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LOUDSPEAKER IMPEDANCE

Vincent Salmon
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This invited technical editorial by the Vice-Chairman of IRE-PGA treats a subject small in scope but of great importance of audio, in a comprehensive and yet fundamental manner. – *Editorial Committee*

INTRODUCTION

In most fields of specialization, the time soon comes when the meanings of terms must be standardized so that all workers may use a common technical language. The initial efforts of technical lexicographers usually run smack into the illogical but accepted usages which have grown up in each field. It then becomes necessary to define old, as well as additional new, terms so as to avoid ambiguity, and then to hope that these precisely defined terms will win some degree of acceptance. Such a situation now exists in the use of the term impedance. In this note we shall discuss it from the point of view of the loudspeaker, and shall comment on some of the proposed new terms and their usefulness. Most of the discussion applies specifically to moving coil loudspeakers, but the concepts are easily transferred to other types.

SPEAKER IMPEDANCE AND SPEAKER RATING IMPEDANCE

What do we do with a loudspeaker? Ordinarily, it is connected to a signal source for operating or for testing. In order to predict the performance during operation, it is necessary to know, among other things, the electrical impedance characteristics of both source and loudspeaker as a function of frequency, acoustical environment, and perhaps power level. On the other hand, engineering tests on loudspeakers are usually made under idealized conditions in which the test source is a constant voltage in series with a pure resistance. In order to relate the two situations, it is necessary to idealize the varying impedance characteristics of a loudspeaker to a single number, which somehow represents an averaged behavior. To get a clue as to how this is done, let us first discuss the operation of a loudspeaker under actual conditions of use.

When a loudspeaker is to be connected to an ordinary amplifier, the first question is usually, "To what terminals should it be connected?" Obviously, this implies an impedance which is a single figure, not the electrical impedance which varies with frequency. In essence, we want numbers on both amplifier output and speaker input terminals such that when terminals bearing the same number are connected together, then the combination will deliver *rated* performance. Accordingly, this magic single number may be called the *rating* impedance of the loudspeaker. The term "speaker impedance" by itself, we then take to be simply the complex electrical impedance, expressed in magnitude-angle or resistance-reactance form, as a function of frequency. What, then, is the relation between "speaker impedance" and "speaker rating impedance"?

The loudspeaker is a load on the driving amplifier, and as such should permit the amplifier to deliver to it the maximum power which is consistent with prescribed distortion limitations. In actuality, the speaker impedance and the spectrum of the audio signal are functions of frequency, and conditions are highly transient. If an ideal transformer of variable turns ratio were interposed between the amplifier and the loudspeaker, then it would be found that the optimum transformer setting for maximum distortion-limited power would depend on the type of signal. Extremes of signal types are represented by those at the output of electric organs and paging systems. The single value of rating impedance commonly quoted applies specifically to a wide range audio signal with spectrum maximizing between 200 and 800 cps. In actuality, the rating impedance is best expressed as a range rather than a single number, because the optimum is so broad. In practice a two-to-one range is acceptable; a loudspeaker with an 8-ohm rating impedance may be connected to amplifier outputs marked from 5.6 to 11.3 ohms. The performance at the impedance extremes will be degraded only at extremes of frequency and power.

AMPLIFIER OUTPUT IMPEDANCE

At this juncture, the term output impedance as applied to amplifiers deserves discussion. It is this writer's opinion that the term should never be used unqualified, since it refers to a position, and does not state in which direction the impedance is to be measured. Facing the amplifier, the amplifier internal output impedance is seen; and facing the load, the amplifier load impedance is seen.

ESTIMATING RATING IMPEDANCE

To return to loudspeakers, it is seen that the rating impedance may be obtained from the loudspeaker impedance if the signal spectrum is used to get a properly weighted average of the impedance-frequency characteristic. For ordinary audio signals, the following rule-of-thumb procedures may be followed for estimating the rating impedance for wide-range, moving-coil loudspeakers; all dc measurements are at the voice coil. In the first procedure, measure the dc resistance of the voice coil. The rating impedance will be about 20 per cent greater for direct-radiator loudspeakers, and about 40 per cent greater for horn loudspeakers. In a second procedure, ac measurements of the magnitude of the impedance are utilized. For direct-radiator loudspeakers, locate the minimum impedance in the 200-800 cps range and add 10 per cent. For horn loudspeakers, the rating impedance is the average value of the magnitude of the impedance taken near the center of the pass band. In view of the two-to-one range in the standard rating impedances of 4, 8, and 16 ohms, these estimates should be close enough. It must again be emphasized that these rough rules apply only to ordinary wide-range, moving-coil loudspeakers; they will be quite different, for example, for loudspeakers designed for battle announce systems.

SPEAKER TEST SOURCE IMPEDANCE AND SPEAKER INPUT

In testing loudspeakers, the rating impedance has another function in the calculation of the input to the loudspeaker. We note first that when an amplifier is properly loaded for maximum output at, say, 2 per cent harmonic distortion, the impedances of the driving source and of the load are rarely equal. Thus amplifiers usually do not operate with a matched load in the sense that the impedance at the output are the same in both directions. The appropriate measure of loudspeaker input is hence not the maximum power available from the source, but rather is the power the source can deliver to its rated load, within prescribed distortion limitations. In testing a loudspeaker we therefore use as a source an adjustable voltage (which is held constant during the test) in series with a fixed resistance which equals the amplifier internal output impedance in the mid-range; this is the test source impedance. To set the power available to the loudspeaker, replace the loudspeaker by a resistor of value equal to the rating impedance. Then vary the adjustable voltage until the desired power is absorbed by the rating impedance. When the loudspeaker is re-inserted into the test circuit, it will actually absorb a different power, but by keeping the source emf constant, conditions of use will be simulated in an idealized fashion.

REGULATION

It is a common tendency nowadays to employ so much negative voltage feedback in amplifiers so that the loudspeaker is driven from essentially a zero impedance source. Whether or not this is desirable is another matter, on which many words have been spilled but for which few facts have been marshaled. Here we shall merely introduce what seems to be a useful measure of the amplifier internal output impedance. First measure the open circuit voltage across the amplifier output, keeping within the linear region. Then connect the rated load across the terminals. The ensuing drop in voltage, expressed in decibels, is termed the regulation. For example, the regulation of a triode output stage is normally 3.5 db. For beam power tubes without feedback, 15 db is perhaps a lower limit. With feedback, values as low as 0.2 db have been stably attained. A matched source has a regulation of 6 db, whereas for a constant voltage source the regulation is zero db. The importance of the regulation is this: at extreme frequencies the loudspeaker impedance usually is very large compared with the rating impedance and hence the loudspeaker approaches an open-circuit load condition. The regulation then represents the maximum increase in voltage across the loudspeaker (and hence the rise in output), expressed in decibels to be expected as a consequence of the rise in impedance. Of course, the presence of reactive components in the actual source and load may somewhat alter this output increase, but the regulation is seen to furnish a good basis for estimating the behavior of a given loudspeaker with different output stages. Also, this points out the necessity for staging the source regulation in giving results of loudspeaker tests, since it is one of the important factors to be considered in comparing results of tests on different loudspeakers.

IMPEDANCE DISTORTION

Amplifiers are commonly tested and rated with a resistive load. Thus, the path of operation in the equivalent plate current-plate voltage family of tube curves is a straight line, assuming a perfect output transformer. However, long before the effect of finite transformer primary inductance is felt at low frequencies, the speaker will have become a load having considerable reactance. Consequently, the path of operation will become an approximate ellipse instead of a straight line. Because this ellipse will extend into regions where the tube characteristics are not so well behaved, there will result a species of distortion for which the term "impedance-distortion" is suggested. In practice it is almost certain that for well designed and stable amplifiers this distortion will be of minor consequence, owing to the vastly greater amount arising from mechanical nonlinearity and magnetic field inhomogeneity in the loudspeaker. Still, it might be instructive to construct a dummy load simulating the loudspeaker for use in amplifier measurements at low frequencies.

CONCLUSIONS

Five impedances are associated with the operation and testing of loudspeakers: the complex speaker impedance; the rating impedance; the test source impedance; the amplifier internal output impedance; and the amplifier load impedance. From these values it is possible to make the proper connections for operation and for testing, and to calculate the power available to the loudspeaker. The rating impedance is not a very sharply defined quantity, and it may be estimated in a relatively simple manner in most cases of interest. The regulation of the test source should always be stated in giving results of tests; it aids in estimating loudspeaker performance at extremes of frequency.

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PGA COMMITTEE APPOINTMENTS

It is a pleasure to announce the following committee appointments:

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We invite any interested PGA members to help these committees. Please communicate with the appropriate chairman or with me.

Marvin Camras
Chairman, IRE-PGA

NEW IRE–PGA CHAPTERS FORMING

Welcome to the new San Diego Chapter of IRE-PGA! Although there was a news item concerning the possible formation of a chapter in San Diego several issues back, the San Diego Chapter is now "official", and we hope to have some news from them in a future issue.

A petition for a new chapter in Houston has been received, and may be acted upon before this issue of *Transactions* of the IRE-PGA is printed.

PGA now has chapters in the following cities:

Albuquerque	Kansas City
Boston	Los Angeles
Chicago	Milwaukee
Cincinnati	Philadelphia
Detroit	Seattle
San Diego	

If the IRE Section in your vicinity has a group of members (or potential members) interested in audio, please communicate with Robert E. Troxel, Chairman of the Committee on Chapters, (Shure Brothers, Inc., 225 W. Huron St., Chicago 10, Illinois).

IRE–PGA AT 1953 NATIONAL ELECTRONICS CONFERENCE

Once again the IRE-PGA is cooperating with the National Electronic Conference by organizing an audio program for the 1953 NEC meeting in Chicago, September 28, 29 and 30. This session has regularly been the fall meeting of the Professional Group on Audio for several years. Mr. R. J. Tinkham has been appointed by Marvin Camras as Chairman of the Program Committee for IRE-PGA.

Another excellent program is expected. Make your plans to attend.

ROOM ACOUSTICS*

Hale J. Sabine
The Celotex Corporation
Chicago, Illinois

This basic summary of room acoustics was prepared by Dr. Sabine specifically for radio engineers. The active discussion which it stimulated at the national IRE Convention indicated its potential general interest to the membership of IRE-PGA.—*Editorial Committee.*

The foregoing papers have described the action of various electrical, mechanical, and electroacoustic links which may form a more or less complex chain between a source of sound and its reception by the ear. The present discussion will be concerned with the transmission of sound through the air in a room and the ways in which the signal may be modified by the acoustical characteristics of the room. In the cases to be discussed, the source of sound in the room may be a live voice or musical instrument, or may be reproduced by a loudspeaker. It may be received either by the ear or by a microphone for further transmission.

