HOW SOUNDS AND SCENES ARE CONVERTED INTO AND REPRODUCED FROM ELECTRICAL SIGNALS

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STUDY SCHEDULE NO. 14

For each study step, read the assigned pages first at your usual speed. Reread slowly one or more times. Finish with one quick reading to fix the important facts firmly in your mind, then answer the Lesson Questions for that step. Study each other step in this same way.

☐ 1. Introduction to Acoustics .... Pages 1-7

Since the entire broadcasting system is created for the purpose of transferring *sound* from one point on the earth to another, you will find this discussion of sound to be highly interesting. This is background information, to help you understand the characteristics of sound as they affect the electrical signal. Answer Lesson Questions 1, 2, 3, 4, and 5.

☐ 2. Technical Facts about Sound and Hearing .... Pages 8-13

Here you will learn the technical differences between noise, speech, and music, and will learn that the human ear has peculiar response characteristics—important to radio systems. Answer Lesson Questions 6 and 7.

☐ 3. Sound Pickups and Reproducers .... Pages 14-17

An introduction to pickups and reproducers, in which you learn how sound is converted into electrical currents, and vice versa.

☐ 4. Requirements of an Audio Amplifier .... Pages 18-25

A practical section showing how an amplifier can distort sound signals. Answer Lesson Question 8.

☐ 5. The Fundamentals of Television .... Pages 26-31

Old and modern ways of converting a scene or picture into electrical signals which can be handled by a radio broadcasting system. Answer Lesson Question 9.

☐ 6. Video Amplifier Requirements .... Pages 32-36

The wide range of frequencies which must be handled in modern television require unique circuits. This section is intended to give you an idea of some of the problems—don't spend too much time on it as you will study television in far greater detail later. Answer Lesson Question 10.

☐ 7. Mail your Answers for this Lesson to N.R.I. for Grading.

☐ 8. Radio Receiver Troubles—Their cause and Remedy .... Reference Text 14X-1

Read the instructions on the inside front cover of this text, then just thumb through the book, reading only the headlines, to get acquainted with its contents and learn how to find any desired subject in it. Once you begin actual servicing of radio sets, you'll really appreciate the value of this carefully cross-indexed summary of receiver troubles.

☐ 9. Start Studying the Next Lesson.

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HOW SOUNDS AND SCENES ARE CONVERTED INTO AND REPRODUCED FROM ELECTRICAL SIGNALS

Introduction to Acoustics

RADIO, as you know, is *a means of sending intelligence through space.* Since it requires no man-made paths between sending and receiving points, it is obviously more desirable in many cases than communication systems which send the same intelligence over land wires or cables. Although land wire telephone, telegraph, and picture-transmitting systems are not of particular importance to radio men, they differ fundamentally from radio systems only in the method used for transmitting the intelligence from the sending to the receiving point. In fact, research work carried out by telegraph and telephone companies has contributed much to the improvement of radio apparatus and technique.

► The three forms of intelligence which can be sent over either wire or radio communication systems are: *1, sound,* such as that used in radio broadcasting, in radiotelephone communication, and in telephone systems; *2, pictures,* either still or moving, such as those transmitted by radio or land wire television and facsimile systems; *3, code,* such as the dots and dashes used in radio and wire telegraphy.

Code transmission can be obtained simply by opening and closing a key in the transmitter circuit which, in the case of radio, causes the carrier current to be fed intermittently to the antenna and which, in the case of wire telegraphy, sends pulses of current over the land wires.

On the other hand, sounds and scenes represent intelligence which must be converted into equivalent electrical signals before being sent over wires or given "rides" through space on radio carriers. At the receiving end, the intelligence signals pass through many electrical circuits before being restored into the original sounds or scenes.

Thus, we must have a device capable of responding to all the characteristics of the sounds or the scenes, so that it can render a faithful electrical reproduction of the desired intelligence. Also, we must have reproducing equipment which can be fed an electrical signal and which will reproduce the original sounds or scenes as faithfully as is economically practical.

Furthermore, amplifiers must be used in both the transmitter and the receiver, in order to increase the voltage and the power of the electrical signal. Since the electrical signal characteristics, in turn, depend upon the sounds or the scenes, we must learn something about the characteristics of sound waves and light waves. First, we will study sound waves.

SOUND IS A WAVE MOTION

You know that sound can be produced by striking metal objects together, by rustling stiff paper, by vibrating your vocal chords, by plucking a stiff wire or spring, or by any one of thousands of different methods. Yet, though the ways we can create sound appear to be many and varied, they are basically identical. *In every*

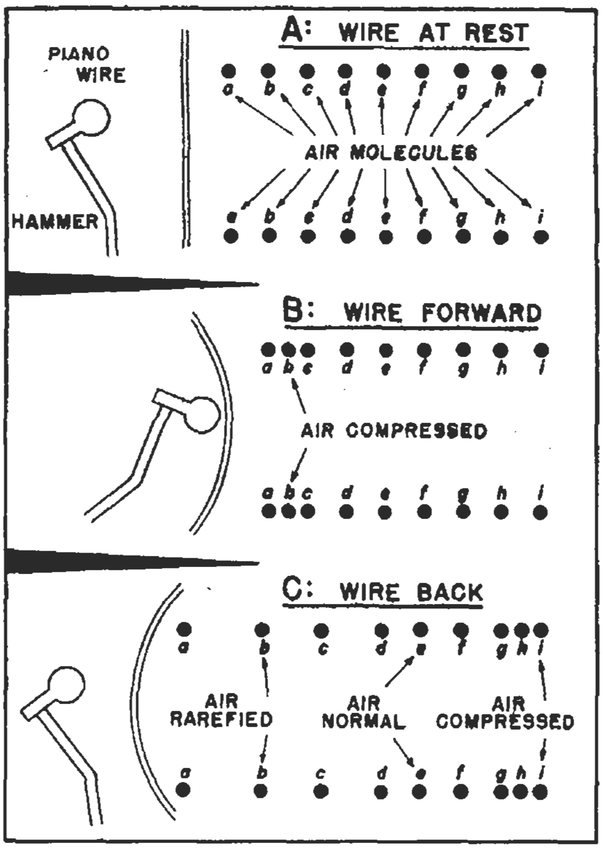


FIG. 1. When a piano key is depressed, a "hammer" strikes a piano wire (actually a group of wires). The wire is set into vibration; this vibration is transferred to the surrounding air molecules. The alternate compressions and rarefactions of the air result in sound waves which travel out from the wire in all directions. (For clarity, only one direction is indicated here.)

case, we create sound only if we produce a vibration. The vibration may be of very short duration, as when two stones are struck together, or it may be prolonged, as when a bell is rung—but, whether short or long, a vibration must exist before sound can be produced.

► However, a vibration alone is not enough to make a sound. The following is a favorite experiment in high-school physics classes: An alarm clock is suspended in a glass jar. The jar is then sealed except for a hose connection leading to an evacuation pump. Provided enough of the air in the jar is removed, no sound is heard when the alarm clock goes off—even though you can actually see the clock hammer strike the bell. This is an illustration of another basic fact about sound: *no sound is produced by a vibration which occurs in a vacuum.* There must be an elastic medium, capable of conducting the sound vibrations, in contact with the vibrating object before a sound is produced. This medium may be a gas, like air, or a liquid, like water, or a solid, like steel—in fact, it may be anything but a vacuum.

We know, then, that we must have some vibrating object and some medium in contact with the object before we can have a sound. Scientists tell us that the sound itself consists of *a vibration of the medium.* The vibrating object is the *source* of sound, the medium is the substance through which the sound is *transmitted,* but the sound itself is not a physical object like either of these; it is a vibratory *motion.*

► Let's make this clearer with an example. Suppose we consider what happens when a piano wire is struck by its hammer. Fig. *1A* shows a piano wire, its hammer, and a schematic representation of a few of the air molecules which surround the wire. If we strike the piano key, the hammer hits the wire and bows it outward. The moving wire strikes the air molecules *a* in front of it and drives them forward; they then strike molecules *b,* which in turn are driven forward to strike molecules *c,* and so on. In this way, the motion of molecules *a* will be transferred all the way along the line of molecules. In fact, if it were not for the inevitable losses of energy which occur in such an operation, the forward motion of the molecules would be transferred indefinitely.

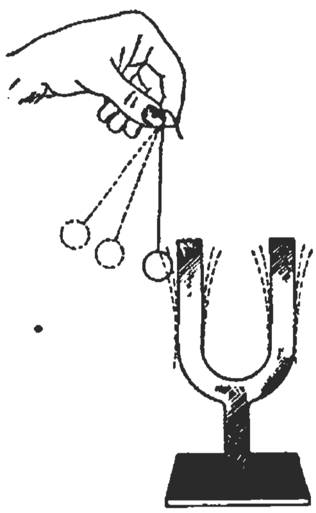
However, it is only the *motion* which is transferred in this manner; the individual molecules lose most of their energy when they strike the next molecules in line, and soon come to a stop.

In other words, *molecules* do not travel from the source of sound to you—*they pass along energy from one to another.*

Fig. *1B* shows the molecules as they would look an instant after the hammer has struck the wire. As you can see, the forward motion of the molecules has produced a compression of the air in the region of molecules *a*, *b*, and *c*, while farther on, where the motion has not yet reached, the molecules are still their normal distances apart (the air is uncompressed).

After it has bowed as far forward as it will go, the piano wire snaps back toward its original position. However, because of its elasticity, it does not just come back and stop; instead, it overtravels and bows backward. Its backward motion drives away the air molecules which were originally behind it, leaving a partial vacuum in its path. Molecules *a*, which by this time have stopped their forward motion, now rush back to fill up this vacuum; molecules *b* then come back to fill the space vacated by molecules *a*, molecules *c* come back to take up the space just vacated by molecules *b*, and so on. We now have a backward motion of the air molecules which, like the previous forward motion, will be transferred all along the line of molecules.

Fig. *1C* shows the distribution of the molecules when the piano wire has reached the end of its backward swing: molecules *a*, *b*, and *c* are now farther apart than normal, because they are still moving to fill up the vacuum created by the backward motion of the wire; hence, the air in this region is rarefied. (Rarefy means to make less dense; it is the opposite of compress.) Molecules *d*, *e*, and *f* have already gone through their forward motion, and are just starting on their backward motion. Therefore, they are about their normal distances apart, and the air in this region is neither compressed nor rarefied. Molecules *g*, *h*, and *i* are still going through their forward motions, so the air in their region is compressed. Thus, distant molecules are made to move back and forth in the same manner as those at the source of the sound.



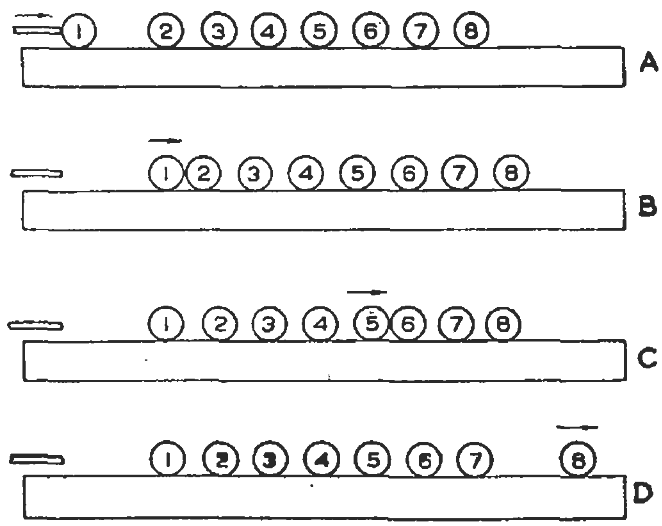
Here is experimental proof that sound-producing bodies vibrate. The tuning fork will produce sounds when struck and forced into vibration. If a ball is suspended so that it touches the vibrating fork, it will bounce away, proving that the fork is in motion.

The piano wire will snap back and forth at least several more times before it comes to rest, and each complete back-and-forth motion will produce two motions which will be transferred through the air molecules surrounding the wire—a *forward* motion (which we can call a compression wave) and a *backward* motion (a wave of rarefaction). Remember: the individual molecules move forward only until they strike other molecules, and move backward only until they are struck by the next wave of forward-moving molecules.

► Let's pretend that the piano wire produces a pure tone having sine-wave characteristics. (Actually it doesn't produce a pure sine wave, but let's see what could happen if this were true.) Then, Fig. 2 illustrates the movements of the molecules after the piano wire has vibrated several times. Instead of picturing just two lines of molecules in this figure, we have drawn curved vertical lines which represent millions of molecules; each molecule on a given line is the same perpendicular distance from the piano wire as every other molecule on that line. The lines which are bunched close together therefore represent compression waves, while those spread out represent waves of rarefaction.

Notice the wave form shown directly below the picture of the sound waves. This wave form is constructed from the picture by plotting the air pressure or air density at various distances from the piano wire, at one particular instant. The horizontal line represents normal air pressure. At points of rarefaction, the air pressure is below normal; at points of compression, it is above normal. The final result, when we draw a curve through all the points constructed, is a simple sine wave.

► However, whether or not the sound wave is a sine wave, regardless of its form, the air molecules will be alternately compressed and rarefied. The plot of air pressure thus follows the vibrations of the source, so that the plot shows the exact wave shape. This means that sound is a *variation of pressure* in the transmitting medium.



Another example of wave propagation. When ball 1 is struck by the cue at the left (in A) it moves forward, striking ball 2. Ball 1 then stops rolling as its energy has been imparted to ball 2, which now strikes ball 3, etc. As shown in B, C, and D, the balls move successively as energy is transferred from ball to ball. Sound energy is transmitted from molecule to molecule in much the same way.

► To put this important fact another way, we can say that a diaphragm struck by a sound will be alternately pushed in by the compression waves and pulled out by the waves of rarefaction. The pressure on the face of the diaphragm will vary above and below normal.

► Actually, a sound wave is very seldom a pure sine wave. The vibrating piano wire we have already discussed would, in addition to the fundamental sine wave pictured, also give off several harmonics of the fundamental, and might have an extremely complex wave shape. You will learn more about this a little later in this lesson.

SPEED OF SOUND WAVES

The speed of sound waves in any solid or liquid material depends upon two things, the *elasticity* of the material and the *density* of the material. You know that a material can be made to stretch or compress by applying force; the amount of compressing or stretching (the decrease or increase in size) is called the strain. *The more force it takes to strain a material a given amount without permanently changing its shape, the greater is the elasticity or springiness of this material.*

Notice—this scientific definition of elasticity is just about the opposite of the way most people use the word. The average man would say, for example, that rubber is more elastic than steel. But according to the scientific definition of elasticity, steel is much more elastic than rubber, for it takes much more force to compress a piece of steel a given amount than it does to compress a piece of rubber the same amount. Keep the scientific definition clearly in mind, for that is what we mean when we refer to the elasticity of a material. As a general rule, the harder a material, the more elastic it is.

► The density of a material is its weight per unit volume; if bricks of identical size are made up from various materials, the material in the heaviest brick will have the greatest density.

