AUDIO, VIDEO, AND CODE SIGNALS

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STUDY SCHEDULE NO. 13

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.

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AUDIO, VIDEO, AND CODE SIGNALS

Radio Communication Services

PEOPLE communicate with each other most commonly by using the senses of hearing and seeing. When information can be heard, it is referred to as aural information; when it can be seen, it is referred to as visual information. Radio makes use of hearing. You are familiar with radio broadcast programs which are examples of the transmission of aural information. Television and facsimile are forms of transmission of visual information.

Radio also conveys information by means of dot and dash (code) signals. Although code signals can be received by a person, as a form of aural information, they can also be received and interpreted by a machine. Automatic code transmission and reception is neither aural nor visual, but it is in a class by itself.

We have, then, three ways of conveying information by radio.

1. Aural, or audio signals.
2. Visual, or video signals.
3. Code signals.

In this Lesson we will study how the sound, scene, or message is converted into electrical signals at the transmitter, and how the electrical signals are converted back into the sound, scene, or message at the receiver. We will also study the nature of audio, video, and code signals so that we can better understand the pickup and reproduction devices which are described in this Lesson.

► This will help us to understand the basic sections and stages of transmitters that we will study about in the next Lesson.

► Of the three types of signals, the code signals are the easiest to transmit. The operator forms dots and dashes with a telegraph key, connected in the radio transmitter circuit, which starts and stops the transmitter. Various combinations of dots and dashes form letters, numerals, and punctuation. A message or other written intelligence must first be translated into dots and dashes for transmission.

► Audio signals are not generated and transmitted as easily as code signals. Sounds are first "picked up" in the broadcast studio or at the radio transmitter by a microphone. The microphone then transforms the sounds into the equivalent electrical form, and since this electrical signal is an alternating current, it is amplified by the audio stages of the radio transmitter.

The audio signals are transmitted as modulations of the transmitter's "carrier." In this process, the power, or level, of the electrical signal from the microphone must be stepped up many times, by means of amplifiers, before they are strong enough to produce the modulations. If the audio signals are transmitted with a high degree of fidelity, the amplifying and transmitting circuits must be designed and operated with considerable care.

► The visual signals required for television transmission are the most difficult to transmit. The picture must be broken up into tiny elements, and each element must be transmitted separately. Besides this, all the tiny elements that form a whole picture must be transmitted in a small fraction of a second, and there must be many whole pictures sent each second for the observers to follow any movement in the original scene. In television, 30 complete pictures are transmitted each second.

In facsimile, the transmission of a picture, document, or written or printed page from one point to another is much less complicated than television. For in this case, the transmission of a complete picture may take several minutes.

In television, the scene or picture is picked up by means of a television camera that transforms the image of the scene into electrical signals which are alternating currents. This electrical signal must be amplified to modulate the transmitter's carrier. The generation, amplification, and transmission of television signals require circuits of greater precision and complexity than those required for the transmission of audio signals.

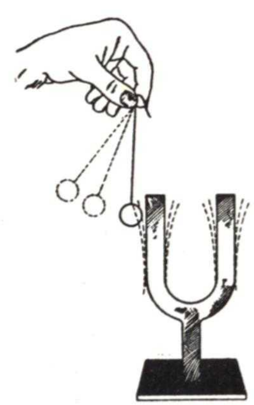
The Audio Signal

A knowledge of sound is essential to the radio broadcast operator in order to understand the radio circuits under his control. In broadcasting, knowledge of acoustics, the science of sound, is necessary in order that the program being produced may be picked up, and transmitted with the best possible realism.

SOUND IS A WAVE MOTION

We know that sound can be produced by striking metal objects together, rustling stiff paper, vibrating vocal chords, plucking a stiff wire or spring, and many other methods. Although these methods are many and varied, they are basically identical. *In every case, we create sound only when we produce a vibration in the air.* 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.

Suppose, for example, that 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 that 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 d in front of it and drives them forward, and they in turn strike molecules e, which are driven forward to strike molecules f, and so on. In this way, the motion of molecules d will be transferred all the way along the line of



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.

molecules. In fact, if it were not for the inevitable losses of energy that occur in such an operation, the forward motion of the molecules would be transferred indefinitely.

However, it is only the *motion* that is transferred in this manner; the in-

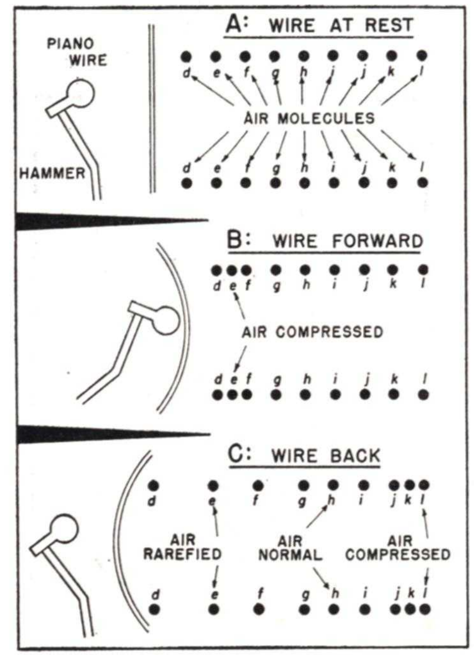
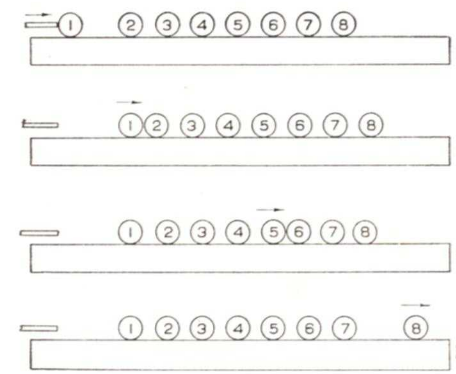