As the simplest possible case, we may consider transmission in a room in which reflection of sound waves is essentially nonexistent. This would be equivalent to the open air, or a so-called "free field". Under these conditions the sound signal is transmitted from source to receiver with no modifications except for simple attenuation due to the decrease in amplitude of the wave as it spreads out from the source. As given by the inverse square law, this attenuation amounts to 6 decibels for each successive doubling of distance from the source, or 20 db per ten-fold increase of distance, and is essentially independent of frequency. Over large distances, high audio frequencies are attenuated somewhat more than low frequencies, due to absorption of sound energy by the air itself. The amount depends on frequency and relative humidity. A 10,000 cycle tone is attenuated 9 db per hundred feet of travel at 20 per cent relative humidity, and 4 db at 50 per cent. In any room this directly transmitted sound is always present as a component of the total sound field, and its intensity or energy density at the receiver depends only on the acoustic power output and directivity of the source and on the distance of the receiver from the source. The energy density of the direct sound in ergs per cubic centimeter is given by

$$E_D = \frac{W}{4\pi D^2 c} \cdot \frac{4\pi}{\Omega_1}. \qquad (1)$$

where W = acoustic power output of source,
D = distance from source,
c = velocity of sound, and
Ω_1 = effective solid angle of radiation of source, in steradians.

For a nondirectional source,

$$E_D = \frac{W}{4\pi D^2 c} \qquad (2)$$

In a typical room sound waves are reflected back and forth many times between the walls, floor, and ceiling, with some loss of energy by absorption at each reflection as shown in Figures 1 and 2. If a steady source is maintained, the rapid accumulation of reflected waves quickly sets up a level of sound energy which is essentially uniform throughout the room and remains constant until the source is cut off. This steady state level is determined by the point at which the rate of energy loss by absorption at the room surfaces equals the rate at which sound energy is supplied by the source. From this it can be shown that the steady state energy density is directly proportional

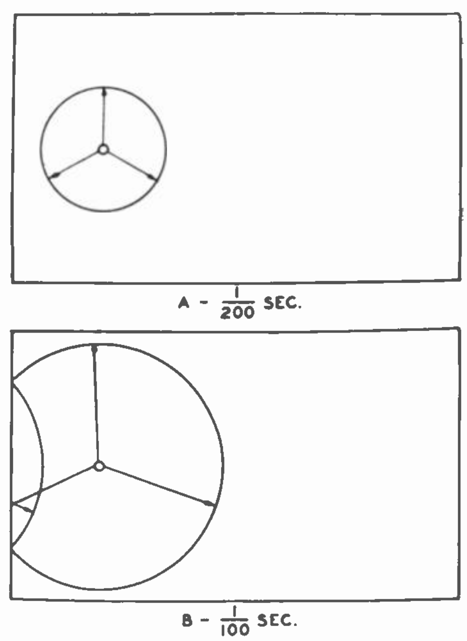


Fig. 1 – Multiple reflection of a single wave front in a closed room.

*Presented as part of the Seminar, "Acoustics for the Radio Engineer", organized by the IRE Professional Group on Audio, IRE National Convention, New York, N.Y., March 25, 1953. Manuscript received June 3, 1953.

to the acoustic power output of the source and inversely proportional to the room absorption, which depends on the area and average sound absorptivity of the room surfaces and furnishings. The value of the reflected energy density is given approximately by

$$E_R = \frac{4W}{ac} \tag{3}$$

The room absorption a, in *sabins*, is the figure obtained by multiplying the area in square feet of each room surface by its sound absorption coefficient, or *absorptivity*, taking the sum of these products, and adding in the absorption supplied by furnishings, seats, and

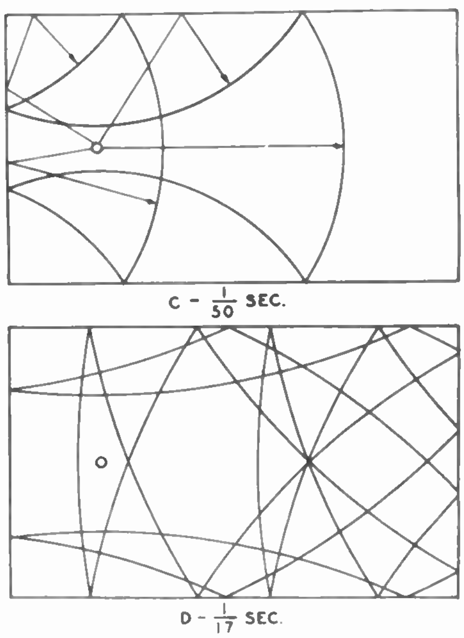


Fig. 2 – Multiple reflection of a single wave front in a closed room.

persons in the room. One sabin, or absorption unit, is defined as the equivalent of one square foot of perfectly absorptive surface, such as an open window. The absorption coefficient of a surface is defined as the per cent of incident sound energy absorbed by the surface. Sound waves are assumed to strike the surface equally from all possible angles. Ordinary hard interior finish materials such as concrete, plaster, wood, glass, etc., are excellent sound reflectors, having absorption coefficients of only 1 to 5 per cent. Carpets and drapes are more or less effective absorbers, depending on frequency. Upholstered seats have absorptions of 1 to 4 sabins each, and a seated person has an absorption of about 4 sabins. Architectural acoustical materials, designed as sound absorbent interior surfaces, have coefficients which may approach 100 per cent at certain frequencies, and in general have coefficients, averaged over the frequency range, of 50 to 90 per cent, depending primarily on thickness.

Sound absorption is fundamentally the transformation of the vibratory energy of air molecules in a sound wave to the random molecular motion of heat energy. This can only take place through friction or viscosity; that is, by *damping*. In the case of porous material, which is the most common type of sound absorber, a sound wave readily enters the material, where the vibratory energy is dissipated by friction of the air molecules against the pore walls or fibers of the material. This results in the reflection of a wave with diminished sound energy, as shown in Figure 3. Sound can also be absorbed by impervious diaphragms, such as thin plywood over an air space. The two basic requirements for effective absorption in this manner are, first, that the diaphragm be light enough and backed by a deep enough air space that it can move relatively freely under incident sound pressure, and second, that there be a frictional or viscous element of the proper value in the system for developing the required transformation to heat energy. This element can be either the internal bending viscosity of the diaphragm or a porous material in the air space.

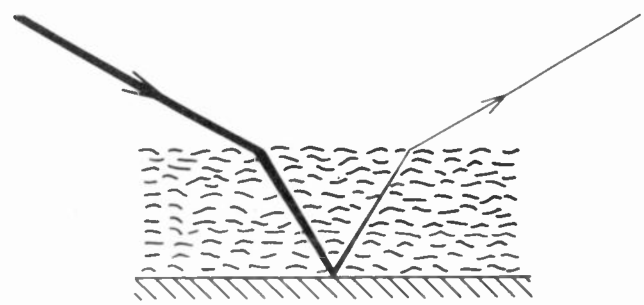
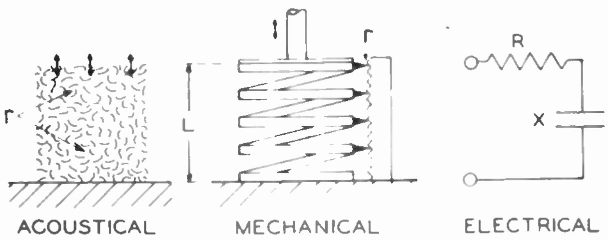


Fig. 3 – Absorption and reflection of sound by a porous material.

The electrical analog of sound absorption is a transmission line having a purely resistive characteristic impedance, representing the air outside the material, terminated by a complex impedance, representing the surface of the material. If the electrical reflection coefficient R of such a line termination is expressed in terms of a power absorption coefficient, $\alpha = 1 - R^2$, its value will be numerically equal to the absorption coefficient of the corresponding acoustical impedance (for normal wave incidence). The simplest type of sound absorber is a porous material on a rigid backing. The acoustic impedance of the surface of such a material is essentially the analog of a resistance and a capacitive reactance in series, as shown in Fig. 4. The resistance corresponds to the frictional resistance offered by the material to the penetration of air, and the capacitive reactance to the stiffness reactance of the air volume contained between the outer surface and the rigid backing of the material. Covering the material with a perforated facing is equivalent to adding a series inductive reactance to the equivalent electrical circuit, and causes an absorption peak at the resonant frequency. (See Figure 5.)



$$R = 1/3\, r L$$

$$X = -\frac{\rho c^2}{2\pi f P L}$$

Fig. 4 – Mechanical and electrical analogs of sound absorption by a homogeneous porous material of thickness L, porosity P, and specific flow resistance r. (H. J. Sabine, "Sound absorption and impedance of acoustical materials," *Jour. Soc. Mot. Pic. Eng.)* vol. 49, no. 3, pp. 262-278; September, 1947. Reproduced by permission.)

Complete transfer of power to the load in the electrical circuit, corresponding to complete sound absorption, will take place only when the load impedance is a pure resistance equal to the characteristic line impedance. Since the impedance of acoustical materials in general contains a reactive component, as they are normally manufactured and installed, their absorption will vary with frequency.

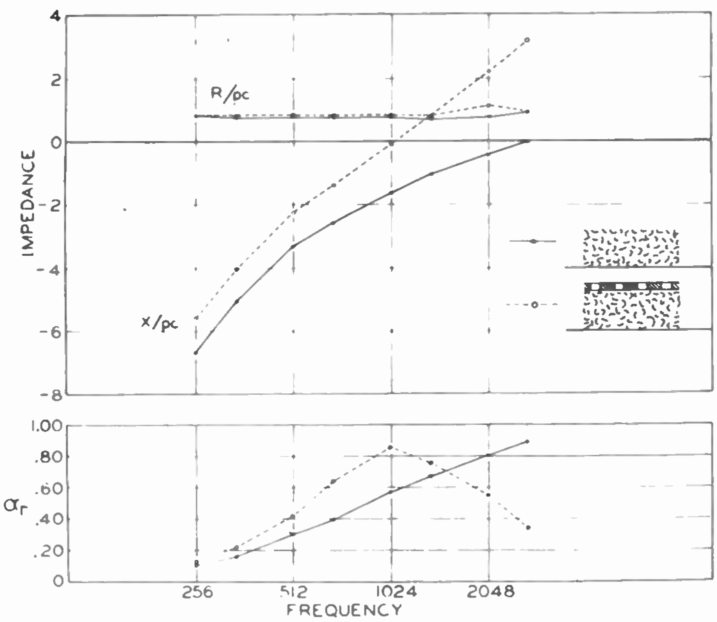


Fig. 5 – Acoustic impedance and absorption coefficient of one-inch porous material with and without perforated facing. (Reproduced by permission from same source as Fig. 4.)