► The more elastic a sound-conducting material is, the faster sound waves can travel through it; the greater the density of a sound-conducting material, the slower will be the speed of sound waves through it. Both elasticity and density must be considered in determining whether one material will transmit sound faster than another. If two material objects have approximately equal densities, you can determine readily which will transmit sound

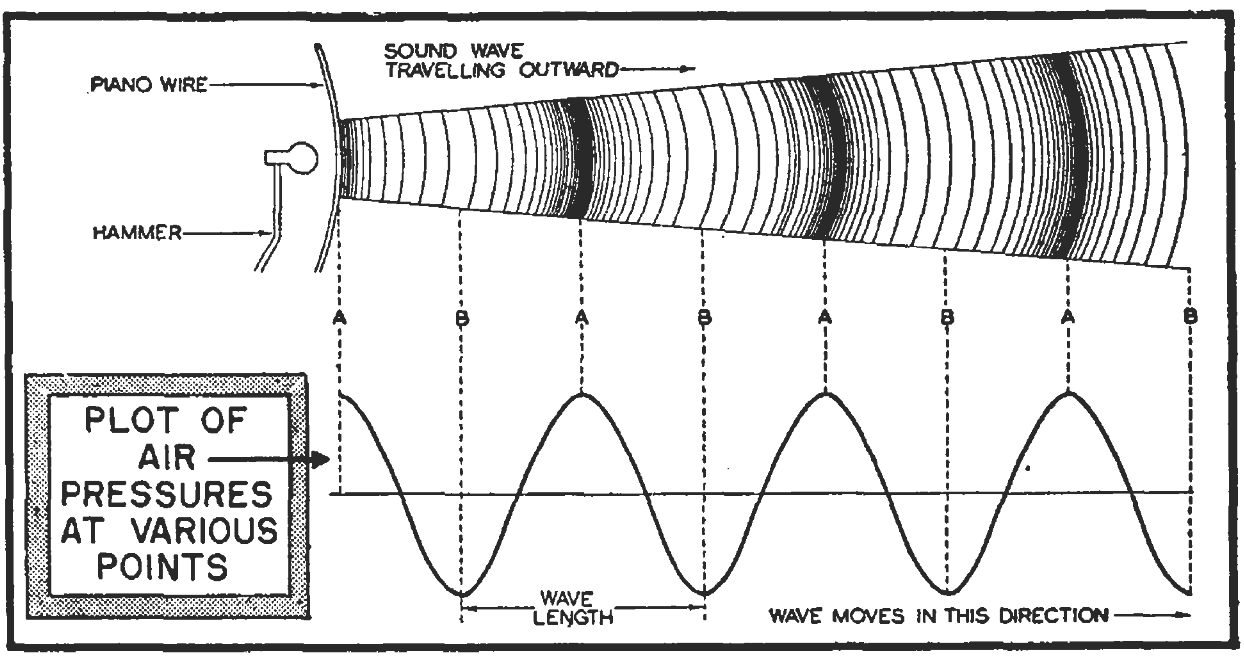


FIG. 2. By plotting the air pressure we would find that the sound waves travel away from the wire as alternate changes in pressure. The actual "plot" will show the wave form of the source. Thus, we get a sine-wave plot if the source produces sine-wave sounds. (A piano wire produces a more complex wave than that shown here.)

waves the faster by comparing their elasticities. For example, sound waves travel faster through aluminum than through soft rubber, because aluminum has a higher elasticity.

The speed of sound waves differs for various materials and even varies considerably with conditions in the same material. With air, for instance, the speed of sound waves increases with barometric pressure and with temperature. Careful laboratory experiments have shown that, at a normal atmospheric pressure and at a temperature of 0° centigrade, the speed of sound waves through air is about 1,089 feet per second. (These measurements are made in still air, since wind naturally affects the speed of sound in air.) Since the speed of electromagnetic waves (radio and light waves) is 186,000 miles per second, it is easy to understand why you can see lightning a long time before you hear the thunder which it produces.

Sound waves travel through water at a rate above 4700 feet per second. Steel is about seven times as dense as water (which would tend to slow up sound waves), but its elasticity is so much greater than water that sound waves will travel through steel at about 16,300 feet per second, almost four times their speed through water.

REFLECTED, TRANSMITTED AND ABSORBED SOUNDS

In radio and in the allied fields of public address and sound movies, we have a great deal to do with sounds produced in air. Ordinarily, these sounds are produced in rooms or in confined places where walls, ceilings or floors prevent the sound from traveling out in all directions. When sound waves strike a material, they are *reflected from the surface of the material, absorbed by the material, or transmitted through the material;* these three factors are shown in Fig. *3.*

Echoes. When sound waves are reflected by a flat surface which is quite far away from the source of sound, we may hear *echoes,* which are distinct but weaker reproductions of the original sound. Echoes are heard in many large rooms or in small rooms having curved walls which reflect the sound waves back to their source in a concentrated form. Echoes can be removed from a room by changing its shape or making it smaller; this clearly is a job for the architect.

Reverberation. When sound waves are reflected repeatedly back and forth

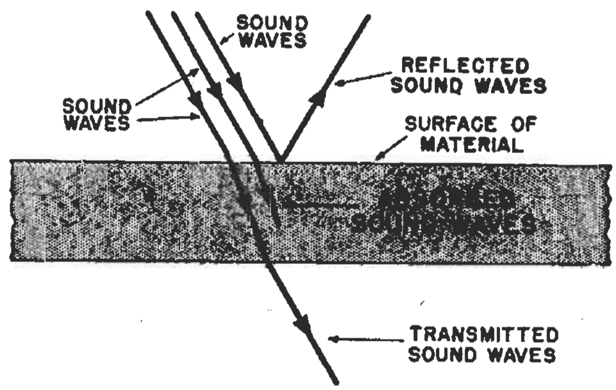


FIG. 3. Three things can happen to a sound wave which hits the surface of a material, as shown here. The kind of material will determine which one (or more) of these effects will occur.

between the walls of a room in a manner similar to that shown in Fig. *4,* we have *reverberation.* The waves which reach the listener after reflection (over paths *2, 3,* and *4*) mix with the waves heard directly (over path *1*) to give a "blurred" sound. The blur is produced because the reflected waves take longer than the direct waves to reach the listener, since they have farther to go. Thus, a sound heard over path *1* is followed a fraction of a second later by the same sound heard over path *2,* then by the same sound heard over path *3,* and finally by the same sound heard over path *4.* This means that sounds coming over path *1* will be interfered with by earlier sounds coming over the other paths. This reduces the understandability of words. If a great many such reflection paths exist, it may take a considerable time for a sound to die out, so that all intelligibility is lost.

The time required for a sound in a room to decrease to one-millionth of its original intensity is called the *reverberation period* of the room. Auditoriums which have reverberation periods of between one and two seconds are considered to have good acoustic qualities. The reverberation period of a room can be reduced by placing sound-absorbing material on the walls and ceiling.

Sound-Absorbing Materials. The better the sound-absorbing material used, the less sound will be reflected or transmitted. A soundproof room must have good sound-absorbing surfaces which keep out external sounds and also absorb sounds produced inside. An open window is sometimes considered to be an ideal sound "absorber" because it allows the sound to pass out of the room; on the other hand, it also allows external sound to enter the room, so you can't soundproof a room by leaving all the windows open! A sound-absorbing material dissipates the sound energy by converting it into heat. There are two ways of doing this: *1*, a porous material has tiny "pockets" such that when sound waves enter, their energy is dissipated by being reflected back and forth in the pocket; and *2*, soft materials will "give" under sound pressures but do not have the elasticity to bounce back and "throw" the sound wave back as a reflection. Thus, energy is absorbed by the "cushioning" effect of the material. Solid dense materials are good sound transmitters, and therefore poor absorbers; soft, pliable, and porous materials like

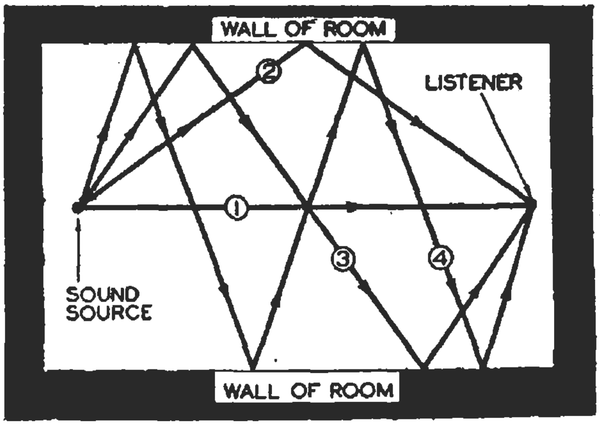


FIG. 4. Reverberations are produced whenever sounds can come to a listener over more than one path. Thus, sound waves reflected from the walls may travel over paths 2, 3, and 4 to the listener. As these paths are longer than the direct path (1), these reflected sounds arrive later than the direct sound waves. If they are of sufficient strength, they may blur the sounds. However, a certain amount of reverberation is desirable when listening to music.

velvet, Celotex, rock wool, cotton, carpet, and porous plaster are good sound-absorbing materials. The thicker the material, the more absorption of sound there will be.

► The fact that an audience absorbs sound waves is recognized by public address technicians; they reduce reverberation and echo effects by using directional loudspeakers which are directed at the audience rather than at the walls of the auditorium. The same thing is done in movie theatres—possibly you have noticed that the sound changes in quality as the size of the audience changes.

► So far, we have shown how sounds are produced, and also indicated some of the problems encountered when sound waves are produced within rooms. Reflection and sound absorption have much to do with the characteristics of the waves which we intend to pick up. Before we go into a study of pickups, however, let us learn more about the wave shapes which we will encounter.

Technical Facts About Sound and Hearing

A vibrating body, as you already know, produces sound waves. Since these sound waves consist of compressions and rarefactions of the particles in the transmitting medium, sound waves are studied best by measuring the pressure which they exert in the medium. Thus, in Fig. *2*, the pressure which the sound waves (produced by a vibrating piano wire) exert at any point varies in exactly the same manner as the motion of the wire. If the diaphragm mechanism shown in Fig. *5A* is placed in the path of a sound wave, the variations in sound pressure will bend the diaphragm in and out. This motion will be traced on a moving strip of paper by the link and pencil mechanism. If the sound wave is of a simple sinusoidal form, the tracing on paper will resemble the sine wave shown in Fig. *5B;* if it is of a complex form, the tracing may be somewhat like that shown in Fig. *5C*.

You will learn shortly that a complex sound wave, like that shown in Fig. *5C*, consists of a basic or fundamental frequency and many overtone or harmonic frequencies, each of which can be considered to be a simple sine wave. Any *simple sine wave* has two important characteristics—*amplitude* and *frequency*—whether it is a sine-wave sound traveling in a sound-conducting medium or a sine-wave current flowing in an electrical circuit. *The amplitude of a sound wave determines the loudness of the sound, while its frequency determines its pitch.*

HARMONICS

Speech, music, and noise are never pure sine-wave sounds; they always consist of a fundamental frequency and many higher frequencies which we call harmonics or overtones. *The harmonics or overtones give certain characteristic qualities (called "timbre")* to a sound. That is, these combinations of fundamental and harmonic frequencies give to speech and music distinguishing characteristics which, through experience, we are able to interpret. Thus we are able to distinguish between a harmonica, a trumpet, a violin, a flute, and a tenor singer, due to the different harmonic values even though all may be producing the same fundamental frequency.

From the foregoing, you can see that any sound has three characteristics: loudness, pitch, and quality or timbre.

How Common Sounds "Look." The wave forms of a number of common sounds are illustrated in Fig. *6*. These waves can be seen on the screen of a device known as a cathode ray oscillograph, which you will study later. This device would trace the wave shown at *A*, if you were to say "ah" before a microphone which was suitably connected to the oscillograph. A greatly different wave form, that shown at *B*, results when the same sound "ah" is sung. The key of C nearest the center of a piano keyboard (known to musicians as "middle C") produces the wave form shown at *C*, while street noise gives the very jumbled wave form shown at *D*. You will notice that with the exception of noise, wave forms of sound appear to repeat themselves at regular intervals; the time of one such interval determines the basic pitch or fundamental frequency of the sound.

► Musical tones, whether produced by stringed instruments, wind instruments, or the singing human voice, all consist of a fundamental frequency and a number of harmonics. For example, one of the C notes on a piano has a fundamental frequency of about 517 cycles per second. This note also contains a second harmonic of 1034 cycles, a third harmonic of 1551 cycles, a fourth harmonic of 2068 cycles, and a fifth harmonic of 2585 cycles. Each harmonic is a pure sine-wave tone.

The amplitudes of these harmonics differ greatly. The second harmonic in the above example is 20% of the amplitude of the fundamental, the third harmonic 25% of the fundamental, the fourth harmonic 10%, and the fifth harmonic about 8% of the fundamental. Another instrument producing the same note also may produce the same harmonics, but the amplitude of these harmonics would not be the same as those produced by the piano. In certain instruments, some of the harmonics are missing and, in some cases, a higher harmonic is stronger than a lower harmonic, or it may be even stronger than the fundamental frequency. It is the harmonics which make musical tones pleasing, while their number and amplitude determine the "timbre" or the characteristic which lets us distinguish between instruments.

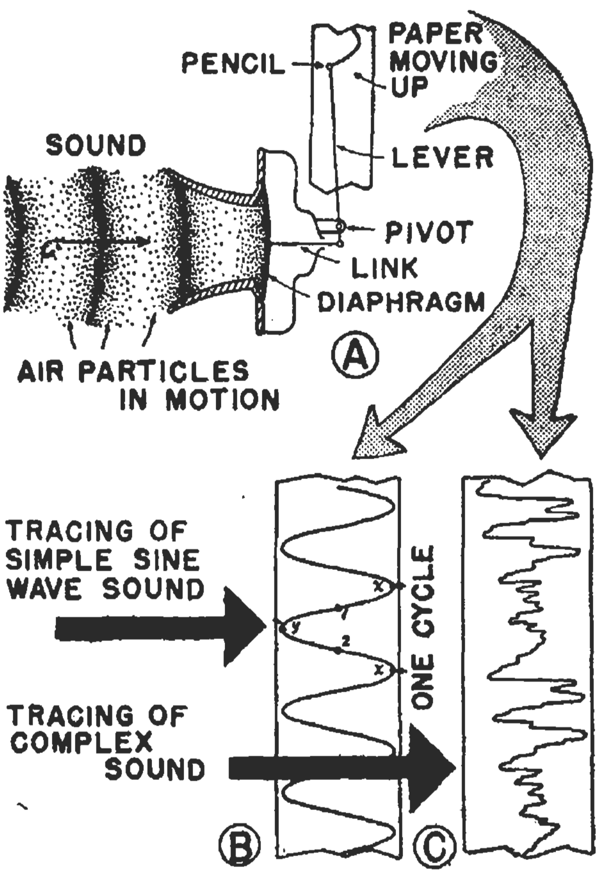


FIG. 5. The simple mechanism at A can be used to trace sound waves, thus proving that sound waves exert a varying pressure on the diaphragm. Typical tracings are shown at B and C.

