The Audio Engineering Society

A group of forward-thinking audio men have banded together to stimulate and improve audio standards.

By LEWIS S. GOODFRIEND
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THE *Audio Engineering Society,* organized in February of this year at New York, has already made rapid strides in bringing together members of the audio profession for the purpose of discussing, organizing the existing knowledge, and promoting the intelligent study of problems in the field of audio engineering. Since most of the readers of RADIO NEWS design or operate audio equipment, the society and its activities are of essential interest. It is with this in mind that the following review is presented.

For the past few years men engaged in audio engineering have found themselves eligible for membership in one of four major professional societies, but in each of these societies "audio" was merely a minor item. Papers and reports on audio subjects seldom found their way into the journals or on the agenda at meetings of these societies. It is a matter of common knowledge that technical developments in the audio field have gone far beyond the material currently available. In order to offset this situation prominent audio men have, from time to time, organized small local groups where they could discuss audio and exchange ideas, the most prominent of these organizations being the *Sapphire Group* in Hollywood. A few publications, among them RADIO NEWS, published articles on audio, but this was an exception, not the general case. However, it was not until a year ago, in the summer of 1947, that any action to form a professional society for audio men received any great impetus. It was at that time that C. J. LeBel, New York consultant, first began to discuss the need for such a society with other prominent members of the audio field. Audio development had made great strides, yet many professional and scientific journals gave even less space than previously to the subject. Mr. LeBel's activity was rewarded by the formation of a committee including four other audio men. They are Norman C. Pickering, John D. Colvin, Charles G. McProud, and C. A. Rackey. It was this committee, whose members took time from their professional duties, that finally succeeded in getting the society started.

In January the committee organized a meeting of audio engineers in the New York area, and on February 17 over 200 audio men met and listened to discussions by LeBel and Pickering on the needs and aims of a professional society of audio engineers. The group in attendance agreed with the two speakers and a motion was carried to form the *Audio Engineering Society.* In order to carry forward the business of the new society, the original committee was empowered to continue its activity.

Since its formation, the society has adopted a constitution and By-Laws, elected officers, and held three technical meetings. Officers of the new Society are the members of the original committee: Mr. LeBel as President; with Mr. Pickering as Secretary, and with the addition of R. A. Schlegel as Treasurer.

The first Technical Meeting featured a paper by Dr. Harry F. Olson, of the *RCA Laboratories,* Princeton, New Jersey. Dr. Olson spoke on "Problems of High Fidelity Reproduction," describing three specific problems involving phonograph transducers, noise suppressors, and loudspeakers. He illustrated his paper with slides showing the dynamical analogies involved, and a demonstration of the equipment resulting from his solution of the problems. At the May Meeting of the Society Herschel Toomin, noted for his work in the study of magnetic recording media, delivered a paper describing a new sweep frequency generator capable of showing the frequency response of an audio system on an oscilloscope screen with true logarithmic frequency scale, and either a linear or decibel scale for vertical presentation. The instrument described has frequency markers and variable high and low frequency limits so that it may be used for specific narrow-band audio systems, as well as wide range equipment, without modification.

The third technical paper presented to the Society was by Colonel Richard H. Ranger, President of *Rangertone,* and Chairman of the Society's membership committee. Colonel Ranger spoke on various problems of the magnetic recording field, and indicated that theory has not yet come abreast of experimental work. However, he pointed out that through experimental work, an extremely high quality magnetic system is possible.

What exactly is the need for any professional Society? The members of the *Audio Engineering Society* decided that there was a need for their organization for the following reasons: "The diffusion and increase of educational and scientific knowledge in audio engineering and the promotion and advancement of this science and its allied arts in both their theoretical and practical applications, and the stimulation of interest in audio engineering, the interchange and intercourse of ideas and the promotion and maintainance of high professional standards among its members." In order to carry out these aims, the Society must hold meetings for the reading of professional papers, publications, communications, and such other professional activity as is necessary.

Among other things the members feel that the Society must remain independent of any other organization or any commercial interest, and although commercial groups may become sustaining members, make donations of space or material, or act to publish the Society's activities, they may have no vote in the operation and conduct of the Society's affairs.

Among the branches of audio engineering in which the Society has a large number of members and about which there is little published information are: speech input system, studio design and use, disc recording, tape and wire recording, hearing aids, public address systems, high quality home reproduction systems, wired music systems, and telephone engineering.

Besides these subjects to be covered by the Society's activity, there are certain obligations that the Society has to the art, including the development of a public appreciation of audio engineering as a separate art, the fostering of research on basic problems in the field, facilities for publication of papers and the holding of meetings on audio engineering, the fostering of education and guidance of education in the audio engineering field, along with the preparation of standards, cooperative research, and standards work not only in audio but with other groups working in closely allied fields. It is with these ideas in mind that the Executive Committee is proceeding with its work. Plans under way call for a meeting of the national organization in October, and the preparation of material to assist in the organization of local sections throughout the country. It is hoped that the local sections will be able to carry on in the fall in a manner similar to the New York section this past spring, holding technical meetings, panel discussions, and contributing papers to the Society's own publication which will be arranged for the fall.

ED. NOTE: *There is no official Society publication, nor affiliation with any magazine at this writing.*

Membership in the *Audio Engineering Society* is open to men of all educational and professional levels from the student through the professional design engineer and manufacturing executive.

Inquiries concerning membership or membership applications should be addressed to Norman C. Pickering, Secretary, Box F, Oceanside, N. Y. —30—

BELL LABS INTRODUCES A REVOLUTIONARY COMPONENT

BELL Telephone Laboratories has demonstrated a new device which may have far-reaching significance in electronics and electrical communications.

Known as the "Transistor," the new unit is capable of performing efficiently nearly all of the functions of an ordinary vacuum tube. The "Transistor" works on an entirely new physical principle which was discovered in the course of fundamental research into the electrical properties of solids.

The whole apparatus is housed in a tiny cylinder less than an inch long. It will serve as an amplifier or an oscillator, yet bears almost no resemblance to the vacuum tube now used to do these jobs. It has no vacuum, no glass envelope, no grid, no plate, no cathode, and therefore no warm-up delay.

Two hair-thin wires touching a pinhead of a solid semi-conductive material soldered to a metal base, are the principal parts of the "Transistor." These are enclosed in a simple, metal cylinder not much larger than a shoelace tip.

Tests have shown that the "Transistor" will amplify at least 100 times (20 db.). Some test models have been operated as amplifiers at frequencies up to 10 million c.p.s.

Key investigations which led to the development of the "Transistor" were carried out by Dr. John Bardeen and Dr. Walter H. Brattain. The general research program leading to the development was initiated and directed by Dr. William Shockley.

Bell Laboratories advises that the device is still in the experimental stage and that no data on cost is presently available. —30—

Relative size of the new Bell Labs "Transistors" can be seen by comparison to pencil.

Fig. 1. A portable audiometer with bone conduction receiver at center and air conduction receiver at right. The small knob atleft is for calibration. The large knobs are for frequency and intensity. (Courtesy Bell Telephone Laboratories)

Problems in Audio Engineering

LEWIS S. GOODFRIEND*

Part III. Articulation, its measurement, and its relation to intelligibility in the determination of the degree of deafness.

When the sounds of speech are joined to form words they are said, by definition, to be articulated and the process is known as articulation. This word—articulation—has been borrowed by audio and communications engineers to describe the ability of a system to transmit articulated speech sounds without reducing the listener's ability to understand them. In adapting the word it has been convenient to express the degree of understanding as the percentage of the original speech sounds fed into the system. This appears to be easy, but after more than thirty years of use of the method there is no standard technique available to determine the percentage of articulation. All of the common systems are similar, and it will suffice to describe only the fundamentals here.

Basically an articulation test consists of a reader who actually makes the speech sounds, the system to be tested, and a listener to record the sound as transmitted by the system. In order to obtain a quantitative result special lists of speech sounds and words have been prepared. In general several lists read by several readers and recorded by several listeners are used.

