CHANGING SOUND INTO ELECTRICITY AND ELECTRICITY INTO SOUND

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NATIONAL RADIO INSTITUTE
EST. 1914
WASHINGTON, D.C.

WHY THE STUDY OF SOUND IS NECESSARY

Sound Engineers have played a great part in the development of Radio. Scientists in the employ of large telephone companies have spent years in studying sound and the means of handling sound electrically.

The problems of the Telephone Companies are often the problems of Radio. In solving them they also aid in the perfecting of Radio. The two fields are linked together by one thing, their common interest in *sound*.

The owner of a Radio receiver expects good sound reproduction from the speaker; the audience in a "talkie" theatre wants to hear the words of the actors clearly; it is essential that a Public Address System reproduce sounds faithfully. From this it can be seen that the Radio Expert must make a thorough knowledge of sound, part of his equipment.

I know you are going to find this lesson, not only one of the most interesting but one of the most useful lessons in the course.

J. E. SMITH.

Changing Sound Into Electricity And Electricity Into Sound

THE NATURE OF SOUND

The study of sound and its relation to Radio is intensely interesting. It is sound that is fed into the transmitter—it is sound that is reproduced in the receiver. Sound waves actuate the microphone in the broadcasting station—and at the receiving end the loudspeaker produces sound waves.

The phenomenon of sound is closely associated with the sense of hearing and sound is usually defined as any effect that can be perceived by the ear. There is an old trick question, usually put something like this, "If there were an explosion in the middle of a desert and no one were there to hear it, would sound be produced?" We won't attempt to answer this question here—it is a question that can be answered in the affirmative and in the negative, and each would be correct depending on the point of view. The reason for mentioning it at all is because it brings out very clearly that there are two ways of considering sound, one as a cause, the other as an effect.

Fundamentally, the idea of sound and the idea of vibration are inseparable. Sound is produced by vibration, and sound travels in the form of successive vibrations, or as we usually say, sound energy travels in the form of sound waves. We become conscious of sound when sound waves strike the eardrums, causing them to vibrate.

Strike a gong—you will hear a musical note which lasts an appreciable length of time. Strike it again and touch the gong gently with your finger—you can feel the vibrations. If you cover the gong with your whole hand and press on it firmly, the sound will stop. You have damped the vibration of the gong, and when there is no vibration there is no sound. You may have watched the trap drummer in an orchestra strike the cymbal, then immediately lay his hand on it. He does this to stop its vibration so that it won't continue to sound.

Vibrations which result in sound are often visible. If you draw a rubber band taut and give it a flip with your finger, you can see it vibrate back and forth and you know it is producing sound because you can hear it. Other more scientific experiments will also show this effect beautifully. If a tuning fork

1

(a device such as piano tuners use as a standard for pitch) is struck and then touched to a small glass ball suspended by a thread, the ball will be set into violent motion. (See Fig. 1.)

We ordinarily think of sound as traveling through air, for it is through the air that most sounds reach the ear. However, sound also travels through other substances—liquids and solids. It may seem peculiar that sound travels through a solid but if you put your ear up against a table and scratch it some distance away you will be able to hear the scratching very distinctly. It is not so difficult to realize that sound travels through a liquid for if you have ever been swimming under water in a pool and someone dove into the water, you heard the splash very distinctly.

If sound travels through various media such as air, solids and liquids, we can conclude that there must be something common to all these. This common property is called elasticity by which we mean that the air or solid or liquid can be *stretched* or *compressed* and when the force applied is removed the material will return to its normal state. The elasticity of a rubber band can easily be seen. If we draw it taut, then give it a flip with the finger, it will vibrate for a while and when the energy imparted to it by the finger has been used up the vibration will stop and the rubber band will return to the position it was in before it was disturbed.

That air has elasticity can be demonstrated with an ordinary tire pump. Close the outlet by holding your finger across the hose connection then push down on the plunger. You will find that you can push it down only part of the way and then if you release the pressure the plunger will fly back. What you have done is that you compressed the air, you caused the air particles to be packed closer together than they are normally. When you released the pressure the air particles resumed their normal position with respect to each other and it is this property that characterizes all substances having elasticity.

That a liquid or a solid such as steel or iron has elasticity is not so apparent and yet sound travels through these much more rapidly than through air. You may at some time or other have constructed a home-made telephone consisting of two tin cans with a string or wire drawn tightly between them, using one tin can as a microphone and the other as a receiver. Evidently the sound traveled through the string or wire. And now we are going to explain this simple phenomenon.

You know that the wire or the string as the case may be

is composed of very small particles called "molecules." Under normal condition these molecules are in a state of equilibrium, that is, they are all packed together to the same degree, depending upon the structure of the substance. If the substance is caused to vibrate in any way, the equilibrium of the molecules is disturbed. When you spoke into the can which was serving as a microphone, the sound vibrations caused the string or wire to vibrate—the molecules in it moved to and fro following your voice vibrations.

This movement of molecules constitutes a transfer of energy. To visualize this transfer of energy imagine a number of pool balls lined up so that all are touching. Now strike another ball with the cue so that this ball strikes the end ball in the line. Each
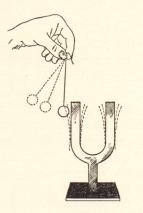

Fig. 1
Vibration
of
tuning fork

ball will move very slightly but the last ball will move away from the line. If done very carefully all the other balls will still be in a straight line and will still be touching each other. What has happened? The energy given to the free ball by the cue has been passed along the line and has been given to the end ball, but the balls which acted merely to transfer the energy moved hardly at all.

From this we can gather that when sound travels through air, the air particles actually move to and fro. This can be proven very nicely in a laboratory experiment in which a small flame is put out by the movement of air particles involved in sound.

From this it can also be seen that sound will travel faster through a medium in which the molecules are packed closely together than through a medium in which the molecules are loosely packed. In a solid the molecules are packed very closely to-

gether. Therefore, sound should travel faster through a solid than through a liquid or air. That this is actually the case will be brought out very clearly in a following section of this book.