► Speech differs from music essentially in that it is less melodic, being produced more nearly as a monotone. Speech sounds, like music, contain a fundamental pitch which distinguishes between the voices of children, women, and men, plus many different higher frequencies with certain frequencies predominating to give the characteristic distinctions between the voices of different persons.

► Noise is somewhat harder to define. Essentially, any unpleasant sound can be called a noise, whether it be just an excessively loud sound or whether it be musical notes combined in an unpleasing manner. Ordinarily, however, a noise is considered to be any irregular, distracting or unpleasing sound which would tend to mar musical reproduction or reduce the intelligibility of speech. Graphs of most noise sounds show irregular, sharp peaks, and are of an unsymmetrical nature. Noise contains overtone frequencies which are not necessarily harmonic frequencies—that is, they are not related mathematically in the same manner as are harmonics. Furthermore, instead of having a single well-defined fundamental frequency, the noise pulse usually will consist of a number of frequencies over a rather wide band.

FUNDAMENTAL FREQUENCIES

The piano, organ, and harp produce the greatest ranges of *fundamental* frequencies—from about 16 to about 4096 cycles. A baritone singer produces fundamental frequencies between about 80 and 400 cycles; a piccolo can "tweet" between about 500 and 5000 cycles; a ukulele has the limited frequency range of between about 300 and 1000 cycles. Thus, each instrument has its own range of fundamental frequencies as well as harmonics or overtones of these fundamentals. Radio apparatus must handle all these fundamentals and the audible overtones, if it is to give high fidelity reproduction.

THE HUMAN EAR

The human ear is by no means an ideal sound-interpreting device. One can control his ears, causing them to hear desired sounds and to disregard others. This is helpful in ignoring noise, but the same mechanism serves to fool one into hearing sounds that are not present! Thus, a person familiar with the timbre produced by, let us say, a piano, will automatically "hear" this timbre although a radio may be reproducing only a part of the harmonics necessary for a true reproduction. This extra effort is tiring, however, and soon one does not wish to listen further.

► In addition, human ears actually can become accustomed to some types of distortion and like it. A boomy radio receiver, with an excessive low-frequency response—sounding as if it were in a barrel—gives distorted reproduction, but many persons like this. In fact, some prefer it to high-fidelity reception. To musicians, however, or to others who appreciate high fidelity, any such distortion may be very annoying.

► Radio technicians should understand this queer behavior of the human ear; it is likewise of vital importance to the designer of radio apparatus. There is no need of spending time and money in reducing distortion when the improvement in fidelity cannot be noticed by the average human ear; nor is there any need to make a receiver respond to frequencies which cannot be heard by the normal human ear.

By studying the responses of thousands of persons to sounds of various

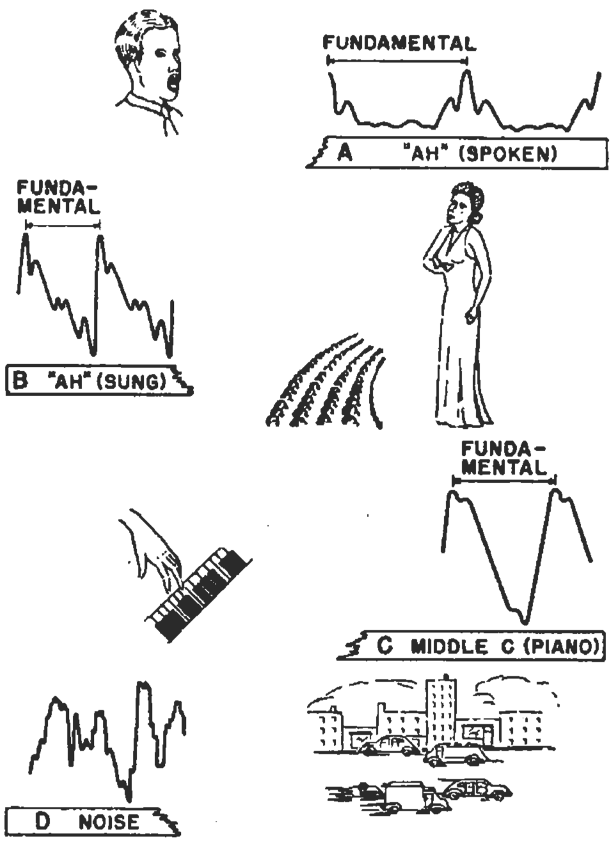


FIG. 6. Sounds can be "seen" as well as heard. By connecting a cathode ray oscilloscope to a microphone, curves like those shown here will be traced on the fluorescent screen of the tube.

frequencies, the Bell Telephone Company has found that the maximum range of frequencies which can be heard by even the best ears extends from 20 cycles to 20,000 cycles; others claim that this range is from 32 to 16,000 cycles. These tests also showed that the human ear is far more sensitive to sounds in the 1000- to 4000-cycle range than to sounds outside this range. Very high sound pressures are required before the ear can detect very low and very high frequencies in the audible range.

Sounds Can Cause Pain. When sound pressures are increased too high, a person stops hearing and actually begins to feel the sounds. At the lower frequencies, the vibration of the air can be felt by all parts of the body, but at the high frequencies, the action is more a sensation of pain in the ear. The response characteristic of the average human ear, obtained by noting when each frequency can just be heard and just be felt, is given in Fig. 7. A pure sine-wave sound which could be varied in frequency from zero to 20,000 cycles was used in this test; the loudness of the sound was measured in terms of the pressure exerted on a flat surface. You will note that *at 2000 cycles* it takes about .0005* bars of r.m.s. sound pressure to make the sound audible, and about 1000 bars (2 r.m.s. pounds per square foot) before the sound can be felt. At this frequency, then, the pressure at the threshold (the beginning) of *feeling* is about 2,000,000 times the pressure at the threshold of *audibility*. This difference in threshold values decreases for higher and lower frequencies, as you can see in Fig. 7.

The Decibel. You might think that a 2000-cycle sound at the threshold of feeling should seem 2,000,000 times louder to the human ear than a 2000-cycle sound near the threshold of audibility, but this is not exactly the way in which the human ear responds. *The amount by which a pure sine-wave sound must be increased before the change in sound level can be distinguished by the average human ear is called a "decibel."* Doubling the pressure of a sound causes a 6-decibel increase in the ear's sensations. Increasing the pressure gradually to ten times its original value results in twenty sensations of increase in sound, or a 20-decibel (abbreviated 20 db) increase. Increasing the pressure one hundred times results in a 40-db increase; increasing the pressure one thousand times gives a 60-db increase; and an increase of 2,000,000 times corresponds to about 126 db. Thus, you can see that increasing the sound pressure of a 2000-cycle note 2,000,000 times gives the equivalent of 126 separate and distinguishable increases in sound, as far as the human ear is concerned.

► The foregoing statements bring out a number of important characteristics of the human ear. Notice that doubling the pressure of a sound causes a 6-decibel increase in the ear's sensation. This does not mean that doubling the pressure results in a sound six times as loud—it means that, as the pressure is gradually raised, there would be six recognizable increases in the volume. In other words, as the sound is increased gradually, there would be a certain point at which it would be possible for one to recognize that the new pressure produces more sound than the original pressure. Then, taking the new pressure as a basis, a further increase eventually would produce another recognizable volume step. There are six of these steps involved when the pressure is doubled.

This fact at once shows that the hu-

*The bar is a term expressing *pressure per unit area;* one bar equals .002 pound of pressure per square foot of surface. Since sound pressure on any material is varying continually, we must deal with the r.m.s. value of the pressure, just as we deal with r.m.s. values of current and voltage. Sound or acoustical power is proportional to the square of sound pressure, just as electrical power is proportional to the square of voltage (electrical pressure). The *acoustic* bar is used here; one bar is equivalent to one dyne per square centimeter when the pressure is measured in metric units.

man ear is far more sensitive to pressure changes *when the original volume is low*. In other words, doubling the sound pressure from one bar to two bars results in a 6-decibel increase, and so would a doubling of pressure from one hundred bars to two hundred bars. Therefore, if the original sound level is low, a small change in pressure will produce a recognizable increase in loudness. On the other hand, the ear becomes less sensitive as the sound pressure increases, so that at high levels large changes in pressure are necessary before the ear can detect any change in volume.

into account. The amount of power fed to the loudspeaker determines the sound pressure the speaker will be able to develop and, in turn, the apparent loudness of the sound. Therefore, it is convenient to speak of the audio power in terms of decibels rather than electrical watts. This allows the designer to see at once just how much an increase in electrical power output will increase the sound output.

Again, a reference value is necessary. Any value can be chosen arbitrarily; radio men normally use the power level of 6 milliwatts, and call this the zero level or zero db. Notice—

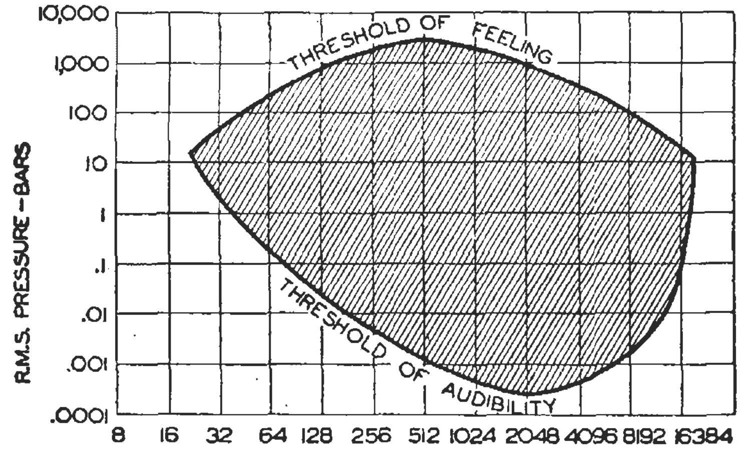


FIG. 7. The response of the average human ear to pure sine-wave sounds of various frequencies is given here. Sound waves having pressures within the shaded area can be heard. The r.m.s. pressure of the sounds is expressed here in "bars"; one bar equals .002 lb. per square foot.

Notice further that the decibel is *a unit of sound change*. It always represents the comparison between one sound level and another. Furthermore, it represents the number of steps in the increase. We cannot say that one sound is just twice as loud as another—but we can say that one sound is a number of steps louder than another.

POWER LEVELS

Because the ear responds to sound in the peculiar way just described, the radio designer must take this factor

this is a unit used in radio when comparing electrical powers, and does not have any relationship to the acoustic reference level. In fact, we cannot say just how much acoustic power will be produced by this amount of electric power, as this depends on loudspeaker efficiencies, the acoustic treatment of the loudspeaker enclosure and a number of other factors. However, we can reasonably expect that a rise of 5 db in electrical power will produce a rise of 5 db in sound power.

From the foregoing, you can see that

the 6-milliwatt level is purely an arbitrary reference level, to which other electrical powers can be compared. Fig. *8* gives db values for various amounts of power in watts, when this reference level is used. You can see that a power level of 19 watts corresponds to about 35 db, 6 watts to 30 db, 1.9 watts to 25 db, etc.

If the power level is less than the reference value, the output is considered to be a *minus decibel* value. Thus, an output of .002 watts is about 5 db below the zero level (commonly called 5 db down, or minus 5 db, and written —5 db).

From the table, you can see that doubling the *power* results in a 3-db increase; multiplying the power by 10 causes a 10-db increase; multiplying by 100 causes a 20-db increase and so forth. Thus, the decibel increases for *power* changes are different from those caused by *pressure* changes.

► Regardless of this, the response of the human ear to sounds follows the decibel scale. As we have said, the average ear can distinguish a 1-db change in sound if the sound is a pure sine wave. If, however, the sound is complex (speech or music for example) the smallest change in level which the average ear can notice is about 3 db (corresponding to a power ratio of 2 to 1).

From this, you can see that giving a power increase or decrease in terms of decibels lets us tell at once whether the increase or decrease will be noticeable to the ear. Thus, if we consider a complex sound, we have to increase the power three decibels, or double it, before the ear is able to tell that the sound has been increased at all. Hence, if an amplifier is feeding 6 watts to the loudspeaker, we must go to 12 watts before the increase would be noticeable. In other words, under the same conditions and for the same loudspeaker characteristics and efficiencies, it is not possible to tell the difference between an amplifier producing 6 watts and one producing 7 watts.

WATTS	DB	WATTS	DB
.00048	−11	.006	0
.00060	−10	.008	1
.00076	−9	.010	2
.00095	−8	.012	3
.0012	−7	.015	4
.0015	−6	.019	5
.0020	−5	.024	6
.0024	−4	.030	7
.0030	−3	.038	8
.0039	−2	.047	9
.0048	−1	.060	10

WATTS	DB	WATTS	DB	WATTS	DB
.076	11	.95	22	11.9	33
.095	12	1.2	23	15.2	34
.12	13	1.5	24	19.0	35
.15	14	1.9	25	23.7	36
.19	15	2.4	26	30.4	37
.24	16	3.0	27	38.0	38
.30	17	3.8	28	47.4	39
.38	18	4.7	29	60.0	40
.47	19	6.0	30	75.9	41
.60	20	7.6	31	94.9	42
.76	21	9.5	32	119	43

FIG. 8. This table shows the relation between electrical power (in watts) and decibels, when the zero db level is set at .006 watt.

► As we will see shortly, the fact that the human ear has this peculiar characteristic is quite helpful since it means that a form of distortion known as frequency distortion is not as noticeable as it would be otherwise.

Sound Pickups and Reproducers

We have shown that one of the important characteristics of a sound wave is that it exists as a variation in pressure. Therefore, it is obvious that a simple and direct means of converting a sound wave into an electrical current is to use a pressure-actuated pick-up unit, so designed that it converts the variations in pressure into electrical current variations.

By the same reasoning, the sound reproducer must change electrical current variations into sound pressure variations.

Let us turn now to typical pickups and reproducers, and learn something of the fundamentals of their operation. At this time, we won't go into all of their characteristics — we'll just see how they operate.

SOUND PICKUPS

There are a number of different kinds of microphones in use today, but the five most common types are shown in Fig. *9*. All of them use some form of diaphragm upon which the sound pressure can act. They differ principally in the method of converting the diaphragm movement into electrical current variations.

Carbon Microphone. Fig. *9A* gives a simple sketch of the single-button carbon microphone. Part *D*, a very thin aluminum disc or diaphragm (about .001 inch thick) is stretched over a fixed metal ring *R*. The pressure of sound waves on disc *D* moves the disc back and forth, alternately squeezing and loosening carbon particles *C* in the telescoping metal sack or button labeled *K*. Carbon itself is a resistance material, but this device depends upon *contact resistance* for its operation. That is, as the particles are squeezed together, more of their surfaces will be in contact, so the resistance between them decreases. Then, when the particles are allowed to separate, they make poorer contact so the resistance increases. The result is that the electrical resistance between disc *D* and container *K* varies continually with the motion of the diaphragm. When this single-button microphone is placed in an electrical circuit containing a d.c. voltage *E*, the current passing through the circuit will be varied by this changing resistance which, in turn, varies in accordance with the wave form of the sound. Audio transformer *T* is placed in this circuit to transfer the variations in current to another circuit.