Neither the ear nor a microphone responds directly to sound energy, but instead to sound pressure (or particle velocity in the case of a velocity microphone), therefore, it is important to know how sound pressure is distributed with a steady source being maintained. The total sound pressure at a given point depends on the vector summation of the random pressure amplitudes and phases of all of the reflected waves passing through that point. The pressure will therefore fluctuate markedly from point to point in a stationary series of peaks and dips termed a "standing wave pattern" or "interference pattern". Successive peaks or dips when excited by a single frequency source are spaced roughly one-half wavelength apart in any direction, the spacing in feet being approximately equal to 560 divided by the frequency. The difference in pressure level between maxima and minima may be as much as 20 or 30 db, but with increasing absorptivity and irregularity of the room surfaces the pattern becomes much less pronounced and also less regular. Interference peaks and dips may be easily heard in any room in which a steady, moderately high-frequency tone is sounding, by stopping one ear and moving the head slowly over a distance of a few feet. A sound level meter will register similarly.

The dips in a standing wave pattern are sometimes referred to as "dead spots", a term which is misleading as far as over-all room acoustics is concerned. For every frequency which would produce a pressure minimum at any given point, there is on the average another frequency which would produce a maximum at the same point. Normal sound sources, such as speech or musical instruments, produce many frequencies at once which are constantly changing. Interference effects therefore tend to average themselves out under actual conditions and there is no evidence of fixed dead spots attributable to interference *per se*. This is shown very clearly in Figure 6,

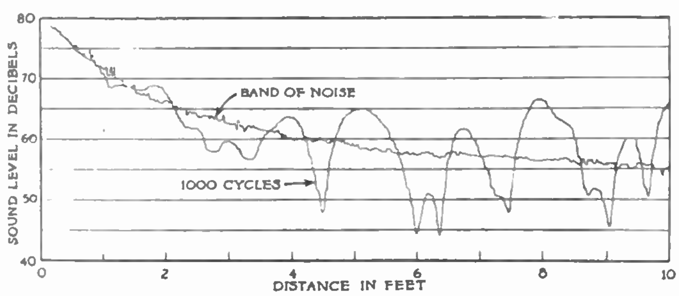


Fig. 6 – Sound pressure level vs. distance from a loudspeaker in a room (Reproduced by permission from V. O. Knudsen and C. M. Harris, "Acoustical Designing in Architecture," John Wiley and Sons, Inc., New York, N.Y., 1950; p. 138.)

in which the interference pattern is virtually eliminated by substituting a continuous band of frequencies one octave wide for a single frequency tone. This figure incidentally illustrates why a band of frequencies rather than a single fixed tone should be used in measuring the average sound pressure level in a room. Interference effects are further minimized by binaural listening, since the chances are against pressure minima coinciding with the positions of both ears. As a possible exception to the above statements, it may be noted that at very low frequencies of less than 100 cycles, where wavelengths are 10 feet or more and interference patterns correspondingly broad, definite areas of poor response to certain bass tones may occur, particularly in small rooms.

Since any change in frequency will cause a change in the spacing of the interference pattern, it can be seen that the sound pressure at a fixed point, due to the reflected wave system, will fluctuate with frequency. Each peak in the pressure-frequency curve is due to an individual resonant response or so-called "normal mode of vibration" of the volume of air enclosed in the room. Roughly, a resonance peak may be thought of as occurring whenever a reflected wave after traveling once around a room (or between two opposite surfaces) is exactly in phase with itself at its starting point.

The response curve of a typical room of living room or studio size and moderate absorptivity is shown in Figure 7. The fluctuations shown are due largely to room resonances, and only secondarily to variation of loudspeaker output. At low frequencies the peaks are clearly defined and quite broad, but at higher frequencies they become more and more closely spaced. In a larger room the low frequency peaks would also be more closely spaced. Since the peaks are caused by reflection, it follows that increasing the absorptivity of the room surfaces should smooth out the response curve. This occurs to a marked degree, as shown by Figure 8, in which the room absorption is increased four times.

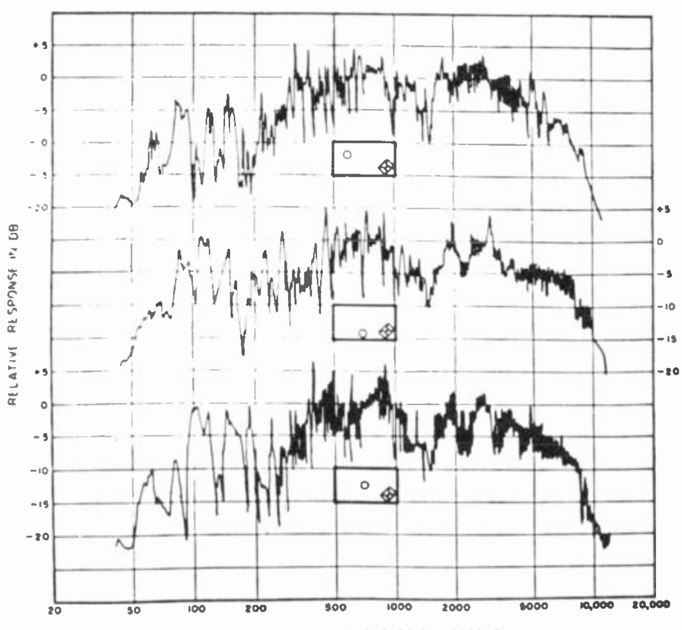


Fig. 7 – Variation of sound pressure level with frequency at a fixed point in a room. (Reproduced by permission from Jensen Technical Monograph No. 1, "Loud Speaker Frequency-Response Measurements," published by Jensen Manufacturing Co., Chicago, Illinois.)

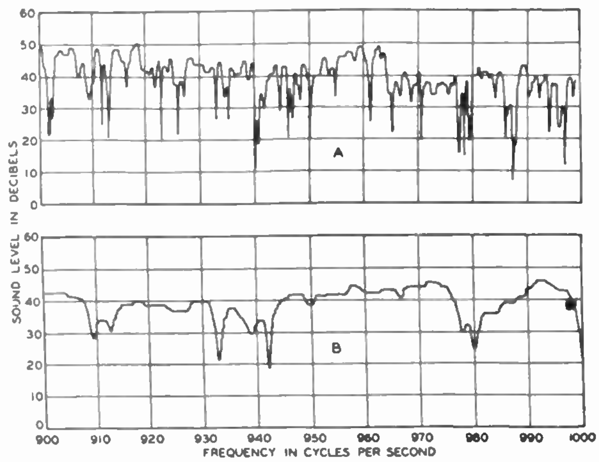


Fig. 8 – Variation of sound pressure level with frequency in a room with low absorption *A* and high absorption *B*. (E. C. Wente, "The characteristics of sound transmission in rooms," *Jour. Acous. Soc. Amer.*) vol. 7, p. 123; October, 1935. Reproduced by permission.)

The frequency irregularity of a room can be easily heard if one ear is closed and a pure tone is varied slowly in frequency. However, in listening to speech and music the frequency peaks are so uniform and so closely spaced that they are not distinguishable as such by the ear. Exception must be made again to the case of low frequencies in small rooms where the irregularities are so broad that they can be readily detected in normal listening or microphone pickup. For example, successive pedal notes of an electronic organ might differ noticeably in audible level.

As far as hearing conditions are concerned, then, we can largely ignore both the spatial and frequency fluctuations of sound pressure due to interference effects, and consider only the average sound pressure which, when squared, will be a direct measure of the reflected sound energy density. As pointed out previously, its steady state level is uniform throughout the room (assuming a generally rectangular shape). The total energy density at the point of pickup with a steady source sounding is the numerical sum of the energy density of the reflected sound and that of the directly transmitted sound. The ratio of direct to reflected sound energy density is, approximately,

$$\frac{E_D}{E_R} = \frac{a}{16\,\pi D^2} \tag{4}$$

where a = room absorption in sabins, and
D = distance from source in feet.
There difference in level in decibels is

$$L_D - L_R = 10 \log_{10} \frac{a}{16\,\pi D^2} \tag{5}$$

If either the source or the receiver, or both, has a directional radiation or response characteristic, the foregoing expression is modified to

$$L_D - L_R = 10 \log_{10} \frac{a}{16\pi D^2} + 10 \log_{10} \frac{4\pi}{\Omega_1} + 10 \log_{10} \frac{4\pi}{\Omega_2} \tag{6}$$

where Ω_1 and Ω_2 are the effective solid angles of radiation and response of the source and receiver, respectively, in steradians.

This relation (Equation 5) is plotted in Figure 9 as a function of distance from the source for the indicated values of room absorption. Both source and receiver are assumed to be nondirectional. If either one were directional, the direct sound in relation to reflected would be correspondingly higher, by the amounts given in (6).

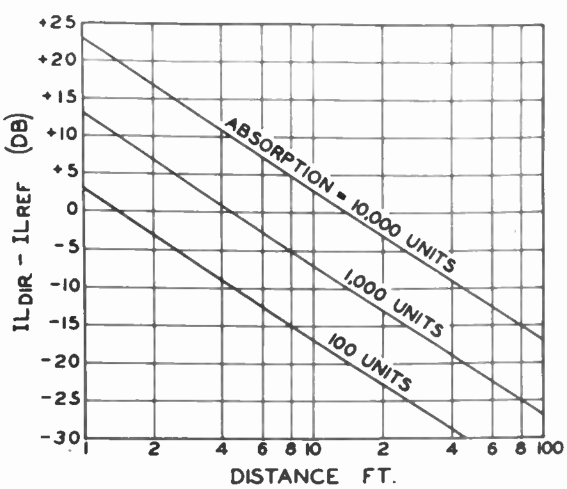


Fig. 9 – Level of direct sound (IL_{dir}) above or below generally reflected sound (IL_{ref}) in relation to distance from source and number of absorption units in the room.

Figure 9 can be used to estimate at what distance from the source the reflected sound will become the major component in rooms of various sizes. If it is assumed that the direct component becomes negligible when it falls to 10 db below the reflected sound, it will be seen that this occurs at only about 5 feet from the source in a room with 100 sabins absorption. This would be typical of an average living room. In a room with 1000 sabins, which would be a small auditorium with an audience of 150 to 200, or a 25 by 40 foot radio studio, the reflected sound would become predominant at about 15 feet from the source. A room with 10,000 sabins would be a fairly large auditorium or concert hall filled to a capacity of 1500 to 2000 people. In this case the reflected sound would not become predominant until about 50 feet from the source.