It must be remembered that while the molecules do actually move when there is a transfer of sound energy, they don't get anywhere, they just move backward and forward in much the same way as the electrons in alternating current which move backward and forward but never get anywhere.

You will remember that an Indian could determine whether anyone were walking in the immediate vicinity by pressing his ear against the ground. Even the very weak sound waves that were transmitted to the earth by a moccasined foot on dry leaves and twigs could be heard in this way over considerable distances.

From all this we can conclude that some medium is necessary for the transfer of sound energy and that the energy is transferred by the motion of the particles or molecules composing that medium. If there are no molecules there can be no transfer of energy. If this is true, sound will not travel through a vacuum. The fact of the matter is that sound does not travel through a vacuum. If you put an electric bell in an air tight glass container and withdraw all the air and gases so that a vacuum exists within the tube, no sound will be heard from the bell even though you can see it vibrating. But allow air to enter the chamber and the sound will appear, growing louder as more air enters the tube.

SOUND WAVES

If you drop a stone into a pond or a still body of water, a wave is created in the water which gradually increases in diameter as it travels away from the point where the stone hit. If there is a cork floating on the surface of the water, when the wave passes it, it will bob up and down with the wave but will not travel forward with the wave.

It could be observed also that as the wave travels it becomes smaller in height, that is, its amplitude becomes less. The reason for this is apparent; when the stone strikes the water it gives to the water its energy which is carried along by wave motion. As the diameter of the wave increases, the energy present must act over a much wider area, consequently the energy that is acting along a particular straight line decreases.

The fact that the cork bobs up and down rather than moving with the wave, shows that the water particles move up and down

4

while the wave form itself moves along the surface. That is, the motion of the particles is at right angles to the direction in which the wave is traveling. For this reason, water waves are said to be transverse waves.

Much that can be said about water waves is true also about sound waves—there is a transfer of direct energy into a wave motion having energy and this energy becomes less as the distance from the source increases. And just as there is a movement of water particles in a water wave, so there is a movement of air particles in a sound wave. At this point, however, the analogy stops for sound waves are not transverse waves, they are longitudinal waves. By this we mean that the air particles do not move up and down, but move to and fro.
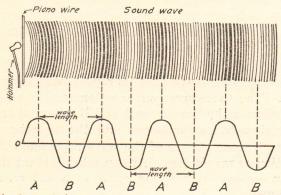

Fig. 2—Radiation of sound waves set up by vibrations of a piano wire struck by a hammer.

When a bell or a piano wire is struck, it will vibrate at the rate of several hundred times a second more or less. Each vibration hits the surrounding air a slight blow and the particles of air in that portion which struck will be pushed together and condensed. At the same time, as the bell or the piano wire moves forward to push the air, there will be a partial vacuum behind the striking surface. In the case of the piano wire, as it moves, air particles rush in to fill the space which had been occupied by the wire. The condition before the air rushes in is a rarefaction, a thinning out, of the air behind the wire. After the wire swings as far as it can in one direction it will swing back past the normal or original position and cause a condensation and rarefaction on opposite sides. Thus if a string vibrates rapidly there will be alternate condensations and rarefactions set up in the air in front of it and behind it—the air particles will be moved back-

5

ward and forward and in this way the energy which was used to cause the bell or piano wire to vibrate is sent out in the form of sound waves.

When the sound waves reach the ear, they strike the eardrums, causing them to move in and out at the same frequency and proportional intensity as the vibrations of the source.

Ordinarily, sound waves travel in all directions and in this respect too they differ from water waves which are confined to the surface of the water. A bell ringing in a steeple can be heard from any angle and if you were 20 feet above it you could hear it as well as if you were 20 feet below it. But sound waves *can* be confined and it is for this purpose that megaphones, loudspeaker horns and speaking tubes are used. By confining the radiation of sound energy to a certain direction, losses due to radiation can be largely prevented and the sounds can be heard from much greater distances than if a special device were not used.

Figure 2 will help us to get a mental picture of the manner in which a sound wave travels. This represents the condition at any instant. The successive condensations and rarefactions of the air are indicated by the curves A and B shown at the bottom, the crests represent compressions and the troughs represent rarefactions. The amplitude of the wave represents the degree of condensation or rarefaction. At any other instant the portion of maximum condensation would occur at a different position than that shown.

In a Radio loudspeaker there is a vibrating system. It is called the diaphragm. This is caused to vibrate by the electrical system of the receiver. As it vibrates successive condensations and rarefactions are set up in the air both in front of the diaphragm and behind it. However, the sound waves in front of the diaphragm are out of phase with the waves behind the diaphragm, that is, at the instant the air in front is compressed, the air behind will be rarefied, and vice versa. This incidentally explains why the two sets of waves must not be allowed to mingle and a so-called baffle must be used. If the front waves are allowed to radiate toward the back, the two sets of waves will be neutralized and the one will balance the other. It is easy to see why the one would wipe out the other, for if a condensation and a rarefaction exist at the same time, the one will make up for the other and there will be a normal distribution of air particles, resulting in no sound wave and no transfer of sound energy.

6

THE SPEED OF SOUND WAVES

You may have noticed that if you are some distance away from a steam whistle that you can see the steam escaping a considerable length of time before you can hear the whistle itself. You no doubt know that you can tell how close you are to the center of a storm by observing the lightning and then noticing how much later you hear the thunder.

Light travels at the rate of 300,000,000 meters per second or roughly 186,000 miles per second. If a man a mile away strikes a gong, you will see his motion in $\frac{1}{186,000}$ of a second, but you won't hear the sound of the gong for some time afterwards, in fact, about 5 seconds later. Sound travels at a much lower speed than light—it doesn't even travel as fast as shells fired by modern high powered guns as was shown during the war. The sound made by a passing shell was heard before the explosion of the gun was heard.