The types of sounds that go into a list such as the Bell Telephone Laboratories' "The Standard Articulation Testing List" are carefully selected to include all the sounds used in speech. These sounds are combined into vowel-consonant, consonant-vowel, and consonont-vowel-consonant forms. These tests, however, can be used only with trained observers. It is therefore necessary to have lists of English words, of either one or two syllables, and in groups such that they will include all the basic speech sounds. Using only the combined speech sounds it is possible to obtain more information in a shorter time than with word lists, but with the word lists, unskilled observers and callers may be used with the possible introduction of memory and association effects that may cause error.

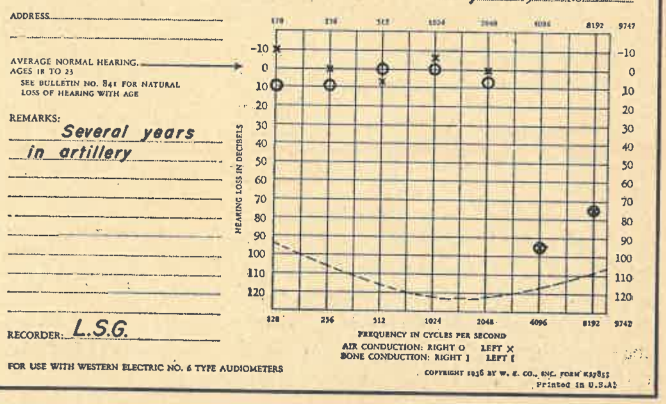
In the Bell Lab's speech sounds list we find groups like hūs, moush, ár and fōn. In the list of monosyllablic words we find these: tie, thy, by, wing, high, wick, and so on. It can be seen that, with either of these or with the new phonetically-balanced word lists prepared by the Psycho-Acoustic Laboratory, if the degree of articulation of the system is poor, many of the sounds will be confused with others and hence the percentage articulation will be low.

Intelligibility Test

Another form of test that is similar to the articulation test is the intelligibility test. In this type test the listener is called upon to derive intelligence from a sentence, either interrogative or imperative. In either case a single word or mark acts as an indication of understanding or intelligibility. However in this type of test the listener receives cues to hard sounds from context. Nevertheless, it appears to be a fairer test of the usefulness of a piece of sound transmission apparatus, since that apparatus will be used to transmit intelligence, not disconnected sounds.

It is only natural that the articulation and intelligibility tests are used to study the characteristics of rooms—which are actually communications systems involving acoustical paths—and in the study of deafness, which involves both acoustical and psychological paths. To see how they are applied in this latter case we must first understand deafness.

Fig. 2. Audiogram of a subject with nerve deafness. High (95db) hearing loss above 2 kc is indicated.



AUDIOGRAM OF S. T. Bronson AGE 42 DATE 6/7 19 42 NO.

ADDRESS

AVERAGE NORMAL HEARING, AGES 18 TO 23

SEE BULLETIN NO. 841 FOR NATURAL LOSS OF HEARING WITH AGE

REMARKS: Several years in artillery

RECORDER: L.S.G.

FOR USE WITH WESTERN ELECTRIC NO. 6 TYPE AUDIOMETERS

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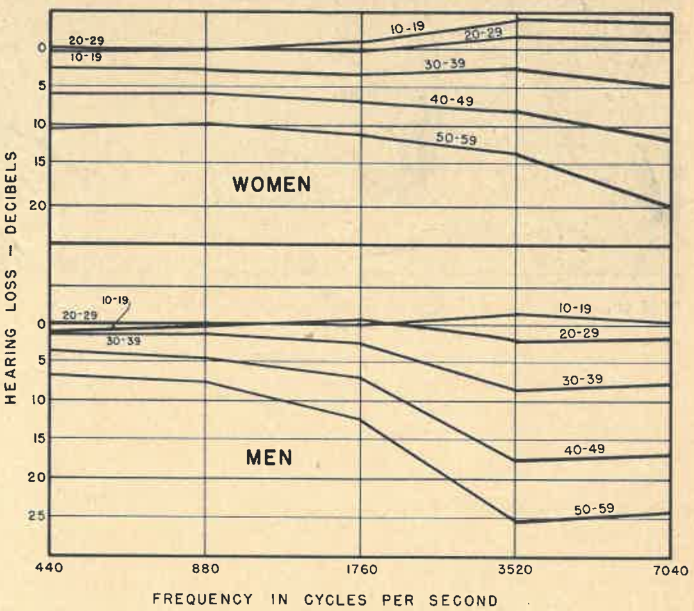
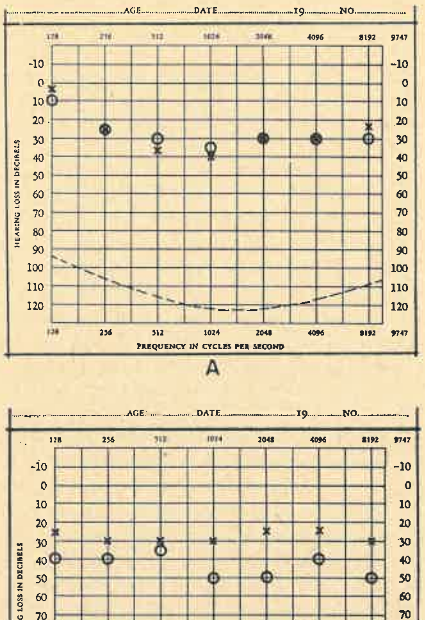


Fig. 4. Results of hearing tests conducted at 1939 - 1940 World's Fair. Hearing loss vs frequency is plotted for various age groups. *(After Steinberg, Montgomery and Gardner; courtesy J. Acous. Soc. Am. and Bell Telephone Laboratories)*

Deafness is the condition that exists when a listener does not perceive sounds of normal character and intensity but has a threshhold of hearing which is considerably above the normal, or when a listener is unable to understand speech at normal levels. Persons who have difficulty in the perception of normal speech and music are assumed to be deaf, but there are many people who are partially deaf and, although their personal efficiency and possibly their social adequacy has been impaired, they are not aware of the condition. To determine qualitatively whether one is deaf it is necessary to measure his threshhold of hearing, and his scores for articulation or intelligibility tests.

There are three common types of deafness: nerve, conduction, and central deafness. In nerve deafness, some of the sensory cells or nerve fibres of the inner ear may degenerate, and a loss of hearing in the frequency region associated with those nerve or cell tissues will result. In conduction deafness, however, the nerve cells are intact but there is some obstruction in the acoustical path which reduces the intensity of the signal as it passes through the outer and middle ear. This can result, for example, when wax blocks the ear canal or gets on the ear drum, or in otosclerosis when a bony material forms about the ossicles and inhibits their motion. In central deafness the nerves and path to the inner ear may function normally, but the brain itself does not take cognizance of the signals from the auditory nerve. This may be the situation in cases of psychological deafness such as battle-fatigue, and in cases of brain diseases or injury.

Fig. 3. Two cases of conduction deafness. Subject of (A) did not require an aid, while subject of (B) did.



Threshold Measurement

In order to measure an individual's ability to hear, early tests used the distance a watch tick or coin click could be heard as an index. However, tests of this type include only narrow bands of frequencies and actually indicate threshold only for this group of frequencies. In order to compare the threshold of hearing of a given subject to the normal threshold, we need an instrument that will indicate how many decibels the subject's threshold is above or below the normal. Such an instrument has been devised and is known as an audiometer. *Figure* 1 shows a typical audiometer of the type used to obtain the data, known as audiograms, which will be discussed. To obtain these data the instrument is calibrated with a self-contained calibrating signal, and is then set to the lowest frequency with the output control set to minimum. The level is then raised until the subject indicates that he just hears the tone. To aid in determining this point a switch that cuts the signal on and off is provided. The level at which the subject first hears the tone is recorded and the data for all the other test frequencies are obtained in like manner. In performing the test the subject has the headphone on one ear for the entire run of frequencies and then repeats the test for the other ear. Several factors must be watched carefully in performing this test. The most important is the background noise, which should be very low. Another is the understanding on the part of the subject as to what he shall indicate. Several authorities believe in using cotton in the ear not under test, but this usually does not increase the comfort of the subject and, therefore, may lead to erroneous results. The audiometer may also be provided with a microphone, bone conduction receiver, sound level indicator, and a masking source or provision for masking input. This last item is used when testing the poorer ear in which case the sound travels to the good ear by bone conduction through the skull. It is necessary to mask this tone in the good ear with noise from the masking source. In conduction deafness where the sound path from the air to the inner ear is obstructed, the results of the bone conduction test using the audiometer should be about the same as bone conduction results for a person with normal hearing, and the bone conduction receiver is used for that test. This is the common test for determining whether conduction or nerve deafness exists.