Condenser Microphone. Fig. *9B* shows a simplified cross-section view of a condenser microphone. The thin aluminum disc or diaphragm marked *D*, mounted on ring *R*, is placed about .001 inch away from the fixed heavy plate *P*, thus forming a simple two-plate air condenser. These two plates are connected into a circuit containing a high voltage d.c. supply *E* and resistor *R*. Varying sound pressures change the distance between *P* and *D*, thus changing the capacity of the condenser. Changing the capacity this way results in a change in the charge stored in the condenser, so that the current through the circuit varies when the microphone picks up sound, and a varying voltage which has the same wave form as the original sound is produced across resistor *R*.

Dynamic Microphone. The dynamic microphone shown in Fig. *9C* has a thin diaphragm *D* on which is mounted a light-weight coil of wire. This coil moves between the poles of a permament magnet when the action of sound moves the diaphragm. As a result, there is induced in the coil a varying voltage whose wave form is a reproduction of the wave form of the sound.

Velocity Microphone. The velocity microphone illustrated in Fig. *9D* operates on much the same principle as the dynamic microphone, except that here sound waves move a thin crimped metal ribbon *M* back and forth through a magnetic field produced by a permanent magnet. A voltage is induced in this metal ribbon; this induced voltage is stepped up by the transformer.

Crystal Microphone. The bending or straining of a Rochelle salt crystal produces charges of opposite sign on the opposite faces of the crystal. The potential difference which exists between these charges is proportional to the strain. This principle is utilized in the crystal microphone illustrated in Fig. *9E*, where two square crystals are mounted back to back to increase the electrical action. This square crystal unit *C* is clamped at three of its corners; its free corner is linked to diaphragm *D*, so that movements of the diaphragm caused by sound waves will bend the crystals. Electrons which flow to the flat surfaces of the crystals under this bending are collected by metal collector plates *P*.

► The electrical output of microphones is very small so it is necessary to amplify this output many times. Therefore, at the transmitter, the microphone will feed into an audio amplifier designed to raise the signal power to the point needed for modulation. In public address systems, a similar amplifier is used to increase the sound power to the level necessary to operate loudspeakers.

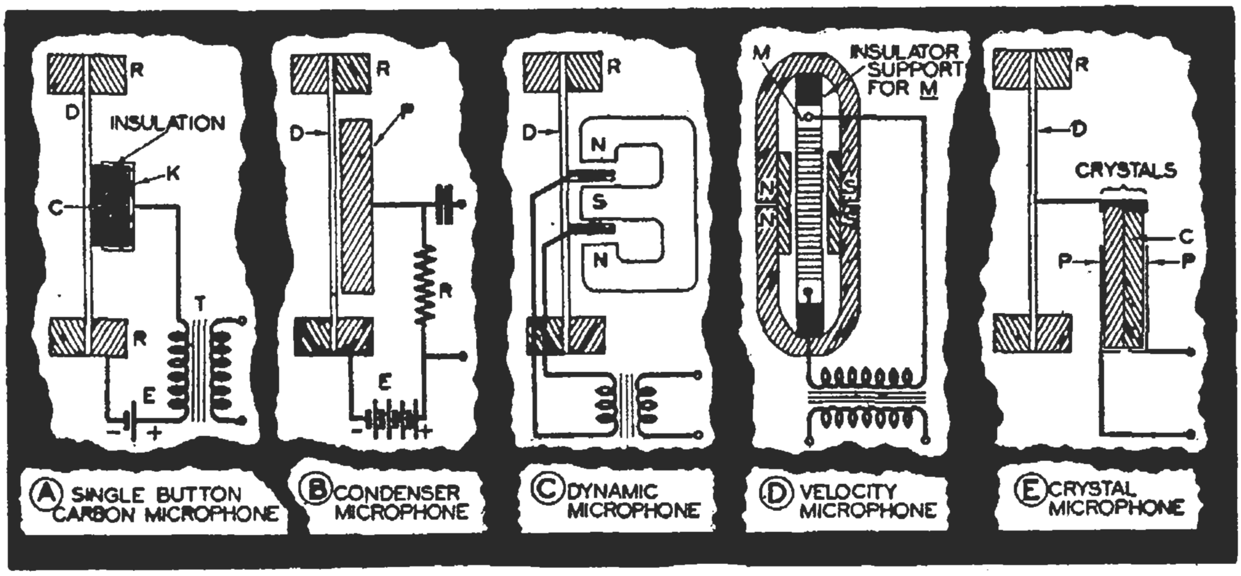


FIG. 9. Simplified diagrams illustrating the operating principles of five different types of microphones. Each converts changes in sound pressure into equivalent changes in electrical signals.

SOUND REPRODUCERS

Headphones and loudspeakers are the devices used for sound reproduction. Although there are a number of types of each of these, they operate on the principle of converting electrical current variations into variations in air pressure, corresponding to sound waves.

After the carrier wave has been demodulated at the receiver, there exists a varying current **which we call an** a.f. current. This current is the electrical counterpart of the sound wave as it was transmitted and as it has been modified by its passage through the radio up to this point. Usually, there is enough power at the output of the average demodulator to operate a pair of headphones satisfactorily, but an audio amplifier is necessary before a loudspeaker can be operated. Thus, there is a miniature counterpart of the transmitter audio amplifier used in the receiver for the purpose of increasing the audio power to the level required by the loudspeaker.

Essentially, both headphones and loudspeakers operate somewhat on the principle of the dynamic microphone in reverse. They utilize electric signals to set into motion a diaphragm or a cone, alternately compressing and rarefying the air in front of the reproducer to reconstruct the original sound wave.

Headphones. In the common headphone unit, illustrated in Fig. *10A*, a thin flexible steel diaphragm is placed over the two poles of a horse-shoe magnet, and a coil having many turns of insulated wire is placed around each leg of this magnet. The audio current flowing through the two coils alternately increases and decreases the attraction which the permanent magnet has on the diaphragm, causing the diaphragm to move in and out, thus producing sound waves.

Magnetic Loudspeaker. If a high sound output is wanted, the reproducer must be able to displace a large amount of air, and so must have a large diaphragm or cone which can be moved appreciable distances. The balanced armature electromagnetic reproducer shown in Fig. *10B* has been used widely for this purpose. In this loudspeaker, a soft steel armature is pivoted between two sets of *N* and *S* poles which are parts of a powerful permanent magnet. Also surrounding the armature is a solenoid or coil which carries the audio current. This coil makes the ends of the armature alternately of op-

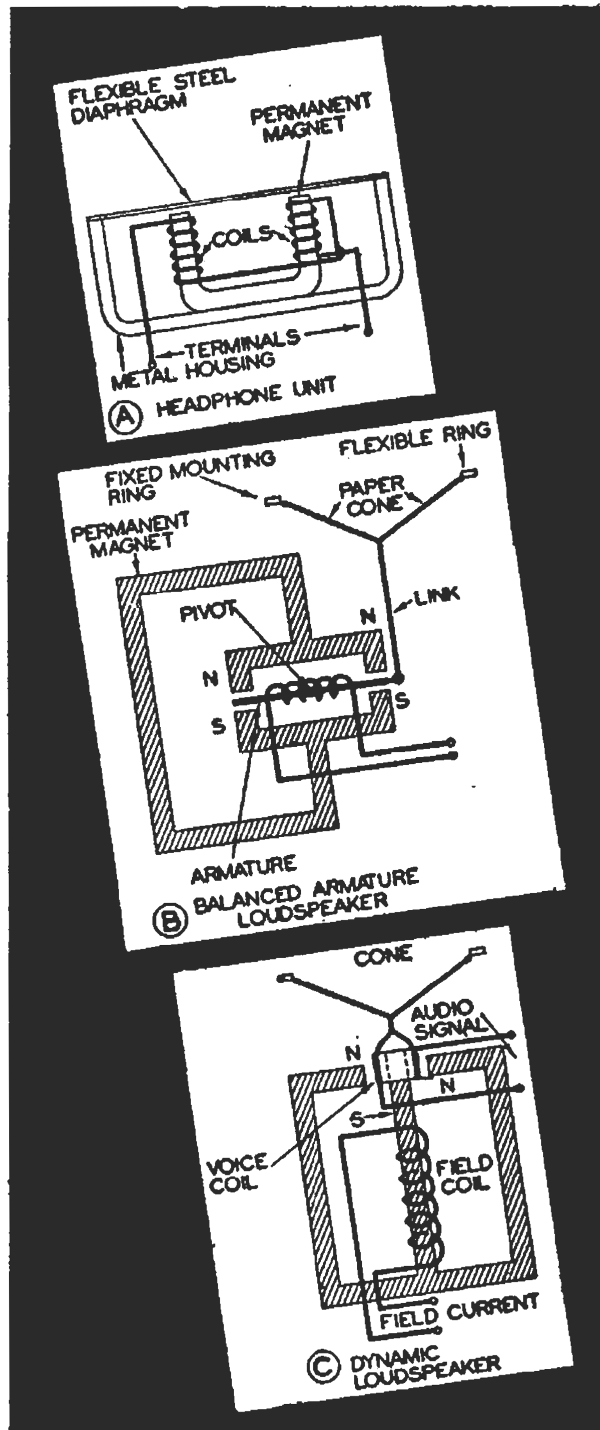


FIG. 10. Simplified diagrams illustrating the operating principles of three different types of sound reproducers. Each converts electrical energy into mechanical energy (motion), which in turn is used to produce sound waves.

posite polarity. The ends of the armature, therefore, alternately move towards and away from the permanent magnet poles. This movement is mechanically relayed to the large paper cone so that the paper cone is pushed in and out, setting the surrounding air into vibration and producing sound which is a reproduction of the original wave form. The outer edge of the cone is designed in such a way that it can be attached to its ring-shaped supporting frame without preventing free movement of the cone in and out under the action of the armature.

Dynamic Loudspeaker. The dynamic loudspeaker shown in Fig. *10C* is the most widely used sound reproducer, for it is capable of delivering high sound outputs when the cone is used with a baffle.* Here, a coil of wire is wound on a thin bakelite or paper tube which is attached directly to a paper cone (as shown) or to a metal diaphragm (not shown). The audio signal passing through this coil causes a varying magnetic field. This field interacts with a fixed field which is produced either by a permanent magnet or by an electro-magnet. The result is that the coil is forced to move in and out of the fixed field, driving the cone and thus moving the air ahead of it.

The amount of movement depends on the field strengths. Hence, the "voice" coil is fed with considerable audio power, and the fixed field is made strong by making the separation between the north and south poles of the magnet as small as possible.

A dynamic loudspeaker can handle from 3 to 50 watts of audio signal power (depending upon its size), while a magnetic loudspeaker can handle from 1 to 3 watts. A headphone unit can handle only about 50 milliwatts.

► You are going to study loudspeakers and headphones in much greater detail in later lessons. However, you can see now that the purpose of the audio amplifier is to increase the power up to the level necessary to operate the loudspeaker. It would appear that the amplifier at both the receiver and the transmitter should be rather simple in its design and operation. Still, this is not altogether true, as this amplifier is largely responsible for the fidelity with which the electrical sound signal is handled.

*A baffle is a large flat surface, a cabinet, or a horn-shaped enclosure which serves to prevent waves from the back of the speaker cone from interfering with those from the front, and also acts to "load" the cone by coupling it to a larger volume of air.

Requirements of an Audio Amplifier

All amplifiers may be divided into two broad groups, high-frequency amplifiers and low-frequency amplifiers. High-frequency amplifiers handle the r.f. carrier signals (which may or may not be modulated), while low-frequency amplifiers handle only the intelligence signals. In this lesson we will learn something about the requirements placed on low-frequency amplifiers. Let's cover first the low-frequency amplifier used in sound radio receivers.

The low-frequency amplifier in a sound radio receiver is called an audio amplifier because the frequencies it handles correspond to audio or sound frequencies. As we have previously mentioned, the audio amplifier is the section in a radio receiver following the second detector or demodulator. A corresponding amplifier is used in the transmitter between the sound pickup device and the modulator, and similar amplifiers are used in public address systems, electronic musical instruments, electrical phonograph amplifiers, and many photoelectric control devices.

The basic requirement of any low-frequency amplifier is, of course, to increase the signal level. We apply a signal to the grid and obtain a change in the plate current. In turn, this plate current flow through the load produces a varying load voltage. This load voltage variation is larger than the grid voltage variation, provided that the value of the load has been chosen properly.

► We now have our amplified signal, but is it *exactly* like the original grid signal? If we have a sine-wave signal, we want to have an enlarged sine-wave signal across the load—if we start with a complex signal, we want the output signal to be exactly like it. However, if the output wave is different in any manner from the input signal (except for being amplified), then the signal is being distorted. The sounds we obtain from the loudspeaker will not be exactly like those originating in the broadcasting studio when the amplifier (or the speaker, for that matter) introduces harmonics which were not in the original signal, or when the relative amplitudes of any of the harmonics in the signal are altered.

From this, the manner in which the signal level is increased is of importance, so we must see how it is possible for an amplifier to distort the audio signal. Before studying the three forms of distortion, let's see how complex waves are formed, so that we can see just what may happen to the signal within an amplifier.

WAVE FORMS

Any wave form differing from a pure sine wave contains one or more harmonics. For example, in Fig. *11*, the wave at *C* can be obtained by combining the fundamental frequency *A* with its second harmonic *B*. You can check this fact by combining the amplitudes along the dotted lines. Thus, adding the distance *1-2* of the fundamental (the distance from the curve to the line *R-R*) to *3-4* of the harmonic gives *5-6* on the resultant.

The distance *9-10* on the second harmonic must be subtracted from the distance *7-8* on the fundamental to get the resultant *11-12*, because the second harmonic is going through the opposite phase to that of the fundamental wave at this particular moment. Thus, you can see that, when both waves are on the same side of the reference lines *R-R*, their amplitudes add. If they are on opposite sides, the smaller is subtracted from the larger to get the resultant. Incidentally, the resultant wave will be changed in shape if the amplitudes of the fundamental and the harmonic are changed.

► In Fig. *12*, we show the result of adding a third harmonic to its fundamental. Again the same procedure is used in combining the fundamental wave *A* to the third harmonic *B* to get the resultant wave *C*. Thus, adding the distance *1-2* to *3-4* gives *5-6*. Subtracting *9-10* from *7-8* gives *11-12*.