Early in the 19th Century some very interesting measurements of the speed of sound were made. In one experiment a gun was fired and the difference in time between seeing and hearing was noted. In another experiment, the velocity of sound in water was found by means of bells sounded under the waters of Lake Geneva in Switzerland.

Since then very accurate laboratory measurements of the velocity of sound waves have been made. It has been found that sound travels through still air at the rate of 332 meters per second. As a meter is approximately 3.28 feet, sound travels through air at the rate of 1089 feet per second. Sound travels through water at the rate of 1435 meters per second or 4707 feet per second. Sound waves travel through steel much more rapidly than through either air or water, the velocity being 4975 meters per second, or 16,318 feet per second.

Distances are often determined by noting the difference in time between sight and sound, especially in war time. For example, if a gun is fired and we hear the explosion six seconds after seeing the flash, then it is evident that the gun is located 6 x 332 or 1992 meters away from the observer.*

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*This would be an exact calculation only if the air is calm. If the sound was aided by the wind the speed of the wind would have to be taken into consideration. If the wind blows toward the sound source it will act as a resistance and it will take longer for the sound to reach us, depending on the velocity of the wind.

CHANGE IN DIRECTION OF SOUND WAVES

As was previously stated, sound waves ordinarily spread out from the source in all directions equally. But what if they strike an obstacle? If the obstacle is of some material that will not absorb much of the sound, as for example, a wall, the sound will be reflected in much the same way that a ray of light is reflected by a white surface.

You are familiar with the phenomenon of echo, you may know some places in your vicinity where if you whistle or call, your whistle or call will be returned to you. The explanation is of course that the sound you make is reflected straight back to you.
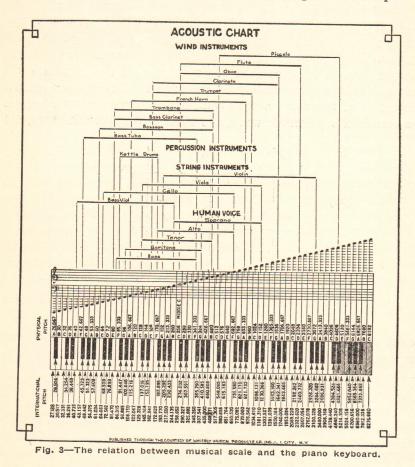
You may also have noticed that in an indoor swimming pool voices sound extraordinarily loud and if a person at the other side of the room is calling to you his words will be hardly understandable. This is because the sound waves are reflected from so many surfaces, the tile floor, the tile walls, the surface of the water, the smooth ceiling. And instead of having a single echo, we have reverberation, a bounding from one surface to another of the sound waves, very much in the same manner as a billiard ball when striking the cushion of a billiard table, will rebound from cushion to cushion until its energy is entirely dissipated and it finally comes to a state of rest.

On the other hand if it were possible to construct a room so there would be no reverberation, sound would appear flat and "dead." In a heavily carpeted, unfurnished room with sound absorbing walls and ceiling, there will still be a little reverberation and yet music or speech will appear very lifeless because our ears have grown accustomed to considerable reverberation and if it isn't there, we miss it. In a room of this sort there is so little reverberation that it is hardly noticeable—and we say the *period of continued vibration* is very short. On the other hand, the period of reverberation in our swimming pool is long.

In a concert hall the acoustics (the arrangement of walls, ceiling and furniture) must be so planned that the period of reverberation is just about right, neither too short nor too long. If too short, the music will lack brilliance—if too long it will be muddled and unintelligible.

When the surfaces of a room or hall are curved, the sound may be reflected by them to a common point called a focus. The familiar bandstand is an example of the reverse of this—it is

built to have a focus and to radiate the sound in a certain direction, very much as an automobile lamp collects light rays from the bulb and concentrates them in a definite direction.
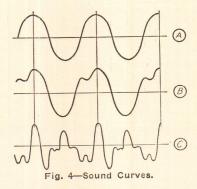
When the ceiling of a hall is arched or the walls curved, it is sometimes possible to hear a whisper at surprising distances as the sound is reflected from the walls or ceiling to a focus point.

Fig. 3—The relation between musical scale and the piano keyboard.

FREQUENCY OF SOUND WAVES

The frequency of sound waves in cycles per second is much lower than the frequency of Radio waves. Of course, the spectrum of sound frequencies necessarily includes only those frequencies which affect the human ear. The lowest tone that can be heard is an organ note of 16 cycles and the highest frequency that can be recognized as sound is about 20,000 cycles.

9

Response to extremely low and extremely high frequencies differs for various persons, depending largely on age. Very few persons over 20 years of age can detect sound frequencies as low as 16 cycles per second, and the same can be said for frequencies over 15,000 cycles per second. In orchestral music the frequency range is from approximately 41 cycles per second, the lowest note on the bass viol, to approximately 4645 cycles, the highest note the piccolo is capable of producing. The frequency range of a piano extends from about 27 cycles per second to 8277 cycles. The frequency ranges of various instruments are brought out in Fig. 3. Notice that the spectrum of the human voice extends from approximately 81 cycles to 1161 cycles per second. Two sets of frequencies are given, one the frequencies accepted as standard in scientific work, the other the frequencies accepted

Fig. 4—Sound Curves.

as standard by musicians. These latter vary slightly for different orchestras, slightly higher frequencies being often used for the sake of greater brilliance.

This leads us to another question—"Why does a voice tone of let us say 200 cycles per second, sound different from a violin tone of the same frequency?" Why is it that we can distinguish sounds produced by different instruments and easily identify sounds produced by a horn, a piano, a violin or any other sound producing instrument? A musician would tell you that the difference in sound as produced by various instruments is due to the presence of overtones, that is, tones of different frequencies than the original or fundamental frequency.

Usually these overtones are harmonics of the fundamental frequency. By this we mean that the overtone frequencies are 2, 3, 4, 5, etc., times the frequency of the fundamental. Thus the second harmonic of middle C, if we consider it as having a

fundamental frequency of 256 cycles per second would have a frequency of 512 cycles per second, the third harmonic would have a frequency of 3 x 256 or 768 cycles per second and so on. Very naturally the presence of harmonics will alter the wave form of the sound wave.