FIG. 1. When a piano key is depressed, a "hammer" strikes a piano wire (actually a group of wires). The wire is set into vibration; this vibration is transferred to the surrounding air molecules. The alternate compressions and rarefactions of the air result in sound waves which travel out from the wire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

dividual 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. Notice that the forward motion of the molecules has produced a compression of the air in the region of molecules d, e, and f, while farther on, where the motion has not yet reached, the molecules are still their normal distances apart (the air is uncompressed).

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



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 a similar way.

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 d, e, and f 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. (To rarefy means to make less dense; it is the opposite of compress.) Molecules g, h, and i 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 j, k, and l are still going through their forward motions, so the air in this region is compressed. Thus, distant molecules are made to move backward and forward in the same manner as those at the source of the sound.

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 that will be transferred through the air molecules surrounding the wire—a *forward* motion (called a compression wave), and a *backward* motion (called a rarefied wave.) Remember: The individual molecules move forward only until they strike other molecules, and then move backward only until they are struck by the next wave of forward-moving molecules.

► Let's assume that the piano wire produces a pure tone having sine-wave characteristics. (Actually, it doesn't produce a sine wave, but let's see what would happen if this were true.) 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 that represent millions of molecules; each molecule on a given line is the same distance from the piano wire as the other molecules on that line. The lines that are bunched together represent compression waves; those that are spread out represent rarefied waves.

Notice the wave form shown directly below the picture of the sound waves. This wave form is constructed from the

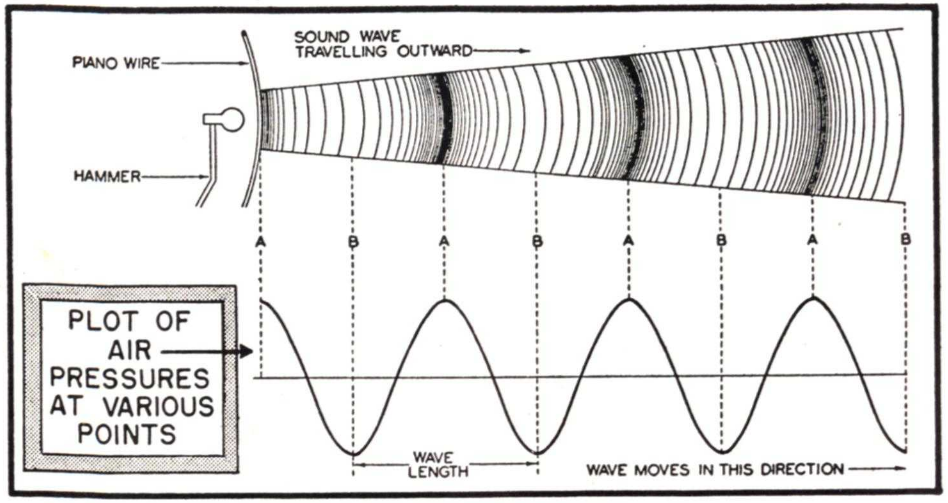


FIG. 2. By plotting the air pressure we find that the sound waves travel away from the wire as alternate changes in pressure. The actual "plot" shows the wave form of the source. Thus, we get a sine-wave plot if the source produces sine-wave sounds. (A piano wire produces a more complex wave than that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

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.

► Notice that the two important characteristics of this simple sine wave are its frequency (pitch) and amplitude (loudness).

► However, regardless of the wave form of the sound wave, the air pressure follows the vibrations of the source, so that a plot of the air pressure shows the exact wave shape. This means that sound is a *variation of pressure* in the transmitting medium. In this case, air is the transmitting medium, although sound will travel through anything but a vacuum.

► A diaphragm, struck by a sound wave, is alternately pushed in by the compression waves and pulled out by the waves of rarefaction. This is the basic principle used in all microphones. Various microphones differ only in the way that this vibration is changed into corresponding electrical signals.

SPEED OF SOUND WAVES.

The speed of sound waves differs with various materials depending essentially on the density (weight per unit volume) and elasticity (the property by virtue of which a body resists and recovers from deformation produced by force). The greater the density of a sound-conveying material, the slower will be the speed of sound through it; the more elastic the material, the faster the sound waves will travel through it. With air, for instance, the speed of sound waves changes with temperature, since the temperature affects the density of the air. Careful laboratory experiments have shown that, at a normal atmospheric pressure, and at a temperature of 0° centigrade (32° Fahrenheit), the speed of sound waves through air is about 1089 feet per second. (These measurements are made in still air, because 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, which is about a million times as fast as sound, it is easy to understand why we can see lightning before we can hear the thunder that it produces.

Sound waves travel through water at a rate of more than 4700 feet per second. Steel is about seven times as dense as water, but its elasticity is so much greater than the elasticity of water that sound waves will travel through steel at about 16,300 feet per second, almost 4 times their speed through water, or 16 times their speed through air.

