NATIONAL RADIO INSTITUTE

Complete Course in PRACTICAL RADIO

NRI

Radio-Trician

(REG. U. S. PAT. OFF.)

Lesson Text No. 30
(2nd Edition)

CHANGING SOUND TO AUDIO FREQUENCY CURRENT

Originators of Radio Home Study Courses
··· Established 1914 ···
Washington, D. C.

SOME GOOD STUDY HABITS

A Personal Message from J. E. Smith

Criticism. This is a valuable study habit, provided it is not carried too far. The ability to criticise the ideas or opinions of another; the power to pass accurate judgment on a piece of work is largely a matter of experience, and experience is habit. Students too often make the mistake of swallowing all that they hear and read, without first passing judgment on its worth.

BF2M121530

Printed in U.S.A.

CHANGING SOUND TO AUDIO-FREQUENCY CURRENT

The entire purpose of a radio broadcasting system consisting of a studio, a radiotelephone transmitter, the ether, a radio receiving installation, and a loudspeaker, is to reproduce in the home of the radio listener speech and music originating in the studio of a distant broadcasting station. Before considering in detail the design and operation of the vast array of apparatus incorporated in the modern broadcasting station, it is, therefore, desirable that we consider in some detail the exact nature of this phenomena which we call sound, and the limitations and possibilities of that human sense which we call the sense of hearing.

FREQUENCY SENSITIVITY OF THE HUMAN EAR

The sensation of sound is produced by wave-motion ordinarily traveling in air. Since sound is a wave-motion, we can discuss the frequency, wavelength of these waves, and the velocity with which they will travel through various media. Not all wave-motions which might be produced in air or similar media are capable of producing the sensation of sound. The human ear is capable of responding to only a limited range of these wavelengths or frequencies. About the lowest frequency which any individual can detect as a musical tone is 16 cycles per second. The highest is around 30,000 cycles per second. However, these are extreme limits. The range of the average human ear is much less than this. Anyone who can detect as sound the frequencies lying between 20 and 20,000 cycles per second may consider that his ears are very good indeed.

POWER REQUIRED TO PRODUCE THE SENSATION OF SOUND

The amount of power required to produce the sensation of sound in the average human ear is very, very small. This amount depends upon the particular frequency of the sound wave striking the human ear. The ear is most sensitive to frequencies lying between 2,000 and 4,000 cycles per second, and only .000,000,000,000,000,4 of a watt is ordinarily necessary to produce the sensation of sound if the frequencies involved lie between these limits. A sound wave striking the ear drum applies a pressure to the drum. Pressure is force per unit of area and pressure on the ear drum is ordinarily measured in terms of dynes per square centimeter. A dyne is a unit of force which is approximately 2 millionths as large as the force exerted by gravity upon a mass of 1 pound.

The sensitivity of the ear varies greatly with the frequency of the sound striking it. Fig. 1 shows this variation of sensitivity for frequencies lying between 20 and 20,000 cycles per second. The curve is labeled "Threshold of Audibility." For

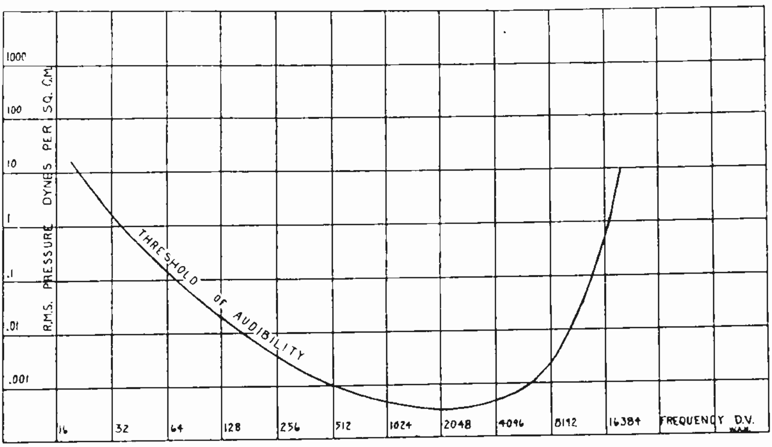

Fig. 1—Sensitivity of the average human ear.

a given frequency, pressures represented by points lying above the curve will produce the sensation of sound unless these pressures are so great as to produce the sensation of pain.

Note that the space between two lines does not always represent the same number of units of pressure. Thus, the next to the top line is labeled 1000 dynes per square centimeter, while the second line from the top is labeled 100 dynes per square centimeter. This space then corresponds to a change of 900 dynes per square centimeter in pressure. The next lower line, however, is labeled 10 dynes per square centimeter and the space between it and the 100 line, therefore, corresponds to 90 dynes per square centimeter. In describing certain phenomena, this type of graph is very useful. It is called a logarithmic graph. Referring to the graph, you will note that while at 512 cycles per second, it requires only .001 dyne per square centimeter of sound pressure to cause the sensation of sound in the average human ear; it requires over 1 dyne, that is, over 1000 times as much pressure at a frequency of 32 cycles per second.

SOURCES OF SOUND

We have seen that the range of frequencies which will produce the sense of sound in the human ear varies greatly with individuals but for the average human ear this range may be considered as lying between 20 and 20,000 cycles per second. We have also seen that the amount of power required to produce the sensation of sound is very, very small, and that the minimum

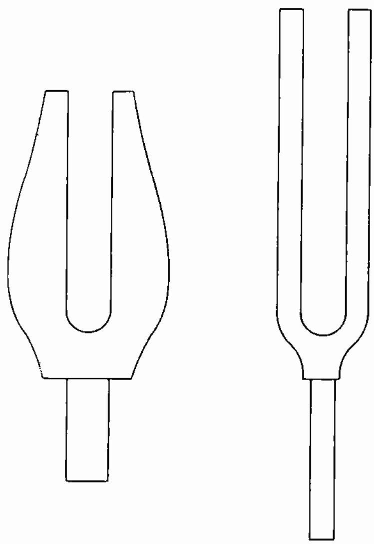

Fig. 2—Cross section drawing of two tuning forks.

amount of power required varies greatly depending upon the frequency of the sound wave striking the ear. With this very brief discussion of sound waves, let us consider just how sound is produced.

Sound waves are produced by vibrating bodies. The rate at which a body vibrates determines the frequency of the wave produced. The types of bodies which may vibrate and produce sound waves are many and include stretched strings, columns of air, stretched diaphragms, rods, plates, vocal chords in the human throat, and many others. The vibration of bodies such as these produce waves in the air which may produce the sensation of sound in a human ear providing the frequencies produced lie within the range of audibility. If the vibrating source is in water or associated with some other medium, then the sound waves may be carried by it. As we shall see shortly, the frequencies produced and their amplitudes are of great importance in determining the character of the sound.

CHARACTERISTICS OF SOUND WAVES

If the vibrating source produces only a single frequency, then the sound is said to be a **pure tone.** Practically pure tones can be produced by tuning forks. Fig. 2 is a drawing showing the outline of two tuning forks. Fig. 3 shows sets of tuning forks for producing various frequencies with a high degree of accuracy. The set of forks to the right will produce a frequency considerably higher than the set of forks on the left. The longer

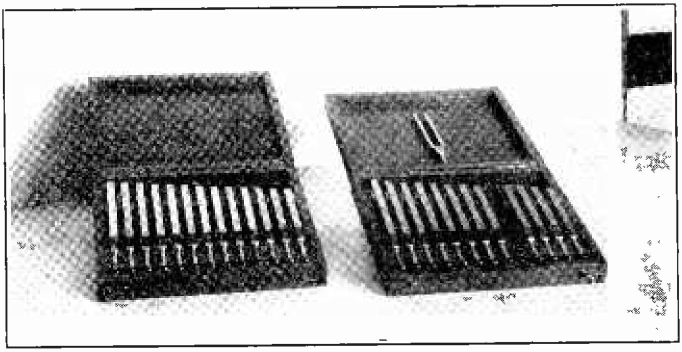

Fig. 3—Sets of tuning forks.

forks are approximately 8 inches long. The tuning fork is set in vibration by striking it with a small padded mallet or by bowing with a violin bow. If the fork is struck softly, the **amplitude** of the vibrations of the fork will be small, and the sound produced will be soft. If, on the other hand, the fork is struck more forcibly, the **amplitude** will be greater and the sound produced is said to be **loud. Loudness** then will vary with the **amplitude** of vibration of the source; the greater the amplitude the louder will be the sound.