Figure 4 will make it easy for us to visualize this distortion. Curve A shows the curve of a sound wave produced by a tuning fork when struck very gently—only the fundamental frequency is present. Curve B represents a very soft violin tone and curve C represents the human voice when singing the vowel ah. In curve B the fundamental wave form is altered by the presence of a second harmonic while Curve C is further distorted by the presence of additional harmonics. Outside of the labora-
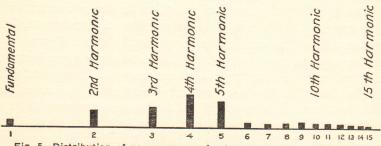
Fig. 5—Distribution of energy among fundamental and harmonics of a musical tone produced by bowing a violin.

tory practically no sounds consist of a fundamental frequency only—most sounds have very complicated wave forms due to the presence of many harmonics.

The quality or as it is usually referred to, the *timbre* of a sound depends not only on the presence of harmonics but the relative distribution of energy in the harmonics. It might naturally be supposed that most of the energy would be in the fundamental frequency but this is not always true. Look at Fig. 5, which shows how the energy is distributed between the fundamental and harmonic frequencies of a sound made by drawing a bow over a violin string. Notice that most of the energy is carried by the harmonic frequencies—that there is more energy in the fourth harmonic than in the second, third and fifth and that the energy of the fundamental frequency is less than any of these. And yet the fundamental frequency determines the pitch of the sound we hear. A trained ear will know that a violin tone having a fundamental frequency of 256 cycles has the

11

same pitch as a tone of the same frequency produced by a piano. But due to the different distribution of energy in the harmonics it is a simple matter to tell which is the violin note and which is the piano note.

The ordinary individual when listening to music recognizes that the timbre of one sound is different from the timbre of other sounds. Trained musicians can actually separate and hear as individual frequencies the harmonic overtones.

The wave forms of various sounds can be photographed and recorded by an oscillograph used in conjunction with a special camera. Figure 6 shows the word "aeolian" in picture form. Note the complex structure of the wave form even though all the sounds in the word "aeolian" are comparatively simple sounds.

The playing of a musical selection by an orchestra or an organ requires the use of a large number of fundamental frequencies and at the same time innumerable harmonic and overtone frequencies are involved. A Radio receiver reproducing an orchestral number must reproduce not only the fundamental frequencies but also the harmonic frequencies if the reproduced music is to sound natural. If the harmonics and overtones are not reproduced, the music will be lifeless, lacking in all brilliance. Realizing the problems involved in the handling of wave forms as complicated as these, containing many fundamental frequencies and many more harmonic and overtone frequencies, we can appreciate modern receiver design and workmanship which gives us receiving apparatus capable of reproducing entire musical programs with remarkable fidelity.

Now let us stop a moment and review briefly what we have spoken of in this chapter. We learned that pitch and timbre enable us to differentiate between sounds. Intensity or loudness is another property of sound by which we can differentiate between notes but this was not mentioned as it is so obvious—everyone knows that a loud tone is different from a soft tone of the same frequency and no one has any difficulty in distinguishing between loud and soft tones. When it comes to pitch and timbre not many people outside of highly trained musicians realize their true significance. But now you know that the pitch of a tone is determined by the fundamental frequency at which the source of the sound vibrates and that the distribution of energy between the fundamental frequency and the harmonics determines the timbre or quality of the sound.

It should be clear that overtones are not necessarily harmonics. Harmonics are always 2, 3, 4 or more times the fundamental frequency whereas overtones often have frequencies between the harmonic frequencies. If you listen closely to the ringing of a large bell, you will hear first the low fundamental frequency and then you will hear certain higher tones. The first overtone might have a frequency one and a half times the fundamental frequency, the second overtone might be the second harmonic, the third overtone might be two and a half times the fundamental frequency and so on. For this reason bell making is really an art—it requires real skill to build a bell whose overtones will combine to form a pleasing tone.
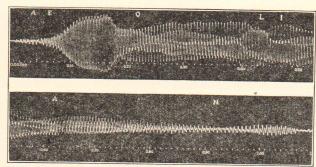

Fig. 6—Picture of a sound wave showing overtones.

INTERFERENCE OF SOUND WAVES

When two adjacent keys of a piano, near the bottom of the keyboard are struck simultaneously, a distinct throbbing of sound can be heard. That is, the sound is loud at one instant and soft the next instant. The frequency of the throbbing is so slow that one can almost count the throbs.

The explanation of this lies in what is known as the phenomenon of beats. The two individual tones have frequencies almost the same, possibly there is only two cycles per second difference between them. Consequently, as the sound is prolonged the faster frequency will run ahead of the slower frequency, then it will gain a cycle and again be in phase with the slower frequency. When the two frequencies are in phase their intensities add, the amplitude of the total sound will be twice that of a single sound and the two combined will sound twice as loud as one by itself. But as the frequency runs ahead of the slower, they will gradually draw out of phase. When completely out of phase, the one will balance the other and practically no sound will be

13

heard. When the two draw together again, the sound will build up until both frequencies are in phase when the sound is again at maximum.

The number of throbs per second is known as the beat frequency and it is always the difference between the original frequencies. If one of our frequencies is 30 cycles per second and the other 32, the beat frequency formed will be 2 cycles per second and we will hear two throbs or two beats each second. Refer to Fig. 7.

Organ tuners make use of this phenomenon of beats in their work very often. They tune a certain set of pipes by ear, using a tuning fork for a standard and then tune all the other sets of pipes by listening for the beat between them and the properly tuned set. If a particular note is out of tune with the tuned note, a beat will appear. As adjustments are made the beat will become less and less and finally disappear. When the beat is no longer present, the two pipes are exactly in tune.

If chimes are played too rapidly there will be considerable beating between overtones and the total effect will be muddled and unintelligible.

