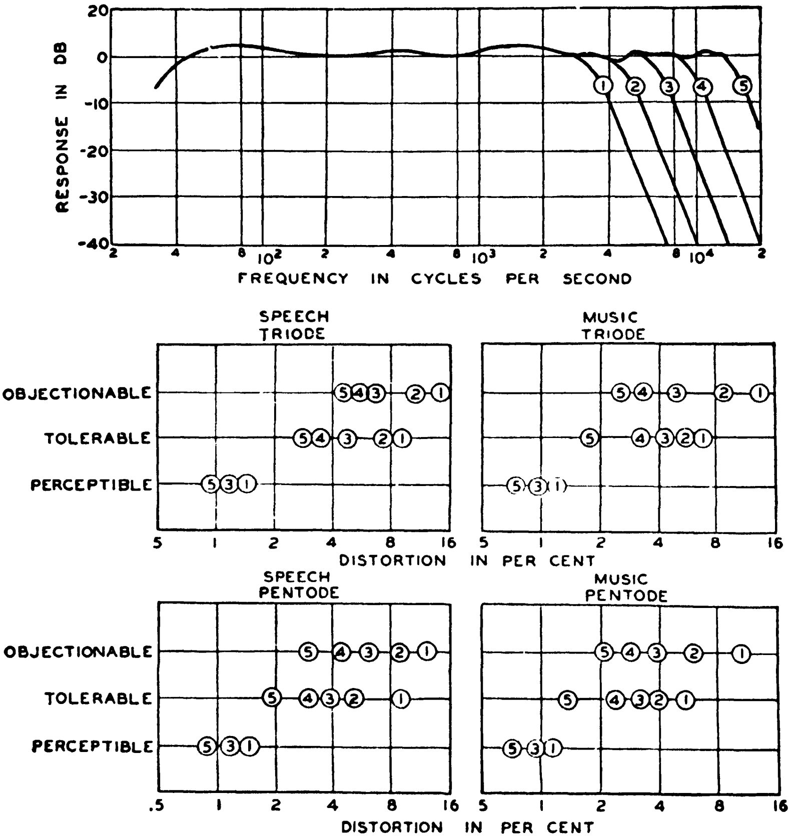

FIG. 12.34. Experimental results of subjective tests of reproduced speech and music depicting objectionable, tolerable and perceptible nonlinear distortion for various high-frequency cutoffs. The numbers in the distortion data points correspond to the numbers which label the response frequency characteristics. The distortion, in per cent, is the ratio of the total r.m s. of the second, third, fourth, etc., components of distortion to the r.m.s. of the fundamental, multiplied by one hundred.

Sec. 10.3*A*4. 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. 12.33.

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 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. 12.34. 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.

12.27. Frequency Ranges of Sound Reproducing Systems. — The frequency ranges of the most common sound reproducing systems are shown in Fig. 12.35. 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. 12.35.

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 receivers refer to commercial models of radio receivers sold during the last decade. A small number of high-quality radio receivers have been built with frequency ranges varying from the commercial models up to the high-quality radio receiver. Most broadcast radio 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

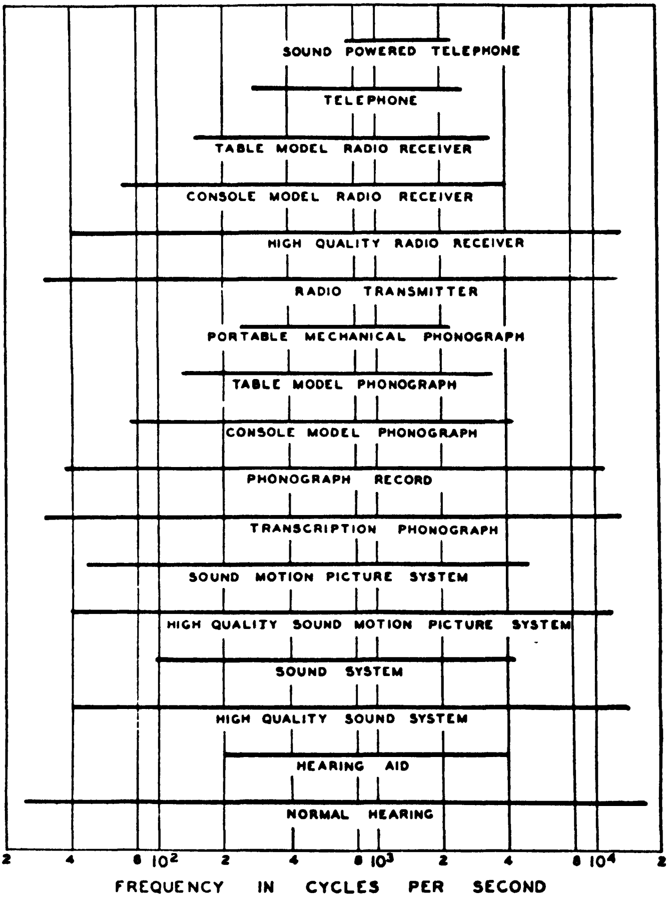

FIG. 12.35. The frequency ranges of sound reproducing systems.

the average of commercial phonograph records pressed at the present time. The frequency range of the transcription phonograph is that of the systems used in broadcast stations and high-quality sound systems.

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.

A small number of high-quality sound motion picture systems are in use with a frequency range varying from the commercial system up to the high-quality system shown in Fig. 12.35.

The average frequency range of commerical sound systems used in public address, announce, paging, reinforcing, intercommunicating and other applications of sound systems, is shown in Fig. 12.35. A small number of 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. 12.35 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 the frequency ranges of the reproducing systems in use today shows that the average upper limit is about 4000 to 5000 cycles. On the other hand, the frequency range of broadcast stations, sound motion picture records, and phonograph records is much greater. In other words, the potential frequency ranges of radio, phonographs, sound motion picture reproducers and sound systems is much greater than those shown in Fig. 12.35.

Referring to Sec. 12.19 and Figs. 12.21 and 12.22 it will be seen that the frequency range of speech, music and noises extends far beyond 5000 cycles. To reproduce speech and music without loss of naturalism requires a frequency range of 40 to 15,000 cycles. This means that existing systems introduce considerable frequency discrimination. Attempts have been made to increase the frequency range but without public acceptance. This substantiates the results given in Sec. 12.28. The reason for these results is, without doubt, due to the distortions listed in Sec. 12.29, because the direct listening tests of Sec. 12.29 indicate that the listener prefers a wide frequency range to a restricted frequency range when listening to speech and music directly without being reproduced.

12.28. Frequency Range Preference for Reproduced Speech and Music.[47] — 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.

[47] Chinn and Eisenberg, *Proc. Inst. Rad. Eng.*, Vol. 33, No. 9, p. 571, 1945.