If either the source or pickup is directional, the above distances will be increased by a factor of $\sqrt{4\pi/\Omega}$. This is equivalent to saying that the effect of a directional source or pickup is the same as moving closer to the source.

One important application of these figures is in determining what effect the absorption-frequency characteristic of the room will have on the sound transmitted to various receiving positions. At points far enough from the source that the total level is due mostly to reflected sound, the frequency characteristic will be essentially that of the room absorption. Close to the source, the variation of absorption with frequency will have very little effect on the transmission-frequency characteristic. The equation relating the total level, in decibels, to distance and room absorption, assuming non-directional source and receiver, is

$$L_T = L_D + 10 \log_{10} \left(1 + \frac{16\,\pi D^2}{a}\right) \tag{7}$$

Since the direct sound L_D is transmitted without frequency discrimination, this equation can be used to determine the effect of variation of absorption a with frequency on the total level for various distances D. In a well-designed room the overall variation in absorption over the audible frequency range will be of the order to 2 to 1, which is a deviation of only 3 db from a flat room response curve. The effect of non-uniform absorption can be further reduced by a directional source or receiver.

Loudspeakers are used as sound sources principally to reinforce the natural voice in large auditoriums, and to reproduce recorded sound, as in motion picture theatres. It is generally desirable that the amplified sound be as close to a uniform level as possible over the seating area. This can be accomplished either by distributing a number of low level speakers around the room so as to blanket the seating area, or by utilizing the directional pattern of a single loudspeaker near the stage. As illustrated in Figure 10, the speaker is oriented so that the drop in level away from the axis is compensated by the closer distances.

The entire discussion has thus far been confined to transmission through the room of sound from a steady, continuous source which does not vary in strength, frequency, or frequency distribution, such as a sustained note from a musical instrument without tremolo or vibrato. The effects of room characteristics on the audible tone quality of such a sustained sound are actually quite minor and difficult to detect, and are apparent only in subtle differences in the harmonic structure of a musical tone due to the interference pattern, and possibly to high frequency attenuation due to air absorption or non-uniform room absorption. The sounds of music, and especially of speech, however, are in general constantly changing and of relatively short duration. The transient responses of the room to these changes have a very marked and unmistakable effect on sound quality and hearing conditions.

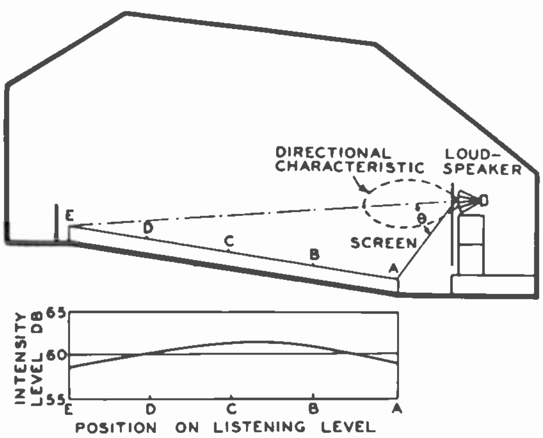


Fig. 10 – Arrangement of the loud speaker for a sound motion picture reproducing system in a theater. (Reproduced by permission from H. F. Olson, "Elements of Acoustical Engineering," D. Van Nostrand Company, Inc., New York, N.Y., 1947; p. 409.)

The most important index of the transient response of a room is its reverberation time. Reverberation is defined as the prolongation and dying out of sound energy in a room after the source is stopped, due to the continued travel and absorption of reflected waves. The sound energy decay follows as exponential curve, which when plotted on a decibel scale against time is a straight line. The reverberation time T of a room is defined as the number of seconds required for a decay of 60 decibels. It is a definite acoustical property of the room which depends only on its volume V in cubic feet and total absorption a in sabins, in accordance with the formula

$$T = \frac{0.05\,V}{a} \tag{8}$$

Since the absorption normally varies with frequency, the reverberation time will be frequency dependent in inverse ratio.

It is important to note that the reverberant sound decay starts from the level of the reflected sound and not necessarily from the total level, as illustrated in Figure 11. That is, if the receiver position is close enough to the source that the direct sound predominates with the source on, there will be an instantaneous drop to the reflected sound level when the source is stopped, after which the decay takes place at the rate given by the reverberation time. At more distant positions where the direct sound is much less than the reflected, the decay starts from the total level with no discernible sudden drop when the source is cut off. The extent of this initial drop-off of direct sound is probably the most important factor in determining "acoustical perspective", or the ability to estimate the distance of a sound source from the point of pickup by listening alone, either directly or through a microphone and loudspeaker. As pointed out

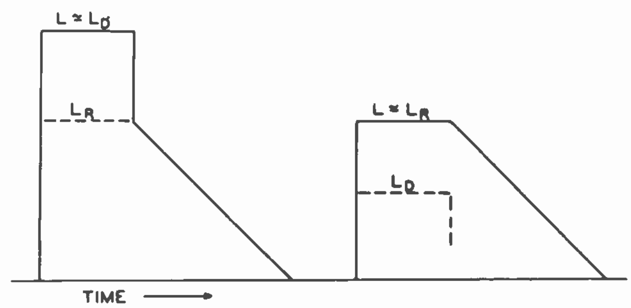


Fig. 11 – Diagram showing that reverberant sound decay starts from level of reflected sound (L_R) and not from level of direct sound (L_D).

above, a directional source or microphone will cause the distance to be estimated as less than it actually is. It is this same effect which makes it possible for directional loudspeakers to partially overcome the effects of excessive reverberation on speech intelligibility. Good results have been obtained in this manner in very large cathedrals, where tremendous volumes and lack of available treatment areas make it impossible to control reverberation sufficiently with acoustical materials.

The term "liveness" has been used to denote the overall subjective effect of reflected sound and reverberation in altering the quality of the directly transmitted sound, and numerous attempts have been made to attach a numerical figure to this property. Maxfield and Albersheim have proposed a "liveness constant" for a given room and pickup location which is proportional to the reverberation time and to the ratio of reflected to direct sound at that location. It is open to question, however, as to whether both of these factors can be properly combined into a single constant. For example, in a room having a four-second reverberation time, a point could be found close enough to the source that the ratio of reflected to direct energy would be 1/4. At this point, therefore, there would be a drop in level of 6 db before the start of the slow decay given by the four-second reverberation time. In another room having a one-second reverberation time, a point could be found distant enough from the source that the reflected energy would equal the direct; that is, their ratio would be 1. In this case there would be a 3 db drop when the source was stopped, but this would be hardly perceptible when followed by the rapid one-second decay time. The liveness constant would be the same for the two cases, but there would be a marked difference in the ear's judgment of room effect. It is especially important to consider the initial drop and the following decay rate separately when adding reverberation artificially to program material either by an echo chamber or electronically.

Generally speaking, the reverberation time must be held within certain limits in order to prevent confusion and overlapping of successive speech sounds, with resulting loss of intelligibility, and to impart a pleasing quality to the rendition of music. The choice of reverberation time representing optimum hearing conditions has been covered extensively in the literature and in standard texts on architectural acoustics, and no detailed discussion will be attempted here. It can be stated that acceptable reverberation times may range from 0.5 to 3 seconds, depending on frequency, room size, the type of sound, that is, whether speech, music, live, or reproduced, and on whether pickup is by the two ears (binaural) or by a microphone (monaural). See Figure 12. It can also be stated that deviations of plus or minus 20 per cent from recommended values will cause no serious harm to the quality of hearing conditions.

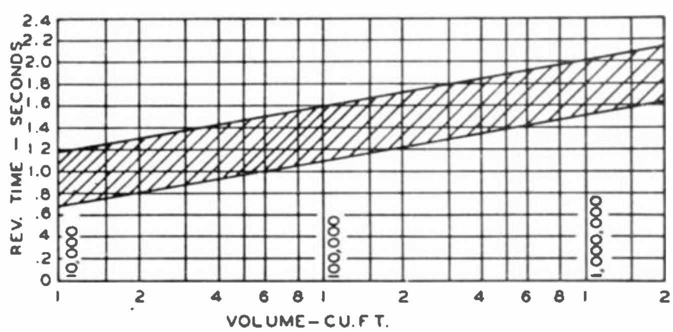


Fig. 12 – Range of acceptable reverberation times at a frequency of 500 cycles for auditoriums of various sizes. Range includes various uses of room. (Reproduced by courtesy of Acoustical Materials Association).

The effect of the frequency variation of reverberation time on hearing conditions may require some explanation. A reverberation curve which is flat above 1000 cycles and gradually rising to about twice the 100 cycle value at 50 cycles is generally considered acceptable. This curve can be justified theoretically on the basis that it causes all frequency components of a complex sound to maintain the same relative loudness to each other throughout the decay. Furthermore, such a curve is fairly typical of most ordinary listening rooms and auditoriums, and can therefore be considered as representing "naturalness" in room quality. Deviations from this ideal curve are objectionable only insofar as they produce "unnatural" sound quality. As a rule, the most readily observed frequency irregularities are excessively long reverberation times at the extreme low and high frequencies, which produce room responses which can be described as "boomy" and "hissy", respectively. Differences in decay rate with frequency are also more easily heard when the average reverberation time is short than when it is long. As a practical matter, reverberation-frequency characteristics cannot always be completely controlled, as for instance in an auditorium with full audience, where the curve is largely fixed by the absorption of the people, and at very high frequencies by the absorption of the air. In broadcasting or recording studios, on the other hand, the reverberation-frequency curve can be adjusted as desired by proper choice of acoustical treatment.

Reverberation as heard by the ear or a microphone has thus far been considered as a smooth, continuous trail of sound dying out to inaudibility. Actually, the decay curve of sound pressure level fluctuates more or less widely about an average straight line, due to the interference pattern. The amount of fluctuation is much less for a mixture of frequencies than for a single frequency tone, and is also reduced by irregular room surfaces, where the dimensions of the irregularities are generally comparable to the wavelength.

A certain amount of fluctuation in decay is common to all rooms and is therefore considered desirable on the basis of "naturalness". Artificial reverberation should therefore provide for some decay irregularity.