In laboratory experiments a way has been found to cancel out a sound by means of an opposing sound of the same frequency. If you had two audio amplifiers, each connected to a power line so as to produce a 60-cycle hum in their loudspeakers, you could take a position midway between them, where a sound would not be heard—the sound waves from one would completely cancel out the sound waves from the other as they would be out of phase. But if you stand in the same position and have the speakers placed side by side the sound waves would be in phase and the sound twice as loud as the hum from one speaker. Their intensities would add. If one speaker emits a 60-cycle note and the other a 64-cycle note you would hear four beats per second.

This phenomenon of beats forms the basis for super-heterodyne reception. In this case, however, the beating is between high frequency A.C. currents one of which carries the sound frequencies.
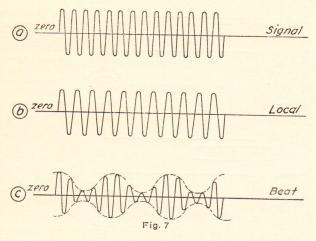
THE CONVERSION OF SOUND ENERGY TO ELECTRICAL ENERGY

The telephone is one of the greatest conveniences of our present age. When we look back to the times before the invention of the telephone we often wonder how people ever got along

without it. There is much that a simple telephone circuit will teach us about the conversion of sound vibrations into electrical vibrations, therefore we shall consider it briefly.

Of course you know that it is an electric current and not sound itself that flows over telephone wires. Yet in some way or other, our voice signal must be impressed on that electric current, otherwise the voice could not be reproduced at the receiving end.

We speak into the telephone transmitter and our voice is carried to a distant city. What happens in the microphone—for the telephone transmitter is essentially a microphone?

Inside the microphone there is a cup shaped chamber the bottom of which is an electrical conductor and in this chamber there

Fig. 7

is a small quantity of fine carbon grains. In the front part of this chamber, a movable circular plunger, is mechanically connected to a large metal diaphragm. When you speak into a transmitter the sound waves hit up against this diaphragm causing it to vibrate, following the sound vibrations exactly. As the diaphragm vibrates in and out it pushes the front metallic plunger in and out also. The result is that the carbon grains in the chamber are compressed and released in accordance with the sound vibrations. This is brought out clearly in Fig. 8.

It is a peculiar property of carbon granules that they offer considerable resistance to current flow when not compressed, that is, when the individual particles lie loose in their chamber. But when compressed, that is pressed together, the resistance be-

15

comes quite small and current flows through them easily. As the movement of the diaphragm causes a variation in the resistance of the carbon granules, more or less current will flow through the circuit depending on the degree to which the carbon particles are compressed. In this way the direct current flowing in the telephone circuit is changed to pulsating direct current having a wave form exactly corresponding to the form of the sound waves impressed on the diaphragm.
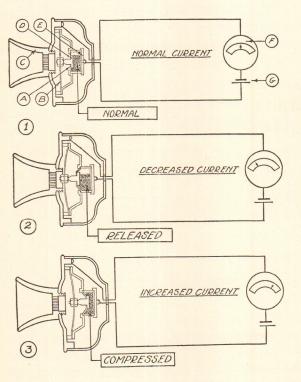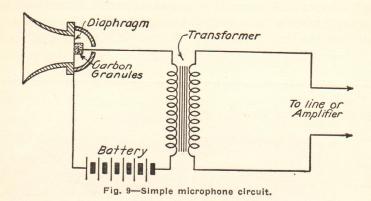

Fig. 8—Graphical representation of microphone action showing how the current varies with the sound waves impressed on the diaphragm. (The movement of carbon granules is exaggerated for purpose of illustration.) (A) Insulating disc, (B) Diaphragm, (C) Collector Horn, (D) Plunger contact, (E) Back end of cylinder containing carbon granules, (F) Ammeter, (G) Battery.

A battery or a D.C. generator is required in series with the microphone. As long as no sound strikes the diaphragm of the microphone, the current flowing through the circuit will be pure direct current just as it comes from the source. In the receiving phone no sound is produced when an unvarying current is flowing through the circuit. The receiver will, however, produce sound whenever the current passing through the circuit varies. The receiver is a device which reconverts the electrical variations into sound variations.

16

The principles underlying the Radio microphone are essentially the same as those used in wire telephony. However, in Radio the fidelity requirements are much higher than in ordinary telephony and to meet the requirements specially designed microphones are used.

Fig. 9—Simple microphone circuit.

The transmitter we described is the single button microphone. Figure 9 shows how a single button microphone would be connected in a circuit. Notice that there is only one chamber containing carbon granules which explains the name single button microphone. That a microphone of this type is not satisfactory for the transmission of music is apparent to anyone who
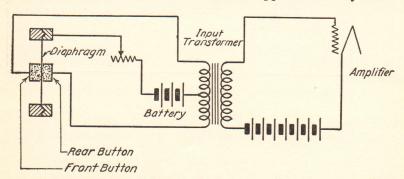

Fig. 10—Double button microphone circuit.

ever used a telephone. You know that a voice over the telephone sounds artificial. Often it is impossible to recognize voices over the phone. If music were transmitted over the telephone using an ordinary single button transmitter, it would sound very strange at the receiving end, a great deal of distortion would be present and the total effect would be very unnatural.

17

The two-button microphone shown schematically in Fig. 10 is much better adapted to the transmission of music. In this case there are two chambers containing carbon granules and the diaphragm lies between them. Because of the way in which these two buttons are connected to the circuit the current variations in the circuit caused by the movement of the diaphragm are considerably smaller than if a single button microphone were used. But the great advantage of this type of microphone is that distortion of the original sound in one button is balanced out in
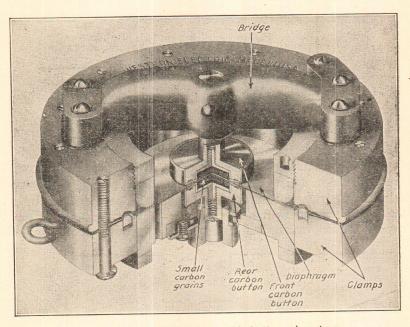

Fig. 11—Cut-away view of a double button microphone.

the other and then even the very complex wave forms that are associated with music are converted into current variations having the original form. Figure 11 shows a cut-away view of a typical double button microphone. This type of microphone is used in most of the broadcasting stations at the present time.