As contrasted to the other data presented, the purpose of this investigation was the determination of the frequency range of reproduced speech

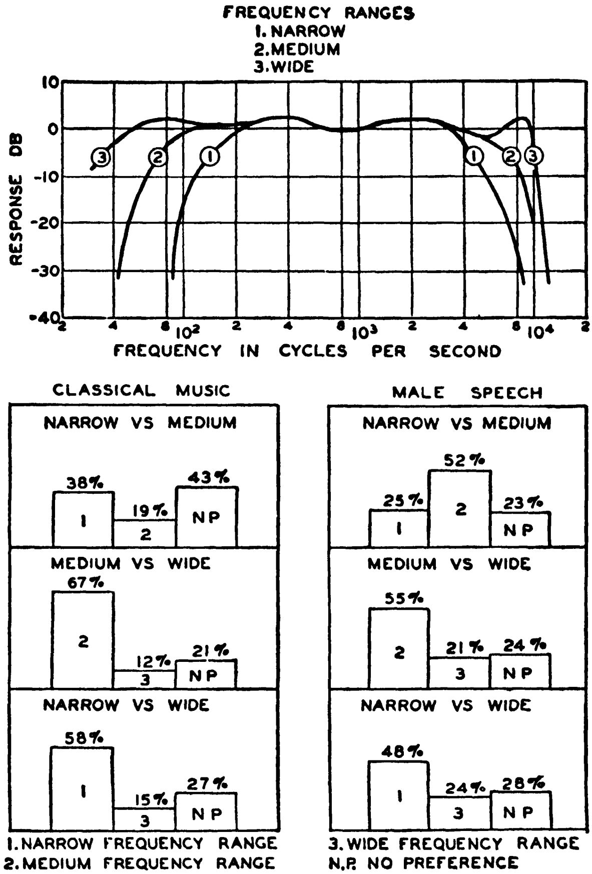

FIG. 12.36. The frequency range preferences of a cross section of listeners for classical music and male speech. The graph depicts the response frequency characteristics for the following: 1 = narrow frequency range. 2 = medium frequency range. 3 = wide frequency range. The block diagrams depict the 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.)

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. 12.36 and were designated as wide, medium and narrow frequency ranges.

The results of the tests are shown in Fig. 12.36. 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 radio receivers, phonographs, sound motion picture systems and sound systems.

12.29. Frequency Range Preference for Live Speech and Music. — The frequency range preference for reproduced speech and music was considered in the preceding section. These tests indicate that listeners prefer a restricted frequency range in monaural reproduced speech and music. There are three possible reasons for the results of these tests, as follows: *A*. The average listener, after years of listening to the radio and the phonograph, has become conditioned to a restricted frequency range and feels that this is the natural state of affairs. *B*. Musical instruments are not properly designed and would be more pleasing and acceptable if the production of fundamentals and overtones in the high-frequency range were suppressed. *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 loud-speaker 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-

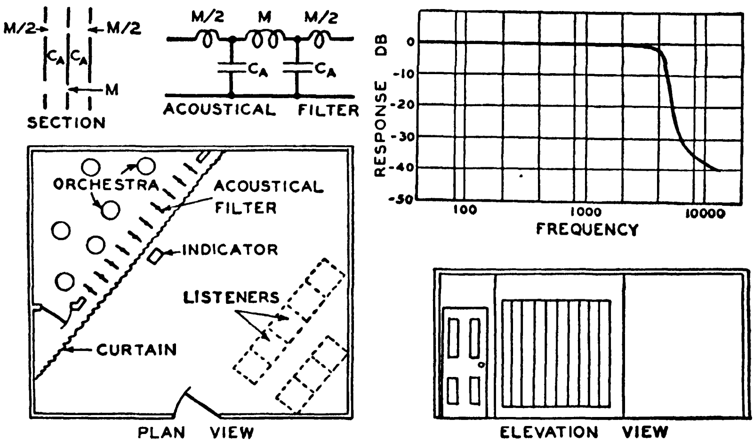

FIG. 12.37. Plan and elevation views of the schematic arrangement of the apparatus for direct frequency preference testing of speech and music. A sectional view, acoustical network and response frequency characteristic of the acoustical filter used in the tests are also depicted.

acoustic test of frequency range preference was made. The general arrangement of the test is shown in Fig. 12.37. 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. 12.37. The response frequency characteristic of the acoustical filter shown in Fig. 12.37 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. 12.37. 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. 12.28. 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. 12.26, 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.

Up to the present time the all-acoustic frequency preference tests have been rather limited in the type of musical material used. A large number of different tests with different cutoff frequencies and musical material must be used before the tests can be considered to be absolutely conclusive.

12.30. Musical Scale. — 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. 12.38.

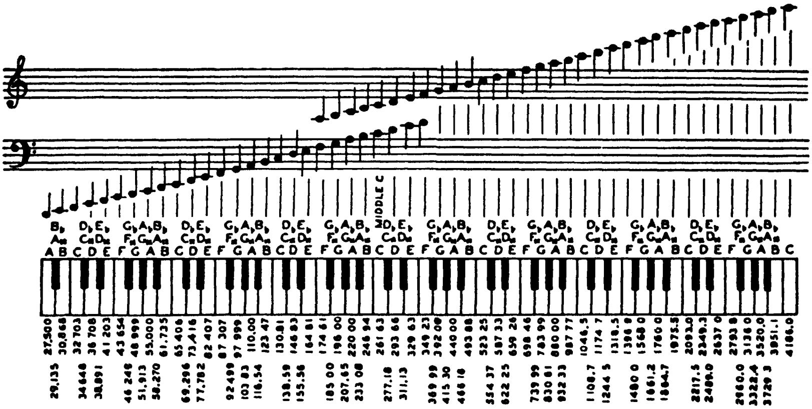

FIG. 12.38. The piano keyboard and the frequencies of the equally tempered scale as used in music according to the American Standard Pitch.

The relative frequencies of the natural or diatonic or just scale and tempered scale are shown below.

	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	1.335	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.

12.31. 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

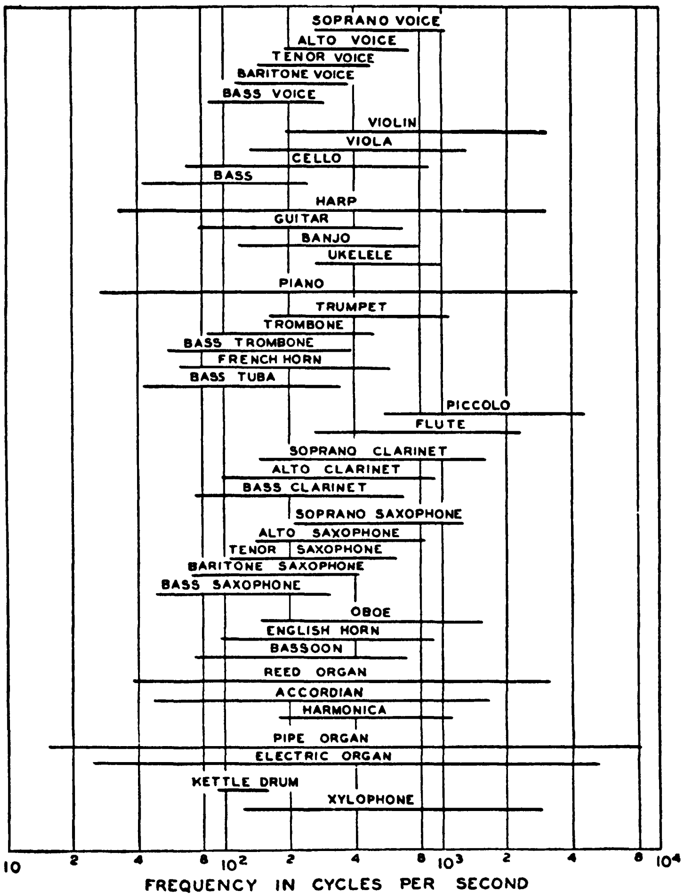

FIG. 12.39. Frequency ranges of the fundamental frequencies of voices and various musical instruments.