Let us assume that we have available a group of ten tuning forks such as is shown in Fig. 4. The frequencies produced by these ten tuning forks are as follows: 128, 256, 384, 512, 640, 768, 896, 1024, 1152, 1280. From the lessons you have already studied, you will readily see that the frequencies produced by striking these ten forks consists of a fundamental and its harmonics. The 128-cycle note is the fundamental, the 256-cycle note, the second harmonic, the 384-cycle note, the third harmonic, etc.

A pure tone as produced by a single tuning fork, such as the one producing a frequency of 128 cycles per second, is not particularly pleasing to the ear. If, however, we strike the ten forks simultaneously or one after the other, we will find that the tone produced is decidedly pleasing to the ear.

The 128-cycle fork by itself will not produce a particularly pleasing tone. However, as the harmonics or **overtones** produced by the other forks are successively added, the tone grows richer until when all ten forks are operating simultaneously, a very pleasing musical tone is produced.* If one were to listen to the sound

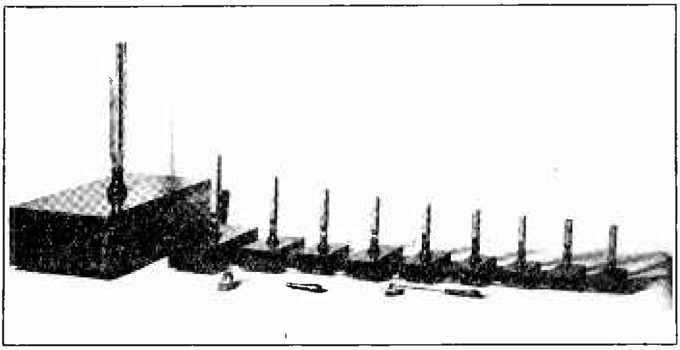

Fig. 4—Set of tuning forks for demonstrating the quality of composite tones.

from these ten tuning forks without seeing them struck, he would hardly be conscious of the fact that it is originating from ten separate sources, but he would rather be inclined to think that the sound came from one source. As a matter of fact, sounds composed of a fundamental and a series of harmonics are very frequently produced by a single source. They are often referred to as **composite tones.**

If a violin string is bowed, or the key to an organ is pressed down, then a composite musical tone is produced. The tone in each case will consist of a fundamental and a series of harmonics such as was produced by the tuning fork. The violin string produces in addition to a fundamental frequency a large number of harmonics. A certain amount of sound energy will be produced in the fundamental, different amounts will be produced in the

*Figs. 2, 3 and 4 are taken from "The Science of Musical Sounds" by Dayton C. Miller, published by Case School of Applied Science, Cleveland, Ohio. This book is highly recommended to those students who are interested in obtaining a comprehensive non-mathematical book on the general science of sound. It should be in the library of every broadcasting station engineer.

various harmonics. The same holds true for the column of air in the organ pipe.

The lower chart in Fig. 5 shows the distribution of energy in the fundamental and harmonics produced by bowing a violin

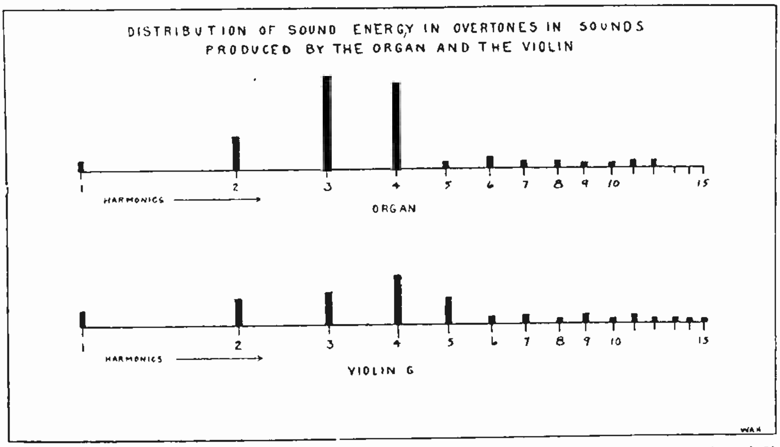

Fig. 5—Distribution of sound energy in overtones produced by the organ and the violin.

string. The upper chart of the figure shows the distribution of energy in fundamental and harmonics as produced by a particular organ pipe.

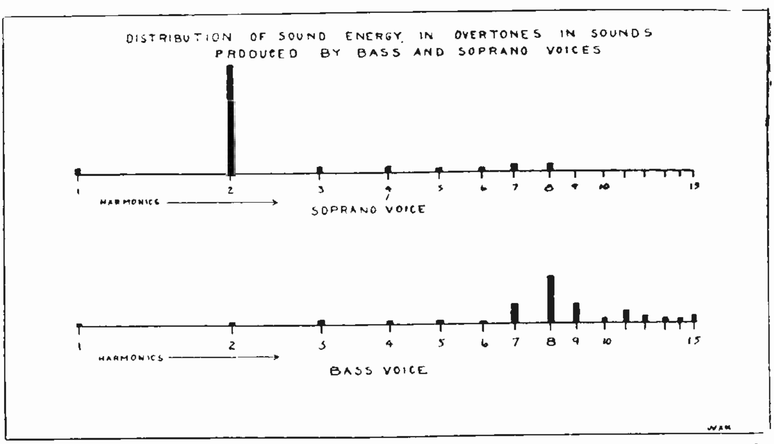

Fig. 6—Distribution of sound energy in overtones produced by bass and soprano voices.

Let us assume that the fundamental frequency produced by both the violin string and the organ pipe is 256 cycles per second. The second harmonic will then have a frequency of 512 cycles per second and the third harmonic a frequency of 768 cycles, etc. Referring to Fig. 5, you will note that there is approximately twice as much energy in the second harmonic produced by the violin as in the fundamental and something more than twice as much in the third harmonic. Referring now to the upper chart in Fig. 5, you will note that there is at least three times as much energy in the second harmonic produced by

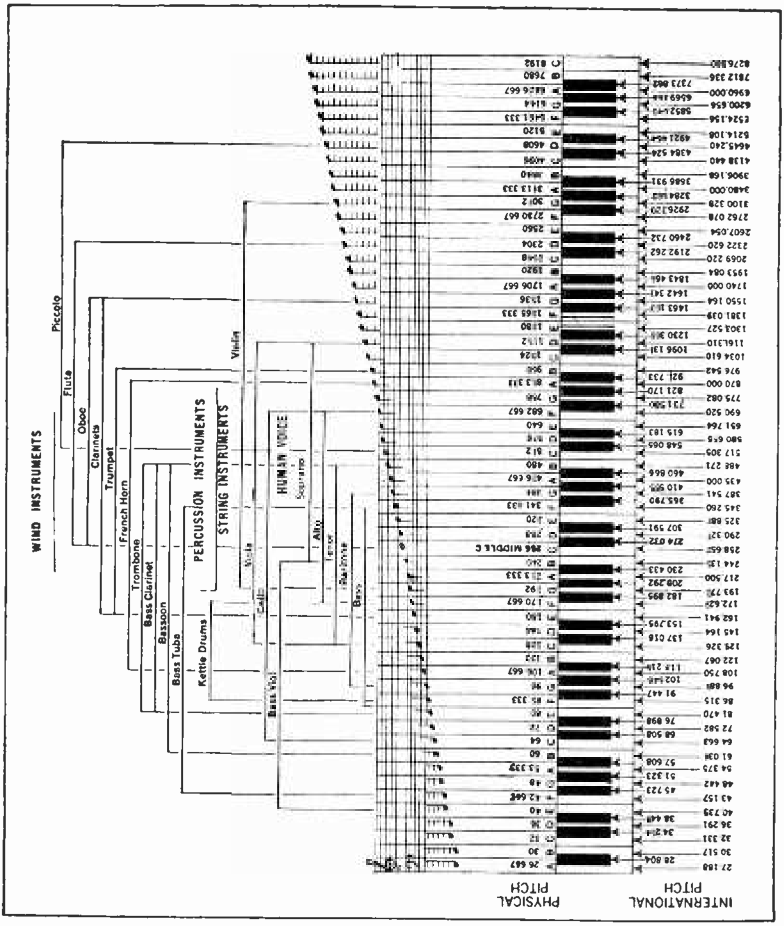

Fig. 7—The relation between musical scale and the piano keyboard.