In testing for the ability of a person to meet the problems of daily life (social adequacy) articulation tests prove very useful. A person with partial hearing loss may not require a hearing aid at all in order to maintain social adequacy, although he may have a flat 20-db loss throughout the entire audio spectrum. Other people with "notches" in their audiogram may find it very difficult to get intelligence out of normal speech although their audiogram is almost normal. The microphone can be used to give audiometer articulation tests to find the speech threshold.

Determining Hearing Aid Need

It is difficult to judge whether a given subject requires a hearing aid or not,

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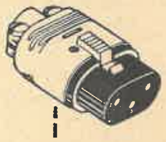
and illustrative audiograms are shown for typical cases. However, these should not be used as indices by the hard of hearing. Each person is subject to a particular set of circumstances and should consult his own physician.

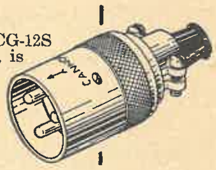
Figure 2 shows the audiogram of a man with high loss above 2000 cps, but who had no difficulty in conversation or work. This is a case of nerve deafness. Examples of conduction deafness are shown in *Fig.* 3. Although the person with the audiogram of (A) has a 30 db loss, he did not require a hearing aid; the subject of the audiogram of (B) did.

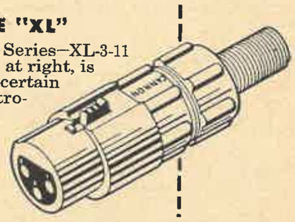
It is of interest to note the results of the tests conducted by Steinberg, Montgomery, and Gardner at the New York World's Fair in 1939 and 1940. They are shown in part in *Fig.* 4. These curves show the average hearing characteristics in the age groups indicated. The increasing loss at high frequencies with increasing age is attributed to degeneration of the sensory cells. In the higher age groups particular cases may require hearing aids; this is not to be considered a strange malady, but a normal occurrence with increasing age.

Two additional effects which accompany deafness are tinnitus and recruitment. Tinnitus is the name given to the "ringing" sound heard when people are fatigued or have been subject to loud sound for long periods. It also occurs as a result of non-acoustic stimulation of the nerves in the inner ear. Such non-acoustic stimuli can be pressure on the eardrum or actual degeneration of the cell tissue or nerves. Recruitment is the name given to the phenomenon occurring with people who have hearing loss for low intensities, and normal sensitivity for high intensities. Often the sensitivity for high tones may be increased. The person suffering from this condition becomes annoyed when people raise their voices. The subject first complains that he cannot hear and then complains that the speaker is shouting.

This article has sketched the uses of articulation and intelligibility tests and has presented a brief outline of certain conditions arising in deafness. The references already listed in the previous articles have valuable material for those who wish to make a complete study of the subject. In particular, Dr. Hallowell Davis's "Hearing and Deafness" gives a good non-technical treatment of these three topics. The next article of the series will begin the study of sound generators.







Problems in Audio Engineering

LEWIS S. GOODFRIEND*

Part IV. A study of sound waves—their characteristics, and their transmission; and an introduction to the wave equation, which governs their propagation.

THIS ARTICLE begins the study of sound waves and their generation and transmission. To establish a starting point we shall define sound and some of its characteristics. According to the "Proposed American Standard Acoustical Terminology," published by the American Standards Association for trial and study, "Sound is an alteration in pressure, stress, particle displacement, particle velocity, etc., which is propagated in an elastic material, or the superposition of such alterations." It may also be defined as ". . .auditory sensation which is evoked by the alterations described above." Since fundamental laws of physics relate the various types of alterations, one of them—pressure—will be discussed and the material then extended to cover the others. Also, a discussion of the superposition of alterations will be left until later.

If a pressure measuring device is placed in the path of a musical sound wave the oscillogram of *Fig.* 1 may be obtained. In order to carry on the analysis of propagation various labels have been placed on the figure and their definitions given below. Those marked with an asterisk are from the proposed American Standard.

Static Pressure*: The static pressure is the pressure that would exist in the medium with no sound waves present. The commonly used unit is the microbar.

Instantaneous Sound Pressure*: The instantaneous sound pressure at a point is the total instantaneous pressure at the point minus the static pressure. The commonly used unit is the microbar.

Maximum Sound Pressure*: The maximum sound pressure for any given cycle of a periodic wave is the maximum absolute value of the instantaneous sound pressure occuring during that cycle. The commonly used unit is the microbar.

Note: in the case of a sinusoidal sound wave this maximum sound pressure is also called the pressure amplitude.

Peak Sound Pressure*: The peak sound pressure for any specified time interval is the maximum absolute value of the instantaneous sound pressure in that interval. The commonly used unit is the microbar. Note: in the case of a periodic wave, if the time interval considered is a complete period the peak sound pressure becomes identical with the maximum sound pressure.

Effective Sound Pressure* (Root-Mean-Square Sound Pressure): The effective sound pressure at a point is the root-mean-square value of the instantaneous sound pressures over a time interval at the point under consideration. In the case of periodic sound pressures, the interval must be an integral number of periods or an interval which is long compared to the period. In the case of nonperiodic sound pressures, the interval should be long enough to make the value obtained essentially independent of small changes in the length of the interval. Note: the term "effective sound pressure" is frequently shortened to "sound pressure."

Periodic Wave: A periodic wave is one in which the values of the pressure amplitude recur for equal increments of time or distance, as shown in *Fig.* 1.

Fundamental Frequency*: The fundamental frequency of a periodic quantity is the frequency of a sinusoidal quantity which has the same period as the periodic quantity. The fundamental frequency of a periodic quantity is also the reciprocal of the period.

Period: The period of a periodic quantity is the smallest value of the increment of the independent variable for which the function repeats itself.

Wavelength*: The wavelength of a periodic wave in an isotropic medium is the perpendicular distance between two wavefronts in which the displacements have a phase difference of one complete period.

Figure 2 shows a circular piston having a plane face, moving in a cylindrical tube. The far end of the tube may for the moment be considered to be completely sound absorbent. This will eliminate any necessity to consider reflections. As the piston moves to the right it compresses air immediately in front of it. This small element of air then compresses the air to its right, thereby sending the compression down the tube. If the displacement of the piston takes place in a short time a steep front pulse will be sent down the tube. A picture of the pulse at successive intervals of time is shown in *Fig.* 3.

If the piston is moved to the left the air immediately in front of it will be

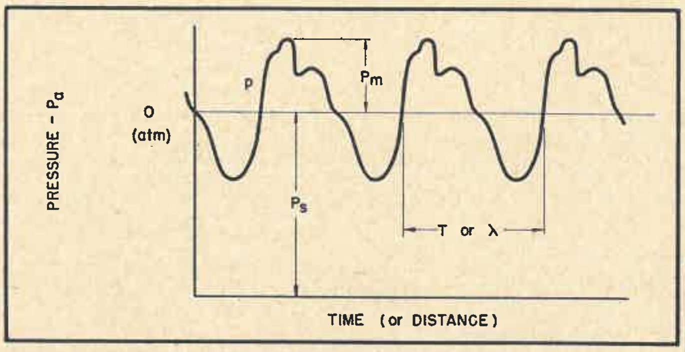


Fig. 1. Oscillogram of a sound wave.

P_a = absolute sound pressure measured above zero
P_s = static sound pressure
p = instantaneous sound pressure
P_m = maximum sound pressure λ = wavelength
T = period ξ = displacement

Fig. 2. Pressure and rarefaction waves in tube excited by piston with plane surface.