therefore be the same as the output of an oscillator and loud-speaker combination. 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. 12.39. 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. 12.39 with the entire frequency spectrum of musical instruments of Fig. 12.22, it will be seen that the overtones of the instruments extend the frequency ranges by a factor of two or more octaves.

CHAPTER XIII

UNDERWATER SOUND

13.1. 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.

13.2. 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

$$I = \frac{P\epsilon^{-\alpha r}}{4\pi r^2} \tag{13.1}$$

where I = intensity, in ergs per square centimeter,

P = power output of the sound source, in ergs per second,

r = distance from the sound source, in centimeters,

$\alpha = \dfrac{2}{3}\dfrac{\mu k^2}{\rho c}$

μ = viscosity, μ = .0114 for water,

$k = \dfrac{2\pi}{\lambda}$,

[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 loud speaker may produce is limited by cavitation at the diaphragm. In general, cavitation occurs at the diaphragm when the pressure, in the rarefraction cycles, is near zero. For a loud speaker operating near the surface of the water the pressure is that due to the atmosphere. Under these conditions the rarefraction pressure is 10^6 dynes per square centimeter.

The output of a subaqueous loud speaker in water is given by

$$P = r_{AW}\,\dot{x}^2 = \frac{p^2}{r_{AW}} \qquad 13.2$$

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

$$r_{AW} = \rho c = 144{,}000 \qquad 13.3$$

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 loud speaker is lowered to a depth of 33 feet the output will be quadrupled.

13.3. Direct Radiator Dynamic Subaqueous[2] Loud Speaker. — The direct radiator dynamic subaqueous loud speaker is a loud speaker designed to operate under water, in which, a diaphragm is driven by a voice coil located in a magnetic field (Fig. 13.1).

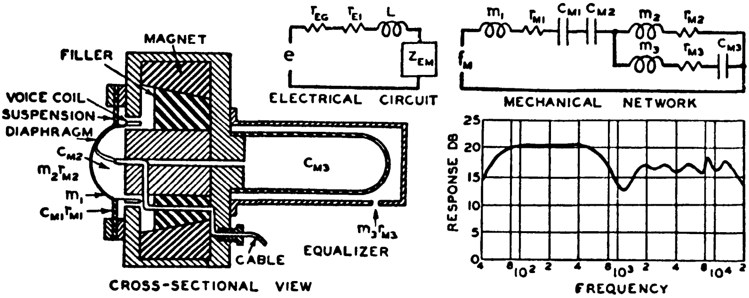

FIG. 13.1. Cross-sectional view, electrical circuit, mechanical network and pressure response frequency characteristic of a direct radiator dynamic subaqueous loud speaker. In the electrical circuit: z_{EM} = the motional electrical impedance. L and r_{E1} = the inductance and electrical resistance of 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.

The theory of the direct radiator dynamic subaqueous loud speaker, except for the high impedance of the medium, is the same as that of the air

[2] In this book the term subaqueous loud speaker will be used to designate a system for converting electrical variations into the corresponding sound vibrations in water and the term subaqueous microphone will be used to designate a system for converting sound vibrations in the water into the corresponding electrical variations. In underwater sound the terms oscillator, transmitter and projector have also been used to designate a subaqueous loud speaker. The terms receiver and hydrophone have also been used to designate a subaqueous microphone. There is a possibility of ambiguity in the use of these terms. For example, when the term projector alone is used it usually means a still or motion picture projecting system. The same confusion exists in the terms receiver, oscillator and transmitter. Therefore, it is necessary to add the adjective subaqueous in the use of the terms to avoid confusion. Under these conditions, it seems logical to use the terms subaqueous microphone and subaqueous loud speaker. As a matter of fact, these terms have been used extensively during the past few years.

direct radiator dynamic loud speaker 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 loud speaker.

Since the acoustical impedance of water is about 3400 times that of air the diaphragm of the subaqueous, direct radiator loud speaker is relatively small compared to that of an air direct loud speaker employing a comparable driving system. In the case of the subaqueous loud speaker 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 loud speaker the response is independent of the frequency in the frequency region where the acoustical radiation resistance is proportional 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 loud speaker, 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 conpensating 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 subaqueous loud speaker shown in Fig. 13.1 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 loud speaker is shown in Fig. 13.1. 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 loud speaker with a comparable driving system.

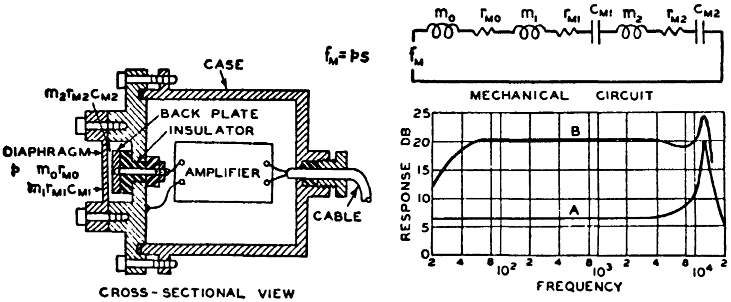

FIG. 13.2. Cross-sectional view, mechanical circuit and voltage response frequency characteristic of a subaquaeous condenser microphone. In the mechanical circuit: m_0 and r_{M0} = the mass and mechanical resistance of the water load upon the diaphragm. m_1, r_{M1} and C_{M1} = the mass, mechanical resistance 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 microphone. B = the output voltage response characteristic of the combination of a two-stage amplifier and condenser microphone for an output impedance of 250 ohms. 0 db = 1 microvolt per dyne per square centimeter.

13.4. Subaqueous Condenser Microphone. — A subaqueous condenser microphone is a microphone designed to operate under water and which depends for its operation on variations in electrical capacitance. A typical subaqueous condenser microphone, shown in Fig. 13.2, 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. 13.2. The clamped plate has been considered in Sec. 3.5*A*. 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. 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 subaqueous condenser microphone is the same as that of the air condenser microphone shown in Fig. 8.4.