REFLECTED, TRANSMITTED, AND ABSORBED SOUNDS

In radio, the sounds to be picked up and transmitted are generally produced in an enclosed room called a "studio." It is designed to keep in the program sounds and to keep out unwanted sounds that interfere with the program or the communications. If it is improperly designed, undesirable echoes and reverberations are produced, altering the characteristics of the sound. By using special materials in the walls and floors, these undesirable effects can be reduced.

Sound waves are *reflected from the surface of certain kinds of materials, absorbed by other materials, or transmitted through still other material.* These three reactions are illustrated in Fig. 3.

Echoes. When sound waves are reflected by a flat surface far from the source of sound, we hear *echoes after* the original sound has stopped, which are distinct but weaker reproductions of the original sound. Echoes are heard in large rooms with hard walls that re-

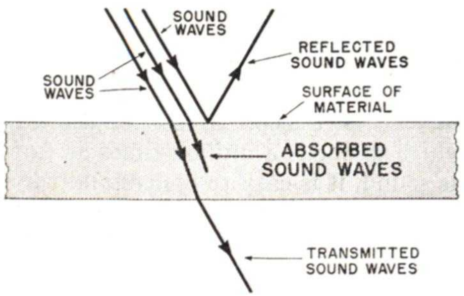


FIG. 3. Three things can happen to a sound wave which hits the surface of a material such as a studio wall. The kind of material will determine which one (or more) of these effects will occur.

flect the sound waves. These echoes can be removed from a room by changing its shape, making it smaller, or covering the walls with some nonreflecting material. This is the architect's job.

Reverberation. We have *reverberation* when sound waves are reflected back and forth between the walls of a room so that the echoes all run together. Reverberations cause the sound, heard at a point away from the speaker, to be "blurred." The blur is produced by the reflected waves which take longer than the direct waves to reach the listener, because they have farther to go. Thus, as shown in Fig. 4, 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.

In a small room, such as a news announcer's booth, paths 2, 3, and 4 will be short, and the sounds travelling over these paths will reach the listener's ear a hundredth of a second or so after the sound that is travelling along path 1. This delay in arrival is so short that the sounds from paths 2, 3, and 4 add to the sound along path 1, so that the sound to the listener is louder than if he were hearing the sound coming by path 1 only.

If the room is large, such as a studio for a large band or symphony orchestra, the sounds arriving along paths 2, 3, and 4 produce distinguishable echoes or reverberations, and the sounds begin to interfere with one another making it difficult to understand speech or to hear the sounds clearly, unless special precautions are taken to avoid this in the original design. A room that has a reverberation time (the time required for a sound to die away to one millionth of its original intensity) of approximately one second, gives a satisfactory and pleasant environment for listening to speech. A longer reverberation time is desirable for listening to music, since some intermingling of the music tones is desirable, and does not detract from intelligibility.

It can be seen from Fig. 4 that if all the sound waves were reflected from the walls with very little loss in their power, they would bounce back and forth from one wall to the other for some time before dying out, and the reverberation time of the room would be long. If, however, we place some sound absorbing material, such as Celotex panels or soft wall coverings on the walls, the sound energy would be absorbed and the reverberation time would thereby be shortened. For this reason, the walls of radio studios are made of special materials that control the reverberation so that a realistic reproduction of the program can be obtained.

Sound-Absorbing Materials. A sound proof room must have good sound-absorbing surfaces that keep out external sounds, and also absorb the sounds produced inside. An open window is sometimes considered an ideal sound "absorber" because it allows the sounds to pass out of the room; on the other hand, it also allows external sounds to enter the room. A room cannot be sound proofed by leaving all the windows open!

A sound-absorbing material dissipates the sound energy by converting it into heat. There are two types of sound absorbers. One type is a porous material that has tiny "pockets" so that when sound waves enter, their energy is dissipated by being reflected back and forth in the pocket. The second type is made of soft materials that "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.

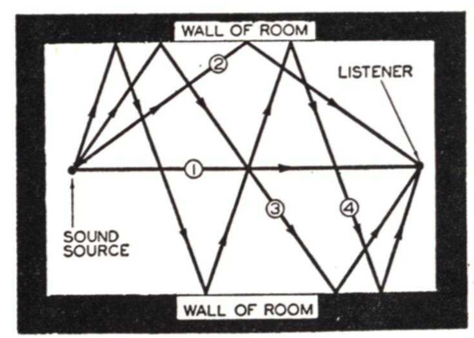


FIG. 4. Reverberations are produced whenever sounds can come to a listener over more than one path. Thus, sound waves reflected from the walls may travel over paths 2, 3, and 4 to the listener. As these paths are longer than the direct path (1), these reflected sounds arrive later than the direct sound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

Solid dense materials are good sound transmitters, and, therefore, poor absorbers; soft, pliable, and porous materials such as velvet, Celotex, rock wool, cotton, carpet, and porous plaster are good sound-absorbing materials. The thicker the material, the more sound it can absorb.

Because we are accustomed to hearing sounds that are reflected from hard-surfaced walls, the sounds that are lacking some reverberation seem dead and unnatural to us. This is very noticeable in a room hung with thick, soft wall coverings. A certain amount of reverberation, then, is necessary in order for sounds to be crisp and natural.

In the design of radio studios, a great amount of attention and effort is applied to getting the reverberation time just right. If it is too short or too long, the program originating in a studio loses naturalness. Some studios are made with sliding panels so that the reverberation time may be reduced for programs consisting mainly of speaking, and may be increased for programs consisting mainly of music.