► When other harmonics are added, the same procedures are followed. As shown in Fig. *13*, adding a number of harmonics gives another and entirely

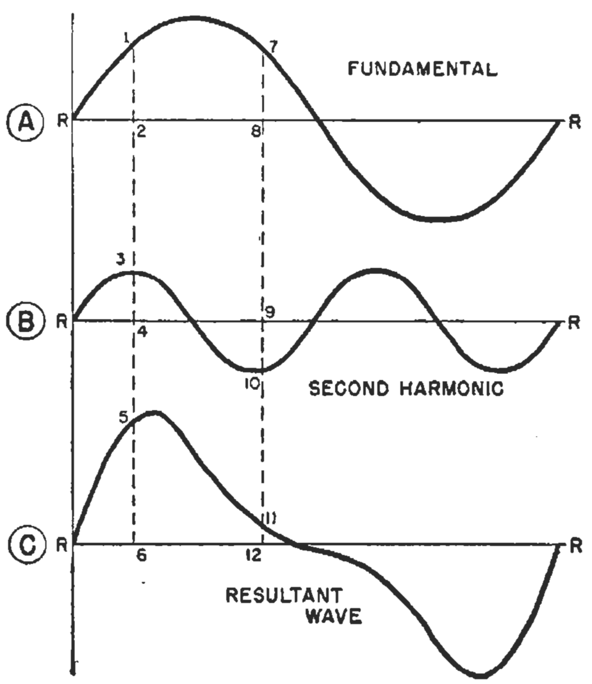


FIG. 11. Here is the result of adding a second harmonic and a fundamental wave. Any wave which is not a sine wave consists of a fundamental and one or more harmonics.

different resultant wave shape. Again the same procedure is followed in that the amplitudes are added and subtracted to give the resultant. Thus, along the dotted line, the amplitude *1-2* is measured, the distance *3-4* is subtracted, the distance *5-6* is added, the distance *7-8* is zero, leaving the resultant, *9-10*.

► From the foregoing, you see that there is an infinite variety of resultant wave forms possible, depending upon the harmonics which are added to the fundamental and upon the amplitudes of the harmonics. (In addition, as we shall learn later, the phase of the harmonics will change the resultant wave shape.)

However, regardless of the shape of the resultant wave, we can always say that any complex wave is a combination of simple sine wave forms. This is the reason that we can study the operation of amplifiers by using simple sine waves—the complex waves which the amplifier must handle are made up from these sine waves.

► As a summary, we know that:

1. Any complex sound or signal consists of a fundamental frequency plus harmonics.
2. If the relative amplitudes of any of these components are altered, we have distortion.
3. If any components are added, which were not in the original signal, we have distortion.

We can now go on to study the kinds of distortion and their relative effects. We will see that practical amplifiers are far from ideal.

FREQUENCY DISTORTION

If one section or stage of a radio *strengthens certain frequencies more than other frequencies, or removes some frequencies entirely,* the resulting sound will not be like the original. This distortion is known as *frequency distortion,* because certain frequencies are favored more than others.

As an example, suppose we have a 1000-cycle fundamental note with a fifth harmonic at 5000 cycles. If the amplifier responds better to 5000 cycles than it does to 1000 cycles, the amplitude of the fifth harmonic will be increased more than its fundamental is, which means that the resultant wave shape will be changed. On the other hand, if the amplifier has a lower response at 5000 cycles than at 1000 cycles, then the fifth harmonic amplitude will be reduced, which again means that the resultant wave shape is changed.

You can see that frequency distortion is produced because an amplifier does not have the same gain for all the frequencies it is supposed to handle.

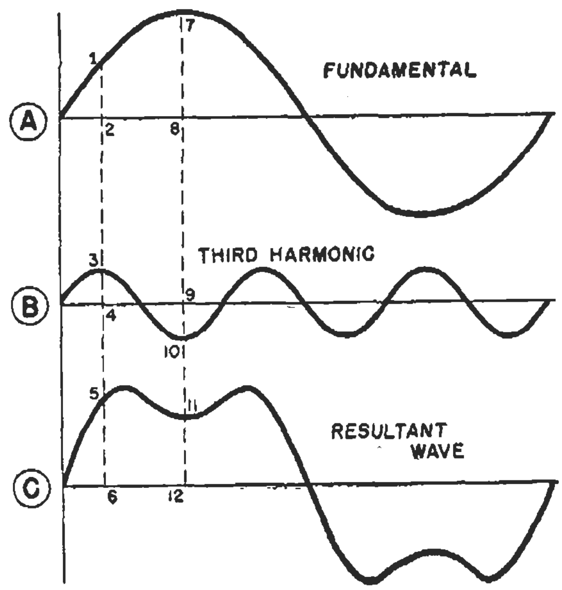


FIG. 12. Adding a third harmonic to a fundamental produces the resultant shown here. Of course, the actual shape of the resultant depends upon the amplitudes of the fundamental and harmonic waves. However, the general shape is correct, and should be compared with the resultant shown in Fig. 11. Notice that odd harmonics (the third, fifth, etc.) produce a resultant with both halves exactly alike, while even harmonics (second, fourth, etc.) produce a resultant which does not have symmetrical halves.

This type of distortion is caused by the circuit parts (coils, condensers and transformers) associated with the vacuum tube in the amplifier stage.

► Earlier, we mentioned that the ear does not notice a power change of less than about 3 db (power ratio of about 2 to 1) when complex sounds are being heard, which means that frequency distortion is less troublesome than would appear at first glance. It is quite permissible to have radio apparatus amplify some frequencies almost twice as much as others, or to convert some frequencies into sound twice as well as others, for the human ear is not able to notice this amount of distortion. This characteristic of the human ear is very much appreciated by the radio designer, since it is quite difficult for him to make radio apparatus amplify very low and very high frequencies equally as well as in-between frequencies.

Fig. *14* shows a typical response curve for an audio amplifier. Notice that the scale at the left is in decibels, because this permits us to compare the response to the hearing capabilities of the ear.

► When making a curve of this kind, the power output at some reference frequency (usually 1000 cycles) is taken as the standard or reference value. Then, the power output at other frequencies is found. If the amount of output is less than that at the reference frequency, then the db output is less, while greater outputs give increases in the decibel level.

As you will notice from the graph in Fig. *14*, the curve is within 3 db of the reference level (24 db) from about 80 cycles to above 4000 cycles. At 5000 cycles there is a peak in the response which is nearly 6 decibels higher than the reference, so the output at frequencies about this value would be noticeably greater.

Above 8000 cycles, the response cuts off sharply, so frequencies above this point are poorly reproduced, if at all.

► When the very high frequencies are cut off by radio apparatus, sound loses some of its fidelity of reproduction. However, not all the high frequencies up to 20,000 cycles are necessary for understandability or appreciation of music. You can cut off sounds at about 8500 cycles and still get high quality speech and music. Why reproduce frequencies up around 15,000 to 20,000 cycles, when only very young people and trained musicians are able, as a rule, to hear these frequencies at all?

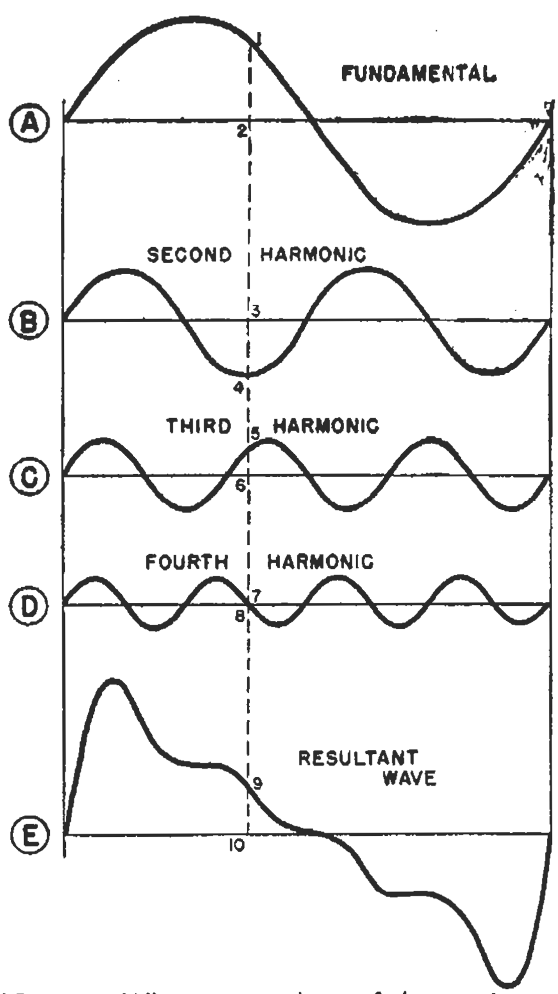


FIG. 13. When a number of harmonics are added, the resultant wave shape depends upon whether the even or the odd ones predominate, and upon the amplitudes of all of them.

With age, one's hearing ability decreases so that elderly people cannot even hear frequencies above 6000 cycles in most cases. Therefore, for all practical purposes, it is not necessary to reproduce these high frequencies.

► Most of the power needed to reproduce music is used to reproduce low-frequency notes, and, since the ear is not sensitive to the lows and highs (low and high frequencies), the reproduction of sound at a volume much below that of the original makes the music sound unreal. It is for this reason that the so-called low (or bass) boosters are included in radio receivers to amplify excessively these lower frequencies and get effects which sound more natural to the ear. Some of these boosters are automatic compensators, while others are tone controls, adjustable at the will of the customer. Thus, it is quite possible that the customer will even introduce a certain amount of frequency distortion, in order to make the sounds more natural to his ears!

Of course, if too many of the high or low frequencies are cut off, then the frequency distortion becomes objectionable.

AMPLITUDE DISTORTION

When a radio circuit or device does not produce current or voltage changes which are *exactly proportional*, at each instant of time, to the changes occurring in the voltage or current values of the incoming signal, we have what is known as *amplitude distortion*.

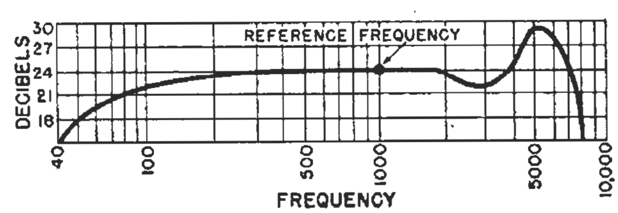


FIG. 14. Here is a typical response curve for an audio amplifier, showing the output over its frequency range. The output is given in db above the .006 watt zero level, so that variations in the response can be judged in terms of their effects on the human ear. Using the db table in Fig. 8, practice converting the db values into watts. Thus, 24 db is equal to 1.5 watts, etc. (Sometimes you will find that a similar curve is used, but the reference frequency output is labelled 0 db. Then values above are written as +db, while those lower are written as —db. That is, when 24 db is called 0, 27 db becomes +3, 30 becomes +6, 21 becomes —3, and 18 becomes —6 db. This allows us to consider the output in terms of the db change above or below the reference level.)

If the original wave is a sine wave, then amplitude distortion will result in the production of harmonics which were not present in the original signal. If the original signal is of a complex nature, additional harmonics are added, or the relative amplitudes of the harmonics are changed by amplitude distortion.

Fig. *15* shows a typical example of amplitude distortion. At *A*, we have a sine-wave signal. We want the amplifier to increase the amplitude of this signal, as shown at *B*, without changing its wave shape. However, if we get the amplified signal with the wave shape changed as shown at *C*, then we have amplitude distortion. As you can see, this wave is not an exact replica of the original sine-wave signal.

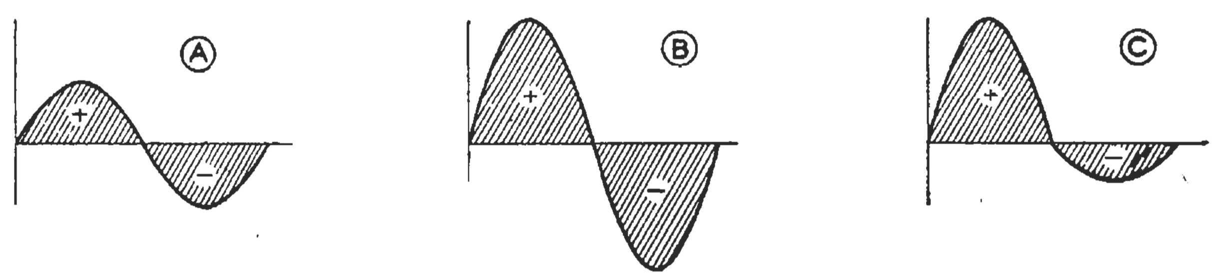


FIG. 15. A typical sine wave is shown at A. When it is amplified, it looks like that shown at B. However, should amplitude distortion occur, it may look like the wave shown at C.

► What could produce this change in wave form? One of the most common sources of amplitude distortion is the operation of a class A amplifier over a curved tube characteristic, as shown in Fig. *16*. Notice that the input signal e_g has more control over the plate current on its positive swing than it does on its negative swing. This results in an output signal (produced by plate current i_p) which is not a sine wave—or rather, it is a pure sine wave plus harmonics. Thus, although we started with a sine-wave signal on the grid, we are obtaining a sine wave plus harmonics in the plate circuit.

Analyzing the wave form of the output signal shown in Fig. *16*, we find that *even* harmonics, such as the second, fourth, sixth, etc., have been added to the fundamental, as shown in Fig. *17*. In other words, the tube's action is approaching that of a half-wave rectifier. From your knowledge of rectifiers, you will remember that this wave form has a d.c. component as shown in Fig. *17*. This d.c. current will be added to the normal d.c. current value. This means that, when a signal is applied to an amplifier tube operating on the lower bend of its characteristic, the resulting distortion will cause the d.c. plate current to rise. This is one of the means whereby a serviceman can determine when distortion occurs—he can apply a sine-wave signal and measure the d.c. plate current before and after the signal is applied. If the stage is a class A amplifier, there should be no change in the d.c. plate current. If there is a change, distortion is taking place.

Should a complex wave be applied to the amplifier having the characteristic shown in Fig. *16*, even harmonics will be added to the wave if they are not already present, or the amplitudes of any even harmonics which are present will be increased by this distortion.

► Going further, a similar distortion can occur if the grid is allowed to swing positive. As shown in Fig. *18*, there is an upper bend in a tube characteristic; this will also produce distortion. The output wave is like that of Fig. *16* except that it is reversed, or

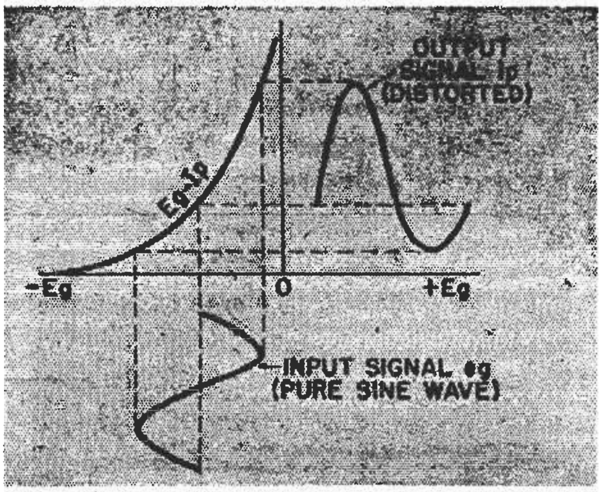


FIG. 16. Amplitude distortion is produced whenever the E_g-I_p curve of a tube is not linear (straight) in the operating region.

is upside down. It has the same components as are shown in Fig. *17*, except that they are shifted in phase 180°. The negative (downward) swings are the larger, so the d.c. component will be of the opposite polarity to that shown in Fig. *17*. In this case, there will be a reduction in plate current when the signal is applied.