The open-circuit voltage response frequency characteristic of the condenser microphone and the output voltage characteristic of the condenser microphone and amplifier combination is shown in Fig. 13.2. It will be seen that the response is uniform below the fundamental resonant frequency.

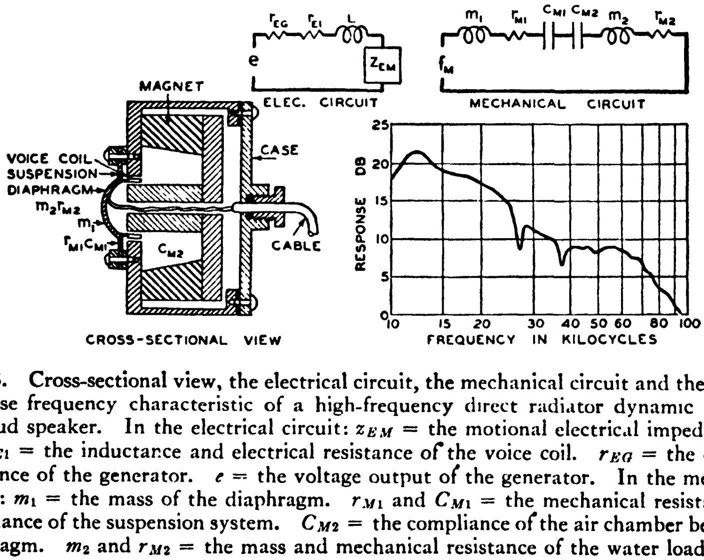

FIG. 13.3. Cross-sectional view, the electrical circuit, the mechanical circuit and the pressure response frequency characteristic of a high-frequency direct radiator dynamic subaqueous loud speaker. In the electrical circuit: z_{EM} = the motional electrical impedance. L and r_{E1} = the inductance 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.

13.5. High-Frequency Direct Radiator Dynamic Subaqueous Loud Speaker and Microphone. — The response of the subaqueous dynamic loud speaker described in Sec. 13.3 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, subaqueous loud speaker is shown in Fig. 13.3. The diaphragm in a typical unit is about three-fourths inch in diameter. The response frequency characteristic is shown in Fig. 13.3. Relatively good output is obtained from 10 to 80 kilocycles.

The same unit may be used as a microphone over the same frequency range.

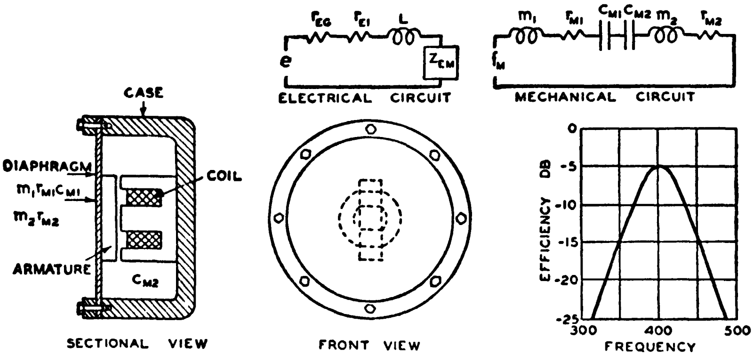

FIG. 13.4. Cross-sectional and front views, electrical circuit, mechanical circuit and efficiency frequency characteristic of a magnetic subaqueous loud speaker. In the electrical circuit: z_{EM} = the motional electrical impedance. L and r_{E1} = the inductance and electrical resistance of the coil. 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.

13.6. Magnetic Subaqueous Loud Speaker. — The magnetic subaqueous loud speaker is a loud speaker designed to operate under water consisting of a resonant diaphragm driven by forces resulting from magnetic reactions. The magnetic, subaqueous loud speaker shown in Fig. 13.4 is the unpolarized armature type. The force [3] on the armature, in dynes, is given by

$$f_M = \frac{C^2 i^2}{4\pi A a^2} \tag{13.4}$$

where i = current in the coil, in abamperes,
$C = 2\pi n A$,

[3] Olson, "Dynamical Analogies," p. 128, D. Van Nostrand Company, New York, N. Y., 1943.

A = area of the center pole, in square centimeters,
n = number of turns in the coil, and
a = spacing, in centimeters.

If the current in the coil is sinusoidal, then the expression for the current can be written

$$i = i_{\max} \sin \omega t \tag{13.5}$$

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 13.5 for the current in 13.4, the force on the armature, in dynes, is

$$f_M = \frac{C^2}{4\pi A a^2} i^2_{\max} \sin^2 \omega t \tag{13.6}$$

$$= \frac{C^2}{4\pi A a^2} i^2_{\max} \left(\tfrac{1}{2} - \tfrac{1}{2} \cos 2\,\omega t\right) \tag{13.7}$$

Equation 13.7 shows that there is a steady force and an alternating driving force of twice the frequency of the impressed current.

The performance of the system can be determined from a consideration of the mechanical circuit of Fig. 13.4. The effective mass of a clamped plate is one third the total mass of the plate (see Sec. 3.5*A*). 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 subaqueous loud speaker is shown in Fig. 13.4.

The motional electrical impedance [4] of the system is

$$z_{EM} = \frac{2\pi^2 n^4 A^2 i^2}{a^2 z_M} \tag{13.8}$$

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.

[4] Olson, "Dynamical Analogies," p. 130, D. Van Nostrand Company, New York, N. Y., 1943.

At resonance the electrical impedance, z_{EM}, becomes an electrical resistance.

The efficiency, in per cent, is

$$\mu = \frac{r_{ED}}{r_{EM} + r_{ED}} \times 100 \qquad 13.9$$

where r_{ED} = damped electrical resistance of the coil, in abohms, and
r_{EM} = motional electrical resistance, in abohms.

From equations 13.8 and 13.9 it will be seen that the efficiency increases with the power input. This characteristic is typical of unpolarized driving systems.

13.7. Magnetostriction Subaqueous Loud Speaker. — A magnetostriction subaqueous loud speaker is a loud speaker designed to operate under water, 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 subaqueous loud speaker is shown in Fig. 13.5. 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 subaqueous loud speaker are shown in Fig. 13.5.

The driving force,[5] in dynes, generated in the rod are given by

$$f_M = \frac{4\pi NiK}{R} \sin \omega t \qquad 13.10$$

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[6] of the system is given by

$$z_{EM} = \frac{16\pi^2 N^2 K^2}{R^2 z_M} \qquad 13.11$$

[5] Olson, " Dynamical Analogies," p. 143, D. Van Nostrand Company, New York, N. Y., 1943.

[6] Olson, " Dynamical Analogies," p. 145, D. Van Nostrand Company, New York, N. Y., 1943.

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

$$z_{EN} = z_{E1} + z_{EM} \tag{13.12}$$

where z_{E1} = damped electrical impedance of the coil, in abohms.