► Electronic devices have been developed to produce synthetic reverberation. These devices are used on radio programs to create the illusion of large echoing halls and great space.

So far, we have shown how sounds are produced, and we have also indicated some of the problems that studio designers must solve in connection with reverberation and absorption. Reflection and sound absorption have much to do with the characteristics of the sound waves which we intend to pick up. Before we go into a study of pickups, however, let us consider the sound waves themselves.

Technical Facts About Sound and Hearing

We know by experience that there is a wide range in the pitch of sounds that we can hear. These sounds range from the lowest note the pipe organ can produce, to notes that are higher in frequency than the shrillest note of the piccolo. Pitch is the musician's term for the rate at which the sound vibrations occur. A low-pitched note has fewer vibrations per second than a high-pitched note. In radio work we use the terms cycles per second, or frequency, in referring to the relative number of vibrations, rather than the term pitch.

HARMONICS

Sound waves are seldom pure tones (single sine waves). They are generally complex waves, that is, they consist of several sine waves of different frequencies. In the case of musical tones, the component frequencies are harmonics of one another, and each of the components are continuously produced. This harmony gives us a pleasant sensation while listening to music.

► In the case of noise, the frequencies are unrelated to one another, and are not continuous. One frequency seldom exists for more than a fraction of a cycle. The unrelated and chopped-up frequencies give unpleasant sensations.

How Common Sounds "Look." In Fig 5A is shown a diaphragm, such as is commonly used in telephone transmitters, which has been connected with a lever that will magnify its movement. This diaphragm is made of light, thin material, so that it will faithfully follow the wave form of the sound wave striking it. One end of the lever is connected to the diaphragm by a link, and the other end is attached to a pencil. The pencil traces a mark on a moving strip of paper so that a time-amplitude picture of the movement of the diaphragm can be obtained. As a pressure wave strikes the diaphragm, it moves inward, the pencil moves to the left, and the amplitude of the movement is recorded. As the rarefied wave strikes the diaphragm, it is pulled outward, the pencil moves to the right, and the amplitude of the movement is again recorded.

If the sound waves striking the diaphragm happen to be a single sine-wave frequency, the trace of the pencil will appear as shown in Fig. 5B. If the sound consists of many different but related frequencies (for example, a musical sound), it will appear as shown in Fig. 5C.

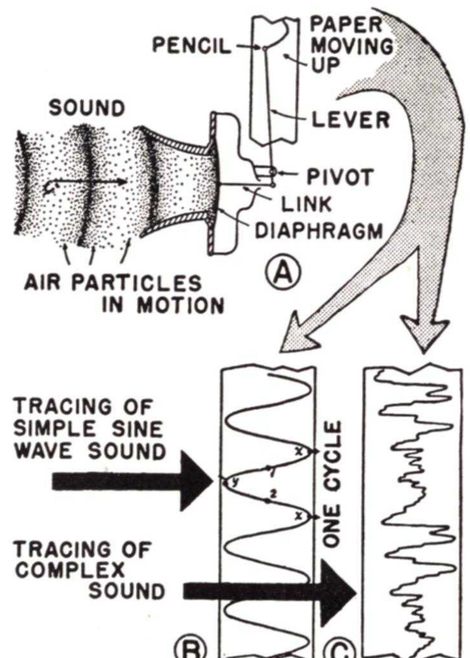


FIG. 5. The simple mechanism at A can be used to trace sound waves, thus proving that sound waves exert a varying pressure on the diaphragm. Typical tracings are shown at B and C.

► The mechanical device, shown in Fig. 5, is not generally used for drawing wave forms. A "cathode-ray oscilloscope" is used in which the electrical output of a microphone causes the wave form of the signal to be traced as a bright line on the face of the fluorescent screen of the cathode-ray tube. All the wave forms that you will see in this Course can be observed on an oscilloscope. Several other familiar sounds are shown in Fig. 6.

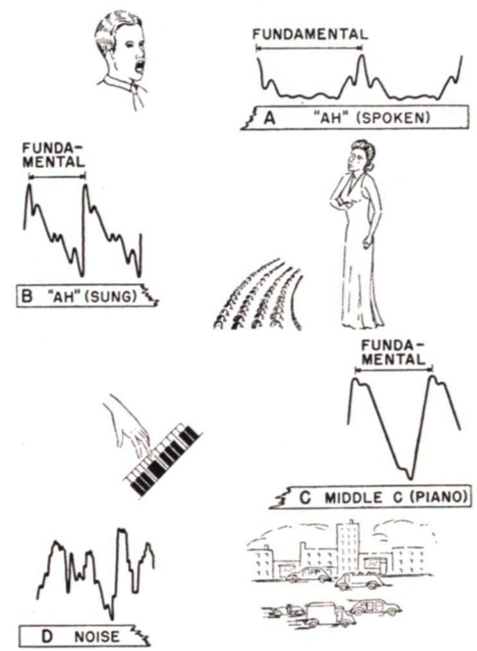


FIG. 6. Sounds can be "seen" as well as heard. By connecting a cathode-ray oscilloscope to a microphone, we can see curves like those traced on the fluorescent screen of the tube.

► 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, the C note above middle C 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 this example is 20% of the amplitude of the fundamental, the third harmonic 25%, the fourth harmonic 10%, and the fifth harmonic about 8%. Another instrument producing the same note may also 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. Harmonics make musical tones pleasing, and their number and amplitude determine the "timbre," or characteristic, that helps us to distinguish between instruments.