Another type of transient room response is associated with the sound which reaches the ear or microphone after only one or two reflections, as shown in Figure 13. These

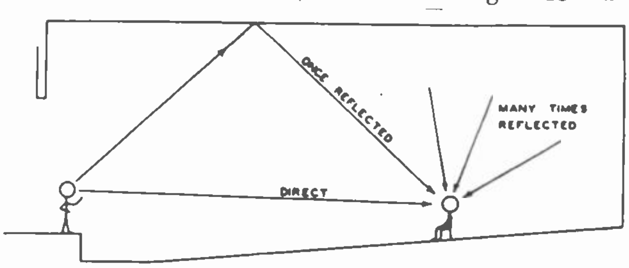


Fig. 13 – Components of sound received by a listener in an auditorium.

arrive immediately after the direct sound, and since they are at a high level and account for comparatively large changes in level at the beginning of the general decay, they may strongly affect the quality of hearing conditions. Probably the most familiar of these effects is an echo, which is defined as a single reflection which can be heard as a distinct repetition of the direct sound. In order for a single reflection to be heard as an echo, it is necessary first that the reflected sound arrive at the ear or microphone at least 1/17th second later than the direct sound, or in other words, that it travel a path from source to listener at least 65 feet longer than the path traversed by the direct sound. If the time lag is shorter than this, the reflection cannot be distinguished by ear, and will instead act as a reinforcement of the direct sound. Secondly, echoes are audible only if the reverberation time is short enough that the sound of the echo is not obscured by the general reverberation. Echoes in auditoriums are most often caused by rear walls and excessively high ceilings, since it is these surfaces which make possible the longest path difference between direct and reflected sound when the source is on the stage.

When caused by flat surfaces, echoes are considered undesirable insofar as they cause a distraction to attentive listening. Concave room surfaces, however, such as curved rear walls or domed ceilings, produce converging or focused reflections which result in greatly intensified echoes. Under the worst conditions, namely, where the ear or microphone is close to a strong sound focus, as

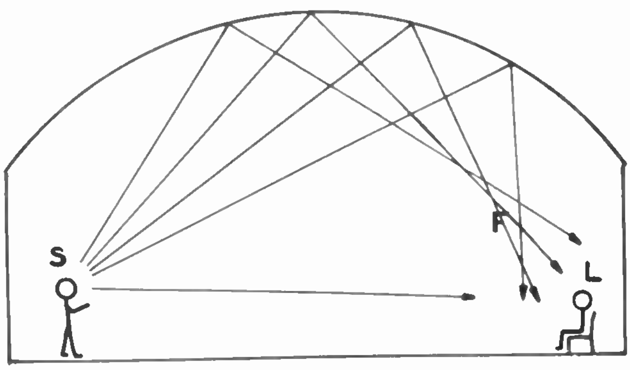


Fig. 14 – Diagram showing focused reflection from a concave ceiling.

shown in Figure 14, and the room is large enough to allow sufficient time lag between direct and reflected sound, echoes may be so severe that speech is unintelligible and listening to music is intolerable. To avoid such difficulties, extended curved surfaces should be laid out so that their centers are far removed from any sound pickup position (see Figure 15), or else should be broken up by large irregularities having dimensions of several feet. Heavy acoustical treatment is helpful, but not completely effective in severe cases.

In designing an auditorium, the shaping of room surfaces and the location and distribution of the acoustical treatment required for reverberation control can be utilized to some extent to take advantage of the first reflections. In general, absorption should be placed on those surfaces which produce the longest delayed first reflections to most listening positions. Such surfaces would be rear walls and excessively high ceilings. In theatres and auditoriums where direct or reproduced speech is at least partly involved, it is desirable to shape the walls and ceilings near the stage so that, as far as possible, first reflections will arrive at each listening position with a path difference of less than about 65 feet and from a direction close to that of the direct sound. These measures result in a marked reinforcement of the direct sound and a heightening of the auditory impression that all of the sound is coming from the stage. The latter effect is especially important in sound motion picture presentation. Obviously, surfaces which provide useful first reflections should not be acoustically treated, or at most lightly and in partial areas.

It is particularly important in a concert hall or scoring stage that the performers be surrounded with reflective surfaces on the stage itself. This is vitally necessary in permitting the conductor as well as each member to hear the group as a whole with the proper blending and balance.

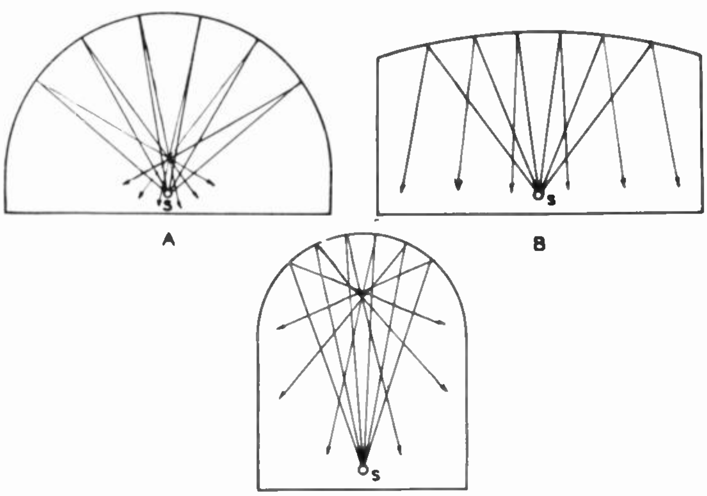


Fig. 15 – Reflection from concave ceilings of various radii. Center of curvature near floor line *A* produces most harmful effects. Conditions *B* and *C* are relatively harmless.

In auditoriums using loudspeakers for reinforcement of live speech, heavy rear wall absorption is of considerable help in preventing acoustic feedback to the microphone, and consequent "howling".

Control of first reflections is effective only in fairly large rooms, such as recording and audience broadcast studios and auditoriums seating at least a few hundred people. In smaller rooms, such as living rooms and small studios, all reflections arrive in such rapid succession that special control of first reflections produces no effects that can be readily appreciated by the ear.

The phenomenon of flutter echo should also be mentioned in connection with room design. This is a multiple reflection back and forth along a single line between two perfectly parallel reflecting surfaces. It can only be detected when the source and pickup are both on the same line perpendicular to the surfaces, and then only when the general room reverberation is short. As the name suggests, it is heard as a buzzing or ringing effect which is undesirable to the extent that it is disturbing or out of place. A flutter can also occur between two diagonally opposite room corners. It can be eliminated by putting the surfaces even very slightly out of parallel, or by making one or both of them absorptive. Serious trouble due to flutter is apt to occur only in small broadcast studios where the foregoing precautions have not been observed.

It is hoped that this brief outline of the essential principles of room acoustics will be helpful to those interested in this field. Reference to the literature and texts as well as personal experimentation and critical listening will establish a background of useful experience for handling the more subtle and involved problems which may be encountered.

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THE UNIAXIAL MICROPHONE*

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Summary—A small unidirectional microphone has been developed with the following features: Maximum sensitivity along the axis of the microphone; a high ratio of electrical output to size; a sharper directivity pattern than a cardioid; a directivity pattern that is independent of the frequency; a blastproof vibrating system. The high discrimination which this microphone exhibits to sounds which originate from the sides and rear makes it particularly suitable for long distance sound pickup in radio, television, sound motion pictures, and sound reinforcing systems.

INTRODUCTION

The general trend in microphones for use in television is in the direction of smaller and more unobtrusive units. One of the first high-quality units of this type was the small "Bantam" velocity microphone.[1] Another example is the "Starmaker," a nondirectional pressure ribbon microphone.[2] However, for boom use and other long distance pickup applications a directional microphone is required. The "77D" unidirectional microphone is almost universally used for these applications. This microphone was developed almost ten years ago. Over this period, research and development work has been carried on with the objective of improving the acoustic and magnetic systems of microphones. An example, incorporating some of these developments, is the "Starmaker" microphone referred to above. A review of the unidirectional microphone indicated that it would be possible to reduce the size and improve the directivity. Furthermore, a new requirement was a blastproof vibrating system capable of withstanding blast from guns, pistols and small explosions. This review also indicated that a blastproof feature could be incorporated. The vibrating system which appeared to be the most suitable for obtaining the above enumerated features was one similar to that used in the Starmaker. In the unidirectional form this system has been termed the uniaxial microphone because the maximum sensitivity corresponds to the axis of the system. It is the purpose of this paper to describe the uniaxial microphone.

THEORY

The motor selected for this microphone is shown in Figure 1. Among the advantages of this motor is the simple and efficient magnetic structure. Simplicity is accomplished by the use of a small number of easily machined parts. High magnetic efficiency is obtained due to the small leakage inherent in magnetic designs of this type.

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[1] L. J. Anderson and L. M. Wigington, "The Bantom Volicity Microphone," *Audio Eng.*, vol. 34, p. 13; January, 1950.

[2] H. F. Olson and J. Preston, "Unobtrusive Pressure Microphone," *Audio Eng.*, vol. 34, p. 18; July, 1950.

The next consideration is a blastproof system. A typical sound wave produced by the firing of a gun is shown in Figure 2. The total time depends upon the type of gun. It appears to range from 1/20 to 1/40 of a second from a 0.45- to a 0.22-caliber gun. The low-frequency components of this wave are not reproduced through the complex chain of elements which constitute the sound

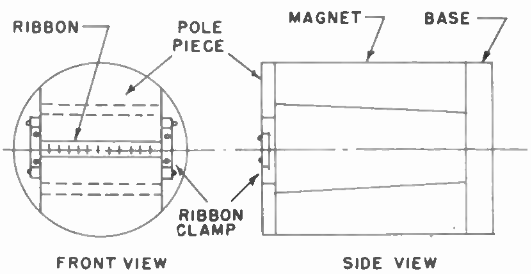

Fig. 1 – Front and side views of the motor of the uniaxial microphone.

channel in recording or broadcasting. However, these low-frequency components in the blast pulse produce the large deflections in the ribbon and stress it beyond the elastic limit and thereby introduce a permanent deformation in the ribbon. What is required is a system which will reduce the low-frequency amplitude of the sound pressure delivered to the ribbon.

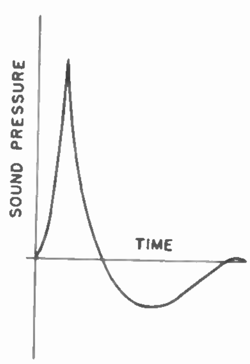

Fig. 2 – A graph of the sound wave produced by the firing of a gun.

The sound pressure delivered to the ribbon in the low-frequency range can be reduced by the use of an acoustical resistance placed over each side of the ribbon as shown in Figure 3. The acoustical curcuit is also shown in Figure 3. The components of the acoustical impedance of the elements of the system shown in Figure 3 are shown in Figure 4. These components are as follows:

The acoustical resistance is

$$r_A = r_{A_1} + r_{A_2}, \tag{1}$$

where r_{A_1} = acoustical resistance of the screen on the front of the ribbon,

r_{A_2} = acoustical resistance of the screen on the back of the ribbon.

The positive acoustical reactance of the ribbon is given by

$$X_{AM} = 2\pi fM, \tag{2}$$

where M = inertance of the ribbon.