There is another microphone used extensively in sound picture studios. This microphone instead of depending upon the variations of resistance when a diaphragm is made to move in and out by a sound wave, depends upon a change in capacity.

It consists of a fixed flat plate with a very tightly stretched diaphragm approximately a thousandth of an inch away from it as shown in Fig. 12. The condenser formed by this particular

arrangement is connected in series with a high resistance between 10 and 15 megohms and in series with a voltage between 180 and 500 volts. Naturally the battery charges the condenser, through the resistance. Until the condenser is charged to full capacity, a certain amount of current will flow. But when the condenser is fully charged, assuming that the diaphragm is stationary, no current will flow in the circuit at all.

Now suppose a sound wave strikes the diaphragm. It will move in and out, vibrating in accordance with the form of the sound wave. But in vibrating it alternately increases and decreases the capacity of the condenser. When the capacity is increased, more current can flow into it. When the capacity is decreased, current will flow out of it. Thus while the diaphragm
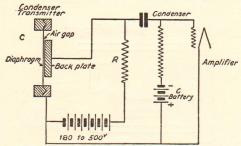

Fig. 12—Condenser microphone circuit.

is vibrating, the capacity of the condenser is continuously changing and there is a continuous varying of current in the circuit, in and out of the condenser.
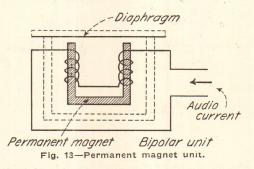
As the current varies, a voltage drop equivalent in intensity to the intensity of the sound waves striking the diaphragm appears across the resistance R. This voltage drop is then impressed across the grid and cathode of an amplifying vacuum tube. In the same way the signal is amplified through succeeding stages of amplification until the voltage variations are large enough to be made use of.

SOUND REPRODUCERS

The devices by which audio signals are converted into sound are just as important in the field of Radio as the microphone. The entire purpose of telephony, Radio, talking pictures or public address systems is to take the sounds produced by a speaker, orchestra or soloist; translate them into an electrical current
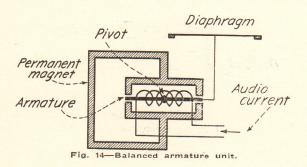
called an audio current; transmit this over distance, either over wires or through the ether by means of Radio waves, reconvert it into audio current and amplify it exactly like the original and then again retranslate the audio current into sound.

The devices used for this purpose are comparatively simple although to build perfect reproducers is quite an art. Figure 13 shows a permanent magnet having projecting pole pieces around which are placed bobbins wound with wire—the bobbin over each pole piece containing thousands of turns of wire. Approximately a few hundredths of an inch away from the pole pieces is placed a circular iron diaphragm which is partially attracted and bent toward the pole pieces by the permanent magnetism in the unit. The audio current is sent through these two bobbins which increases the magnetic attraction on the

Fig. 13—Permanent magnet unit.

diaphragm causing it to be pulled closer to the pole pieces. When the current is reversed the diaphragm swings back and beyond its normal position for the reversed current field reduces the permanent field. The magnetic flux produced by the audio current will vary exactly like the audio current and the motion of the diaphragm will follow it quite closely if the motion is not too large. Of course, on the front side of the diaphragm, the air is condensed and rarefied in the manner similar to the original sound wave. Built into a small unit, the device is placed to the ear and the sound wave directed to the inner portion of the human ear. In the case of loudspeakers, a horn is used with its throat close to the diaphragm. This horn confines the sound waves so that all the sound energy is directed toward the listener and losses due to radiation to the sides and back are reduced.
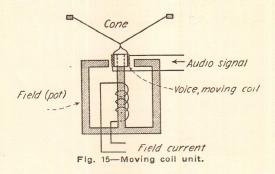
The swing of the diaphragm is naturally limited for in order to obtain good sensitivity, the diaphragm must be placed close to the pole pieces. In Fig. 14 is shown a development of the bi-

polar unit known as a balanced armature loudspeaker. Notice that this device has two permanent magnets which, of course, are considerably larger than the magnets used in earphones, something like magneto magnets, to which is connected a double prong pole piece. Between these two pole projections is placed an iron armature, pivoted in the center. Around this armature is placed a coil made up of two sections which is connected electrically to the source of audio current. When current flows through this coil, the armature is magnetized north and south. One side of the armature swings up and the other down. When the current reverses, the soft armature iron easily reverses its polarity and the motion is in the opposite direction. As the armature is connected by means of a link to a diaphragm the rocking action of the armature is mechanically transferred to the diaphragm, which vibrates following the variations in audio current.

Fig. 14—Balanced armature unit.

The balanced armature unit has limitations however. Its permissible swing is not large enough to produce loud signals. It therefore lacks fidelity when reproducing low notes, which as you will learn later, requires large motions of the diaphragm. For this reason, the dynamic unit or as it should be called the moving coil unit was developed. A diagram of this is shown in Fig. 15. Notice that in this particular case, the magnetic field is produced by means of an electromagnet which is made in form of a pot having a center stud around which is wound many turns of wire and which is excited from a source of direct current, independent of the audio signal. At the top a voice coil is connected which may have as low as one and as high as a hundred turns. This coil is connected to the source of audio signal, which produces a magnetism of its own, and reacts with the field magnetism, tending to push the coil in and out over the iron stud. The swing of the coil is large—it may move as much as five

21

eighths of an inch in high powered dynamic moving coil units. This particular moving coil may be mechanically connected to a diaphragm of special construction or it may be connected to a cone. In one case we have what is known as a dynamic unit which is used extensively in sound picture horn reproducers and in the other the dynamic cone used almost exclusively in modern Radio receivers and public address as well as sound picture installations.