► Therefore, if the d.c. plate current changes when a signal is applied to a class A stage, distortion is occurring. If the d.c. plate current *rises*, the tube is operating on the lower bend of its characteristic. (This operation is caused by excessive bias or low plate supply voltage.) If the d.c. plate current *falls*, the tube is operating on the upper bend in its characteristic. (Operation here is caused by too little bias or by an excessive plate supply voltage.) Notice the value of *knowledge* to the serviceman. Upon observing that distortion is occurring, he can take a meter reading and, *from his knowledge of circuit actions,* he can reason directly to the probable causes of the trouble!

► There is another kind of amplitude distortion in which *odd* harmonics are added instead of *even* harmonics. This distortion will be produced when the signal applied to the grid is excessive, so that operation occurs over *both* tube curvatures as shown in Fig. *19*. Here, the shape of the output wave indicates that the third, fifth, seventh, and other odd harmonics are being added to the fundamental. This distortion is very noticeable to the ear, but it will not cause a change in the d.c. plate current if both halves of the output wave are exactly alike. The serviceman must use a cathode ray oscilloscope to actually "see" the wave form before he can find the stage producing this diffi-

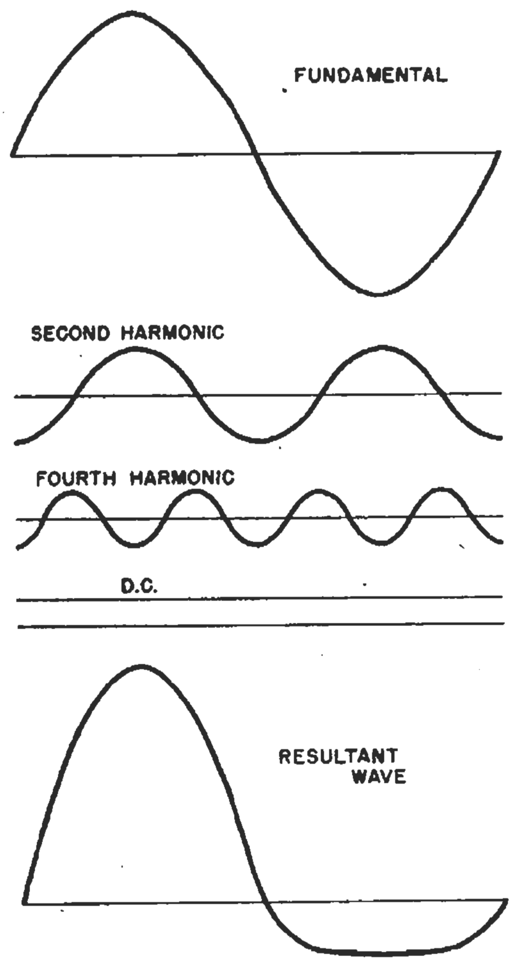


FIG. 17. Here is an analysis of the wave produced by the E_g-I_p curvature which was shown in Fig. 16. Notice that even harmonics are added, plus a d.c. component. Although we have shown only the second and fourth harmonics, there are actually small amounts of other even harmonics added. A phase shift is also involved in that the harmonics are added 90° out of phase with the fundamental. This phase shift cannot be heard, but the added harmonics are noticeable. (In television, the phase shift is of importance also.)

culty, unless there are other clues in addition to the distortion.

► Incidentally, it is possible to tell from the wave shape whether the distortion is an even or an odd harmonic addition. If the two halves of the resultant wave are *exactly alike,* only odd harmonics have been added. If two halves are not symmetrical, however, then even harmonics have been added. Where both even and odd are added, it is not so easy to determine this unless one or the other happens to predominate.

Amplitude distortion is readily detected by the ear. Strangely, the ear seems more critical of odd harmonic distortion than it is of even harmonic distortion. An odd harmonic distortion causing a change in harmonic levels of more than 5% begins to be noticeable,

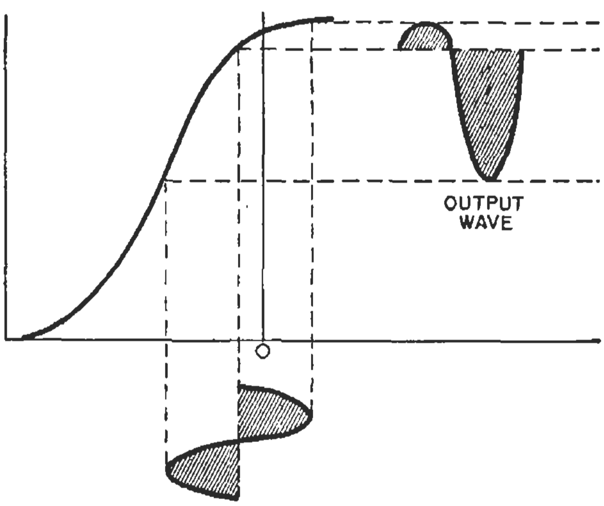


FIG. 18. Amplitude distortion will occur also if the tube is allowed to operate at the upper bend of its characteristic.

while an even harmonic distortion of as much as 10% is relatively unnoticeable.

PHASE DISTORTION

When a radio circuit or part changes the phase relationship between different frequencies in the signal, we have phase distortion. The distance (phase) between peaks of the different signal components is changed, with the result that there is a change or distortion in the wave form of the output signal. (Later in this lesson we will show an example of phase distortion.)

The human ear is relatively unconscious of even large amounts of phase distortion, being far more critical of amplitude or frequency distortion. The reason for this is that the ear hears the individual components of the wave rather than the resultant wave. That is, the sound wave is "taken apart" by the ear. Hence, the ear is quite conscious of added or deleted frequencies, and of changes in amplitude. It cannot detect changes in *time* in the arrival until such change approaches the time length of a syllable or of a musical note.

However, phase distortion is far more important in television. As we shall learn later in this lesson, the time delay caused by even a small phase shift in a television signal will be noticed readily by the eye.

AUDIO AMPLIFIER REQUIREMENTS

From the foregoing, we can begin to see just what the requirements are for an audio frequency amplifier. In order, we may well list them as follows:

Voltage Amplification. A certain amount of voltage amplification is necessary in all radio receivers, as the voltage output of the average second detector is too low to operate the power output stage. Therefore, one or more voltage amplifying stages are used between the second detector and the power stage.

Power Amplification. Raising the voltage level is not enough—*power* is required to operate a loudspeaker. Therefore, the output stage of the audio amplifier in a radio receiver will always be a power stage, designed to deliver sufficient power to operate the loudspeaker. Actually, this stage can be said to be converting signal voltages into high signal powers.

As you have learned in earlier lessons, one of the important differences between power stages and voltage amplifying stages lies in the choice of the

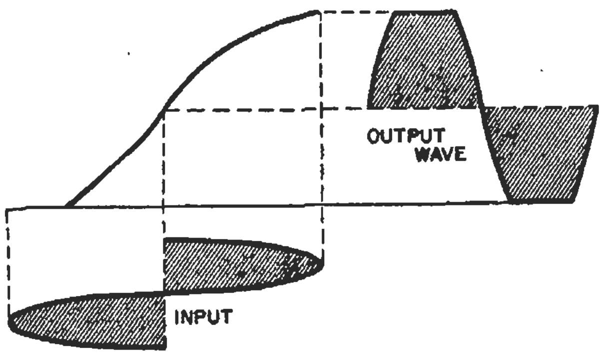


FIG. 19. When the input signal is allowed to become so large that it exceeds the normal operating region, both halves of the output wave will be distorted. This shows that odd harmonics have been added. Strangely, this distortion is far more noticeable than is even harmonic distortion.

load values. However, there are other important differences as we shall soon learn.

In public address systems, where large amounts of signal power are required, medium power stages may be used to drive high power stages. Similarly, in transmitters, the power levels may be quite high. In these two instances, true power amplification does occur, for the final power stage is actually *driven* by a low power stage.

Distortion. From the foregoing, you can see that every effort must be made to cause tubes to operate on the straight-line portion of their characteristic curves. Inductance and capacity values must be carefully chosen and controlled to prevent the undesirable formation of low-pass and high-pass filters which would tend to limit the frequency response. Even a matter like the choice of the load value is a compromise between the amount of amplification desired and the fidelity of response wanted.

► In your next lesson, you will learn much more about these important requirements and about how they affect the serviceman and his choice of replacement parts.

The Fundamentals of Television

Television involves the transmission of intelligence which reaches our brain through our eyes. First, let us consider what the eye sees when it looks at an object. Ordinarily, it looks at reflected light, made up of electromagnetic waves; occasionally, it looks directly at light sources such as electric lamps, a fire, or the sun. The eye sees color because the electromagnetic waves in the visual band have different frequencies, each frequency or group of frequencies giving, through the action of the brain, a color sensation. The human eye serves as a complicated lens (much like the lens in a camera), for it projects these electromagnetic waves on the *retina,* a surface at the back part of the eye. This retina is composed of millions of nerve ends, each of which is connected to the brain. These nerve ends interpret the strength of each electromagnetic wave which hits them (determined by the brightness of the object) and they also interpret the frequency of the wave (the color of the object). *Each nerve end "sees" only a tiny portion of the entire scene;* the brain reconstructs the over-all picture by assembling all the nerve impulses. Thus, the eye breaks up the scene into elements, each of which is transmitted over a separate nerve channel to the brain.

One scientist calls the human eye nature's own television system. The object viewed acts as the transmitter in the system, sending out electromagnetic waves which are picked up by the eye acting as a receiver, and then are relayed to the brain to give us the sensation of seeing.

A Suggested Television System. This action of our visual mechanism immediately suggests a method of constructing a television system. Why not arrange thousands or millions of tiny electric eyes on a screen to pick up the light waves, and connect these by thousands of wires or radio frequency transmitters to a receiver containing thousands of tiny glow lamps? Each of these would reproduce the amount of light picked up by its corresponding electric eye, so the combination of all the lamps would reproduce the object viewed by the transmitter. Yes, a television system like this has actually been tried for land wire television, but only on a small scale. The scheme was found to work after a fashion, but obviously was far from practical, as entirely too many wires were necessary.

Practical Television Systems. The television systems in use today do not attempt to pick up a complete scene and transmit it to a receiver all at once. Instead, television takes advantage of an eye characteristic known as *persistence of vision*—the ability of the eye to retain an impression of an object for a short time after the object has disappeared from view. This makes it possible to send a portion of a scene at a time, so long as the entire scene is transmitted before the eye "forgets" the first of it.

The scene is broken up into elements by *scanning* or by viewing a small portion at a time. Scanning is an operation very like what you do when you read this page. You don't look at the page and attempt to read every word at once. Instead, you read the first line from left to right, swinging quickly back to the left-hand side of the second line, read the second line, go back to the start of the third, and repeat the process until you have taken in every word by itself.

That is just about what a television "camera" does. (This camera is the pickup device of a television system, corresponding to the microphone in a radio system.) In effect, an "eye" in the camera travels over the top edge of the scene from left to right, swings quickly back to the left-hand side, moves down slightly, travels horizontally over the scene again, and repeats the process until the whole scene has been scanned. As you no doubt know, or have guessed, this "eye" is really a light-sensitive surface which converts the light received from the scene into an electric current. This current, which of course varies as different parts of the scene come into view of the scanning eye, is then transmitted by radio

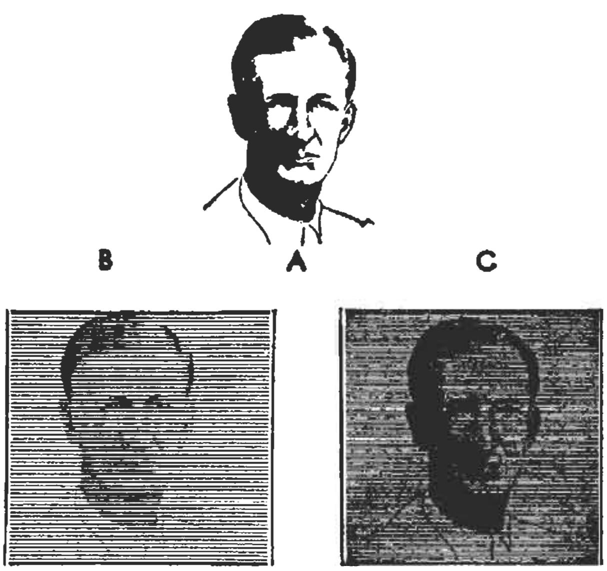


FIG. 20. The drawing at A is reproduced as a series of lines at B and C. Greater detail is obtained by using more lines, as at C.

to the receiver. At the receiver, the process is reversed, and the original scene is traced out line by line.

This is a highly simplified version of how a television system works, but it will serve to show you the basic idea of operation. Right now, the important fact for you to grasp is that a scene is televised "bit by bit," not as a whole.

► Fig. *20* illustrates the general effect produced when a scene is scanned. Suppose we wish to televise a picture like that in Fig. *20A*. After it has been scanned by the camera, transmitted to the receiver, and reproduced on the receiver screen, it will have something of the appearance shown in Figs. *20B* and *20C*. That is, it will consist of a series of lines; these lines will vary in brightness along their length, and so make up the picture we see. The more lines we have in a given area, the greater the detail of the final picture. Fig. *20C*, which has 120 lines, exhibits more detail than Fig. *20B*, which has only 60.

Note that, as you move the illustrations in Fig. *20* farther and farther away from you, a point is reached for each illustration where the details seem to blend into a complete and nearly perfect reproduction of the original. This brings out an important fact about television: if a reproduced picture is made larger without increasing the number of lines, the picture will have to be viewed from a greater distance to get a satisfactory eye impression.

HOW SCENES ARE SCANNED AND REPRODUCED

Before considering the technical details of breaking up a scene into a number of lines, it will be valuable to get clearer ideas of how a scene is taken apart or *scanned*, and how a scene is reproduced. Only the basic schemes will be considered; naturally, different television experts have different ways of accomplishing the desired results.

Mechanical Scanning Methods. Even though mechanical methods of scanning are considered inadequate today, we will consider them first since they are easier to understand. Punch a hole in the center of a small business card with a pin and hold the card up to one of your eyes, so you can look through the hole. Turn to some object or scene. Notice that you can see only a small part of this scene through the tiny hole. Now move the card horizontally from left to right; you see all the portions of the scene along the line which you are scannning. Move the card back and forth horizontally while shifting it vertically downward a little at the end of each line, and your eye will scan the entire scene, piece by piece.