Fig 4. Instantaneous pressure along tube as result of oscillating (sinusoidal) motion of piston of Fig. 2.

Fig. 2

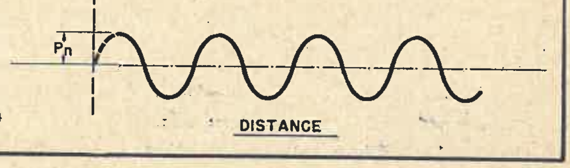


Fig. 4

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*Rangertone, Inc., 73 Winthrop St., Newark 4, N. J.

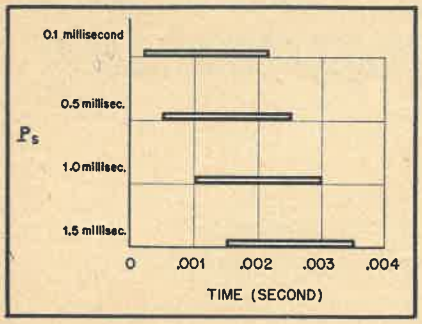

Fig. 3. Movement of pressure wave down tube of Fig. 2.

rarefied and a rarefaction wave will be sent down the tube.

If the piston is driven by a mechanism which will impart to it a sinusoidal motion the pressure in the tube after a short time will vary sinusoidally with distance, as shown in *Fig.* 4, as a train of waves is sent down the tube. Since the waves are being generated by a plane surface, it may be seen that the pressure in any one plane parallel to the piston face, neglecting the frictional drag near the surface of the tube, will be equal, so that the waves are said to be plane waves.

Plane waves and the theory of their propagation offer an excellent means of analyzing many audio problems. They permit simple studies of loudspeakers, the transmission of sound through panels, sound at long distances from the source, and sound in bars and rods.

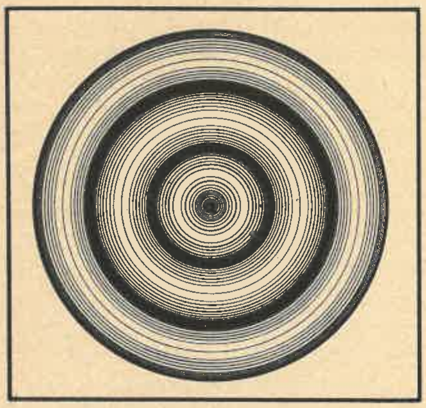

Fig. 5 Showing pressure and rarefaction in two-dimensional sound wave.

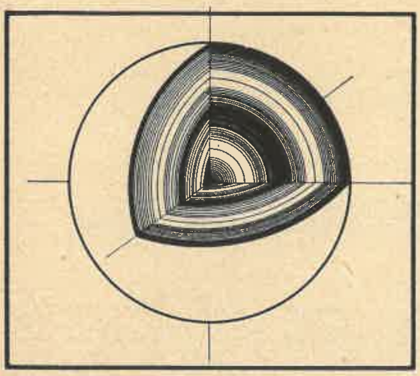

Fig. 6. Pressure and rarefaction existing in a three-dimensional sound wave.

As an example of plane waves in bars consider what happens when a steel bar is struck on the end with a hammer. When a thin circular plate is struck in the center, waves similar to those in the tube are sent out through the plate in all directions, but they are restricted to the plate. This is a case of a two-dimensional wave and is illustrated in *Fig.* 5. When a source of sound radiates in all directions, three dimensional or spherical waves exist as is shown in *Fig.* 6, and in the simplest form would consist of a small hollow, spherical ball fed with air through a small tube. As the sphere gets larger and smaller it successively compresses and rarefies the air. If we place a loudspeaker on the outside wall of a building and measure its radiation pattern we can see that it will generate spherical waves over a hemisphere.

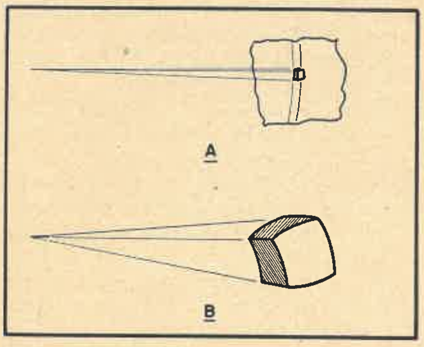

Fig. 7. Elements of pressure waves in (A) two-dimensional plane, and in (B) three-dimensional space.

The extension of plane wave theory to both the spherical and two-dimensional cases simplifies many problems and greatly reduces the mathematical work involved. If we examine portions of the spherical wave and two-dimensional wave at long distances from the source as shown in *Fig.* 7, we see that a small element of either is very much the same as an element of the plane wave, and it is possible to prove this extension mathematically.

Some of the properties which we would like to know about sound waves are: velocity through the medium, velocity of the particles of the medium, particle amplitude, pressure (rms or peak), energy, and intensity. From the physical and thermodynamical principles of continuity, adiabatic expansion and compression gases, and the equation of motion, it is possible to derive an expression relating the particle characteristic under study (displacement, pressure, etc.) to the distance the wave travels down the tube and to time. This equation is the so-called wave equation, and for pressure in plane waves the form

$$\frac{\partial^2 p}{\partial t^2} = \sqrt{\frac{\gamma p_s}{\rho_s}} \quad \frac{\partial^2 p}{\partial x^2}$$

The constant term, which we may call *c*, may be shown to be the velocity with which the wave travels down the tube. It is also the velocity of waves propagated in either two or three dimensions. In the wave equation the symbols are as follows:

P_s = Static pressure in dynes per square centimeter
ρ_s = Static density of the medium in grams per cm^3.
γ = the ratio of the specific heats of the gas (1.4 for air at room temperature)

Assuming that the displacement ξ varies sinusoidally according to $\xi = a \sin(\omega t - 2\pi x/\lambda)$, a generalized form for sinusoidal motion, the particle velocity u may be obtained from the above expression giving

$$u = \omega a \cos(\omega t - 2\pi x/\lambda) \text{ cm. per sec.,}$$

with the maximum particle velocity $u_m = \omega a$ cm. per sec. The maximum particle amplitude is $\xi_m = a$ cm.

The instantaneous pressure in the wave can be determined from the other quantities giving

$$p = p_s \gamma k a (\cos \omega t - k x) \text{ dynes per cm}^2.$$

where $k = 2\pi/\lambda$.

Determining the energy density (energy per unit volume) in the wave involves the use of calculus, but when the operation is performed we obtain the following expression:

$$E = \rho_s u_m^2/2 \text{ ergs per cm}^3.$$

If we rearrange some of the terms and make the correct substitutions it is possible to obtain the energy equation in another form, $E = \rho_o u^2$ ergs per cm^3. The total energy in a given volume may, of course, be obtained by multiplying the energy density by the volume.

Finally, the intensity of the sound is obtained by multiplying the energy

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TABLE I

1 microbar = 1 dyne per cm^2.
1 dyne per cm^2. = 1.45×10^{-5} lbs./in^2.
1 erg = 1 dyne-centimeter
= 7.38×10^{-8} ft. lbs.
1 erg per sec. = 1×10^{-7} watts
1 erg/sec./cm^2. = 1×10^{-7} watts/cm^2.

TABLE II

For air at 20° C. and 760 mm. pressure
$P_s = 1.013 \times 10^6$ dynes per cm^2.
$\rho_s = 1.21 \times 10^{-3}$ grams per cm^3.
$\gamma = 1.40$ (dimensionless)
$c = 34{,}300$ cm. per sec.

standards. Frequency response is ± 2 db from 50 to 15,000 cps at 15 inches per second; and ± 2 db from 50 to 7500 cps at 7.5 inches per second. The signal-to-noise ratio is better than 60 db measured in accordance with the NAB definition (ratio of peak recording level to total unweighted noise level recording zero signal; peak recording level is that point at which the overall total harmonic distortion does not exceed 3 per cent measured on a 400 cps tone).