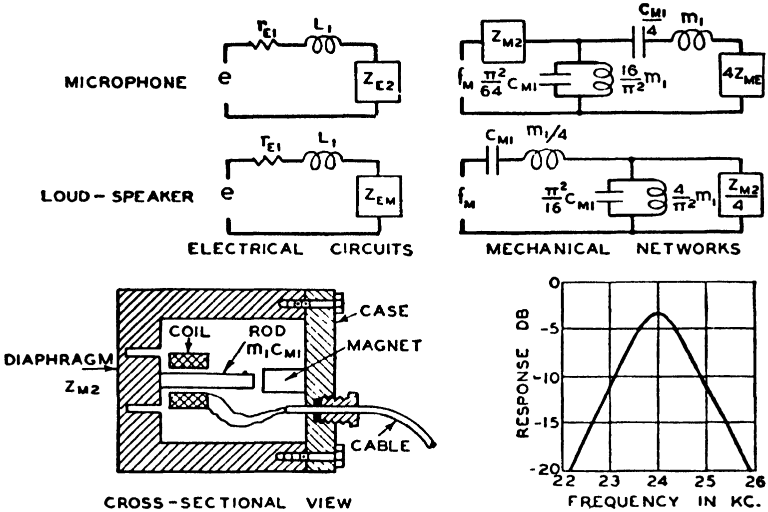

FIG. 13.5. Cross-sectional view, electrical circuits, mechanical networks and response frequency characteristics of a magnetostriction subaqueous loud speaker or microphone. In the electrical circuit: z_{E2} = the electrical load when used as a microphone. z_{EM} = the electrical motional impedance when used as a loud speaker. L_1 and r_{E1} = the inductance and electrical resistance of the coil. e = the open-circuit voltage output when used as a microphone or the voltage input when used as a loud speaker. 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.

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. 13.5.

The lumped constant representation of the system as depicted in the mechanical network of Fig. 13.5 is valid in the region at and near the resonant frequency of the rod. The mass,[7] m_1, in Fig. 13.5 is given by

$$m_1 = \frac{\rho l A}{2} \qquad 13.13$$

where ρ = 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,[8] C_{M1}, in Fig. 13.5 is given by

$$C_{M1} = \frac{8l}{\pi^2 \mathcal{Q} A} \qquad 13.14$$

where A = cross-sectional area of the rod, in square centimeters,
l = length of the rod, in centimeters, and
$\mathcal{Q}$ = 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

$$\mu = \frac{r_{EM}}{r_{ED} + r_{EM}} \times 100 \qquad 13.15$$

where r_{ED} = damped electrical resistance, in abohms, and
r_{EM} = motional electrical resistance, in abohms.

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. 13.5.

Magnetostriction loud speakers of the type shown in Fig. 13.5 are suitable for frequency band response in the frequency range from 10 to 50 kilocycles.

A single-rod system is shown in Fig. 13.5. Loud speakers 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.

13.8. Magnetostriction Subaqueous Microphone. — A magnetostriction subaqueous microphone is a microphone designed to operate under water, in which a voltage is generated in a coil surrounding a rod having magnetostrictive properties. A cross-sectional view of a magnetostrictive subaqueous microphone is shown in Fig. 13.5. The mechanical network of

[7] Olson, " Dynamical Analogies," p. 145, D. Van Nostrand Company, New York, N. Y., 1943.

[8] Olson, " Dynamical Analogies," p. 146, D. Van Nostrand Company, New York, N. Y., 1943.

the vibrating system and electrical circuit is also shown in Fig. 13.5. The mass m_1 and the compliance C_{M1} are given by equations 13.13 and 13.14.

The driving force, f_M, in dynes, is

$$f_M = pS \qquad 13.16$$

where p = sound pressure, in dynes per square centimeter, and
S = area of the diaphragm, in square centimeters.

The mechanical impedance[9] due to the electrical system is

$$z_{ME} = \frac{16\pi^2 N^2 K^2}{z_E R^2} \qquad 13.17$$

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.

The voltage,[10] in abvolts, developed in the coil due to deformation of the rod is

$$e = \frac{4\pi NK}{R}\dot{x} \qquad 13.18$$

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 13.18. 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 loud speaker shown in Fig. 13.5.

13.9. Quartz Crystal Subaqueous Loud Speaker. — A quartz crystal subaqueous loud speaker is a loud speaker designed to operate under

[9] Olson, " Dynamical Analogies," p.163, D. Van Nostrand Company, New York, N. Y., 1943.

[10] Olson, " Dynamical Analogies," p. 162, D. Van Nostrand Company, New York, N. Y., 1943.

water, in which the quartz crystal is driven by mechanical forces generated in the crystal due to converse piezoelectric properties. A cross-sectional view of the quartz crystal subaqueous loud speaker is shown in Fig. 13.6. The electrical circuit and the mechanical network of the quartz crystal subaqueous loud speaker are shown in Fig. 13.6.

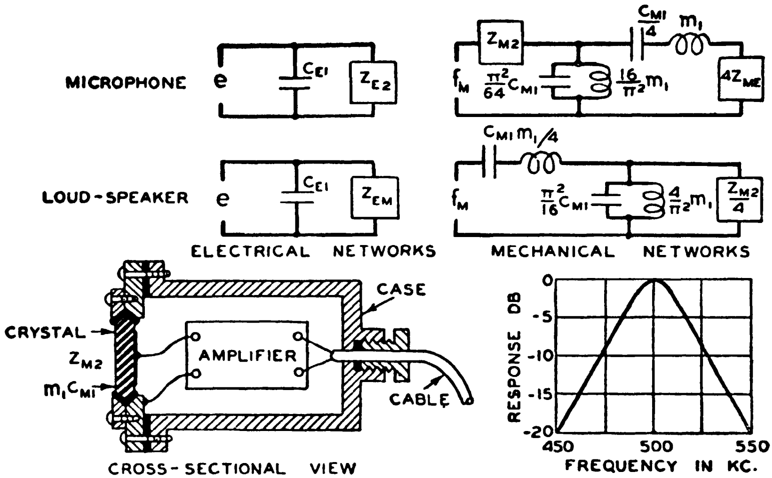

FIG. 13.6. Cross-sectional view, electrical networks, mechanical networks and response frequency characteristic of a quartz crystal subaqueous microphone or loud speaker. In the electrical circuit: z_{E2} = the electrical impedance of the load when used as a microphone z_{EM} = the electrical motional impedance when used as a loud speaker. C_{E1} = the electrical capacitance of the crystal. e = the voltage output when used as a microphone or the voltage input when used as a loud speaker. 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.

The driving force,[11] in dynes, generated in the crystal is given by

$$f_M = \frac{K\mathcal{Q}Ae}{l} \tag{13.19}$$

where K = constant of the crystal, 6.4×10^{-8} for quartz,
$\mathcal{Q}$ = 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.

[11] Olson, "Dynamical Analogies," p. 149, D. Van Nostrand Company, New York, N. Y., 1943.