The subject of loudspeakers, their sensitivity, fidelity, and response, characteristics, their limitations and the proper methods of making corrections in them is vast. Two whole lessons in the latter portion of your course are given over entirely to loudspeakers.
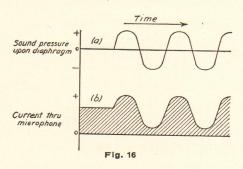

Fig. 15—Moving coil unit.

MODULATION

Whenever the current flowing in a circuit is changed, either by varying the resistance in the circuit by means of a microphone actuated by sound waves, or by mixing two alternating currents of different frequencies, we say that modulation has taken place. Thus when we talked about impressing sound wave forms on a direct current in the microphone circuit we were really talking about modulation.

Curve (a) in Fig. 16 represents a pure sound wave without any harmonics, the sort of wave form you would get if you tapped a tuning fork lightly. The portion of the curve above the line represents a condensation of air particles and below the line, the rarefaction. As you know, when a condensation strikes a microphone diaphragm it pushes the diaphragm in and decreases the resistance of the carbon granules behind the diaphragm. When a rarefaction reaches the diaphragm, the diaphragm swings in the opposite direction increasing the resistance of the carbon

22

granules. Curve (b) shows the effect of diaphragm movement on current flowing in the circuit. Now we can say that the current in the microphone circuit is carrying the audio signal and for this reason we say that *audio current* is flowing in the microphone circuit or more frequently that the current is pulsating at *audio frequency*.

When the audio frequency current flows through the primary of the transformer, the magnetic field about the primary expands and collapses, following the variations in current. But as the field collapses and builds up, an alternating voltage is induced into the secondary of the transformer so that in the secondary circuit there is an alternating current of the same frequency as the frequency of the pulses in the microphone circuit. In other words, the current flowing in the secondary circuit is *audio frequency alternating current*.
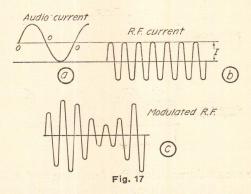

Fig. 16

It must be remembered that the frequency of A.F. current is comparatively low, the frequency range extending from about 80 to 5,000 cycles per second under ordinary circumstances. We can consider the average audio frequency as about 1,000 cycles per second. Frequencies used for Radio transmission are above 20,000 cycles and these high frequencies travel for great distances. The broadcast band of frequencies includes all frequencies between 550 kilocycles (550,000 cycles) and 1,500 kilocycles.

To get our audio signals on the air, we must combine the audio frequency current with a radio frequency current. Now how is this done?

A separate vacuum tube stage in which the tube acts as an oscillator, generates a high frequency current. The frequency of this current determines the frequency on which the signals are broadcast by the transmitter. The output of this oscillator

23

and the audio frequency current from the microphone transmitter which has been greatly amplified are both led to a "modulator" tube. In this tube a second modulation takes place*—the audio frequency current modulates with the radio frequency current from the oscillator tube. In Fig. 17 (a) represents the variation in audio current. For the sake of simplicity we take a current representing a pure tone; (b) represents the R.F. current which varies from maximum to minus many times a second. Notice that its height is I. When the audio current is zero the R.F. current has a value of I, but when the audio current rises in value the height of the R.F. current increases, and when the audio current decreases, the R.F. current diminishes. The process of modulation is so arranged that the R.F. current amplitude *swells* and contracts as the audio current increases and decreases. A modulated R.F. current if we were to photograph

Fig. 17

it by a oscillograph would look like (c). There are other ways of modulating an audio and R.F. current, but this method developed by Heising is usually used in Radio.

It is in this form the signal is transmitted and it is in this form the signal is picked up by the receiving antenna. In the receiver the audio frequency signal must be separated from the radio frequency carrier. This is done in the detector stage in which the vacuum tube acts as a demodulator. The detector tube acts as a one-way valve. It allows current to flow through it in only one direction. Consequently, if a high frequency alternating current is passed through a detector tube all the negative

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*When talking about modulation in a Radio transmitter, this modulation is usually referred to.

24

alternations will be cut off and the output of the detector will be pulsating direct current.

In this book on Sound we can give only a general introduction to the subject of modulation and demodulation. In later lessons we shall learn how the voice current in the transmitter is applied, where in the transmitting system modulation takes place; we shall study the action of detection or demodulation and the various types of detectors that can be used.

WAVE FORM AND DISTORTION

It has been mentioned that sound waves can be photographed. However, one way this can be done is by first converting the sound variations into electrical variations. Then these are photographed by means of an oscillograph with a sensitized photographic plate.

To get an exact picture of sound waves, a perfect condenser microphone or double button microphone is placed before a source of sound. Microphones can be designed to translate sound waves into their electrical equivalents without any distortion. The oscillograph is placed in the microphone circuit. If the varying currents are too small for convenience they can be amplified through several stages of audio amplification until they are large enough to be of use.
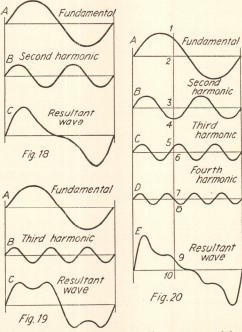
Let us start our study of audio frequency wave form with a very simple wave form. Suppose that a single tuning fork is struck lightly in front of our condenser microphone in the oscillograph arrangement. Figure 18A shows how the electrical equivalent would look as recorded by the oscillograph. This is also the way any sound would be recorded if only its fundamental were impressed on the current in the microphone circuit.

Now suppose we have another tuning fork which vibrates just twice as frequently as the original tuning fork. If both were struck simultaneously in front of the microphone the current variation curves would appear as in Fig. 18C. Notice that the curve does not show two individual sound waves but only one, shaped to conform with the combined sound waves.