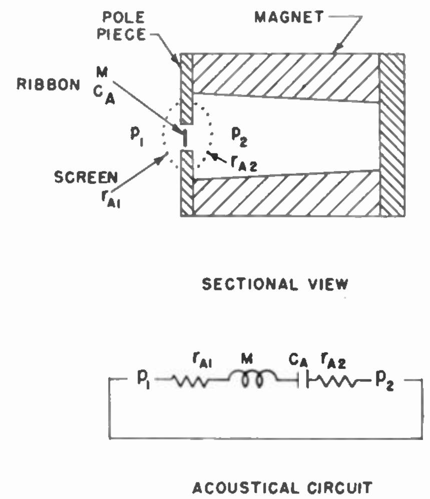

Fig. 3 – Sectional view and acoustical circuit of a simplified vibrating system of a uniaxial microphone to show the effect of the blast baffles. In the acoustical circuit: p_1 is the sound pressure on the front of the microphone; r_{A_1} and r_{A_2} the acoustical resistances of the blast baffles; M and C_A the inertance and acoustical capacitance of the ribbon; and P_2 the sound pressure on the back of the microphone.

The negative acoustical reactance of the ribbon is given by

$$X_{AC} = \frac{1}{2\pi fC_A}, \tag{3}$$

where C_A = acoustical capacitance of the ribbon.

The volume current in the system of Figure 3 is given by

$$\dot{X} = \frac{p_1 - p_2}{r_A + jX_{AM} + jX_{AC}}, \tag{4}$$

where p_1 = pressure on the front of the ribbon, and

p_2 = pressure on the back of the ribbon.

The driving pressure, $p_1 - p_2$, is proportional to the frequency. The reason is as follows: The magnitudes of p_1 and p_2 are the same. However, the phase difference between p_1 and p_2 is proportional to the frequency. Therefore, in the frequency range in which the distance between the front and back is small compared to the wavelength, the difference in pressure between the front and back will be proportional to the frequency.

Referring to Figures 3 and 4 it will be seen that the low-frequency response will be attenuated by the addition of the acoustical resistances. The response–frequency characteristics with and without the acoustical

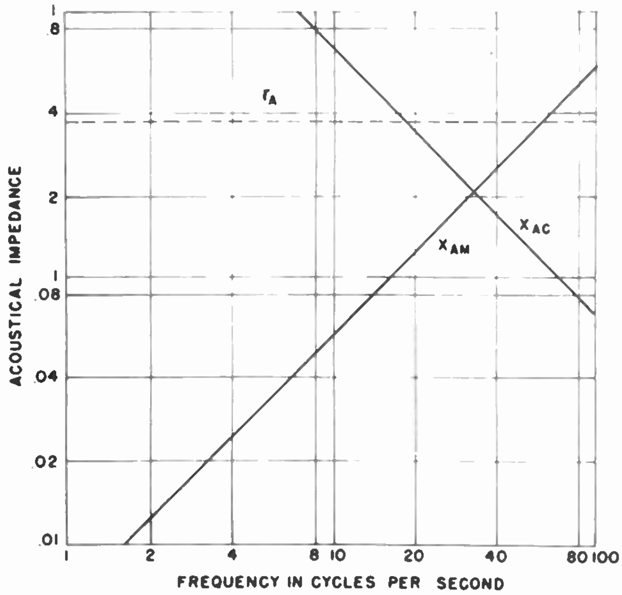

Fig. 4 – The components of the acoustical impedance of the system of Fig. 3. r_A is the total acoustical resistance, x_{AM} the positive acoustical reactance, and x_{AC} the negative acoustical reactance.

resistances are shown in Figure 5. These characteristics show the high attenuation in the low-frequency region due to the addition of the acoustical resistance.

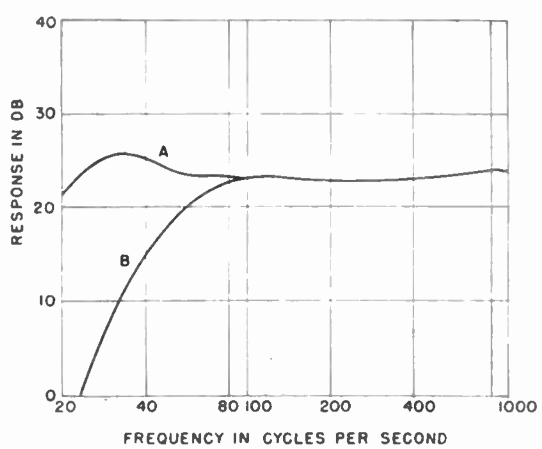

Fig. 5 – Response frequency characteristic of the system of Fig. 3: A. Without acoustical resistance; B. With acoustical resistance.

A sectional view of the complete vibrating system is shown in Figure 6. The magnetic system is similar to the one shown in Figure 1. The ribbon is connected to the damped folded pipe or labyrinth by means of a connector which couples the rectangular cross-sectional area at the ribbon to the circular cross-sectional area at

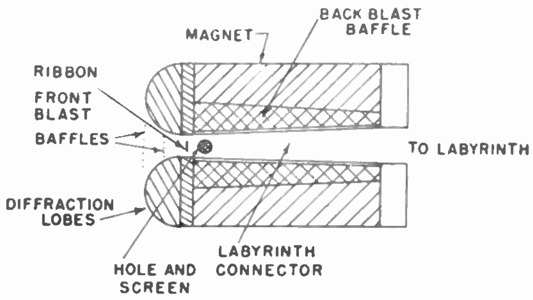

Fig. 6 – A sectional view showing the elements of the motor of the uniaxial microphone.

the labyrinth. The connector is provided with two holes, one on each small side near the connection to the ribbon. These two holes provide the essential portion of the phase-shifting acoustical network so that the directional pattern will be of the unidirectional type. The front face of the microphone is equipped with two lobes. The lobes perform three functions: the reduction of the deleterious effects of diffraction, the accentuation of the high-frequency response, and the support of the blast baffles. The front of the microphone is equipped with two blast baffles held in place by the lobes. The side of the microphone is equipped with a single blast baffle. The action of the different elements and the complete microphone will be described in the discussion which follows.

The front face of the microphone is circular. Therefore, the response will be nonuniform in the high-frequency range due to the diffraction effects produced at the face of the cylinder. The sound-pressure response-frequency characteristic at the center of the cylinder for normal incidence of the impinging sound wave of uniform sound pressure in free space is shown in Figure 7A. The response-frequency characteristic of the system shown in Figure 6 operating as a pressure microphone and without the lobes is shown in Figure 7B. The deviation of this response-frequency characteristic from that of Figure 7A is due to the fact that the ribbon covers a length of about one inch on the surface rather than a single point at the center. The response-frequency characteristic of the system of Figure 6 operating with the lobes is shown in Figure 7C. The addition of the lobes reduces the effects of diffraction. The net result is a smoother response-frequency characteristic. The lobes also act as a small horn which accentuates the response in the extreme high-frequency range.

A sectional view and acoustical network of the complete microphone are shown in Figure 8. It is of the single-ribbon type in which the back of the ribbon is coupled to a damped pipe and an inertance in the form of an aperture in the pipe.

The action of the microphone can be obtained from a consideration of the acoustical network. The sound pressure on the open side of the ribbon may be written

$$p_1 = p_0 \epsilon^{j\omega t}, \tag{5}$$

where p_0 = amplitude of the pressure, in dynes, per square centimeter,

ω = $2\pi f$,

f = frequency, in cycles per second,

t = time.

The sound pressure acting upon the air load of the aperture and the aperture may be written

$$p_2 = p_0 \epsilon^{j(\omega t + \phi_1)}, \tag{6}$$

where ϕ_1 = phase angle between the pressure p_1 and the pressure p_2.

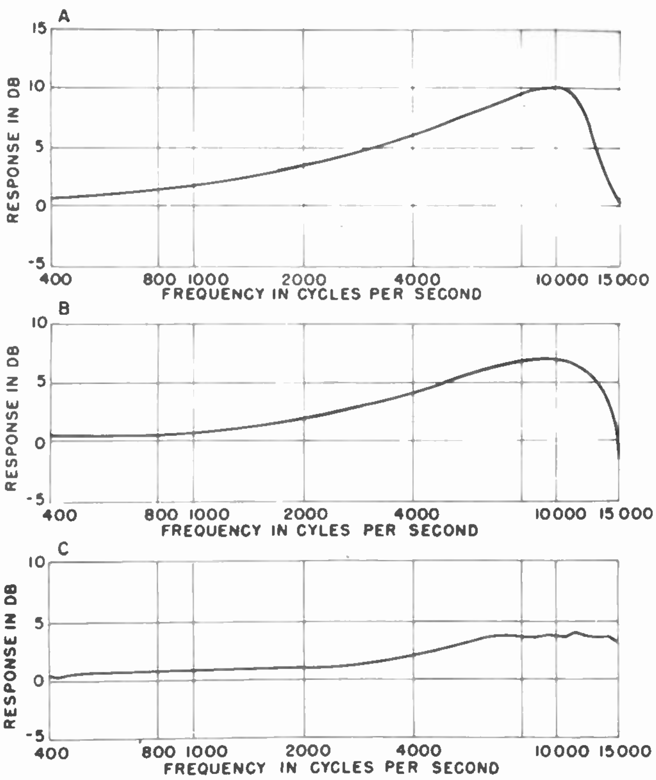

Fig. 7 – A. The response at the center of a cylinder; B. The response of the uniaxial microphone without the diffraction lobes; C. The response of the uniaxial microphone with the diffraction lobes.