the organ as in the fundamental and approximately ten times as much in the third harmonic as in the fundamental. In other words, although the frequencies produced in fundamental and harmonics by the violin string and the organ may be the same, the relative amounts of energy may be quite different. You are well aware of the fact that if a frequency of 256 cycles (middle C) is produced by a violin and then the same frequency is produced by an organ, you will in general have no difficulty in telling which note is produced by the organ and which by the violin. The two tones possess different characteristics. It is the varying amount of energy in the harmonics of the two notes which determines the difference and enables us to distinguish between the musical instruments. In other words, it is the relative amounts of energy in the fundamental and harmonics of a musical note which determine its **quality** or **timbre.**

There is one other characteristic of musical tones with which you should be familiar. The **pitch** of a musical tone depends upon the frequency of the fundamental. We may, therefore, use alternately the terms **pitch** and **frequency,** but when we do, it must be remembered that we are referring to the fre-

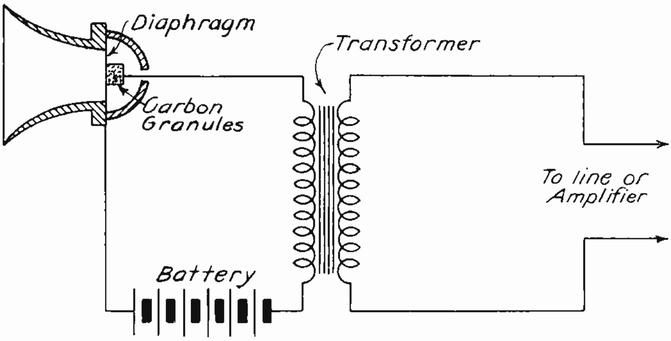

Fig. 8—Simple microphone circuit.

quency of the fundamental even though, as is shown in Fig. 5, the amount of energy in the fundamental is relatively small in comparison to that in the harmonics.

The range of fundamental frequencies produced by musical instruments varies greatly. Most fundamental frequencies produced by a violin lie approximately between 193 and 3900 cycles, by a harp between 32 and 3100 cycles, by a piano, between 27 and 4138 cycles, etc. Fig. 7 is a splendid graphic picture of the frequencies produced by various musical instruments shown together with the musical scale and with a picture of the piano keyboard. This figure contains much valuable information and is worthy of considerable study. It should be remembered that the range of frequencies for various instruments and for the human voice as shown in Fig. 7 cover only the fundamentals produced, as we have seen in addition to a fundamental, each of these instruments will produce a large number of harmonics, each containing different amounts of energy. Since the harmonics are always frequencies higher than the fundamental, you can readily see that the frequencies produced by musical instruments extend much higher than 4000 cycles. Many of the frequencies produced in music extend as high as 15,000 or 16,000 cycles. It is important that this fact be borne in mind. As we have seen, it is the relative amounts of energy in the harmonics and fundamentals which enable us to determine the **quality** or **timbre** of a musical note, and enable us to differentiate between sounds as produced by various musical instruments. Unless a radio broadcasting station provides for the transmission not only of the fundamental frequencies found in music, but also of the harmonics of these frequencies, then the quality of the program will be lost.

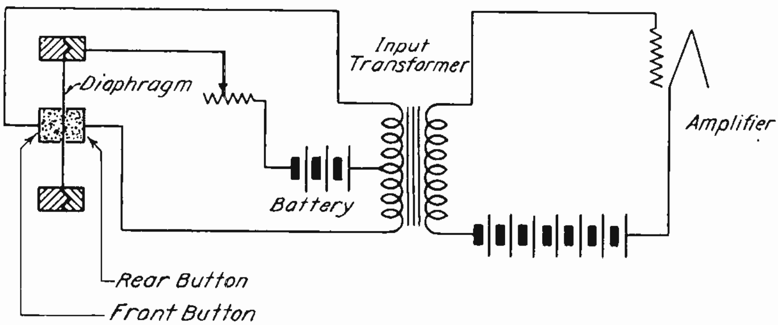

Fig. 9—Double button microphone circuit.

THE CONVERSION OF SOUND TO ELECTRIC ENERGY

Radio broadcasting makes use of electromagnetic waves. Electromagnetic waves are produced by high frequency alternating electric current which have been modulated at audio frequency. It is, therefore, essential that a broadcasting system contain apparatus which will convert sound energy as produced in a broadcasting studio to electric energy, and that the electric frequencies correspond to the sound frequencies it is desired to broadcast.

The device which performs this function in a broadcasting station is called a **microphone** or a **transmitter.** The sole function of the microphone is to convert sound energy to electric energy.

Fig. 8 is a schematic circuit drawing designed to show the method of operation of an ordinary single button microphone such as is used in the ordinary wire telephone system. Fig. 9 shows the connections for a two-button microphone such as is used in the great majority of radio broadcasting stations. Both the one and two-button microphones make use of carbon buttons. These are shown graphically in the figures. The buttons in both types are filled with small particles of carbon. The principle of operation depends upon the fact that if the carbon granules in the button are compressed, the resistance offered to an electric current is decreased. If, on the other hand, this pressure is removed, the resistance is increased.

Fig. 10 shows a cutaway picture of a two-button microphone such as is used in most broadcasting stations. The two carbon buttons can be clearly seen in the center of the picture.

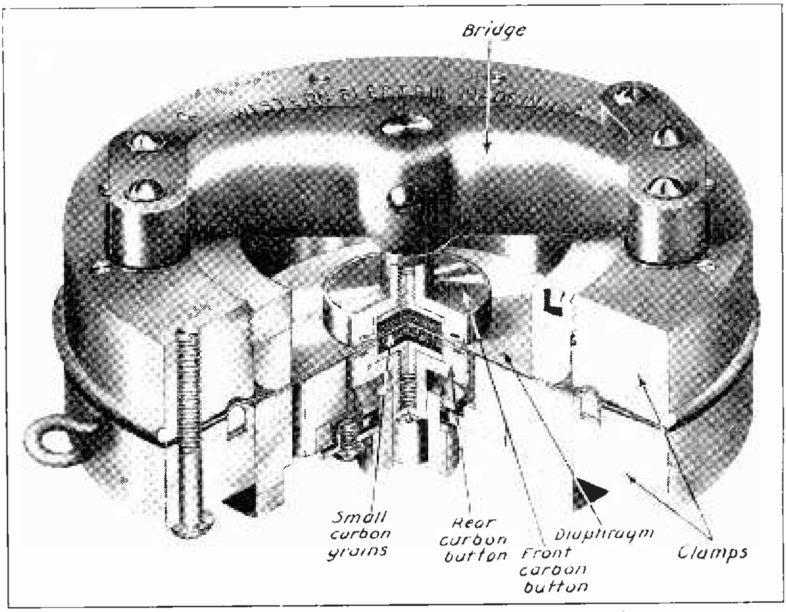

Fig. 10—Cutaway view of a double button microphone.

Referring to Fig. 8, let us assume that no sound is striking the microphone diaphragm. A steady current will then flow through the microphone from the battery and through the primary of the transformer. This is as shown in the left-hand portion of curve (b), Fig. 11.

Let us now assume that a pure tone from a tuning fork is produced in front of the carbon microphone. A sound wave such as would be represented by a sine wave will then strike the diaphragm of the microphone. This will cause pressure to be exerted on the diaphragm first in one direction and then in the other as is shown by curve (a) Fig. 11. This pressure upon the diaphragm first in one direction and then in the other as shown will cause the resistance offered by the carbon button to the passage of electric current to first decrease and then increase. The electric current which will flow through the circuit will then be as shown in curve (b) Fig. 11. This current will consist of an alternating component superimposed upon a direct component.