At the 15-inch speed, flutter and wow are well below 0.1 per cent. This measurement is made on a bridge which measures all flutter components from zero to 300 cycles, using a signal of 3000 cycles. At 7.5 inches, the flutter and wow are less than 0.2 per cent.

The final test of performance is in listening. The most critical test that can be made is the "A-B" test, whereby the output of the recorder is compared with the input by rapidly switching the monitoring equipment between input and output. Because of the minutely detailed comparison which can be made at the time of switchover, slight discrepancies between two systems can be detected by means of the A-B test which would never be noticed if the systems were listened to separately. When making critical A-B tests on the Model 300, using live program material of the most exacting nature, and with the highest quality amplifier and speaker systems, it is impossible to distinguish recorded material from the live program.

PROBLEMS IN AUDIO

[*from page* 20]

by the velocity of the sound. This gives the expression I = c E or $2\pi^2\rho_s{}^2c^3a^2/\lambda^2$ or I = $2\pi^2\rho_s{}^2c\, f^2a^2$ ergs/sec/cm².

In all of the above equations the units have been given in the metric system, which is the standard system of units. However, since many authors still use other systems, Table I gives the relation of metric units to the others, and Table II lists the reference values of physical constants. It is important to remember that the constants of the medium change with temperature and it is therefore necessary to use the appropriate values for the temperature at which the computation is to be made. The values given here are for room temperature, 20° C. being equal to 68° F., which is close to the normal working temperature of most places in which audio work is carried out and for which calculations are of importance. In the following articles, consideration will be given to generators and sources, and to applications of the material in this article.

Problems in Audio Engineering

LEWIS S. GOODFRIEND*

Part V. A study of sound generators.

Any study of sound must necessarily include a study of the principles involved in the generation of sound whether from an original source, or as a reproduction by means of some form of electro-acoustical transducer, such as a loudspeaker. In the previous article, considerable space was devoted to the wave equation and to its derivation, and before continuing further it is desirable to derive the simplified equation for the intensity of a sound wave as an illustration of the mathematical methods used. The general form of the equation for intensity is

$$I = \frac{2\pi^2 \rho_s c^3 a^2}{\lambda^2}.$$

Multiplying the numerator and denominator by c/c—which is equivalent to multiplying by 1—gives

$$I = \frac{2\pi^2 \rho_s c^3 a^2}{\lambda^2} \times \frac{c}{c} = \frac{2\pi^2 \rho_s c^4 a^2}{\lambda^2 c}$$

Then, by substituting $P_s\gamma/\rho_s$ for c^2 in the numerator,

$$I = \frac{2\pi^2 \rho_s P_s^2 \gamma^2 a^2}{\lambda^2 c \rho_s^2}$$

The terms may now be rearranged, giving

$$I = \left[\frac{P_s^2 \gamma^2 2\pi^2 a^2}{\lambda^2} \right] \frac{1}{\rho_s c}$$

The part in the brackets is equal to the effective sound pressure P. We now have a simplified form of the equation for intensity expressed in terms of two constants for the medium—density and wave velocity—and the effective sound pressure

$$I = \frac{P}{\rho_s c}$$

Returning now to the radiation from a plane piston in an infinite wall, let us compare the radiation pattern for the piston to that for an ideal point source. It will then be possible to examine the pattern for the loudspeaker. While the radiation from a plane circular piston in a wall is almost identical with that from a point source, actually there are several differences in the respective radiation patterns. However, these differences are small at long distances from the piston and at low frequencies, as shown in *Fig.* 1. If the wavelength of the sound being radiated is less than four times the diameter of the piston, the pattern narrows down and the intensity along the axis perpendicular to the center of the piston is much greater than at an angle off to one side. For the case of the loudspeaker it will be noted that the curve shown in *Fig.* 2 is similar, with the loss at 90° being about double that for the piston of the same diameter. This is the result of a conical surface radiator in the loudspeaker and the fact that the cone is not absolutely rigid. However, in working problems in sound distribution it is not illogical to assume that the loudspeaker will behave as a plane piston, and for a rough estimate of the distribution at low frequencies, the point source treatment may be used. The computation of the high frequency case is not simple, and the published curves for the distribution of sound from pistons are of great value here. In order to overcome the normal concentration of sound along the axis at high frequencies,

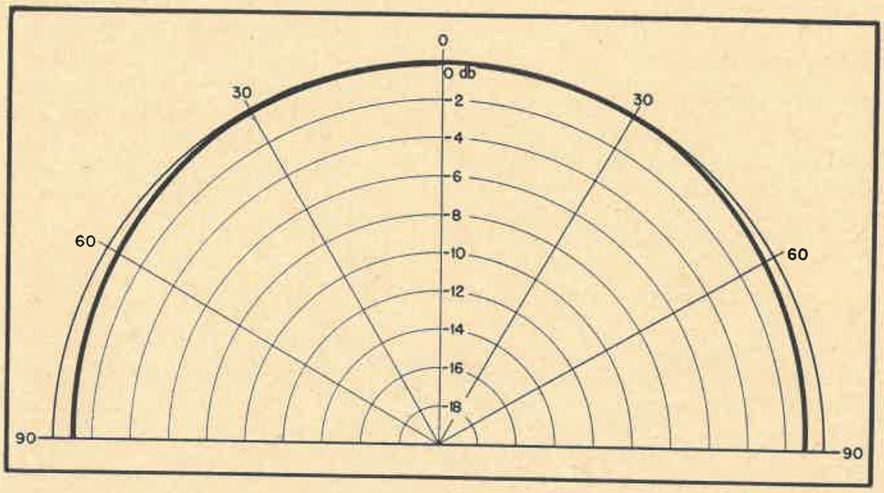

Fig. 1 (above). Polar pressure radiation pattern for a plane circular piston in an infinite wall. The piston is small compared to a wavelength and the pressure was measured at a long distance from the piston.

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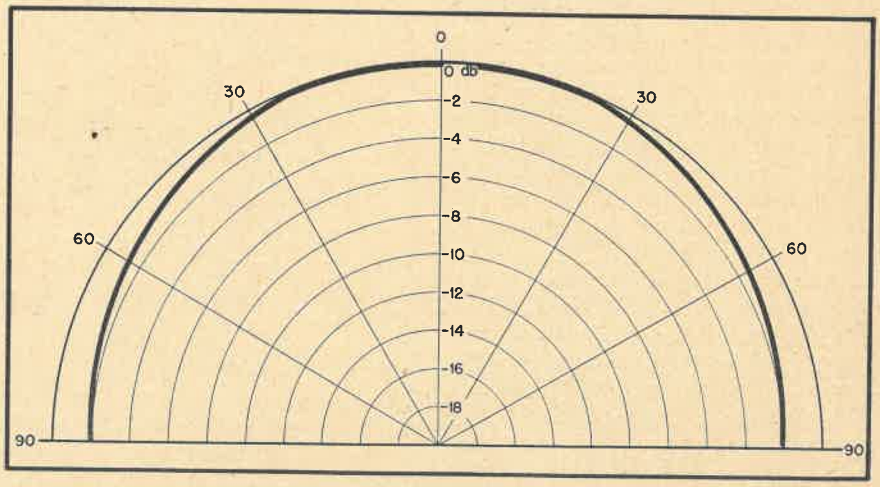

Fig. 2 (right). Polar pressure radiation for a loudspeaker in an infinite baffle. The diameter of the speaker is small compared to a wavelength.

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audio engineers use multicellular horns or directing vanes which help to distribute the energy more evenly. In addition it is known that a good low-frequency unit is seldom efficient as a high-frequency reproducer. Therefore, a separate high frequency horn designed to have a high efficiency and good distribution pattern is not a whim of the manufacturer or the theoretician, but the logical solution to two problems.