Fundamental Frequencies. The piano, organ, and harp produce the greatest ranges of *fundamental* frequencies—about 16 to 4096 cycles. A baritone singer produces fundamental frequencies from 80 to 400 cycles; a piccolo can produce notes with frequencies ranging from 500 to 5000 cycles; a ukulele has the limited fundamental frequency range of 300 to 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.

► Speech differs from music in that

it is less melodic, and it is produced more nearly a monotone. Speech sounds also contain a fundamental pitch that distinguishes between the voices of men, women, and children plus many different, higher frequencies with certain predominating frequencies that give the characteristic distinctions between the voices of different persons.

A Complex Wave Form Broken into Elements. We have said that common musical sounds are more complex in wave form than a sine wave, although they are themselves composed of numerous sine waves. We shall now demonstrate how a complex wave, either sound or video, can be produced by combining harmonic sine waves with a fundamental.

Let us assume that we have the fundamental frequency sine wave shown in Fig. 7A, and that we combine it with the second harmonic shown in Fig. 7B. The wave shape of the combined waves is given in Fig. 7C. If the fifth harmonic, shown in Fig. 7D, is added to the fundamental and second harmonic, the wave form shown in Fig. 7E is obtained. To obtain this combined wave shape we add the amplitudes of these two waves algebraically at every instant. For instance, the amplitude k-e is added to amplitude m-n to produce amplitude o-p in the resulting wave, and s-t is obtained by adding k-e, m-n, and q-r. Notice that the combined wave shape is quite different from any of its components, but it has the same frequency as the fundamental component.

We could continue adding harmonics to the fundamental indefinitely, and each harmonic would change the wave shape as the second and fifth harmonics have done.

By changing the amplitude, frequency, and phase difference of the various components, it is possible to obtain any desired signal wave shape.

It is also true that any periodic (repeating) wave form consists of a number of sine-wave components that differ from each other in frequency, amplitude, and in phase relationship. This is a very important fact for we can determine the response of any radio apparatus to a complex wave input by knowing how it responds to each of the components of the complex wave, and then adding these components together to obtain the output wave form.

► For accurate reproduction of either audio or video signals, amplifiers must be designed to amplify all these components an equal amount, without adding any other components, and without changing the phase relationship of the components. In other words, the frequency distortion, amplitude distortion, and phase distortion must be at a minimum. When we study amplifiers in detail, we will find out what causes these distortions, how they are corrected, and the amount of distortion permissible for different types of radio signals.

HOW THE EAR INFLUENCES COMMUNICATIONS SYSTEMS

Since hearing is essential to audio and code transmission, it is important that we know something about the characteristics of the human ear.

The human ear is by no means an ideal sound-interpreting device. One can train his ears to hear desired sounds, and to disregard others. This is helpful in ignoring undesirable noises, but it also leads one to believe that he hears sounds that are not present. One who is familiar with the timbre produced by 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 to "hear" what is not actually present is tiring,

and soon one does not wish to listen any longer.

► What is more, it is possible for human ears to 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 individuals like this type of reproduction. In fact, some people prefer it to high-fidelity reception. To musicians, however, or to others who appreciate high fidelity, such distortion is very annoying.

By studying the responses of thousands of persons to sounds of various frequencies, the Bell Telephone Company has found that the maximum range of frequencies that can be heard by even the best ears extends from 20 cycles to 20,000 cycles; others claim that this range is more limited, from 32 to 16,000 cycles. These tests also revealed that the human ear is far more sensitive to sound 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.

The frequency range of the average ear determines the range of frequencies that must be reproduced to make the reproduced sounds seem natural. If all frequencies between 20 and 20,000 cycles are reproduced, all of the frequencies distinguishable to the ear are present, and the reproduction sounds very natural. It has been found in practice that a frequency range from 40 to 10,000 cycles is adequate for satisfactory high fidelity reproduction, since the ear is not as sensitive to frequencies from 10,000 to 16,000 cycles, and from 20 to 40 cycles as it is to the range from 40 to 10,000 cycles. Broadcast stations and reproducing systems try to reproduce, as far as possible, all the frequencies from 30 to 15,000 cycles.

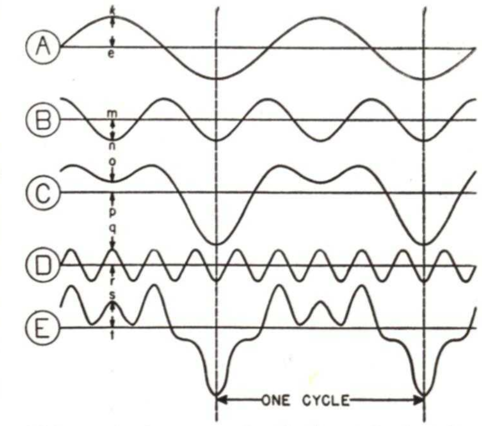


FIG. 7. A sine-wave signal, shown in A, when combined with its second harmonic, shown in B, produces the complex wave shown in C. If, for example, the fifth harmonic shown in D, is also added, the complex wave in E is obtained. In fact, any complex wave can be shown to be composed of sine-wave components by this method. The distance o-p, at any point on curve C, is obtained by adding algebraically k-e and m-n. To obtain s-t on curve E, q-r is added algebraically to m-n, and k-e.

For voice communication or talking circuits, such as aircraft, police, telephone, and taxicab communication services, a reproduction range from 200 to 2500 cycles carries enough of the speech sounds to make the talking understandable, although not natural.