The effect of the alternating component flowing through the primary of the transformer will be to produce an alternating voltage in the secondary of the transformer as is shown by curve

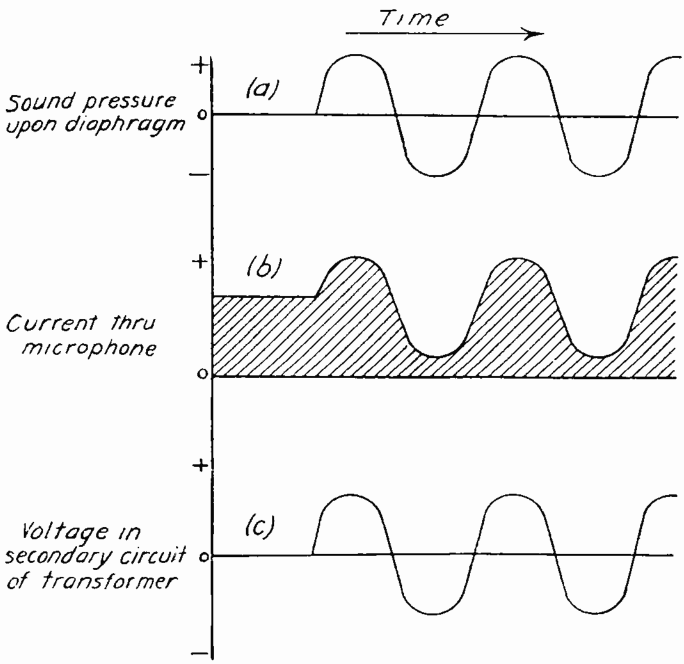

Fig. 11—Curves illustrating the conversion of sound to electric power using a single button carbon microphone.

(c) Fig. 11. The electric voltage generated in the secondary may be used to cause currents to flow over a wire line and to actuate a telephone receiver at the other end, or the voltage may be applied to a vacuum tube amplifier and its effect amplified.

If instead of a sound wave possessing the characteristics of a pure tone striking the diaphragm of the transmitter, we have complex sound waves such as would be produced by speech and music, then the alternating currents and voltages resulting will, if the system is operating properly, truthfully represent the complex waves which strike the microphone diaphragm. If, on the other hand, the system is not operating properly, then the electric currents and voltages produced by the sound wave will not truthfully correspond in wave form to the sound wave striking the diaphragm and distortion will result.

The theory of operation of the two-button microphone shown in Fig. 9 is identical with that of the single button microphone shown in Fig. 8, with the exception that now as the diaphragm vibrates back and forth, due to the effect of the sound wave striking it, it increases the resistance of the carbon button on one side and at the same time correspondingly decreases the resistance of the carbon button on the other, etc. The effect of this is shown graphically in Fig. 12, curve (b). The upper portion of curve (b) shows graphically the variation in current through the circuit connected to the rear button while the lower half of the curve shows the variation in current through the front button. During the part of the cycle when the resistance of the rear button is **decreasing** and the current through it **increasing,** the resistance of the front button is **increasing** and the current through it **decreasing.** The direction of winding of the transformer is such that the effects of the two buttons add and a voltage such as is shown in curve (c), Fig. 12, will be produced in the secondary circuit of the transformer.

While the operation of the single button carbon microphone such as is used in the ordinary telephone system depends upon the same principle as does the operation of the two-button microphone such as is used in broadcasting stations, the details of construction and the characteristics of the two types are entirely different. For instance, the amount of electric energy produced by the single button microphone is far greater than that produced by the two button. On the other hand, the distortion introduced by the single-button microphone is far greater than that introduced by the two-button microphone. Consequently, the single-button microphone as described can only be used satisfactorily for the transmission of speech, but it produces sufficient energy to permit the transmission of this speech by wire over considerable distance. On the other hand, the two-button microphone, because of its better frequency characteristics, will satisfactorily reproduce both speech and music, but because the amount of energy obtained from it is so much less than that from the single-button microphone, amplifiers must always be associated with it. Figure 13 shows a two-button microphone connected to the input of a two-stage amplifier. The voltage for the operation of the microphone is obtained from the filament battery. There are many other points which might be mentioned with respect to the characteristics of these two microphones. Inasmuch as we are now concerned primarily with the theory of operation of microphones and not with the details of their construction and practical operation, we will leave this subject for further consideration in later lessons designed to teach you the details of operation of modern broadcasting installations.

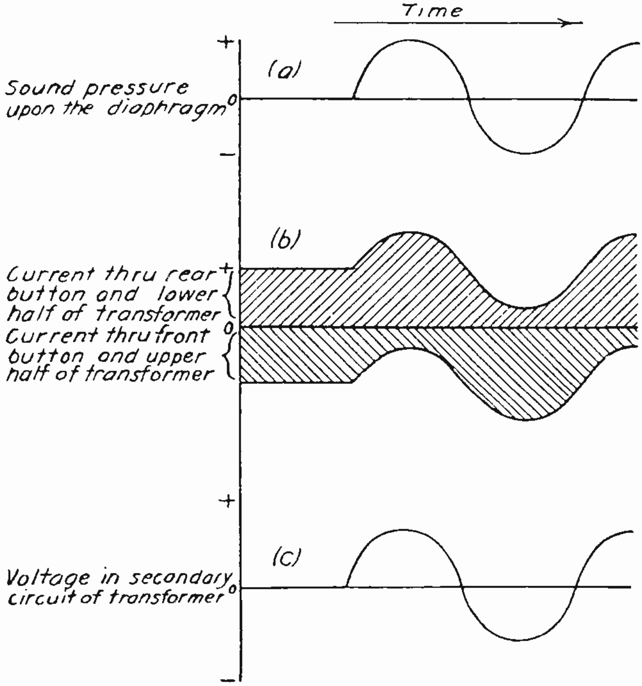

Fig. 12—Curves illustrating the conversion of sound power to electric power using a double button carbon microphone.

Carbon microphones are not the only devices which can be used to convert sound to electric energy. Another device which is enjoying increasing use in broadcasting station installations is the **condenser** transmitter. The condenser transmitter consists of a small block of metal called a back plate such as is shown in Fig. 14 in front of which is stretched very tightly a thin metal diaphragm. The front surface of the back plate must be as nearly plane as possible and the diaphragm must be stretched very tightly, as the diaphragm must be placed very close to the back plate although it must not touch it. This space between diaphragm and back plate is usually about .001 inch.

As can be seen from Fig. 14, one side of the 130-volt battery is connected to the diaphragm, while the other side is connected through an iron choke coil to the back plate. An electric potential of 130 volts will then exist between the back plate and the diaphragm. The back plate and diaphragm of the transmitter constitute two plates of a small condenser. A charge will, therefore, exist. Sound waves striking the diaphragm will cause it to move back and forth thereby varying the distance between the diaphragm and the back plate. This variation in distance between diaphragm and back plate results in a change in the capacitance of the condenser. As the diaphragm moves closer to the back plate, the capacitance increases, while during that half of the cycle when it moves away from the back plate, the capacitance decreases.

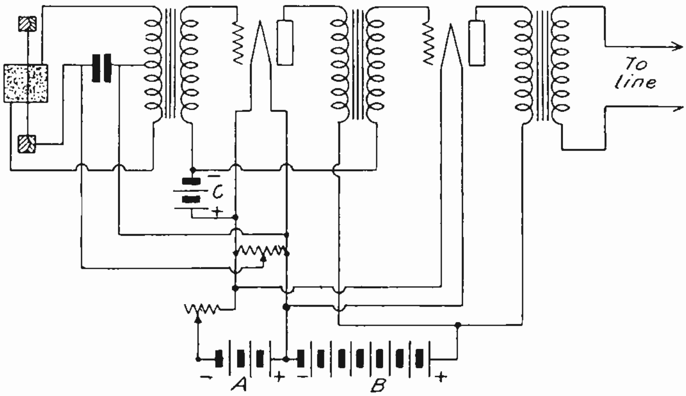

Fig. 13—Two button microphone connected to the input of a two stage amplifier.

The effect of the large iron choke coil is to oppose any change in current. The voltage across the condenser, the charge on the plates of a condenser, and capacitance of the condenser are always related by the formula—**charge = capacitance** times **voltage**. Algebraically, this equation is given as follows:

$$Q = e \times V \qquad (1)$$

A change in the amount of charge upon the condenser plates can only result by a flow of current through the choke coil. The inductance of this coil is, however, so great that practically no current will flow through it at audio frequency. As the capacitance of the condenser increases due to motion of the diaphragm, towards the back plate, the voltage across the condenser must decrease since the charge remains constant. As the capacitance of the condenser decreases, due to motion of the diaphragm away from the back plate, the voltage across it must increase. In other words, the product obtained by multiplying capacitance by voltage must always be constant. If one increases, the other must decrease, etc.