The motional electrical impedance[12] of the system is

$$z_{EM} = \frac{l^2}{K^2 \mathfrak{Q}^2 A^2} z_M \qquad 13.20$$

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

$$z_{EN} = \frac{z_{EM}}{1 + j\omega C_{E1} z_{EM}} \qquad 13.21$$

where z_{EM} = motional electrical impedance, equation 13.20, 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. 13.6.

The lumped constant representation of the system as depicted in the mechanical network of Fig. 13.6 is valid in the region at and near the resonant frequency of the crystal. The mass,[13] m_1, in Fig. 13.6 is given by

$$m_1 = \frac{\rho l A}{2} \qquad 13.22$$

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.

The compliance,[14] C_{M1}, in Fig. 13.6 is given by

$$C_{M1} = \frac{8l}{\pi^2 \mathfrak{Q} A} \qquad 13.23$$

where A = cross-sectional area of the crystal, in square centimeters,
l = length of the crystal, in centimeters, and
$\mathfrak{Q}$ = Young's modulus, in dynes per square centimeter.

[12] Olson, " Dynamical Analogies," p. 150, D. Van Nostrand Company, New York, N. Y., 1943.

[13] Olson, " Dynamical Analogies," p. 151, D. Van Nostrand Company, New York, N. Y., 1943.

[14] Olson, " Dynamical Analogies," p. 151, D. Van Nostrand Company, New York, N. Y., 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

$$\mu = \frac{r_{EM}}{r_{ED} + r_{EM}} \times 100 \qquad 13.24$$

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 loud speaker is very high, being very close to 100 per cent.

A typical response frequency characteristic is shown in Fig. 13.6.

Quartz crystal loud speakers of the type shown in Fig. 13.6 are suitable for frequency band response in the frequency from 100 kilocycles to 100 megacycles. It appears that the quartz crystal loud speaker is the only system suitable for high conversion in the ultra high frequency range.

13.10. Quartz Crystal Subaqueous Microphone. — A quartz crystal subaqueous microphone is a microphone designed to operate under water, in which a voltage is generated in a crystal having converse piezoelectric properties. A cross-sectional view of a quartz crystal microphone is shown in Fig. 13.6. The mechanical network of the mechanical system and the electrical network is also shown in Fig. 13.6. The mass m_1 and the compliance C_{M1} are given by equations 13.22 and 13.23.

The driving force, f_M, in dynes, is

$$f_M = pS \qquad 13.25$$

where p = sound pressure, in dynes per square centimeter, and
S = area of the diaphragm, in square centimeters.

The mechanical impedance [15] due to the electrical circuit is

$$z_{ME} = \frac{K^2 \mathfrak{Q}^2 A^2}{l^2} z_E \qquad 13.26$$

where K = constant of the crystal, 6.4×10^{-8} for quartz,
$\mathfrak{Q}$ = Young's modulus, in dynes per square centimeter,
l = length of the crystal, in centimeters,

[15] Olson, "Dynamical Analogies," p. 167, D. Van Nostrand Company, New York, N. Y., 1943.

A = area of the electrode, in square centimeters,

$$z_E = \frac{z_{E1} z_{E2}}{z_{E1} + z_{E2}},$$

$$z_{E1} = \frac{1}{j\omega C_{E1}},$$

where C_{E1} = electrical capacitance of the generator, in statfarads, and
z_{E2} = electrical impedance of the external load, in statohms.

The generated electromotive force,[16] e, in statvolts, across the electrical impedances z_{E1} and z_{E2} is

$$e = \frac{KQA\dot{x}}{l} z_E$$

where $\dot{x}$ = velocity of the crystal, in centimeters per second.

The voltage response frequency characteristic is usually the same as that of the loud speaker shown in Fig. 13.6.

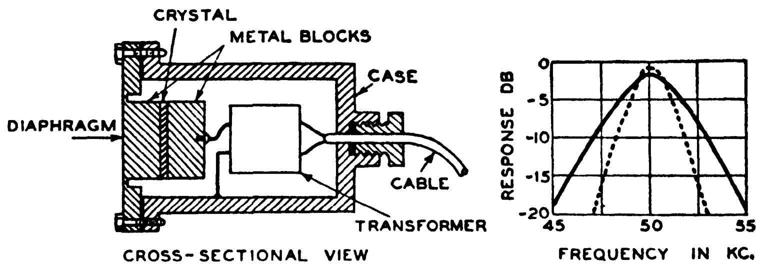

FIG. 13.7. Cross-sectional view and response frequency characteristic of a quartz crystal sandwich subaqueous loud speaker or microphone. The solid-line response frequency characteristic depicts the response for aluminum blocks and the dotted-line response frequency characteristic depicts the response for steel blocks.

13.11. Quartz Crystal Sandwich Subaqueous Loud Speaker and Microphone. — A quartz crystal sandwich subaqueous loud speaker is a subaqueous loud speaker consisting of two blocks of metal cemented to the two sides of a relatively thin quartz crystal (Fig. 13.7). One of the metal blocks is terminated in water and the other is terminated in air. The maximum output occurs when the overall effective length is one-half wave-

[16] Olson, "Dynamical Analogies," p. 169, D. Van Nostrand Company, New York, N. Y., 1943.

length. 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 frequency characteristics for steel and aluminum blocks are shown in Fig. 13.7. As would be expected, blocks with a lower density yield a wider frequency band. The quartz crystal sandwich subaqueous loud speaker conserves quartz since only a thin layer is used as compared to the large block for the all-quartz subaqueous loud speaker. 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. 13.7 may also be used as a microphone. The shape of the response frequency characteristics is approximately the same for the microphone as that shown in Fig. 13.7 for the loud speaker.

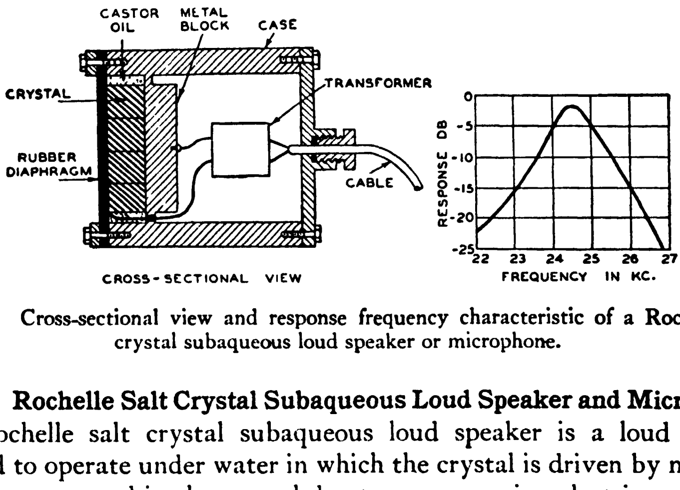

FIG. 13.8. Cross-sectional view and response frequency characteristic of a Rochelle salt crystal subaqueous loud speaker or microphone.