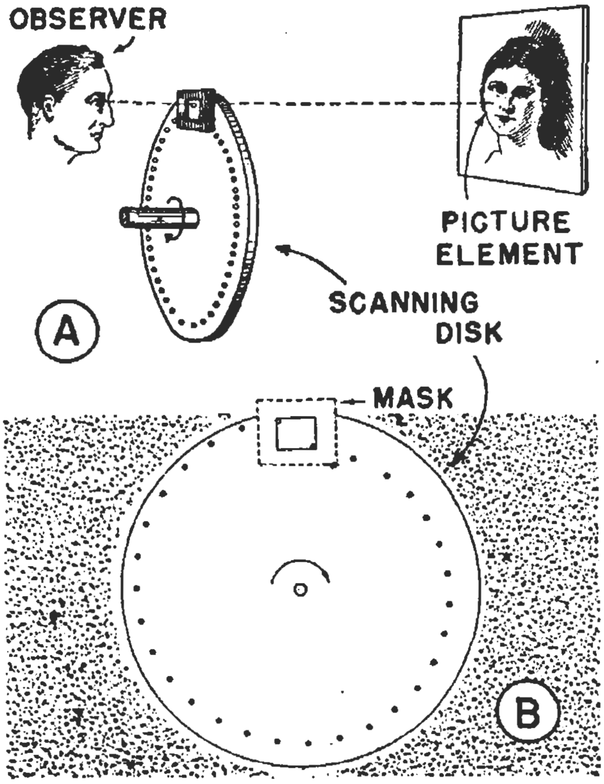


FIG. 21. This diagram shows an elementary mechanical scanning disc system. If the disc is rotated rapidly enough, the observer will be unconscious of its presence, as persistence of vision will allow him to apparently see the entire scene, although he is actually viewing only a tiny spot at a time.

The Scanning Disc. In place of this crude scanning device, we can use the system shown in Fig. *21A*, in which a large number of holes are arranged in a spiral fashion on a rotating disc called the *scanning disc*. This disc really replaces the business card used in our previous example: one complete revolution of the disc gives one complete scanning of the entire picture, for each hole on the disc scans one line. If the disc is revolved fast enough, the visual sensation is the same as if the entire picture were being seen at one time.

The exact arrangement of the holes on the scanning disc is shown more clearly in Fig. *21B*. The observer views the scene through the mask, a rectangular opening in a piece of black cardboard. As the disc is rotated, each hole moves across the opening in this mask, the outermost hole in the spiral moving across the top of the opening and each succeeding hole moving across one line down. Finally, when the innermost hole has moved across the bottom of the opening, the outermost hole again scans the top line and the entire scanning process starts over again.

Mechanical Television Transmitters. If the observer in Fig. *21A* is replaced with a light-sensitive cell, this cell will deliver a varying electric current which is at all times proportional to the amount of light reaching the cell, and therefore proportional to the shade of lightness or darkness of the element of the picture being scanned at a particular instant. This arrangement gives us a means of converting a picture or scene into a varying electrical current. This current or picture signal can be amplified and placed on a radio carrier for transmission through space. At the receiver, the carrier can be demodulated and the picture signal amplified sufficiently to operate a picture reproducer.

Mechanical Television Receivers. In the early television receivers, the amplified picture signal was fed to a neon glow lamp like that shown in Fig. *22A*. This lamp consisted of a wire anode and a rectangular flat metal piece (the same size as the reproduced picture) which served as a cathode.

These elements were enclosed in a gas-filled glass envelope. A red glow of light formed on the plate when sufficient voltage was applied between the electrodes; the intensity of this glow varied with the applied voltage. The amplified picture signal was made to change the applied voltage, thus changing the intensity of the glow.

A pin-hole scanning disc was rotated before the glow lamp in such a way that the holes scanned the glowing plate. The transmitter and the receiver were so synchronized that when the scanning disc at the transmitter started to scan the top line of the scene, the receiver scanning-disc likewise started to scan the top line. Line by line the scanning discs were kept in step or in synchronism, so that the intensity of the glow lamp at any instant corresponded to the intensity of the light reflected from that same element on the actual scene. The arrangement of the scanning disc and glow lamp are shown in Fig. *22B*. The lens shown is a magnifying glass used to enlarge the image to three or four times the size of the glow lamp plate.

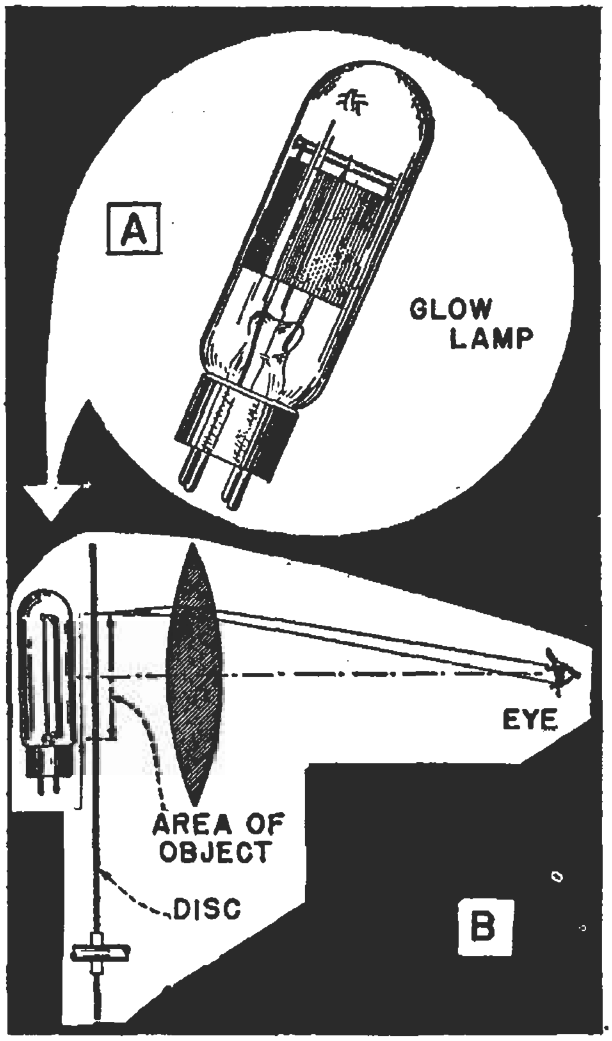


FIG. 22. An early type of mechanical television reproducer. The glow-lamp light depended on the brilliancy of the spot being scanned at the transmitter at that moment. The scanning disc is synchronized (in step) with the transmitter disc so that it arranges the light elements in their proper sequence.

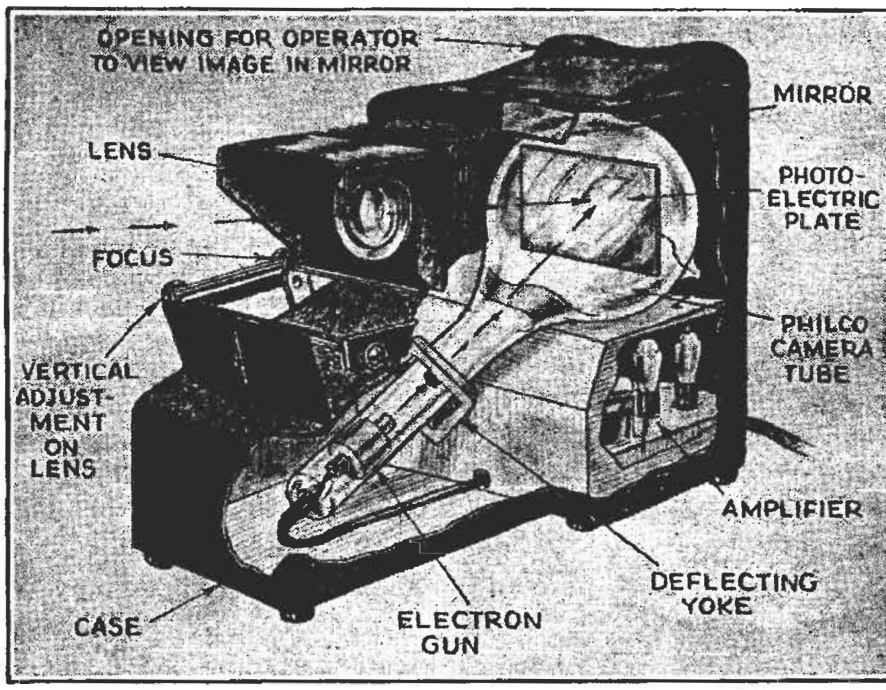
Courtesy Don Lee Broadcasting System

The engineer is holding an electronic pickup tube such as is used in television studios. Scanning is accomplished within this tube by electronic means.

Electronic Television Transmitters. Although present day methods of scanning and picture reconstruction differ greatly from the method just described, the principle of breaking up a picture into a number of elements which are scanned line after line is still used. Figure 23 illustrates a modern electronic television camera. The scene is focused on the photoelectric plate by a high-grade camera lens combination. This light-sensitive photoelectric plate consists of millions of tiny light-sensitive spots, each insulated from the others and each scarcely larger than the point of a pin. Under a microscope this plate looks as if it were covered with grains of sand.

When a scene is projected on the photoelectric plate by the lens, the action of light drives out electrons from each of the tiny light-sensitive units. These electrons pass through the space in the tube to a conducting surface on the inside of the glass envelope, which is at a high positive voltage and therefore attracts the electrons. The action of light thus leaves the photoelectric plate elements more or less positively charged (because they have lost electrons).

Naturally, the amount of electrons lost from any given section of this photoelectric plate depends upon the amount of light reaching that section. Thus, some spots on the plate are more positively charged than others, and we actually have an electronic image of the scene. An electron gun now shoots a fine stream of electrons at the photoelectric plate. Electromagnetic deflecting coils (here designated as "deflecting yoke") shift this electron beam horizontally and vertically, one line at a time, to scan the entire photoelectric plate from top to bottom. When this electron stream strikes a positively charged surface, that surface recovers its electrons and, in so doing, relays the charge to a flat metal supporting electrode which is back of, but insulated from, the photoelectric plate.



Courtesy Radio-Craft

FIG. 23. A cut-away sketch showing the arrangement of parts inside one type of electronic television camera.

In this manner, an electronic impulse is relayed from each spot which is hit by the electron beam. The size of each impulse corresponds to the amount of light striking the spot, so the sum of all the impulses (sent one at a time) constitutes a picture signal. The supporting electrode collects the picture signal and, after a great deal of amplification, the picture signal is placed on a carrier wave and transmitted through space, just as in the mechanical television system. In addition, impulses are sent at the beginning of each "frame" or new picture, to keep the image-reconstructing devices in step with the scanning mechanism at the transmitter.

Electronic Television Receivers. Figure *24* shows a simplified diagram of a typical electronic picture reconstructor. This employs an electron gun and two sets of electromagnetic deflecting coils. Special oscillators generate the current pulses which flow through these coils; the oscillators are controlled by the synchronizing impulses sent out by the transmitter. A spot of light appears on the special screen at the end of the tube when it is hit by the electron beam produced by the electron gun; the brilliance of the spot increases with the speed of the electrons in the beam and with the number of electrons in the beam. The picture signal voltage controls the speed and number of the electrons in the beam by means of a special grid electrode, and the deflecting coils control the scanning of the beam across, and up and down, the screen. The combined action is such that while the beam is sweeping across the screen, its intensity is changing continually in accordance with the picture signal, and the effect of "painting" light on the screen is secured. The picture size is controlled here by the size of the screen; a 12-inch diameter tube gives a 7.5" × 10" picture.

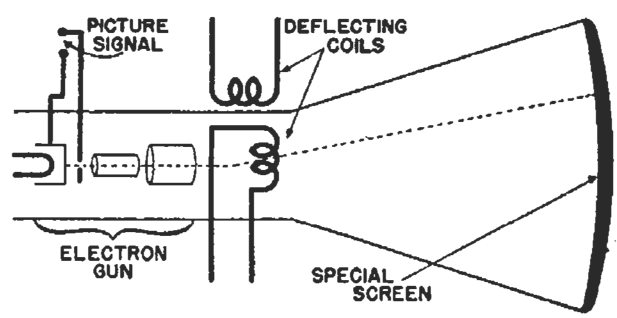


FIG. 24. A simplified diagram of an electronic picture reconstructor tube.

Video Amplifier Requirements

Television, like sound systems, requires both low- and high-frequency amplifiers. However, the requirements of both are far more exacting than those on the corresponding sound amplifiers. Basically, there must be voltage and power amplification, but the very wide frequency band which must be handled with a minimum of distortion requires special tubes and unique circuits, as we shall point out in later lessons.

VIDEO FREQUENCIES

The low-frequency television amplifier (called a video amplifier) appears almost misnamed, as it must handle frequencies which are well up in the r.f. range of sound systems. However, as in sound systems, the video signal is only that which was removed from the carrier by the demodulator; the television carrier is far higher in frequency.

First, let's see just what range of frequencies must be handled by the video amplifier. To begin with, we want a picture with the greatest possible amount of detail. People are familiar with the details of photographs and the movies and naturally expect television to be as good. As technical problems are solved, details become better, so that today the fidelity is acceptable. Even so, still greater fidelity is being sought.

► Another problem is *flicker*. To give the sensation of motion, the scene must be scanned over and over, with one scanning rapidly replacing another, just as in the movies. Each scanning is called a "frame" and, the more frames there are per second, the less chance there is for the eye to see them individually—they blend together better. If too few are transmitted, a flickering of the image becomes noticeable.

► Greater picture detail can be obtained by increasing the number of lines per frame; increasing the number of frames per second gives less flicker of the reproduced picture. Both of these factors contribute to what is called high-definition (or high-fidelity) reproduction. However, there are definite limits to the number of lines and frames which can be handled.

Picture Elements. The maximum frequency of the picture signal current is of great interest to every television engineer, for all television equipment must be designed to handle this frequency. In order to have a way to calculate the maximum frequency of the picture signal current, it is assumed that the picture being scanned consists of a checker-board pattern of black and white squares, with each square being equal in size to one of the sensitized spots on the photoelectric plate of the television camera. (Since each of these sensitized spots is the smallest part, or element, that the camera can "see" of a scene, these spots are usually called "picture elements.")

The signal current is said to go through one cycle each time the electron scanning beam passes over one light and one dark picture element, because the signal current goes through a maximum and a minimum value each time this happens. For our checkerboard scene, the picture signal current goes through a cycle each time the scanning beam passes over two consecutive picture elements. To find the maximum frequency of the picture signal current, then, all we have to do is compute the number of picture elements scanned per second and divide by two (since it takes two elements to make one cycle). Let's go through the simple computations involved and see just how high this maximum frequency may be.

If each picture element is considered to be as high as it is wide, it is easy to compute the number of elements in one complete picture. Assuming a square picture with N lines, then there will be N picture elements per line, or N times N picture elements in the complete square picture, which is known technically as *one frame*. For example, at present the television standards call for a 525-line picture. Hence, in a square picture there will be 525 times 525 or 275,625 picture elements. For ordinary calculations, 276,000 elements will be sufficiently accurate.

Aspect Ratio. The pictures commonly involved in television are not square, however; they are wider than they are high. The width of a picture divided by its height is called the *aspect ratio*. In order to conform to motion picture standards, the aspect ratio has been standardized at 4/3 or 1.33. This means that the number of elements in each line has been increased by the aspect ratio, which we will designate as a. Now the number of picture elements per frame or picture will be N times N times a. For the example just considered, the total number of elements will therefore be 276,000 times 4/3 or 368,000.