As can be seen from Fig. 14, the condenser microphone, choke coil, and battery are so connected that the alternating voltage produced by the motion of the diaphragm is applied to the grid of the first tube of an audio-frequency amplifier. These voltages are then amplified and may be used to modulate a radiotelephone transmitter.

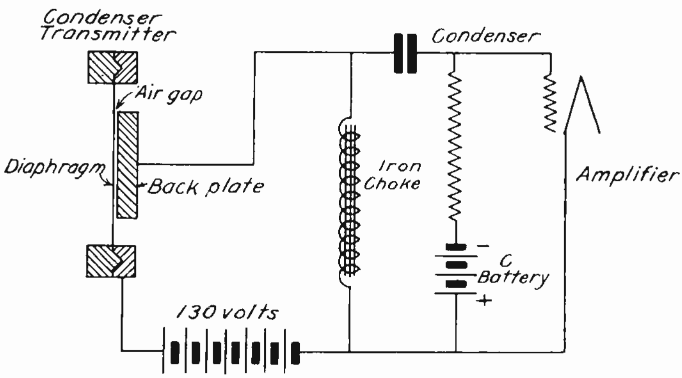

Fig. 14—Schematic circuit showing the use of a condenser transmitter.

The amount of electric energy produced from transmitters of this type is even less than that produced by the two-button carbon microphone. However, the frequency characteristics of a conversion system such as this are superior even to those of the two-button microphone and consequently, the condenser transmitter is particularly applicable to handling speech and music. It is, however, necessary to use more stages of amplification than with the ordinary two-button type.

There is one other device which should be discussed at this time although it is not used to convert sound waves directly to electric current. This is what is known as the magnetic pick-up. This device is used to convert energy mechanically from a phonograph record to electric energy which can then be amplified and used either to operate a loudspeaker or to modulate the output of a radiotelephone broadcasting set. Briefly described, the theory of operation of the magnetic pick-up is somewhat as follows: As the needle of the device traverses the channels cut upon the phonograph record, the motions imparted to it cause

4 6 9

a small armature to move in a magnetic field. The motions of this armature in turn cause variations in the number of magnetic lines which link with a coil of wire. These in turn set up voltages in a coil of wire which correspond to the sound wave as recorded upon the record. The electric energy produced may then be amplified and used to operate a reproducer or to modulate a radio-telephone transmitter as the case may be.

As has been stated above, the electric energy obtained from either the two-button carbon microphone as used in broadcasting stations or from the condenser transmitter is so small that it must be amplified. Accordingly, these conversion devices are connected to the input terminals of multi-stage vacuum tube amplifiers known as **input** amplifiers.

Fig. 13 is a schematic circuit diagram showing how a two-button carbon microphone might be connected to the input terminals of a two stage vacuum tube amplifier. This is not meant to be a working diagram but only to illustrate the theory of operation of the microphone and amplifier. A practical amplifier would have to have associated with it a device for controlling the **amplification** or **gain** of the device. Detailed information concerning input amplifiers, mixing systems, volume indicators, signalling systems, and monitoring systems will be given in a later lesson devoted to the operation of practical broadcast equipment.

THE TRANSMISSION UNIT AND ITS USE

In the first part of this lesson we have studied in some detail the nature of sound, particularly such sounds as are produced by the human voice and by musical instruments. We have considered the range of frequencies to which the human ear is ordinarily sensitive. We have also considered and discussed the small amounts of power which are involved in sound waves. We have found that **amplitude** determines **loudness,** **frequency** determines **pitch** and that the relative amounts of energy in fundamental and harmonics determine **quality** or **timbre.** Sound waves as ordinarily received by the human ear travel through air with a velocity of 1080 feet per second at ordinary room temperature. In radio broadcasting stations, it is essential that the energy contained in sound waves be converted to electric energy so that it may be used to modulate the output of the high frequency transmitting device. We have now reached a stage in our discussion where we must digress from the consideration of fundamental principles and discuss methods of measuring the electric power resulting from the operation of the microphone and the apparatus associated with it.

You are undoubtedly quite familiar with the general terms used in the measurement of electric power, potential and current in electric power systems such as are used for house lighting. Electric pressure is measured in volts. Electric current is measured in amperes. Power is measured in watts. Energy delivered to a house is measured in watt-hours. These are all comparatively familiar terms and it might seem that in referring to the power produced by a microphone, the power delivered to a telephone line by an amplifier, etc., that we might speak in terms of watts or some sub-multiple such as the microwatt (millionth

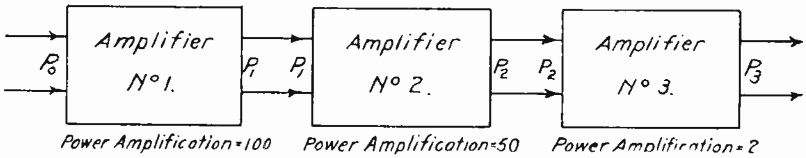

Fig. 15—Block diagram showing three amplifiers connected in series.

watt). For instance, we might say that a carbon microphone produces 10 microwatts of electric power, an amplifier delivers .5 watt to a line, or a loudspeaker requires 25 watts to produce satisfactory volume in a given room. Although this might seem a logical course of procedure, as a matter of fact, an entirely different set of nomenclature and an entirely different way of expressing the characteristics of amplifying apparatus, etc., has been developed. There are, as you will find after having had some experince with this new system, certain very decided advantages which it holds as compared with the system used in connection with power apparatus. The new system which we are about to describe in some detail is known as the Transmission Unit System, commonly known as the TU. Recently the term "Decibel" or "DB" was adopted by nearly all countries for use instead of TU. Wherever TU occurs in place of DB in older writings they are interchangeable, as the values are the same.

TRANSMISSION UNIT OR DECIBEL

Let us assume that we connect a microphone to the input of the three amplifiers shown in Fig. 15 and that a given amount of sound power is then delivered to the microphone. Let us assume that this power (P_o) is 1 microwatt, (.000,001 watt). Let us further assume that the power output of the first amplifier (P_1) is 100 microwatts, of the second amplifier (P_2) 5000 microwatts and of the third amplifier (P_3) 10,000 microwatts.

The power amplification factor for the first amplifier can be obtained by dividing the power output (P_1) by the power input (P_o), or

$$\text{Factor for amplifier No. 1 } (F_1) = \frac{P_1}{P_o} = \frac{100}{1} = 100 \tag{1}$$

$$\text{Factor for amplifier No. 2 } (F_2) = \frac{P_2}{P_1} = \frac{5000}{100} = 50 \tag{2}$$

and

$$\text{Factor for amplifier No. 3 } (F_3) = \frac{P_3}{P_2} = \frac{10{,}000}{5{,}000} = 2 \tag{3}$$

Suppose now we wish to obtain the power amplification factor for the entire combination. This is given as follows:

$$\text{Factor for all three } \left(F_{1+2+3}\right) = \frac{P_3}{P_o} = \frac{10{,}000}{1} = 10{,}000 \tag{4}$$

This may be obtained from the individual amplification factors by multiplying them together.

$$\left(F_{1+2+3}\right) = 100 \times 50 \times 2 = 10{,}000 \tag{5}$$

We must now turn our attention to certain algebraic expressions which are called **logarithms.** If a number is multiplied by itself, we say that the number has been squared. Thus 4 is the square of 2 since $2 \times 2 = 4$. This is sometimes written 2^2 which is read "2 to the second power." If we multiply the number by itself again, we have $2 \times 2 \times 2 = 8$. This may be written 2^3, that is "2 to the third power." Likewise, $2 \times 2 \times 2 \times 2 = 2^4 = 16$ and $2 \times 2 \times 2 \times 2 \times 2 = 2^5 = 32$, and so on.

Now let us consider the number 10.

$10^1 = 10$ (10 to the first power is 10).
$10^2 = 10 \times 10 = 100$ (10 to the second power is 100).
$10^3 = 10 \times 10 \times 10 = 1000$ (10 to the third power is 1000).
$10^4 = 10 \times 10 \times 10 \times 10 = 10{,}000$ (ten to the fourth power is 10,000), etc.