Acoustical Dipole

If there are two point sources of sound spaced by an infinitesimal distance and radiating 180° out of phase, we have what is known as a dipole or doublet source of sound. For purposes of analysis it is also possible to use a small spherical source, but instead of pulsating, as in the case of the point source, the sphere oscillates back and forth. The radiation from a dipole is not uniform in all directions, but is maximum along the axis through the two points or on the axis along which the oscillating sphere moves, *Fig.* 3. This is the case of the loudspeaker when not in a baffle of any kind. The equations for the radiation from the point source and the dipole differ in one important respect. Radiation from a point source varies directly with the square of the frequency, and that from the dipole varies as the fourth power of the frequency. This means that the low-frequency response of a loudspeaker outside of a baffle drops much more rapidly than that of a speaker in a baffle.

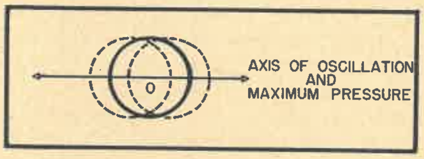

Fig. 3. The oscillating sphere moves transversely along the axis, which is also the axis of maximum pressure.

The phenomenon associated with a point source can also be associated with a loudspeaker in an infinite baffle or a wall. However, if the speaker is in a finite square, flat baffle there will be some frequency at which the wavelength of sound is long enough so that the sound diffracted around the edges of the baffle travels one wavelength before reaching the front of the cone. At this frequency, destructive interference (cancellation) will result. Below this frequency the system acts in a manner similar to a dipole source, while above, it will act in a manner similar to a point source. Further information on this subject will be found in the references. Although there are other types of sound sources, they will not be dealt with here, as they enter into audio only in the design of special instruments.

Mechanical Vibration and Sound Generators

Having examined the various forms of radiation most common in the audio field, let us turn our attention to the generation of sound waves by considering the application of some perturbing force to strings, pipes and plates. It shall be assumed that the reader is familiar with some of the fundamentals of physics or will consult a physics text. The generation of harmonic oscillations may be studied by taking an ideal, frictionless case of the simplest mechanical vibrating system, a spring and a weight, *Fig.* 4. The spring is fastened at one end to a wall, at the other end to a ball which rolls on a perfect bearing. It is found from physics that if the weight is pushed toward the wall and then released it will oscillate back and forth, and the frequency of oscillation is dependent only on the mass of the weight and the stiffness or compliance of the spring. (Stiffness k is the reciprocal of compliance C.) The equation for frequency is

$$f=\frac{1}{2\pi}\sqrt{\frac{k}{m}}$$

where k = stiffness in dynes per cm.
m = mass in grams.

This frequency is called the natural frequency of vibration of the system, and as defined by the Proposed American Standard is "a frequency of free oscillation." In other words, the natural frequency is the frequency of vibration of any system which is excited by a small perturbing force which is then removed. For the practical case of a plucked string the natural frequency may be the fundamental or any harmonic, the fundamental being expressed as

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

where f = fundamental resonant frequency in cps.
l = length of the string between supports in cm.
T = tension along the string in dynes.
m = mass per unit length in grams.

If a sinusoidal driving force is now applied to either the string system or the weight-spring system at the resonant frequency and in phase, standing waves are set up in the system. Actually the term "standing wave" comes from the case of a wave that is transmitted down a string, pipe or bar, and is reflected so as to reinforce the next wave coming from the transmitting end. When standing waves are set up in a vibrating system the amplitude of the vibrations is likely to increase, being limited by either the resistance forces within the system or the amplitude at which the system will destroy itself.

When the string is excited at some frequency other than the fundamental it will produce the fundamental and the harmonically related overtones up to and including the exciting frequency, and some of the higher overtones. In musical instruments it is phases and relative intensities of these overtones, in addition to the sound box amplifier, that greatly affect the quality or timbre of the instrument. Without any sort of sound box the note from the string would be almost inaudible.

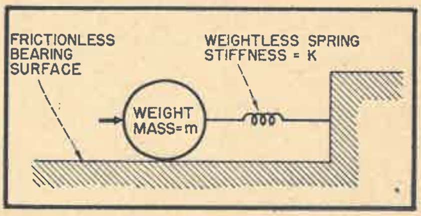

Fig. 4. The ideal, frictionless, case of a weight and a spring fastened to a rigid wall.

Air Column Generator

The next generator to be examined is the air column. Fundamentally the air column is not a sound generator, but a resonator for the sound made by a vibrating reed, double reed, or edge tone. Organ pipes, clarinets, trumpets and trombones are a few of the instruments that use this method to produce amplification of the sound generated by one of the three methods named. The usual example—the organ pipe—will be discussed here. When a sound wave travels the length of a tube which is terminated by a closed end, it is reflected back to the other end of the tube. If the wavelength of the sound is equal to four times the length of the tube (the tube is then said to be a quarter-wavelength long) the wave will be reflected back to the transmitting end in the correct phase to reinforce the original wave and we have the case of a standing wave in a tube. It is also possible to have a standing wave in the tube for any wavelength whose frequency is an odd multiple of the fundamental frequency.

In the case of the pipe with the far end open, standing waves will exist at a frequency whose wavelength is twice the length of the pipe or at any multiple of this frequency. Therefore it may be stated that the tone produced by a closed pipe will contain only the odd harmonics of the fundamental while the tone of an open pipe will contain all the harmonics.

[*Continued on page 46*]

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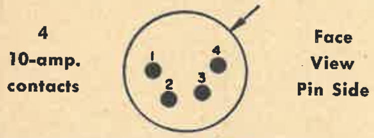

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Finally, the log element should be so proportioned that the maximum output voltage is of the order of a volt. Too small an output will require more amplification in the vacuum tube voltmeter and in the output circuit, amplification which will handle a 15000 cps square wave without distortion. This is costly and unnecessary, for the log output can be made as much as a volt without trouble.

It is believed that the stable logarithmic system, perhaps combined with an oscillograph, has very interesting possibilities in audio instrumentation.

PROBLEMS IN AUDIO

[*from page 19*]

If the source of the sound which the pipe amplifies or resonates, contains numerous other frequencies, but none that are harmonically related, then only the one frequency will be amplified. This is the case with the organ. In either the reed or the flue-pipe organ, the tones produced by the reed or edge contain numerous tones, some that are harmonically related and others that are not. However, because the pipe is a selective device it will deliver a fairly pure harmonically related series of tones.

The final two cases of sound generators that will be considered in this article are the circular membrane and the circular plate. There have been numerous analyses made of plate and membranes of various shapes. However, they are extremely complex and result in a series of equations for each case which express the frequency of the fundamental and the series of overtones for the different modes of vibration of the plate and the membrane.

Basically, if a stretched circular membrane is struck in the center, a two dimensional wave will be gener-

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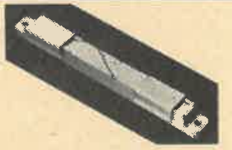

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ated and the air in front of the membrane will be set in motion thus causing a spherical wave to be generated in the air. This is the situation that occurs when a drum is struck. However the form of the enclosure of the drum also affects the modes of vibration thus determining the overtone structure. The frequency spectrum of the stretched circular membrane may be expressed by the general equation.

$$f = \frac{C}{R}\sqrt{\frac{T}{m}}$$

where C = a constant determined by the overtone to be found and the mode of vibration.
R = radius of membrane in cm.
T = tension on membrane in dynes per cm.
m = mass in grams per square cm. of area.

In the references, the book by Dr. H. F. Olson contains an excellent list of the numerical values of the constant C.

For the clamped circular plate which appears in audio work as the diaphragm of the carbon microphone, the telephone receiver, and certain types of loudspeakers, the derivation and the resulting equations of its fundamental frequency and overtones are even more complex than those for the membrane. The general form of the equation is

$$f = \frac{Ct}{R^2}\sqrt{\frac{M}{\rho\ (1-\sigma^2)}}$$

where C = constant determined by the overtone to be found and the mode of vibration.
R = radius of the plate between clamps in cm.
ρ = density of the plate in grams per cm.^{3}
M = Young's modulus of elasticity in dynes per cm.^{2}
σ = Poisson's ratio.
t = thickness of plate in cm.

Again the reader is referred to Dr. Olson's book for the values of the constants.

Having discussed the general background of sound generators, two special cases will be left for the next and concluding article. These are the generation of speech sounds and the generation of sound by musical instruments.