13.12. Rochelle Salt Crystal Subaqueous Loud Speaker and Microphone.—A Rochelle salt crystal subaqueous loud speaker is a loud speaker designed to operate under water 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 subaqueous loud speaker is shown in Fig. 13.8. 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. 13.8.

The system shown in Fig. 13.8 may also be used as a microphone. The shape of the response frequency characteristic is approximately the same for the microphone as that shown in Fig. 13.8 for the loud speaker.

There has been another type crystal[17] developed which may displace Rochelle salt for many applications. This crystal is ammonium dihydrogen phosphate and has been given the abbreviation ADP.

ADP crystals provide high electromechanical coupling. The 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.

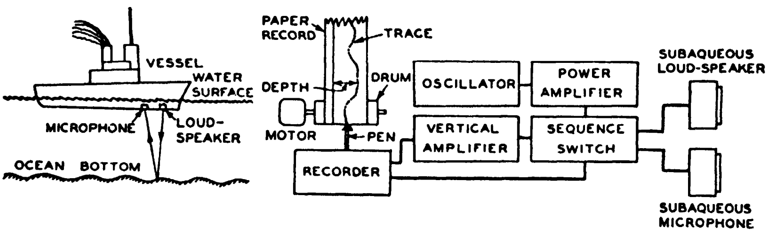

FIG. 13.9. Echo depth sounding sonar system.

13.13. Echo Depth Sounding Sonar.[18] — 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. 13.9. A high-frequency oscillator operating at a fixed frequency in the range from 10 to 100 kilocycles is coupled to a power amplifier. The subaqueous loud speaker and microphone are mounted flush with the hull of the vessel and directed downward. The sequence switch connects the power amplifier to the subaqueous loud speaker for about a few milliseconds. The pulse sent out by the subaqueous loud speaker is reflected from the ocean floor and received by the subaqueous microphone. The output of the subaqueous microphone is coupled to the vertical amplifier by means of of the sequence switch. The output of the vertical amplifier is fed to the

[17] Mason, W. P., *Bell Labs. Record*, Vol. 24, No. 7, p. 257, 1946.

[18] 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.

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 subaqueous microphone 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 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 subaqueous loud speakers and microphones have been used for sonic depth indicators.

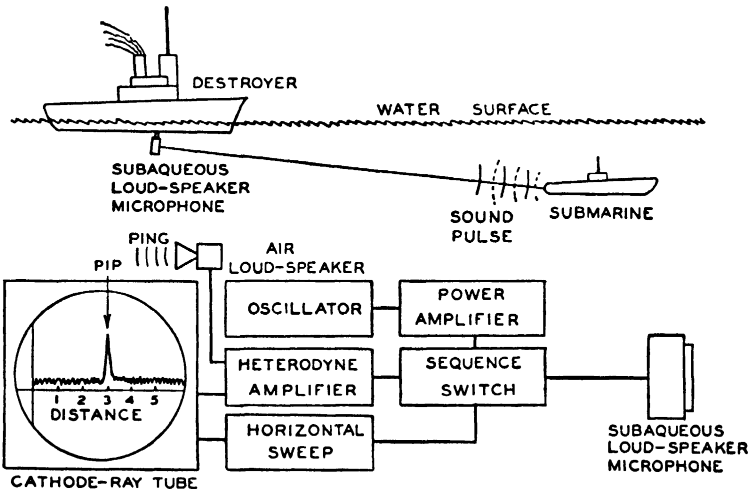

FIG. 13.10. Echo direction and ranging sonar system.

13.14. Echo Direction and Ranging Sonar.[19] — 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. 13.10. 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 subaqueous loudspeaker microphone used as a loud speaker for a few milliseconds. Then

[19] Evans, R. J., *Electronics*, Vol. 19, No. 8, p. 88, 1946.

the sequence switch connects the amplifier to the subaqueous loud-speaker microphone used as a microphone. 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 loud speaker. 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 subaqueous loud speaker microphone is usually about 5 to 10 wavelengths. Since the same unit is used for both sending and receiving, the ordinates of the directional patterns shown in Fig. 2.10 must be squared. It will be seen that under these conditions the directional pattern of a 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.

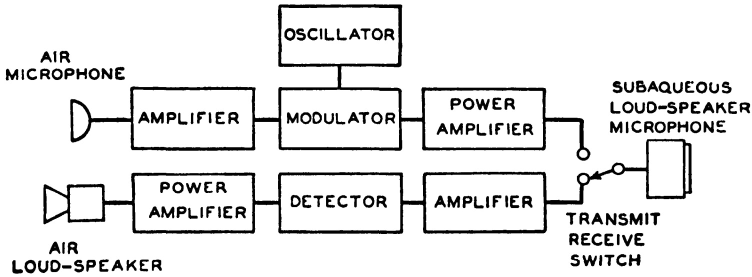

FIG. 13.11. Communication sonar system.

13.15. 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. 13.11. In general, the same underwater transducer is used as both a loud speaker and microphone. Both magnetostriction or crystal transducers have been used for the subaqueous loud-speaker microphone. The frequency band width of the underwater must be sufficiently broad to accommodate the modulated signal. The system shown in Fig. 13.11 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.

CHAPTER XIV

ULTRASONICS AND SUPERSONICS

14.1. Introduction. — The terms ultrasonic and supersonic have been used loosely to designate frequencies above the audio-frequency range. For this reason, it is advisable to define the terms, ultrasonic and supersonic. Ultrasonic is a term used to designate a sound above the audible frequency range. Supersonic is a term used to designate any very intense sound regardless of frequency.

In the earliest experiments when sound vibrations were applied to various physical or chemical bodies, frequencies of 100 kilocycles to 1 megacycle were used. However, in recent years it has been found that the same results can be obtained with lower frequencies.

The effects of intense sound vibrations have been known for about two decades, yet they have scarcely been applied in industrial processes. A few of the reasons for this are as follows: Supersonic transducers and the associated equipment are complex and costly. The transduction efficiency of most supersonic transducers is low. Equipments for large-scale processes have not been developed. Chemical, metallurgical and mechanical engineers are not familiar with the rather complex electrical and acoustical equipment.

Many advances were made in the conversion of electrical energy into sound energy during World War II. As a result, the efficiency of conversion has been increased. Furthermore, the power outputs have also been increased to the point that it should be possible to handle materials on a large scale. There appears to be no doubt but that supersonics will find many uses in industrial fields. It is the purpose of this chapter to outline a few of the salient applications of supersonics in industry.

14.2. Supersonic and Ultrasonic Generators.[1,2,3,4,5,6,7] — Intense sound energy may be applied to gases, liquids or solids to produce desired changes or effects. The most important applications are obtained in the case of liquids and solids. For these applications, the generators designed for use in underwater sound, and described in Chapter XIII on Underwater

[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.

Sound, are particularly suitable. For gases, air loud speakers may be used. Since the theory of both air and underwater loud speakers has been considered at some length in preceding chapters there is no object in discussing these systems again because the adaption of these transducers to the treatment of materials is merely straightforward design.