Referring to Figures 6 and 8 it will be seen that there is a cavity between the magnets filled with damping material. This cavity is coupled to the aperture as shown in the acoustical network. This cavity is also acted upon by a sound pressure which may be written

$$p_3 = p_0 \epsilon^{j(\omega t + \phi_2)} \tag{7}$$

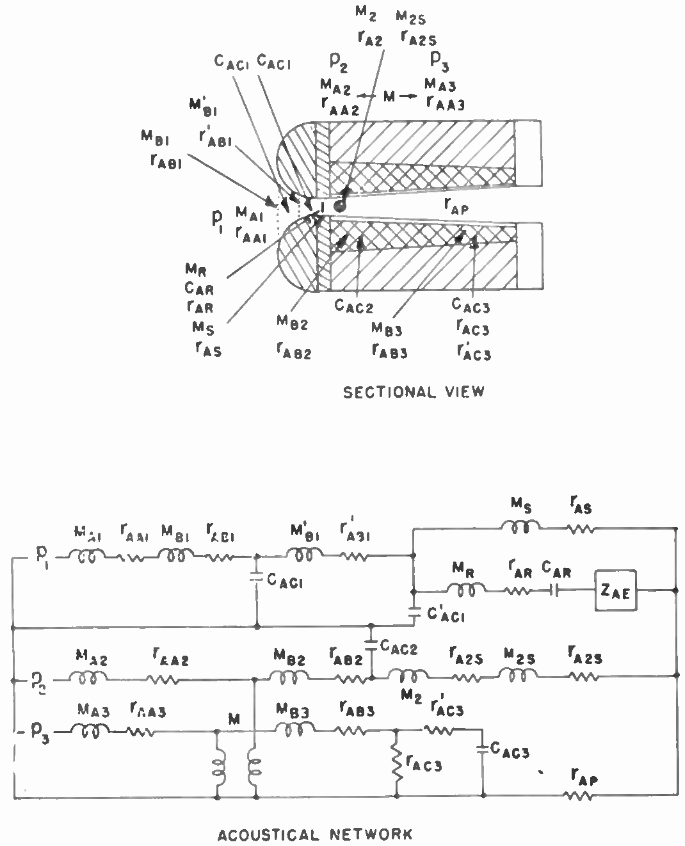

Fig. 8 – Sectional view and acoustical network of a uniaxial microphone. In the acoustical network: p_1 is the sound pressure on the front of the microphone; M_{A_1} and r_{AA_1} the inertance and acoustical resistance of the air load on the front of the microphone; M_{B_1}, r_{AB_1}, M'_{B_1}, and r'_{AB_1} the inertances and acoustical resistances of the blast baffles on the front of the microphone; C_{AC_1} and C'_{AC_1} the acoustical capacitances of the volumes between the blast baffles; M_S and r_{AS} the inertance and acoustical resistance of the slit between the ribbon and pole pieces; M_R, r_{AR}, and C_{AR}, the inertance, acoustical resistance and acoustical capacitance of the ribbon; Z_{AE} the acoustical impedance due to the electrical circuit; p_2 the sound pressure at the apertures in the labyrinth connector; M_{A_2} and r_{AA_2} the inertance and acoustical resistance of the air load at the apertures of the labyrinth connector; M_{B_2} and r_{AB_2} the inertance and acoustical resistance of the blast baffles on the side of the microphone; C_{AC_2} the acoustical capacitance of the volume behind the blast baffle; M_{2S} and r_{A_2S} the inertance and acoustical resistance of the screen covering the hole in the labyrinth connector; M_2 and r_{A_2} the inertance of the hole in the labyrinth connector; r_{AP} the acoustical resistance of the labyrinth; p_3 the sound pressure at the damped cavity between the magnets; M_{A_3} and r_{AA_3} the inertance and acoustical resistance of the air load upon the damped cavity; M_{B_3} and r_{AB_3} the inertance and acoustical resistance of the blast baffle over the damped cavity; C_{AC_3}, r_{AC_3} and r'_{AC_3} the acoustical capacitance and acoustical resistances of the cavity between the magnets; M the coupling between the cavity and the apertures.

The phase angles ϕ_1 and ϕ_2 are a function of the angle of the incident sound as follows:

$$\phi_1 = \Phi_1 \cos\theta, \tag{8}$$

and

$$\phi_2 = \Phi_2 \cos\theta, \tag{9}$$

where θ = angle between the normal to the surface of the ribbon and the direction of the incident sound wave,

Φ_1 = constant phase angle for the pressure p_2, and

ϕ_2 = constant phase angle for the pressure p_3.

The coupling M between the aperture and the damped cavity is complex function of the dimensions and configuration of the cavity and the relation of the cavity to the aperture. In addition, the coupling is a function of the direction of the incident sound. Under these conditions the acoustical network of Figure 8 is extremely

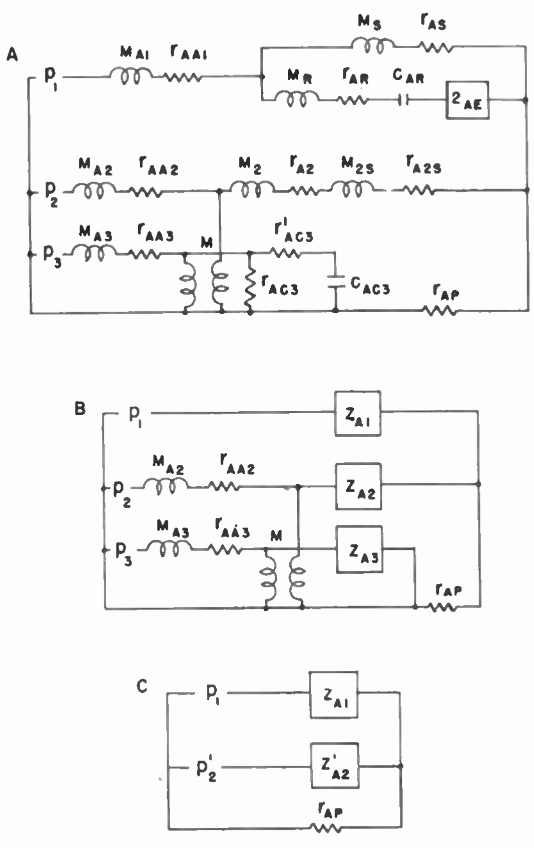

Fig. 9 – A. Acoustical network of Fig. 8 with the blast baffles removed; B. Acoustical network of A with the equivalent lumped acoustical impedances; C. The approximate equivalent acoustical network of B with the mutual coupling system and associated acoustical impedances replaced by a single pressure and acoustical impedance.

complex. If the blast baffles are removed, the acoustical network is reduced to the one shown in Figure 9A. A further simplification can be obtained by lumping the elements into the impedances as shown in Figure 9B. The acoustical network of Figure 9B can be reduced to the acoustical network of Figure 9C in which the pressure

$$p'_2 = f_1(p_2)\, f_2(p_3)\, f_3(\theta), \tag{10}$$

and the acoustical impedance

$$z'_{A_2} = f_4(z_{A_2})\, f_5(z_{A_3}). \tag{11}$$

In the acoustical network of Figure 9C, the volume current in the acoustical impedance z_{A_1} due to pressure p_1 is

$$\dot{X}_1 = \frac{p_1(z'_{A_2} + r_{AP})}{z_{A_1}z'_{A_2} + z_{A_1}r_{AP} + z'_{A_2}r_{AP}}. \tag{12}$$

The volume current in the acoustical impedance z_{A_1} due to the pressure p'_2 is

$$\dot{X}_2 = \frac{p'_2 r_{AP}}{z_{A_1}z'_{A_2} + z_{A_1}r_{AP} + z'_{A_2}r_{AP}}. \tag{13}$$

The resultant volume current in the acoustical impedance z_{A_1} is the difference between the volume currents

$$\dot{X}_R = \dot{X}_1 - \dot{X}_2. \tag{14}$$

If the volume current through the slit between ribbon and the pole is negligible, the volume current $\dot{X}_R$ is the volume current of the ribbon.

The velocity of the ribbon is

$$\dot{x}_R = \frac{\dot{X}_R}{A_R}, \tag{15}$$

where A_R = area of the ribbon.

The voltage generated by the motion of the ribbon is given by

$$e = Bl\dot{x}_R, \tag{16}$$

where B = flux density in the air gap, and l = length of the ribbon.

The performance of the microphone can be predicted from (5) to (16) inclusive. The directivity pattern of the uniaxial microphone shown in Figure 6, under a given set of constants, can be approximately expressed as

$$e = \left(K \quad 0.3 + 0.7\cos\theta\cos\frac{\theta}{3}\right), \tag{17}$$

where K = sensitivity constant of the microphone.

The directivity pattern of the uniaxial microphone obtained from (17) is shown in Figure 10.

It is interesting to compare the directional patterns of the uniaxial microphone with the conventional unidirectional microphone. If the damped cavity is omitted, then

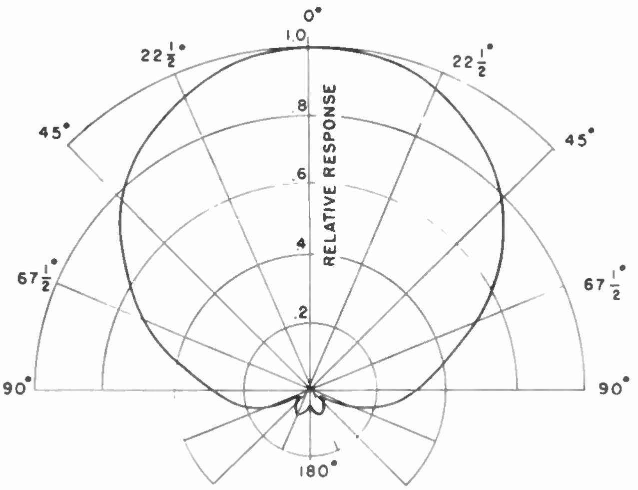

Fig. 10 – Theoretical directional pattern of the uniaxial microphone.

the microphone becomee the conventional unidirectional type with the acoustical network of Figure 11A. The acoustical network of Figure 11A can be reduced to the acoustical network of Figure 11B.

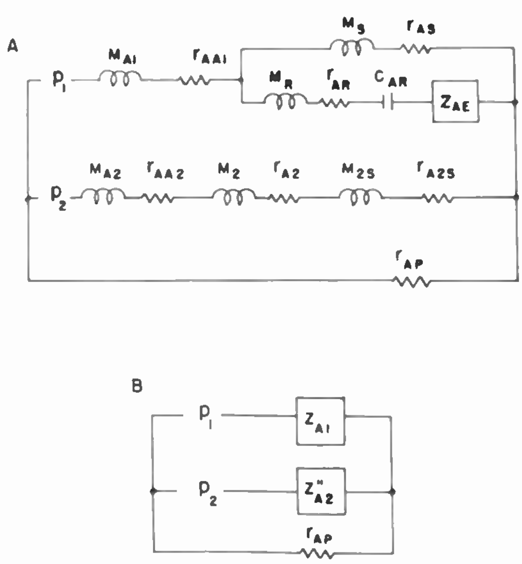

Fig. 11 – A. The acoustical network of the acoustical system of Fig. 9 with the damped cavity omitted; B. The equivalent acoustical network of A with the equivalent lumped acoustical impedances.

The resultant volume current through the acoustical impedance z_{A_1} is given by

$$\dot{X}_R = \frac{p_1(z''_{A_2} + r_{AP}) - p_2 r_{AP}}{z_{A_1} z''_{A_2} + z_{A_1} r_{AP} + z''_{A_2} r_{AP}} . \qquad (18)$$

The voltage generated by the ribbon can be obtained from (15), (16), and (18). The directivity pattern of this microphone is given by

$$e = K\,(a + b\cos\theta)\,,$$

where K = sensitivity constant of the microphone, a and b = limaçon constants, $a + b = 1$.