Now the logarithm of a number is the power to which 10 must be raised to obtain the number. Thus, the logarithm of 10 is 1 since $10^1 = 10$; the logarithm of 100 is 2 since $10^2 = 100$, and the logarithm of 1000 is 3 since 10^3 is 1000, etc., etc. Now, if the logarithm of 10 is 1 and the logarithm of 100 is 2, then the

TABLE NO. 1

LOGARITHM TABLE

Numbers 1 to 100

No. Logarithm	No. Logarithm	No. Logarithm	No. Logarithm	No. Logarithm
1—0.000	21—1.322	41—1.612	61—1.785	81—1.908
2—0.301	22—1.342	42—1.623	62—1.792	82—1.913
3—0.477	23—1.362	43—1.633	63—1.799	83—1.919
4—0.602	24—1.380	44—1.643	64—1.806	84—1.924
5—0.698	25—1.398	45—1.653	65—1.813	85—1.929
6—0.778	26—1.415	46—1.663	66—1.819	86—1.934
7—0.845	27—1.431	47—1.672	67—1.826	87—1.939
8—0.903	28—1.447	48—1.681	68—1.832	88—1.944
9—0.954	29—1.462	49—1.690	69—1.839	89—1.949
10—1.000	30—1.477	50—1.698	70—1.845	90—1.954
11—1.041	31—1.491	51—1.707	71—1.851	91—1.959
12—1.079	32—1.505	52—1.716	72—1.857	92—1.964
13—1.114	33—1.518	53—1.724	73—1.863	93—1.968
14—1.146	34—1.531	54—1.732	74—1.869	94—1.973
15—1.176	35—1.544	55—1.740	75—1.875	95—1.978
16—1.204	36—1.556	56—1.748	76—1.881	96—1.982
17—1.230	37—1.568	57—1.758	77—1.886	97—1.987
18—1.255	38—1.579	58—1.763	78—1.892	98—1.991
19—1.278	39—1.591	59—1.770	79—1.898	99—1.996
20—1.301	40—1.602	60—1.778	80—1.903	100—2.000

logarithm of a number between 10 and 100 must be some number between 1 and 2. For instance, it can be proven that the logarithm of 20 is 1.301; this is written—

$$\log 20 = 1.301 \tag{6}$$

Similarly, it can be proven that—

$$\begin{aligned} \log 40 &= 1.602 \\ \log 60 &= 1.778 \\ \log 80 &= 1.903 \end{aligned} \tag{7}$$

and so on. Note that we have not derived the above equations. While it is, of course, possible to derive the expressions for the logarithms of numbers, the mathematical processes are so long that tables known as **logarithm tables** have been published. These can be used to obtain the logarithms of numbers to a high degree of accuracy. Some of these carry the decimal as far as seven places.

Table 1 gives the logarithms for all numbers from 1 to 100, to the third decimal place. Note that the logarithm for any number between 1 and 10 lies between 0 and 1, while the logarithm for any number between 10 and 100 lies between 1 and 2. Note also that while the logarithm of 2 is 0.301, the logarithm of 20 is 1.301. Similarly, the logarithm for 200 would be 2.301 and for 2000 it would be 3.301, etc. Thus, if we know the log for any figure between 10 and 100, we can easily calculate the logarithms of any number obtained by adding zeros to this number. Thus, the logarithm of 45 is 1.653.

$$\begin{aligned} &\log 450 = 2.653 \\ &\log 4500 = 3.653 \\ &\log 45{,}000 = 4.653 \\ &\text{etc.} \end{aligned} \tag{8}$$

In a logarithm, the figure to the left of the decimal point is called the **characteristic** while the figures to the right make up the **mantissa.** Thus 4 is the **characteristic** of the log of 45,000 while 653 is the **mantissa.** Logarithm tables usually give only the mantissa.

Logarithms are very useful in solving certain kinds of mathematical problems. If you are interested in them you would find it well worth while to obtain a copy of a book on algebra and to read the chapters on logarithms. We are concerned with logarithms in this lesson because the **gain** and **loss** characteristics of amplifiers, lines and apparatus used to transmit speech and music are measured in terms of units which can be calculated **only** by the aid of logarithms.

The **Decibel** method of expressing the characteristics of speech and music transmission apparatus is based upon the logarithms of power ratios. (By power in this case, we usually mean electric power.) The **gain** in Decibels or **DB,** as they are called, introduced by an amplifier, **is obtained by taking the logarithm of the ratio of the POWER OUTPUT of the amplifier to the POWER INPUT and multiplying this logarithm by 10.** The mathematical expression for this is:

$$\text{DB gain} = 10 \log \frac{\text{P (output)}}{\text{P (input)}} \tag{9}$$

The ratio of power output to power input for amplifier No. 1 as shown in Fig. 15 is 100, the logarithm of 100 is 2, and

$10 \times 2 = 20$. The gain in DB introduced by this amplifier is 20 DB. Mathematically, this may be written:

$$\text{DB gain for amplifier No. 1} = 10 \log \frac{P_1}{P_0} = 10 \log \frac{100}{1} = 20 \tag{10}$$

Similarly—

$$\begin{aligned} \text{DB gain for amplifier No. 2} &= 10 \log \frac{P_2}{P_1} = 10 \log \frac{5000}{100} \\ &= 10 \log 50 \\ &= 10 \times 1.698 \\ &= 16.98 \end{aligned} \tag{11}$$

(The logarithm of 50 can be obtained from Table 1 or from Fig. 16.)

Now 16.98 is approximately 17, so we would probably say that the gain introduced by this second amplifier is 17 DB. Also

$$\begin{aligned} \text{DB gain for amplifier No. 3} &= 10 \log \frac{P_3}{P_2} = 10 \log \frac{10{,}000}{5{,}000} \\ &= 10 \log 2 \\ &= 10 \times .301 \\ &= 3.01 \end{aligned} \tag{12}$$

or since 3.01 is approximately 3, we would say the gain introduced by the amplifier No. 3 is approximately 3 DB.

Let us now consider the three amplifiers together. The total power amplification as a whole is $\frac{P_3}{P_0} = \frac{10{,}000}{1}$ or 10,000. To obtain this factor from the power amplification factors for the three individual amplifiers, we must **multiply** them together.

$$\begin{aligned} &\text{Factor for No. 1} = 100 \\ &\text{Factor for No. 2} = 50 \\ &\text{Factor for No. 3} = 2 \\ &\text{Factor for three amplifiers in series} = 100 \times 50 \times 2 = 10{,}000 \end{aligned} \tag{13}$$

Now the DB gain for the combination of the three amplifiers is computed as follows:

$$\text{DB gain} = 10 \log \frac{10{,}000}{1} = 10 \log 10{,}000 = 10 \times 4 = 40 \tag{14}$$

Since the logarithm of 10,000 is 4 as we have seen. Now note how we obtain the gain in DB for the three amplifiers used together making use of the DB gain for each individual amplifier.

Gain introduced by Amplifier No. 1 = 20 DB
Gain introduced by Amplifier No. 2 = 17 DB (15)
Gain introduced by Amplifier No. 3 = 3 DB
Gain introduced by combination of three amplifiers = 40 DB

You can, therefore, see that by the use of logarithms we have substituted **addition** for **multiplication.** To obtain the power amplification factor for a series of amplifiers, the factors for the individual amplifiers must be **multiplied** together. To obtain the DB gain for a series of amplifiers, the DB gains for the individual amplifiers must be **added** together. I think you can now begin to see some of the advantages of the DB system of expressing **gain** and **loss.**

The three amplifiers such as are shown in Fig. 15 are presumably so located that the power output of amplifier No. 1 is equal to the power input of amplifier No. 2 and the power output of Number 2 is equal to the input of Number 3. This would not be the case in the event the amplifiers were separated by appreciable distances. For instance, amplifier No. 1 might be at a baseball park from which it is desired to broadcast a description of a World Series game. No. 2 might be in a studio control room three or four miles away, while No. 3 might be at the radio transmitter fifteen or more miles from the studio. With such an arrangement, power losses would take place in the lines connecting these three points.