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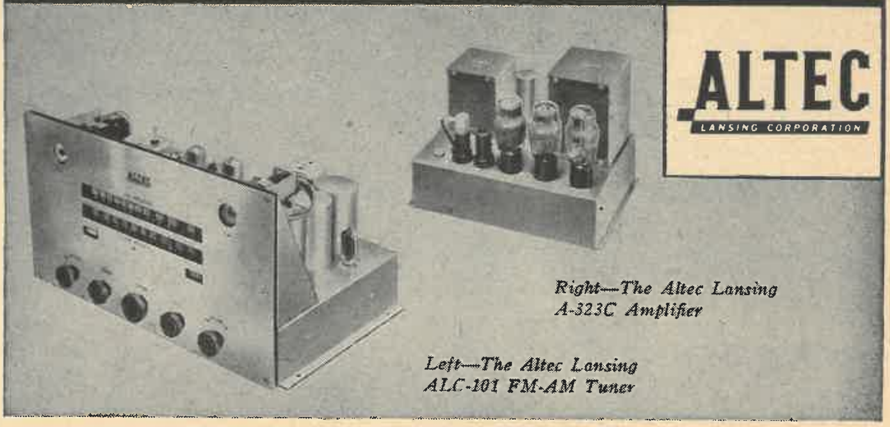

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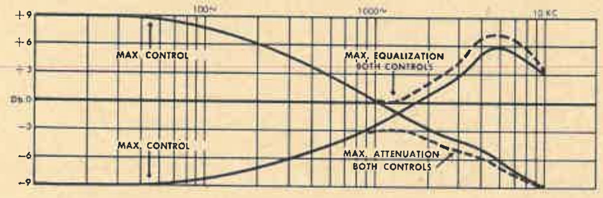

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Problems in Audio Engineering

L. S. GOODFRIEND*

In this concluding article, the author describes musical instruments and the human speech organs as sound generators.

As pointed out in the previous article, musical instruments and the organs of speech constitute a special group of sound generators. This article will discuss at length the manner in which various instruments produce musical sounds and will tabulate the more common musical instruments with their method of sound production and reinforcement or amplification.

Sounds generated by strings, reeds, or edges in air streams produce not only the fundamental and harmonics, but also generate inharmonic partials. These generators usually do not produce loud tones, but must have some form of amplification before they are clearly audible as instruments in a concert hall. The method by which this amplification takes place is usually the forced vibration of an air column, as in the case of reed and edge-tone instruments, or the forced vibration of a sounding board, as in the string instruments. In the first case the standing waves set up in an air column selectively reinforce the desired resonant frequency to which the column is tuned. In the case of the string instruments, the strings are coupled to an air chamber which has a particular resonant frequency, and to a sounding plate or membrane. In a piano there is a large wood sounding board which acts to couple the string to the air. This occurs because the string sets the sounding board in motion, and while the displacement of the board may be small, it moves a large volume of air. This is in contrast to instruments that require an air column resonator. For example, when resonance in the true sense (as shown by the simple mechanical system described last month) takes place in the string instruments, it causes a selective amplification which is greatest at one frequency. In the violin this is the so-called "wolf note," a loud unpleasant note, and one not easily controlled by the player. In the wind instruments and brasses additional coupling of the air column to the surrounding air is accomplished through the use of end bells which, like exponential horns on some types of loud speakers, couple the end of the pipe to the atmosphere with an action similar to a transformer coupling two lines of different impedance. The classification of the various types of instruments is shown in Table I.

It is not the purpose of this series to deal with the aesthetic and musical factors involved with bowing violins or tonguing wind instruments. The manner in which strings produce tones has been discussed earlier. A brief glance at the generation of tones by sharp edges placed in an air stream and by blown reeds, however, is in order.

Air Stream Generators

If a jet of air from a slit is aimed at a sharp edge, as in *Fig.* 1, the stream will divide and flow past the edge. As the air stream leaves the edge it forms little whirlpools, or vortices, alternately on each side of the edge. In other words, the stream tends to oscillate back and forth across the edge. The frequency at which these oscillations occur is generally within the range of audibility for most constructions, and is dependent on the velocity of air across the edge and on the distance from the slit to the edge. The tones produced by this method are not very loud and have a large number of overtones present. However, by placing the edge and slit at one end of a pipe whose fundamental is the same as the edge tone or one of the overtones, the fundamental will be reinforced and clearly heard. This is the method used to produce tones in the modern flue-pipe organ. The flute is also an edge-tone instrument, with the lips used as the slit, and the mouthpiece forming the edge.

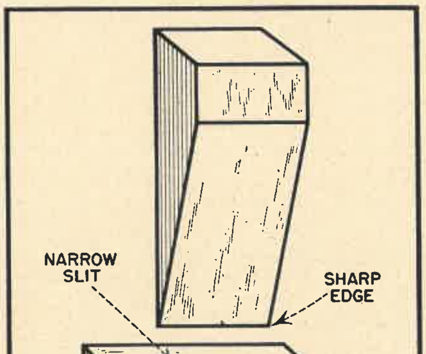

Fig. 1. Basic apparatus for producing edge tones. Narrow stream of air is "split" by sharp edge, which might be that of an organ pipe.

Reed instruments, on the other hand, depend on the motion of air about an obstacle. When we consider that sound consists of alternate compressions and rarefactions of the air, it is obvious that we can generate a sound wave by alternately turning a stream of air on and off. When a strip of metal or bamboo is placed at the end of a pipe, the reed will tend to vibrate at the frequency to which the pipe is tuned. When the reed and pipe are tuned to the same frequency, large amounts of power may be generated, and in the woodwind instruments this usually produces the squeaky sound associated with the playing of beginners.

Reed Instrument Construction

There are several types of reed structures used to produce sound in musical instruments. These are shown in *Fig.* 2. The clarinet and saxophone reeds operate against the opening of the mouthpiece, completely stopping the flow of air when it is against the opening. On instruments of the oboe family there are two reeds which vibrate against each other, alternately opening and closing. When they are open they form a tube through which the air passes. Other instruments that use this double reed construction are the English horn and the bassoon. The reeds on the instruments named are of the striking reed type as shown in (*A*) of *Fig.* 3; that is, they close the aperture through which the air passes by striking against an opening or against another reed. There is another type of

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reed construction in which the reed stops the flow of air by moving back and forth through a slot in a plate as in (*B*) of *Fig.* 3, and as the reed passes through the plate it effectively cuts off the air. The brass instruments, including the trumpet, trombone, and French horn, are really reed instruments in which the player's mouth acts as a vibrating double reed, and the piping, of which the horn is made, acts as a resonator for the tones thus produced.

Although it might appear from the descriptions so far that each of the instruments described produces tones at the fundamental, this is not altogether true. In the brass instruments, in particular, the ratio of the width of the pipe to its length causes the second and higher harmonics to be resonated with greater ease than the fundamental, and it is only in the trombone that the fundamental is more prominent than the overtones. Pictures and charts of the structure of musical tones produced by the various instruments are so common in the contemporary literature that they will not be presented here.

Since it is possible to force a generator of edge or reed tones to generate sound having the pitch to which the resonating pipe is tuned, it is only necessary to change the length of the pipe to change the pitch. (The term pitch is used here because the tones produced by musical instruments are not simple, and do not have the characteristics of pure tones). In the woodwinds the length of the pipe is varied by opening and closing holes spaced along the length of the instrument. In the simpler instruments this is done by placing fingers over the holes, while in the more complex ones, the holes are covered by pads which are controlled through lever, key, and cam assemblies. In the brasses the length of the pipe can be changed through the use of valves which bypass the sound through longer or shorter lengths of pipe. One exception to this method is the slide trombone. In this case the length of the pipe itself is increased or decreased.