Sounds Can Cause Pain. When sound pressures are increased too high, one stops hearing, and actually begins to feel the sounds. At the low frequencies, the vibration of the air can be felt by all parts of the body, but at the high frequencies, the vibration of the air becomes a sensation of pain in the ear. The response characteristic of the average human ear, obtained by noting at what frequency the thresholds of hearing and of feeling occur, is shown in Fig. 8. A pure sine-wave sound that could be varied in frequency from zero to 20,000 cycles was used in this test to determine the response of the average human ear; 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* bar of r.m.s. sound pressure to make the sound audible, and about 1000 bars (2 r.m.s. pounds per square foot) to make the sound felt. At

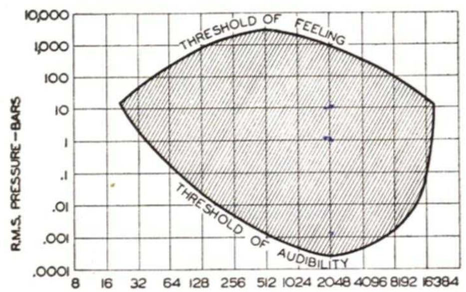


FIG. 8. The response of the average human ear to pure sine-wave sounds of various frequencies is given here. Sound waves having pressure within the shaded area can be heard. The r.m.s. pressure of the sounds is expressed here in "bars"; one bar equals .002 lb. per square ft.

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 shown in Fig. 8.

The Decibel. It has been found that the human ear responds to variations in loudness in an approximately logarithmic manner. This fact enables us to use decibels in measuring sound levels. Let us first study what is meant by a logarithmic scale.

*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 varies 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.

The scale of r.m.s. pressure measured in bars, as shown in Fig. 8, is a good example of a logarithmic scale. Starting at 1, the first interval represents 9 units (one to ten). The next interval represents ninety units (ten to one hundred) and so on.

If we start with a 2000-cycle tone with a pressure of 1 bar, and increase it to ten bars in one step there would be a definite increase in loudness apparent to the ear. The next step in sound pressure that would cause the same increase in loudness would have to be 100 bars. The next step would be to 1000 bars, and so on.

This rate of increase is based on a ratio. The ratio in the examples given was ten to one from one step to the next.

In order for scientists and engineers to have a convenient unit to measure the logarithmic response of the ear, the unit of measurement called the "bel" was invented, and named for Alexander Graham Bell, the inventor of the telephone. Technically, the bel is the logarithm of the ratio of two sound powers or electric powers. For example, the ratio between 10 watts and 100 watts is equal to $\log_{10}\frac{100}{10} = \log_{10}10 =$ 1 bel. In other words, a power ratio of 1 to 10 is equivalent to one bel.

The bel unit proved too large to handle easily, and so another unit, one-tenth the size of the bel was decided upon. Thus the commonly used measuring unit is the decibel, the prefix "deci" meaning one-tenth. A *power* ratio in decibels is defined as

$$\text{db} = 10 \times \log_{10}\frac{P_1}{P_2}.$$

P_1 and P_2 are the two sound powers or electrical powers to be compared.

You will notice that the above relationship refers to power ratios only. It is common in radio work to refer to voltage ratios, especially when calculating or discussing the amplification of amplifiers or the pressures of sounds. When the decibel ratio between two voltages is calculated or referred to, we must modify the decibel equation to take care of the fact that power ratios are proportional to the squares of voltage ratios. (Since $P = \frac{E_1^2}{R}$.) The decibel equals $10 \log \frac{P_1}{P_2} = 10 \log_{10} \frac{E_1^2/R_1}{E_2^2/R_2}$. If the resistance in the two circuits is the same then R_1 and R_2 will cancel out, and this will leave $10 \log_{10}\frac{E_1^2}{E_2^2}$, or

$$2 \times 10 \log_{10}\frac{E_1}{E_2}.$$

Therefore, when we know the *voltage* ratios, the formula for decibels is

$$\text{db} = 20 \log_{10}\frac{E_1}{E_2}.$$

In like manner, when the *currents* are given,

$$\text{db} = 20 \log \frac{I_1}{I_2}.$$

As it happens, and conveniently so, *a change of one decibel in sound power level of a sine-wave signal is the smallest amount of change that can be distinguished by the average human ear. When the signal is a complex wave, however, a 3-db variation in level is necessary before the average human ear can detect a change.*

Gradually increasing the sound *pressure* until it has been doubled (equivalent to doubling the *voltage*) would cause a *6*-decibel increase in sound power, and the average human ear should notice six distinct increases in loudness. However, if the *power* is doubled, only a *3*-decibel increase in sound will result.

Increasing the pressure gradually to 10 times its original value results in 20 sensations of increase in sound or a 20-decibel (abbreviated 20-db) increase. Increasing the pressure 100 times results in a 40-db increase; increasing the pressure 1000 times gives a 60-db increase, and an increase of 2,000,000 times corresponds to 126 db. Thus, 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 6 times as loud, but it means that, as the pressure is gradually raised, there are 6 recognizable increases in the volume. In other words, as the sound is increased gradually, there is a certain point at which it is 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 produces another recognizable volume step. There are 6 of these steps involved when the pressure is doubled.

This fact shows that the human ear is far more sensitive to pressure changes *when the original volume is low.* In other words, doubling the sound pressure from 1 bar to 2 bars results in a 6-decibel increase, and likewise, a doubling of pressure from 100 bars to 200 bars also produces a 6-decibel increase. 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.