14.3. Cavitation Due to Supersonics. — 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. 13.2. 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 and chemical effects in the application of high intensity sound to various mediums.

14.4. Dispersion Due to Supersonics.[8, 9, 10, 11] — 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

[8] Solner, K., *Jour. Phys. Chem.*, Vol. 42, p. 1071, Nov. 1938.
[9] Freundlick and Gillings, *Faraday Soc. Trans.*, Vol. 35, p. 319, Feb. 1939.
[10] Bergmann, "Ultrasonics," John Wiley and Sons, New York, N. Y., 1938.
[11] Herzfeld, K. F., *J. Chem. Phys.*, Vol. 9, p. 513, July, 1941.

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.

14.5 Emulsification Due to Supersonics.[12, 13, 14, 15, 16, 17, 18] — 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 supersonics 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 supersonics 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.

Supersonics have also been applied to photographic emulsions with an improvement in homogeneity and stability.

Supersonics 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 supersonics.

14.6. Coagulation Due to Supersonics.[19, 20, 21, 22, 23] — In spite of the fact that supersonics 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 pre-

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[12] Wood and Loomis, *Phys. Rev.*, Vol. 29, p. 373, 1927.
[13] Chambers, L. A., *Jour. of Dairy Sci.*, Vol. 19, p. 29, 1936.
[14] Bondy and Sollner, *Faraday Soc. Trans.*, Vol. 32, p. 556, March 1936.
[15] Sokoloff, S., *Acta Physicochimico*, Vol. 3, p. 939, 1936.
[16] Schmid and Ehret, *Zeit f. Electrochem*, Vol. 43, p. 869, Nov. 1937.
[17] Bergmann, " Ultrasonics," John Wiley and Sons, New York, N. Y., 1938.
[18] Wood, " Supersonics," Brown Univ. Press, Providence, R. I., 1939.
[19] Sollner and Bondy, *Faraday Soc. Trans.*, Vol. 32, p. 616, April, 1936.
[20] Number of Authors, *Faraday Soc. Trans.*, Vol. 32, p. 104, August, 1936.
[21] Behr, A., *Metal Ind. Lon.*, p. 422, Dec. 31, 1943.
[22] Sorenson, Ch., *Ann Phys. Lpz.*, Vol. 27, p. 70, 1936.
[23] Brandt and Freund, *Z. Phys.*, Vol. 94, p. 348, 1936.

cipitated. The action depends in some degree on the wavelength and intensity.

Degassing of molten metals by the application of supersonics is another example of coagulation. Small bubbles form at first which join to form larger ones. The larger ones rise to the surface and are expelled. This use of supersonics should lead to an improvement in castings where the presence of bubbles is very objectionable.

14.7. Chemical Effects of Supersonics and Ultrasonics.[24, 25, 26, 27] — 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.

14.8. Biological Effects of Supersonics.[28, 29, 30, 31, 32, 33, 34, 35, 36, 37] — Supersonics have a very destructive effect upon small living organisms. Small fish have been killed by high-power echo ranging and sounding devices.

Supersonics 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 supersonics. The bacteria

[24] Porter and Young, *Jour. Amer. Chem. Soc.*, Vol. 60, p. 1497, 1938.
[25] Benthe, H., *Zeit. f. Phys. Chem.*, Vol. 163, p. 161, Feb. 1933.
[26] Schmitt, Johnson and Olson, *Jour. Amer. Chem. Soc.*, Vol. 51, p. 370, 1929.
[27] Barrett and Porter, *Jour. Amer. Chem. Soc.*, Vol. 63, p. 3434, Dec. 1941.
[28] Wood and Loomis, *Phys. Rev.*, Vol. 29, p. 373, 1927.
[29] Harvey, E. N., *Amer. Jour. Physiology*, Vol. 91, p. 284, 1929.
[30] Harvey and Loomis, *Jour. of Bacteriology*, Vol. 17, p. 373, 1929.
[31] Harvey, E. N., *Biological Bulletin*, Vol. 59, p. 306, 1930.
[32] Flosdoff and Chambers, *Jour. of Immunology*, Vol. 28, p. 297, 1935.
[33] Bergmann, " Ultrasonics," John Wiley and Sons, New York, N. Y., 1938.
[34] Wood, " Supersonics," Brown Univ. Press, Providence, R. I., 1939.
[35] Gregs, E. C., O. Glasser's " Medical Physics," p. 1591.
[36] Lynn, Zwemer, and Chick, *Science*, Vol. 96, p. 119, 1942.
[37] *Electronics*, Vol. 16, No. 5, p. 154, 1943.

in milk have been reduced by the application of supersonics. This indicates that milk can be sterilized by supersonics.

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.

As in the case of chemical effects the biological effects are somewhat obscure but very interesting.

14.9. Thermal Effects of Supersonics. — 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 supersonics are applied are also obtained by the application of heat. The practical value of heating by supersonics remains to be seen. It has been suggested for use in medicine for diathermy.

14.10 Supersonics as a Detergent. — Supersonics 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.

14.11. Television System Modulation by Ultrasonics.[38] — The modulation of ultrasonic waves in liquids forms the basis of a system of television. Television systems other than those employing the Kinoscope depend upon the modulation of a steady light beam. The sound cell consists of a glass container with parallel walls. A quartz crystal, placed at the bottom of the cell, excites sound vibrations in the liquid. The velocity of light through the liquid depends upon the rarefactions and condensations in the liquid. By means of an optical system the amount of illumination at the point of the screen is a function of the sound intensity at the corresponding point in the liquid. In other words, the light image on the screen depicts the sound intensity distribution in the liquid. If the image of the cell is reduced to a line, this line then constitutes a one-line element of a television image. The light intensity also varies along the line itself. The line is moved normal to itself by means of a mechanical rotating mirror to form the entire picture.

[38] Okolicsanyi, F., *Wireless Engineer*, Vol. 14, No. 169, p. 527, 1937.

14.12. Testing of Materials by Means of Ultrasonics.[39,40,41] — 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[42] 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. 13.13 and 13.14. A quartz crystal loud-speaker microphone (see Secs. 13.9 and 13.10) 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 crystal used as a loud speaker. The reflected pulse is picked up by the crystal used as a microphone. The output of the microphone 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 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 [43] 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 subaqueous loud speaker and subaqueous microphone 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.

[39] Bergmann, "Ultrasonics," John Wiley and Sons, New York, N. Y., 1938.
[40] Andrews, A., *Electronics*, Vol. 17, No. 5, p. 122, 1944.
[41] Hayes, H. C., *Jour. Acous. Soc. Amer.*, Vol. 8, No. 4, p. 220, 1937.
[42] Firestone, F. A., *Jour. Acous. Soc. Amer.*, Vol. 17, No. 4, p. 363, 1946.
[43] *Jour. Appl. Phys.*, Vol. 15, No. 3, p. XIV, March, 1944.

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