Frame Frequency. The number of pictures sent per second is the *frame* (or picture) *frequency;* let us designate it as F. By multiplying the number of picture elements in a frame by the frame frequency, we get the total number of picture elements per second. The total number of picture elements per second is then $N \times N \times a \times F$. Since it takes two picture elements to make a cycle, we get the maximum number of cycles per second by dividing the preceding formula by 2. The standard frame frequency is 30; in our example, then, we get the maximum frequency involved by multiplying 368,000 by 30 and then dividing by 2; the result is 5,520,000 cycles per second.*

Factors Affecting Frequency. In actual television practice, the picture elements are being scanned only about 85% of the time. The remainder of the time is used for sending line synchronizing impulses. This fact increases our maximum picture frequency because we must crowd our picture elements into 85% (85/100) of a second. We must, therefore, multiply our computed value by 1.17, making the maximum picture frequency 1.17 times 5,520,000 or approximately 6,458,400 cycles.

► Up to this point, our analysis of the maximum frequency has been based upon the assumption that there is always a sharp contrast between adjacent elements of the picture or scene, one dark and the next light. Of course, this is not true in actual practice, for no scene is made up of perfectly arranged checkerboard squares. Several adjacent picture elements in a line may reflect the same or nearly the same amount of light. Also, in most scenes, especially those having action, it is not necessary that slight variations between the shades of adjacent picture elements be transmitted. The average scene thus contains considerably less than the maximum possible number of cycles (changes from light to dark).

This is quite fortunate, for it reduces the maximum frequency required. Tests and experience have

*The formula now is: Maximum theoretical picture frequency $f_p = \frac{1}{2} N \times N \times a \times F$.

shown that apparatus capable of sending about 60% of the maximum frequency is fairly satisfactory. Since the maximum number of cycles was assumed in our example, we multiply 6,458,400 by .6 and get about 3,900,000 cycles (3.9 megacycles) as the final maximum frequency for a 525-line picture scanned 30 times per second, with the standard aspect ratio of 4/3. Any increase in this frequency up to the extreme limit of 4.5 megacycles permitted within a television band gives a definite improvement in picture fidelity.

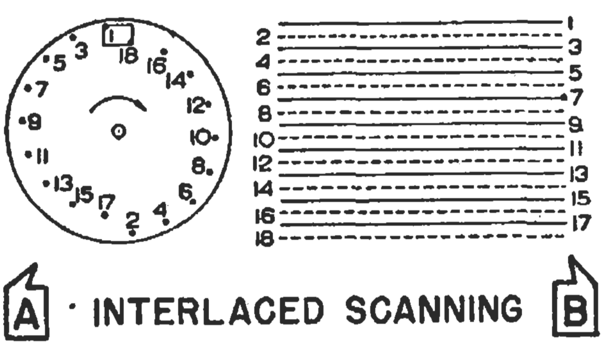


FIG. 25. The double-spiral type of scanning disc at A illustrates the principle of interlaced scanning. The arrangement is such that the successive holes on one spiral skip every other line of the image, and the holes on the other spiral scan the lines missed by the first spiral. The result is shown at B. Whether produced by the double-spiral disc at A or by an electronic television system, the lines of the picture are scanned (and reproduced) in the order 1-3-5-7-9-11-13-15-17-2-4-6-8-10-12-14-16-18.

The Minimum Frequency. The upper part of the average outdoor scene (usually the sky) is bright, while the lower part is considerably darker. In scanning such a scene, the picture elements are varying in light intensity at a high level for the upper half of the picture and at a low level for the remainder of the picture, giving one cycle of change from light to dark for each scanning of the picture. Transmitting these changes properly calls for a minimum frequency corresponding to the vertical scanning frequency (the frame frequency). Satisfactory reproduction of slow changes in background illumination requires, however, that frequencies down to at least 10 cycles be passed, so we should consider 10 cycles as the minimum frequency for a practical high-fidelity television system.

► For a 525-line picture having an aspect ratio of 4/3 and a frame frequency of 30, the picture frequency ranges from a minimum of 10 cycles to a maximum of about 3.9 megacycles. Compare this with the frequency range of high fidelity sound, which extends from about 35 to 8500 cycles per second!

Interlaced Scanning. Increasing the number of frames per second reduces flicker, but it also steps up the maximum signal frequency. For example, a 525-line picture with an aspect ratio of 4/3 and 30 frames per second gives a maximum frequency of about 3.9 megacycles. At 60 frames per second, the maximum frequency becomes about 7.8 megacycles, which is way beyond the present ability of television apparatus. But scientists were not to be balked; they resorted to interlaced scanning, an old principle of television which was used on many of the scanning disc systems. In these systems, the holes were arranged in two sets of spirals, as shown in Fig. *25A*. There was one set of holes scanning or reconstructing every other line, as illustrated in Fig. *25B*. Although the number of lines in the picture is not increased, the frame is scanned vertically *twice* for each complete scanning of all the elements. This does not change the maximum signal frequency, but reduces flicker considerably. Each of the "scannings" are called a "field" to distinguish it from the complete image which is called a *frame*. There are two fields in each frame, in the system just described.

► This same principle of interlaced scanning has been applied to electronic television methods. It was discovered that at a field frequency of 60, by using an odd number of scanning lines per picture, the picture would automatically interlace. This is why 525 lines is the present standard rather than 500 lines or 520 lines or any other *even* number.

PHASE DISTORTION

Radio apparatus designed for sound signals must handle the required range of frequencies in the sound, and must not introduce either frequency or amplitude distortion. Television amplifiers also have these requirements, as well as one other, that there be negligible phase distortion. With sound signals, phase distortion is of little consequence because the human ear is concerned only with the frequency of a sound and with its strength, but in television, the phase distortion may tend to shift an impulse from one picture element to the other which definitely will cause distortion.

The effects of phase distortion are illustrated in Fig. 26. When the signal consists of a fundamental frequency *A* and a third harmonic *B*, the resultant wave *C* will cause, when fed into a suitable reproducing device, light and dark areas to appear on the screen in exact reproduction of the light and dark areas of the original scene. If, however, radio apparatus were to shift the third harmonic by 90 degrees, as at *D* and *E*, the resultant wave at *F* will no longer represent the original scene. This can easily be proved by comparing corresponding points, such

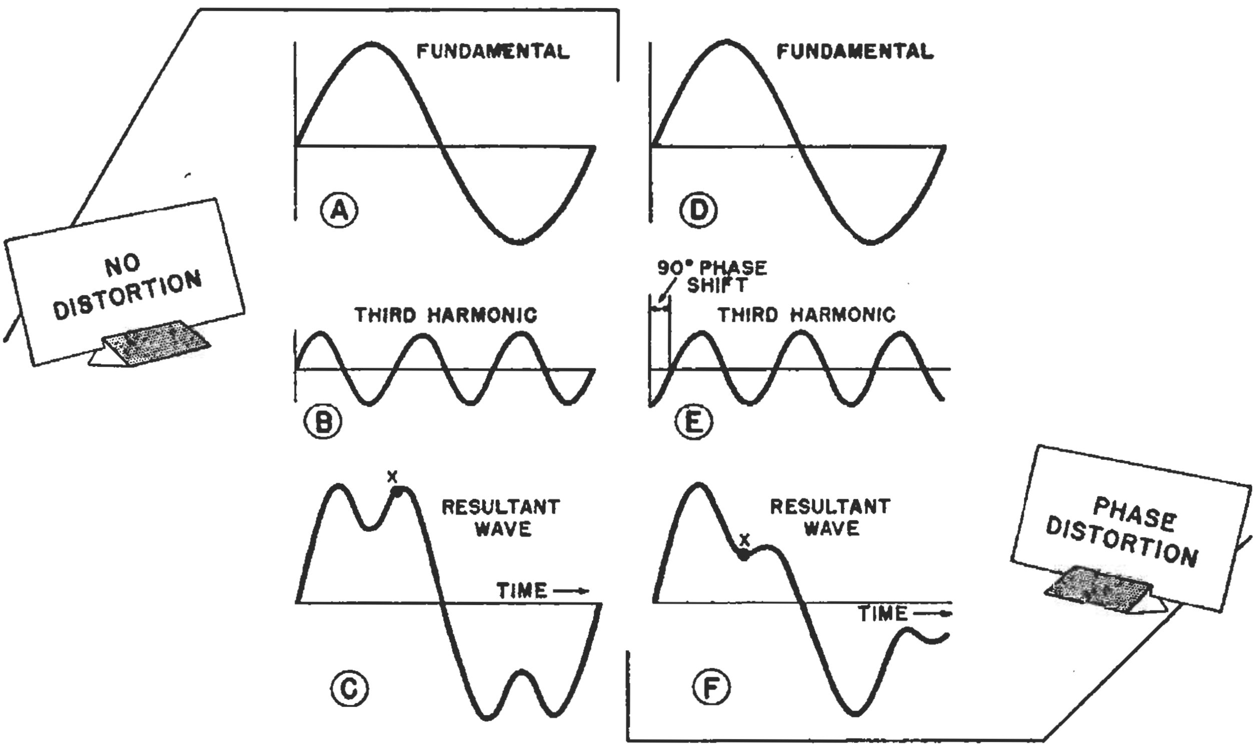


FIG. 26. A comparison between the resultant waves will show that phase distortion will produce a different wave shape. In television, this means the spot brilliancy at the receiver will not correspond to that of the scene at the transmitter.

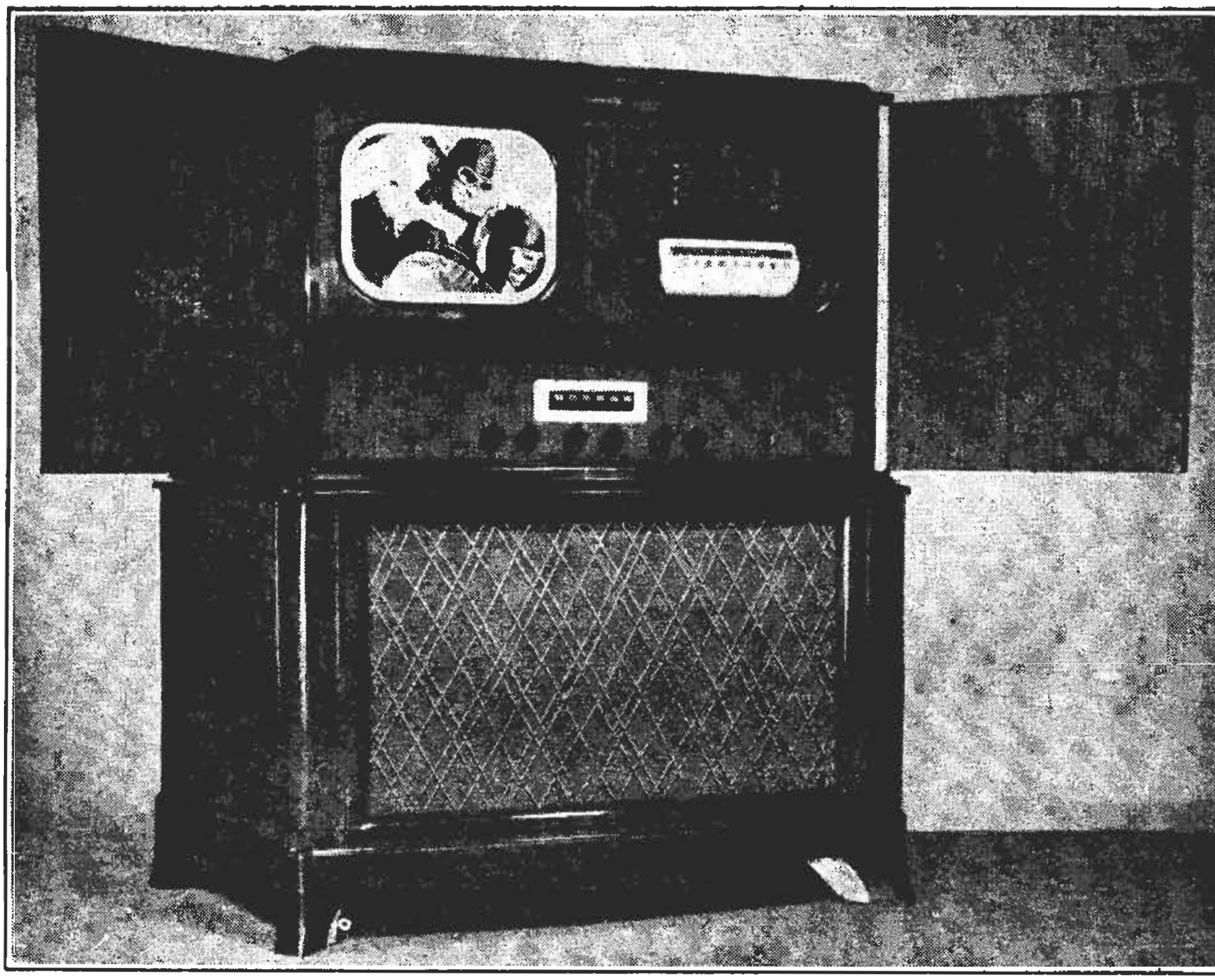
Lesson Questions

Be sure to number your Answer Sheet 14FR-2.

Place your Student Number on *every* Answer Sheet.

***Send in your set of answers for this lesson immediately after you finish them,* as instructed in the Study Schedule. This will give you the greatest possible benefit from our speedy personal grading service.**

1. Name the three forms of intelligence which can be sent over either wire or radio communication systems.
2. Why isn't sound transmitted by a perfect vacuum?
3. What is the speed of sound waves through air under normal conditions?
4. What three things can happen to sound waves which strike a material?
5. How can the reverberation period of a room be reduced?
6. What are the two important characteristics of a sound having the form of a simple sine wave?
7. Can the average human ear detect changes of less than 3 db in the intensity of a complex sound?
8. Which TWO of the following are the most objectionable types of distortion insofar as sound signals are concerned: *1, frequency distortion; 2, amplitude distortion; 3, phase distortion?*
9. What characteristic of the human eye allows it to retain an impression of an object for a short time after that object has disappeared?
10. What is meant by the aspect ratio of a television picture?



Courtesy General Electric Co.

A combination television and all-wave receiver. The television reproducer tube is at the upper left and the television receiver controls are in the center. The broadcast-short wave receiver controls are at the upper right. The loudspeaker (used for both the all-wave set and for the television sound program) is behind the grille cloth at the bottom of the cabinet front.

as x, on the waves C and F. Since the amount of light produced on the screen of the image reconstructor tube depends on the amplitude of the wave, this difference in amplitude may easily make the image element far lighter or far darker than the original.

Looking Ahead. Now that you have some idea of the requirements placed on low-frequency amplifiers, you are ready to study practical amplifiers. In your next lesson, you will learn how low-frequency amplifiers are made to meet the requirements you have just studied. Then, in other lessons, you will go on to high-frequency amplifiers, demodulators, and other radio sections and stages. Soon you will have a complete understanding of *all* the sections in modern receivers.