Two notable characteristics of woodwinds should be mentioned here. In the clarinet the bore of the pipe is straight, while in the oboe and saxophone it is conical. This difference causes the clarinet's harmonic structure to contain only the odd harmonics in its overtones, which is the structure normally associated with a closed pipe. The conical bore of the oboe permits it to produce even harmonics and an overtone structure that contains the complete series of harmonics. In addition to varying the pipe length, it is possible to overblow an edge or reed tone, causing some higher harmonics to become predominant. This also occurs in the brasses, and it is only through overblowing that the higher octaves may be produced in wood and brass wind instruments. If the reader would like to study this phenomenon without purchasing an instrument, it is only necessary to take a cola bottle and first blow the fundamental. Then, if the lips are pressed together to form a narrower slit, harder blowing should produce one or more of the overtones quite clearly, without the fundamental.

Percussion Instruments

The percussion instruments—drums, cymbals and xylophones—incorporate membranes, plates and bars. The mechanism of sound production of the

TABLE I

INSTRUMENT	GENERATOR	RESONATOR	CLASS
Piano	String	Sounding board	Percussion-string
Violin		Air chamber & plates	String
Violoncello		Air chamber, plates & floor	
Double Bass			
Harp			
Mandolin		Air chamber & plates	
Guitar		↓	
Banjo	↓	Air chamber & Membranes	↓
Flute	Edge tone, lips used to form slit	Cylindrical bore pipe	Woodwind
Piccolo			
Clarinet	Striking reed	↓	
Saxophone	↓	Conical bore pipe	
Oboe	Double reed		
English horn			
Bassoon	↓	↓	↓
Trumpet	Lips	Cylindrical valved pipe	Brass
Cornet			
Bugle			
French horn		↓	
Trombone		Cylindrical pipe sliding section varies in length	
Tuba	↓	Valves	↓
Tympanum	Membrane	Air chamber	Percussion
Drums	↓	↓	
Xylophone	Bars	None	
Chimes	Bars (Hollow cylinders)	Air column	
Triangle	Rod	None	
Celesta	Plate		
Glockenspiel			
Cymbals			
Gong	↓	↓	↓
Bell chimes	Bell		Bell

first two types has been covered in an earlier article, and the relation of the bar to the string is analagous to the relation of the plate to the membrane. It is the use of the bar with which we shall concern ourselves here. If a series of bars are supported at the point where there is no motion in their standing wave pattern, and the bars are of increasing length, they may be struck in succession to produce a series of rising tones. It is possible to select a set of these bars having tones corresponding to the musical scale; thus a xylophone is the result if the bars are of wood, or a hollow chime, if the bars are hollow metal cylinders. The triangle and the tuning fork, are bent or split rods. The plate is used in the celesta, cymbal and gong.

Bells are often considered as special cases of plates and are extremely complex in theory. They are uesd musically as chimes, carillons and single bells.

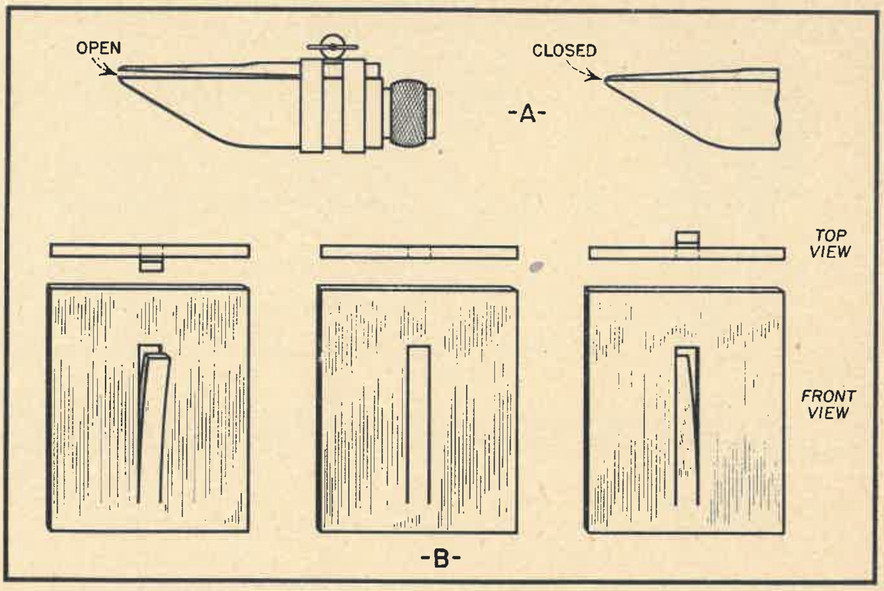

Fig. 3. (A) Side view of clarinet reed. (B) Metal reeds of the type employed in reed organs and harmonicas.

Speech

The study of speech has been the concern of three separate groups: teachers of speech and English, medical groups, and audio and acoustical engineers. Since better transmission of speech is the primary aim of a large group of the audio engineering profession, it is important to understand its production and characteristics. Speech is characterized by two distinct types of sound—buzz or hum tones, and the hiss sound. The vowels and their associated diphthongs, semi-vowels and transitionals, are of the buzz tone type; while the unvoiced fricative consonants, *f* and *s* are hiss sounds.

Buzz energy in the speech sounds is produced by a pair of muscularly controlled membranes in the larynx. These membranes interrupt the flow of air from the lungs at a periodic rate, and the tone thus produced is resonated in the mouth and nasal cavities. Hiss energy is produced by the flow of air through slits or past sharp edges in the oral cavity. Speech sounds are, in general, a combination of the two types of energy.

Fig. 2. Woodwind reeds differ in construction. The clarinet uses the single reed shown at the left, while the oboe and bassoon reeds at the right are actually two reeds placed back to back and wrapped at the stem to hold them in proper position.

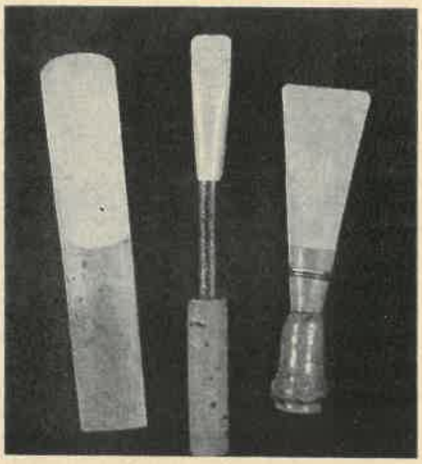

TABLE II

1. **Pure Vowels (11)**
   Long: ū (tool), ō (tone), ó (talk), a (far), ā (tape), ē (team).
   Short: u (took), o (ton), á (tap), e (ten), i (tip).
2. **Diphthongs (4)**
   ī, ou, oi, ew.
3. **Transitionals (3)**
   w, y, h.
4. **Semi-vowels (5)**
   l, r, m, n, ng.
5. **Fricative Consonants (8)**
   Voiced: v, z, th (then), zh (azure).
   Unvoiced: f, s, th (thin), sh.
6. **Stop Consonants (8)**
   Voiced: b, d, j, g.
   Unvoiced: p, t, ch, k.

The names used to tag particular combinations of sounds are the subject of a continual discussion among the various learned groups investigating speech phenomena. Table II shows an arrangement of speech sounds as worked out by Dr. Harvey Fletcher. The table is in order from buzz sounds to hiss sounds.

In contrast to musical instruments, which have hard walls and well defined smooth shapes, the oral and nasal cavities, which act to resonate speech sounds, have compliant, poorly defined, varying shapes. It is, therefore, the vocal cords that determine the frequency and not the resonator. These cords force air in the oral cavities into vibration, and by changing the manner in which energy is released through the mouth by movement of the lips and tongue we may combine the two forms of speech sounds to produce the full range of speech sounds or words. The similarity of the vocal cords to the reeds used in musical instruments should be apparent, and the overtone structures of each bears out the likeness. The sounds from reeds and vocal cords are in pulses which, since they are not purely sinusoidal, contain many harmonics and have distinctive qualities, depending on the phase and amplitude of their harmonics.

It is not difficult to see that all sound generators involve principles which are of a physical nature, and to recognize the importance of including physics as an element of the background for audio engineering. These same physical principles also form the basis for electronics and for architectural acoustics, which too many audio engineers consider as separate fields unrelated to the other fields of audio work.

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