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 twice as loud as another, but we can say that one sound is a number of decibel steps louder than another.

POWER LEVELS

Because the ear responds to sound in the way just described, the radio designer must take this factor into account. The amount of power fed to the loudspeaker determines the sound power that the loudspeaker will be able to develop and 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

DB PADS - Inserted in Book 22

DB	Voltage Ratio	Power Ratio
0	1.0	1.0
1	1.12	1.26
2	1.26	1.58
3	1.41	1.99
4	1.58	2.51
5	1.78	3.16
6	1.99	3.98
7	2.24	5.01
8	2.51	6.31
9	2.82	7.94
10	3.16	10.
20	10.0	100.
30	31.6	1,000.
40	100.0	10,000.
50	316.2	100,000

FIG. 9. This table shows how to convert voltage and power ratios to decibels.

how much increase in electrical power output will increase the sound output.

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

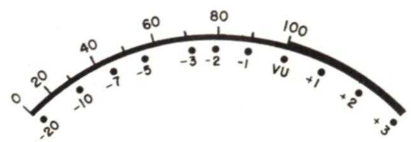


FIG. 10. The standardized scale for VU meters.

power, if the efficiency of the reproducer does not change.

Fig. 9 is a table of db values for various ratios of power and voltage. A power ratio of 5 corresponds to about 7 db; a voltage ratio of 100 corresponds to 40 db, etc.

If the power level is less than the reference value, the output is a *minus decibel* value. Thus, a power ratio of 1/5 is a reduction in amplitude of 7 db, and is written —7 db.

► Remember that *doubling the power is an increase of 3 decibels,* and that *multiplying the power by 10 causes a 10-db increase.* These two facts enable us to change decibels to power ratios, or power ratios to decibels without using tables. For example, suppose that the power input to an amplifier is 5 watts, and the output is 2000 watts. This is a power gain of 400 (2000 divided by 5). Since 400 is $2 \times 2 \times 10 \times 10$, the db gain would be $3 + 3 + 10 + 10$, or 26 decibels.

Doubling the voltage (or current) in a circuit *causes a 6-db gain,* and *multiplying the voltage (or current) by 10 causes a 20-db gain.* Suppose, for example, that the voltage input to an amplifier is 3 volts, and that the output is 240 volts. This is a voltage gain of 80. Since 80 is $2 \times 2 \times 2 \times 10$,

the decibel gain is $6 + 6 + 6 + 20$, that is, 38 decibels.

VOLUME LEVEL MEASUREMENT

In radio circuits it is necessary to maintain the audio signal at a reasonably constant level. This is done to prevent the distortion that would occur if the amplifier or transmitter circuits were overloaded.

A VU (Volume Unit) meter with a scale like the one shown in Fig. 10, is used to measure the level of audio signals. The pointer of this meter moves back and forth over the meter below one milliwatt). One milliwatt is produced when the meter reading is at the zero VU mark (0 dbm). The remainder of the scale is calibrated directly in dbm with respect to this reference level (from —20 dbm to +3 dbm). However, when an audio signal with a complex wave form is applied to a VU meter, the pointer, because of its inertia, will not be able to follow the variations of the audio signal. Because of this lag, the meter reading cannot be referred to as decibels or dbm. Instead, the signal level is referred to in terms of "volume units"

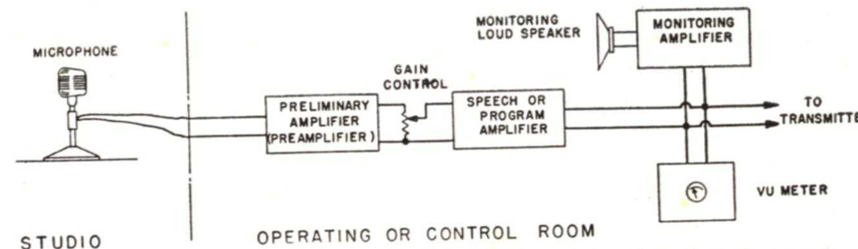


FIG. 11. The audio signal channel from microphone to transmitter. Note that when a microphone having a low output is used, a "pre-amplifier" is inserted between the microphone and the main amplifier for additional amplification.

scale according to the average strength of the audio signals being amplified.

In the amplifier input circuit there is a volume control or "gain" control with which the signal level can be adjusted. When the meter indicates that the signal level is so high that the transmitter is in danger of being overloaded, the operator reduces the level with the gain control. When the meter indicates that the signal level is too low for efficient transmission, he uses the same control to increase the signal level. This operation is called "riding the gain."

A VU meter is a special-made copper oxide rectifier-type voltmeter. When it is connected across a 600-ohm circuit with a steady sine-wave output, the meter readings are in decibels with respect to 1 milliwatt, and are referred to as dbm (read as decibels above or above or below zero level. Thus, for example, when measuring an audio signal on a VU meter, the signal is "+3 VU" instead of "+3 dbm." The term volume unit refers only to audio signal levels, measured with meters which have the standardized dynamic characteristics (inertia) of the VU meter.

Fig. 11 shows a typical input arrangement for the control and monitoring of a broadcast. In addition to adjusting the input to the transmitter by means of the VU meter, the operator also listens to the program through a monitoring loudspeaker which reproduces the program just as the radio listener receives it.