Fig. 17 illustrates graphically the situation which might exist. Figures given on the diagram show average power levels at particular points throughout the system. Thus the average input power to amplifier No. 1 is 1 microwatt, its output 100 microwatts. The attenuation of Line No. 1 connecting the output of amplifier No. 1 with the input of amplifier No. 2 is such that the input power to amplifier No. 2 is only 20 microwatts. The power amplification factor for amplifier No. 1 is therefore $\frac{100}{1} = 100$. This factor is obtained by dividing the power **output** by the power **input.**

The **attenuation** factor for Line No. 1 is obtained by dividing the power **input** to the line by the power **output** from the line. The power **input** to Line No. 1 is the power delivered to it by

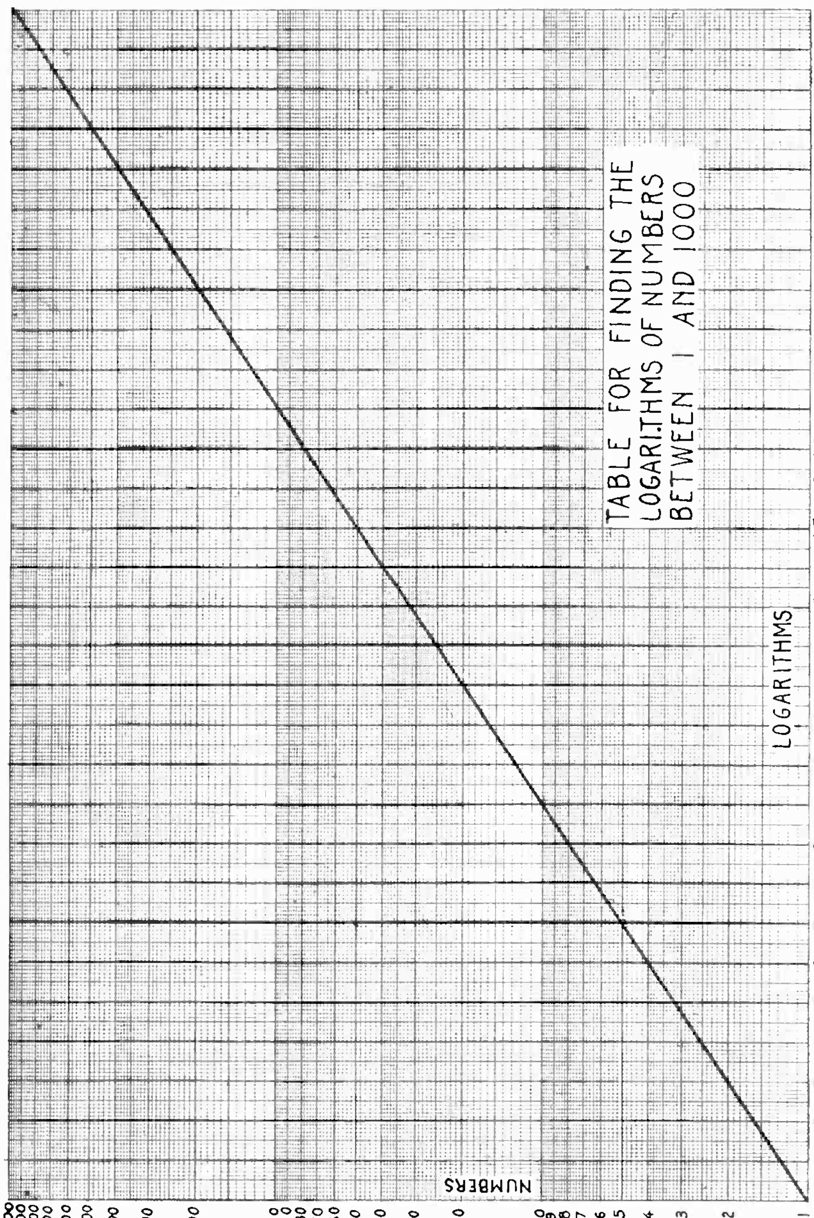

Fig. 16—Graph for producing the logarithms of numbers between 1 and 1000.

amplifier No. 1 which is 100 microwatts. The power output from Line No. 1 is the power the line delivers to amplifier No. 2 which is 20 microwatts. The attenuation factor for Line No. 1 is therefore $\frac{100}{20} = 5$.

The **loss** in DB introduced by a line or any piece of apparatus which introduces attenuation is calculated in exactly the same way as **gain** except that the logarithm of the ratio of the power **input** to the power **output** is used instead of the ratio of the power **output** to the power **input.** Thus, the loss introduced by Line No. 1, Fig. 17, is obtained by taking the logarithm of $\frac{100}{20}$ and multiplying it by 10

$$
\begin{aligned}
\text{Loss in DB (Line No. 1)} &= 10 \log \frac{100}{20} \\
&= 10 \log 5 \\
&= 10 \times .698 \\
&= 6.98 \\
&= 7 \text{ approximately}
\end{aligned}
$$

(16)

In exactly the same way, the loss introduced by Line No. 2 may be calculated.

Suppose now that we desire to find the total **gain** or **loss** due to the combination of lines and amplifiers between microphone and the output of amplifier No. 3. We first add the **gains** in DB, then separately add the **losses** in DB. If the sum of the **gains** exceeds the sum of the losses, then we obtain the total amplification by subtracting the losses from the gains.

Loss Line No. 1 = 7 DB
Loss Line No. 2 = 20 DB
Total Loss = 27

Gain Amplifier No. 1 = 20 DB
Gain Amplifier No. 2 = 17 DB
Gain Amplifier No. 3 = 3 DB

Total Gain = 40 DB
Less 27 DB

Total gain in system = 13

The graph shown in Fig. 18 provides a very convenient method for obtaining gain or loss in DB from power ratios or for obtaining the power ratios from gain or loss in DB. It can be proven that if the input impedance of an amplifier is equal to the impedance of the load across the output, the **gain** in DB introduced is given by

$$\text{Gain in DB} = 20 \log \frac{\text{I out}}{\text{I in}} \tag{17}$$

in which I out is the alternating **current** delivered to the load and I in is the alternating input **current.** Also, it can be proven that under the same condition

$$\text{Gain in DB} = 20 \log \frac{\text{E out}}{\text{E in}} \tag{18}$$

in which E out is the alternating **voltage** across the output and E in is the alternating voltage across the input. Curve 1, Figure 18, is for use with power ratios while Curve 2 is for use with current or voltage ratios.

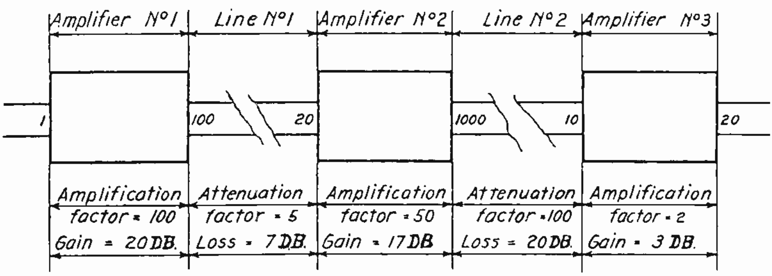

Fig. 17—Graphic representation of Decibel system including amplifiers introducing gain and lines introducing losses.

REFERENCE VOLUME

We have discussed the Transmission unit or Decibel System as applied to the measurement of **gain** and **loss.** In other words, we have used this system to describe the characteristics of amplifiers, apparatus and lines. In our discussions, we have assumed that the energy amplified has been uniform in character such as might be the case if the electric voltage applied were a combination of pure sine waves all of constant amplitude. We may also use the Decibel System as a means of expressing **power level.** If the energy under discussion consists of pure sine waves of constant amplitude, then the amount of power involved is a very definite and constant quantity. If on the other hand we are concerned with the electrical energy produced by a broadcast program consisting of speech and music, then the amount of power involved varies over a very wide range. The amount of power may be very small during a **soft** passage in the musical scale but may be many times this during some of the louder passages. In specifying the amount of power involved in a broadcast program, we usually mean an approximate average value, the measurement of which will be described later when we discuss volume indicators.