A few of the directivity patterns which can be obtained from this microphone are shown in Figure 12.

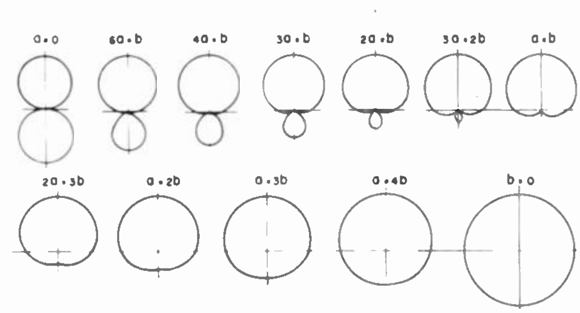

Fig. 12 – A few of the single infinity of directional characteristics obtainable with the polydirectional microphone.

Comparing Figures 10 and 12 it will be seen that it is not possible to obtain, in the conventional microphone, the same directivity in the front hemisphere as that in the uniaxial microphone without sacrificing high discrimination in the rear hemisphere. For example, referring to Figure 12, the directivity pattern of the unidirectional microphone in the front hemisphere for $3a = b$ is approximately the same as the uniaxial microphone in the front hemisphere. However, the discrimination in the rear hemisphere for the unidirectional microphone for the same constants is only 6 decibels for 180 degrees as compared to 26 decibels for the uniaxial microphone.

PERFORMANCE CHARACTERISTICS

The response frequency characteristics of the uniaxial microphone for the angles 0°, 45°, 90°, 135° and 180°, where 0° corresponds to the axis of the microphone are shown in Figure 13. The average polar directivity pattern of the uniaxial microphone is shown in Figure 14.

In order to show the effect of the damped cavity upon the directivity pattern, the cavity was covered with thick sheets of copper. The directivity pattern obtained under these conditions is shown in Figure 15. The directivity pattern under these conditions is slightly broader in the front hemisphere. Also, a large lobe has now appeared in the rear hemisphere. The directivity pattern shown in Figure 15 is not satisfactory as a unidirectional microphone because of the inadequate discrimination in the rear hemisphere. With the cavity covered, the holes in the connector were reduced in size until the

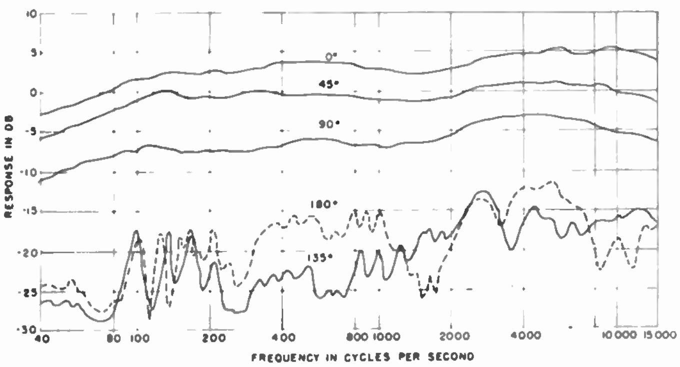

Fig. 13 – The measured response frequency characteristics of the uniaxial microphone sound incident at 0°, 45°, 90°, 135° and 180°. 0° corresponds to the axis of the microphone.

cardioid pattern shown in Figure 16 was obtained. This is the typical pattern for a unidirectional microphone with high discrimination against sound coming from the rear. Now the directivity pattern is much broader in the

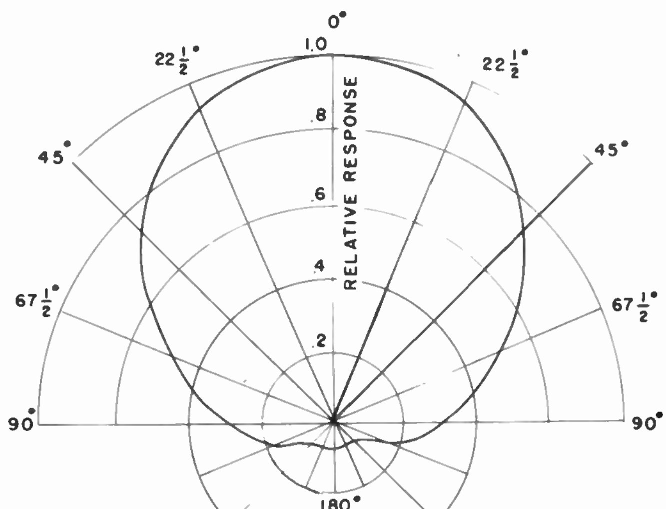

Fig. 14 – The average of the measured directional patterns of the uniaxial microphone.

front hemisphere. The response of a unidirectional microphone with a cardioid directivity pattern to random sounds, with all directions equally probable and of equal strength, is 1/3 that of a nondirectional microphone. The response of the uniaxial to random sounds is 1/5 that of a nondirectional microphone. This means that the response of the uniaxial microphone to random sounds is 60 per cent that of a microphone with a cardioid pattern. From the standpoint of sound pickup distance, the uniaxial microphone will operate at 30 per cent greater distance for the same reverberation or undesirable sounds or noise.

Tests have been made of the effectiveness of the blastproofing. The uniaxial microphone will stand the blast of a 0.45 pistol firing blanks at a distance of 5 feet indoors, with the direction of firing at right angles to the

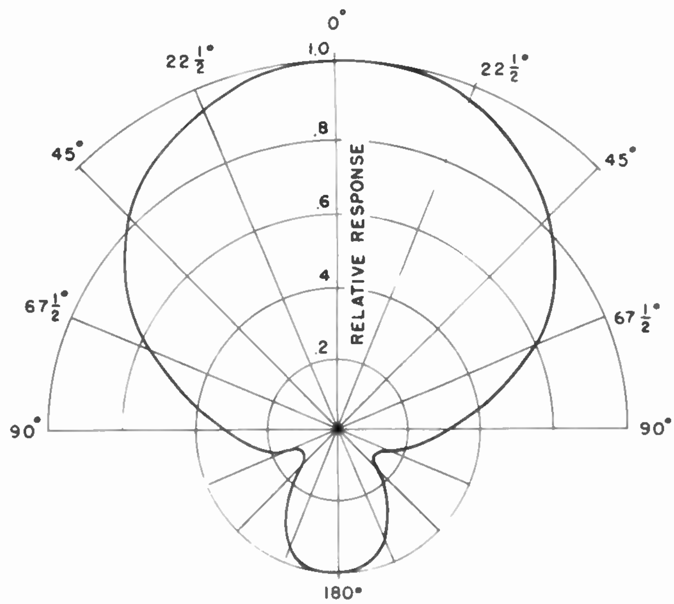

Fig. 15 – The average of the measured directional patterns of a conventional unidirectional microphone set for the pattern $e = .3 + .7 \cos \theta$.

microphone. It will withstand the same blast at a smaller distance outdoors. However, a distance of 5 feet should be ample for practically all conditions of use because the effective operating pickup distance of the uniaxial microphone is greater than the conventional unidirectional microphone.

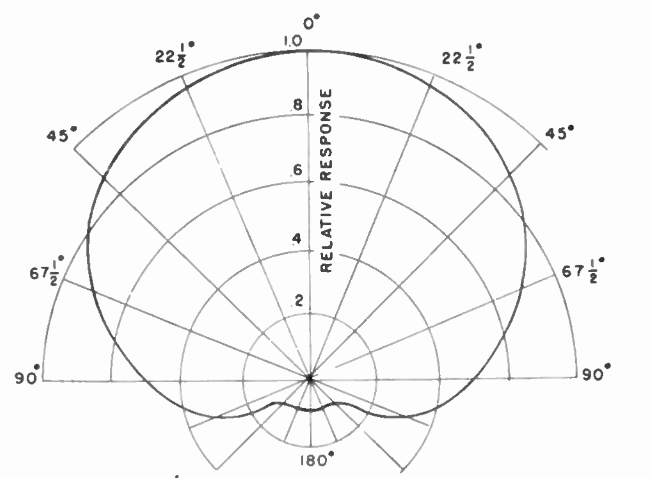

Fig. 16 – The average of the measured directional patterns of a conventional unidirectional microphone set for a cardioid pattern.

APPLICATIONS

One of the applications for this microphone is the pickup of sound in television where the microphone is mounted on a boom and kept out of the picture. The directional properties of the microphone are particularly suitable for this application because the discrimination against unwanted sounds which originate at the sides and the rear is very high. A photograph of the cradle which has been developed for this microphone for use on a boom is shown in Figure 17. The cradle design provides an underslung mounting of the microphone so that the microphone is the lowest part of the system. This condition allows for minimum distance between the source of sound and the microphone.

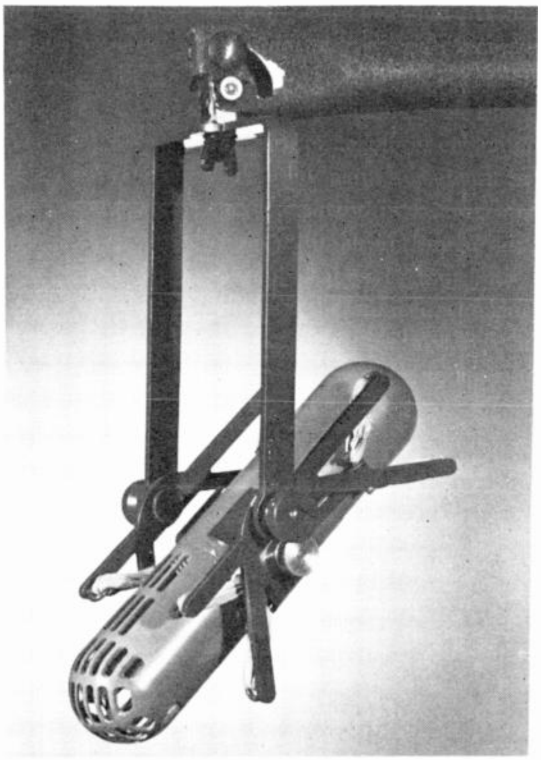

Fig. 17 – The uniaxial microphone mounted in a cradle for boom operation.

Other applications for this microphone are the pickup of sound in television where the microphone is seen in the picture. Either floor or desk stands are used for these applications. A desk stand is shown in Figure 18. The uniaxial microphone is particularly suited for these applications because the maximum response occurs on the axis. Under these conditions the projected area of the microphone is a minimum as viewed by the camera.

Fig. 18 – The uniaxial microphone mounted on a desk stand.

The uniaxial microphone is also useful for standard broadcast and sound reinforcing applications where the unidirectional microphone has already been established as the solution for sound pickup problems. It has been extensively field tested, including use at the 1952 political conventions.

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