In Fig. 19 curve A represents the fundamental frequency of a sound and B represents a harmonic frequency having three times the frequency of the fundamental while C represents the audio current which carries both the fundamental and the third harmonic. Notice the double hump in the positive and negative alternations.

25

In Fig. 20 curve A represents a fundamental frequency, B the second harmonic, C the third harmonic and D the fourth harmonic. When all these are simultaneously impressed on a microphone, the electric current in the microphone circuit assumes a wave form like E in the same figure.

The telephone companies have done a great deal of experimental work along the lines of sound analysis. First they convert the sound into its electrical equivalent, then by means of very complicated apparatus they analyze this audio current. As

Figs. 18, 19 and 20—Curves representing waves and harmonics.

the result of this analysis, they can tell how the energy in various sounds is distributed among the harmonics. The practical value of this analysis is that the input and output of an amplifier or any transmission medium can be compared and they can tell to what extent distortion has been introduced into the signal by the amplifying or transmitting apparatus.

By distortion we do not necessarily mean that a sound is distorted beyond recognition. If the output of an amplifier shows that there is more energy in the third harmonic than in the second, whereas in the original there was more energy in the second harmonic than in the third, sound engineers consider that

26

distortion has taken place, even though it could hardly be recognized in the sound output.

Distortions which are particularly prominent in audio devices, both in Radio and public address systems, are first due to the elimination of a certain frequency or frequencies. By this we mean that if the original sound consisted of a fundamental and harmonics up to the tenth, and the seventh and ninth harmonics were eliminated, while the sound could be recognized as a reproduction of the original sound, it would lack the timbre and quality characteristics of the original. This kind of distortion caused by frequency elimination is often referred to as "selective attenuation."

Another form of distortion which is prevalent in practically all electrical devices in circuits is intensity distortion. Again let us say that we had a sound which originally contained a definite amount of fundamental, second, third, fourth and fifth harmonic. The final audio current might have all these frequencies but they may be in different proportions. Although you would be able to recognize the original sound from the loudspeaker, it would not have the original timbre or quality characteristics of the original sound.

DECIBELS

When comparing the loudness of two sounds, it is not possible to say that one sound is two or three times as loud as the other. Careful study of the behavior of the ear to sound, has clearly shown that the ear responds to perceptible increases in sound. If in quiet surroundings, such as a sound proof room, the sound energy from a pure tone (let us say 1,000 c.p. s. sound) is increased, when the first perceptible increase is heard by the ear, it is said that the sound has been increased one decibel, often called a *db*. For a complex sound, as instrumental, voice or orchestral music, the sound must be increased 3 db before the increase is apparent.

The decibel is a unique way of expressing the ratio of sound energy. For example, if the sound energy is increased

	2	3	4	5	6	7	8	9	10 times
The db increase is	3	4.8	6.0	7.0	7.8	8.5	9.0	9.5	10*

If the sound pressure is increased 2, 3, etc. times, the decibel increase is twice as much or 6, 9.6, etc. The unit of pressure as

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*In a later reference text you will learn how to compute power ratios in terms of db. level.

27

used by sound men is the bar. It is worth remembering that under ordinary atmospheric conditions, the atmosphere presses on the ear drums with a pressure of about 1,000,000 bars*. Experience with a large number of individuals has shown that a pressure variation .0005 bars will be just audible and a pressure variation of about 500 bars will cause the sound to be painful. In practice a pressure of .0005 bars is referred to as the threshold of audibility with a level of zero db. A pressure variation of 500 bars when we compare it to zero level is said to have a level of 120 db.

Although the decibel is strictly a sound interpreting unit, it is convenient and desirable to express audio signals and even radio signals in terms of decibels. Again it is a way of expressing the ratio of power. For example, the ratio of power output and power input to a receiver may be expressed in decibels. The power fed to a loudspeaker may be expressed as so many db. provided we have a reference level. In radio, public address and television systems this level is 6 milliwatts. To be complete it is 6 milliwatts fed to a 600 ohm load in which case the potential difference will be 1.9 volts.
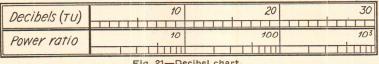

Fig. 21—Decibel chart.

The decibel is extensively used by expert radio men because it tells us whether an increase in output will be perceptible to the ear. For example, if the electrical power output of a radio rceiver is increased from 25 to 26 db. more than likely it will not be apparent (complex sound). If the sound is increased to 28 db the increase will be heard. A receiver feeding 5 watts of power to the loudspeaker is said to deliver 28 db. Figure 21 is a simple scale which will help you to convert power ratio to decibels.

If the power delivered by a device, a phonograph pick-up, microphone is less than 6 milliwatts, the decibel output will be designated as minus so many *db*. In such cases the ratio of the large to small power is computed and the ratio expressed as minus so many db. A condenser microphone delivers about -60 db, a carbon microphone about -30 db, a phonograph pick-up

*This pressure which is equal to about 15 pounds per square inch is balanced in the ear by an equivalent internal pressure.

about -15 db, and a loudspeaker for a small hall should be fed at a level of about 35 db. If sound to a carbon microphone is to be amplified so as to feed 35 db to a loudspeaker, the amplifier used should be able to amplify 15 + 35 or 50 db.

TEST QUESTIONS

Be sure to number your Answer Sheet 14 FR.
Place your Student Number on every Answer Sheet.

Never hold up one set of lesson answers until you have another set ready to send in. Send each lesson in by itself before you start on the next lesson.

In that way we will be able to work together much more closely, you'll get more out of your course, and better lesson service.

1. How are sound waves produced?

2. Will sound waves travel through a vacuum?

3. What is the velocity of sound waves through air?

4. Explain the difference between echo and reverberation.

5. State what range of sound wave frequencies can be heard.

6. Why is it that different musical instruments producing the same fundamental frequency, sound different?

7. What factors determine the *pitch;* and the *timbre* of musical sounds?

8. Explain how a beat note is produced.

9. Explain briefly what is meant by modulation.

10. What unit is used for expressing loss or gain of sound intensities?