Let us select a specific amount of power for reference purposes. This power we will call **reference** volume. Ordinarily, 1 milliwatt (.001 watt) is referred to as reference volume. All other powers can be referred to as so many DB above or below **reference volume.** Thus, a power of 10 milliwatts (.01 watt) may be referred to as a volume 10 DB above reference volume. This can be determined from the curve shown in Fig. 18 or from the formula—

$$\begin{aligned}\text{Volume} &= 10 \log \frac{10 \text{ milliwatts}}{1 \text{ milliwatt}} \\ &= 10 \log 10 \\ &= 10 \times 1 \\ &= 10 \text{ DB above reference volume}\end{aligned} \tag{19}$$

In a like manner, the volume represented by .1 milliwatt (.0001 watt) would be calculated from the formula—

$$\begin{aligned}\text{Volume} &= 10 \log \frac{1 \text{ milliwatt}}{.1 \text{ milliwatt}} \\ &= 10 \times \log 10 \\ &= 10 \times 1 \\ &= 10 \text{ DB below reference volume}\end{aligned} \tag{20}$$

You will note that we refer to powers greater than reference volume as so many DB **above** reference volume and powers less than reference volume as so many DB **below** reference volume. In our computations for power above reference volume, we take the ratio of the power in question to the power corresponding to reference volume (1 milliwatt), while in computations where power is less than reference volume, we take the ratio of the power corresponding to reference volume to the power in question as shown by equations 20 and 21, respectively. 10 DB above reference volume is often written + 10 DB, 10 DB below reference volume is often written — 10 DB.

On the basis of the information given in the paragraph above, let us compute the **power levels** existing at various points

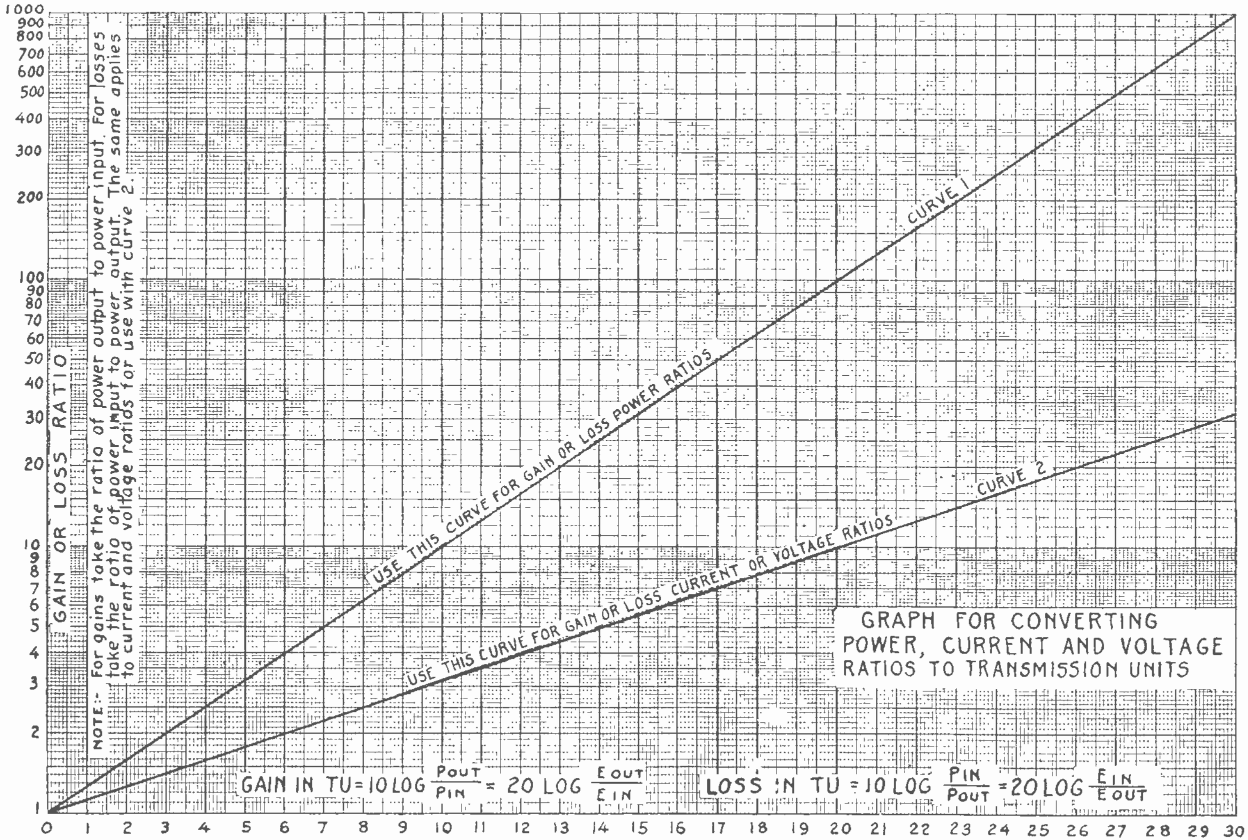

Fig. 18—Graph for converting power, current and voltage ratios to transmission units. TU or DB.

in our amplifier system as shown in Fig. 17. The results are shown in a following table.

1 microwatt	= 30 DB below reference volume	(—30 DB)
100 microwatts	= 10 DB below reference volume	(—10 DB)
20 microwatts	= 17 DB below reference volume	(—17 DB)
1000 microwatts	= 0 DB or reference volume	

The following table shows the volume level for a number of powers using .001 watt as reference volume. (Note: 1 milliwatt = .001 watt, 1 microwatt = .000,001 watt.)

Power in microwatts	Power in milliwatts	Level
1	.001	—30 DB
10	.01	—20 DB
100	.1	—10 DB
1,000	1	0
10,000	10	+10 DB
100,000	100	+20 DB
1,000,000 (1 watt)	1000	+30 DB

USEFUL CONSTANTS APPLICABLE TO SPEECH INPUT EQUIPMENT

We are now prepared to understand and make use of certain constants with respect to speech and music and apparatus designed to handle them. The information contained in the following paragraphs has been gleaned from several sources but mostly from articles published by members of the Bell Telephone organization. The engineers of this organization, because of their extensive studies with respect to the transmission of speech and music, have probably contributed more to the knowledge of mankind on this subject than any other group.

The range of audibility may be said to extend from about 20 to 20,000 cycles. This range, as has been pointed out, varies with individuals.

If a transmission system is to handle only those frequencies necessary for the transmission of intelligible speech, then only the band of frequencies lying between 200 and 2000 cycles need be accommodated. The "naturalness" of the voice will, however, to a certain extent be lost if this band only is accommodated.

The fundamental frequencies produced by the human voice and by musical instruments lie approximately between 30 and 4,000 cycles. However, some of the harmonics generated lie above 10,000 cycles per second. The faithful reproduction of speech and music requires provision for a band of frequencies lying between 30 and 10,000 cycles. However, fairly satisfactory fidelity will result if the band lying between 50 and 5,000 cycles is satisfactorily accommodated. Most broadcast installations and the lines connecting them are engineered to accommodate the band lying between 50 and 5,000 cycles.

The average human ear is most sensitive to sounds having a frequency lying between 2,000 and 4,000 cycles per second. For this frequency, the approximate minimum sound power which will produce the sensation of sound is .0,000,000,000,004 milliwatt per square centimeter. This is a power level approximately 124 DB below reference volume (assuming reference volume to be one milliwatt).

The ratio of the amount of sound power produced by an orchestra playing very loudly to the sound power produced while playing some of the softer portions of a selection is very great. The loud portions possess 30,000 times the power possessed by the soft portions. That is to say, the difference in level is approximately 45 DB. The average electric power output from a two-button carbon microphone such as is used for broadcasting is about 10 milliwatts (10 DB above reference volume) for a speaker close to the microphone and about .001 milliwatt (30 DB below reference volume) if the speaker is five feet away. The output from a condenser transmitter under similar circumstances will be considerably less.

TEST QUESTIONS

Number your answers 30—2 and add your **Student Number.**

1. State at what frequencies the human ear is most sensitive.
2. How are sound waves produced?
3. What is the function of a microphone in a Broadcasting Station?
4. Draw a circuit diagram showing the connections for a single button microphone.
5. What is the advantage and disadvantage of a single button microphone?
6. Name several devices which can be used to convert sound waves to electric current.
7. Draw a circuit diagram showing a double button microphone and 2-stage amplifier.
8. Upon what does the loudness, pitch and quality or timbre of musical sounds depend?
9. Explain the action of a magnetic pick-up.
10. What unit is used for expressing transmission loss or gain (amplification)?

RADIO BY MAIL

National Radio Institute

STUDENTS ALL OVER THE WORLD