REVIEW OF AUDIO SIGNALS AND THE EAR'S RESPONSE

We have seen that although the ear

can be made, under ideal conditions, to respond to frequencies of from 20 cycles to 20,000 cycles per second, most individuals' hearing is limited to the frequency range of 30 to 8500 cycles per second. However, we learned that in high fidelity radio broadcasting, or in high fidelity sound reproduction, a transmission of frequencies between 30 and 15,000 cycles per second is generally considered necessary. However, for ordinary communication, or voice, with the sole purpose of transmitting intelligible, understandable speech, a frequency range of 200 to 2500 cycles per second is necessary.

We have noted that the ear responds to changes in level in db steps, and we find that it is advisable to make all level measurements in decibels or volume units.

We will see as we continue to study the subject of communications that amplifiers are necessary to raise both the voltage and the power of the audio signal. It is important that the amplifying and transferring equipment introduce as small an amount as possible of the types of distortion recognized by the human ear.

Although we will discuss distortion in detail when we study communication equipment, for the time being, a quick preview of this problem will be beneficial.

Frequency Distortion. If an audio signal reproducer or amplifier amplifies one group of frequencies more than another group, it introduces frequency distortion into the signals.

If, for example, the high audio frequencies are amplified more than the low frequencies, the sound or music will sound "tinny"; if the low frequencies are amplified more than the high frequencies, the response will be "boomy." For high-fidelity reproduction, all audio frequencies should be amplified equally.

Amplitude Distortion. If an audio signal reproducer or amplifier does not accurately reproduce the amplitude of all signals with the same proportions with respect to the original signal, it introduces amplitude distortion into the signals.

This means that the output signal will contain harmonics that were not in the input signal. These harmonics, if they are large enough, will be noticed by the ear, and the sound will be "distorted." Small amounts of amplitude distortion may not be noticed, but listening to distorted music or sound soon becomes tiring.

Phase Distortion. Phase distortion occurs when the phase relationship of the components of a complex wave in the output of an amplifier is different from the phase relationship of the components of the input. This means that the output wave shape will not be the same as the input. The human ear, however, is generally unable to detect phase distortion. The ear will hear the sounds properly even though some of the components are delayed more than others (as they are in phase distortion).

In television, however, phase distortion must be minimized, for it appears as a distortion of the television picture, and is easily noticed by the eye.

Sound Pickups and Reproducers

The invention of the telephone transmitter and receiver* made audio radio and telephone communication possible. They convert the sound waves into electrical signals, and electrical signals back to sound waves. The diaphragm, which we used as an illustration in a preceding section, is the basis of both these instruments. In all types of microphones, the sound causes the diaphragm to move in synchronism with the sound wave. This movement is converted into an alternating electric current, or voltage, which has the same wave *shape* as that of the sound waves. Several different methods are used to convert this movement into electrical energy.

In reproducers, such as headphones and loudspeakers, the reverse process takes place. The alternating current or voltage causes a diaphragm to vibrate, and this vibration produces the sound waves.

► Specialized uses for microphones, loudspeakers, and headsets have caused many different types of these instruments to be developed. The broadcasting and movie fields have required high-fidelity reproduction in microphones and loudspeakers at a sacrifice in efficiency and ruggedness. High-fidelity microphones and reproducers for these applications are delicate and expensive.

Microphones for common telephones, police radio, aircraft radio, and other uses that do not require high-fidelity reproduction are constructed so that they are more rugged, but less faithful in their reproduction. For some applications, microphones are "peaked," that is, made more sensitive, to the frequencies between 200 and 2500 cycles for better intelligibility, for this frequency range includes the voice frequencies, but eliminates extraneous sounds of other frequencies. The same limited frequency response is sometimes used in headsets and loudspeakers for the same reason.

Let us turn now to typical pickups and reproducers, and learn something about the fundamentals of their operation. We will not discuss all their characteristics here, but we will 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. 12. 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 variations.

Carbon Microphone. Fig. 12A gives a simple sketch of the single-button carbon microphone. A very thin aluminum disc, or diaphragm D, (about .001 inch thick) is stretched over a fixed metal ring R. The pressure of the 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 K. As the particles of carbon are squeezed together, more of their surfaces are in contact, so that the resistance between them decreases. When the particles are allowed to separate, they make poorer contact so that the resistance increases. Consequently, the electrical resistance, called "contact" resistance, between disc D and container K varies continually with the

*In telephone practice, the microphone is usually called the transmitter; the earphone, or sound reproducer, is referred to as the receiver.

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 circut so that the variations in current induce a voltage in the secondary of the transformer.

The carbon microphone is universally used in telephone transmitters, and in the microphones for police and aircraft radio communications systems, and similar systems. It is rugged, has good intelligibility, has the highest output of all the microphone types (about .006-watt output), and is inexpensive. Its disadvantages are that it generates a hissing sound, because of slight but continuous changes in the contact resistance between the carbon particles, and it must always have a low voltage local battery for operation.

Carbon microphones are low in internal impedance, and are generally connected to 200- to 500-ohm input circuits.

Condenser Microphone. A simplified cross-sectional view of a condenser microphone is shown in Fig. 12B. The thin aluminum disc, or diaphragm 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 in 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 that has the same wave form as the orginal sound is produced across resistor R.

Condenser microphones are used principally in broadcasting and movie studios to pick up programs. They provide high-fidelity reproduction, but they are very delicate and expensive.

The output from a condenser microphone is very low, and requires 40 to 60 db more amplification following it than do carbon microphones. Because they operate as a variation in the capacity of the condenser plates, they cannot be operated more than a few

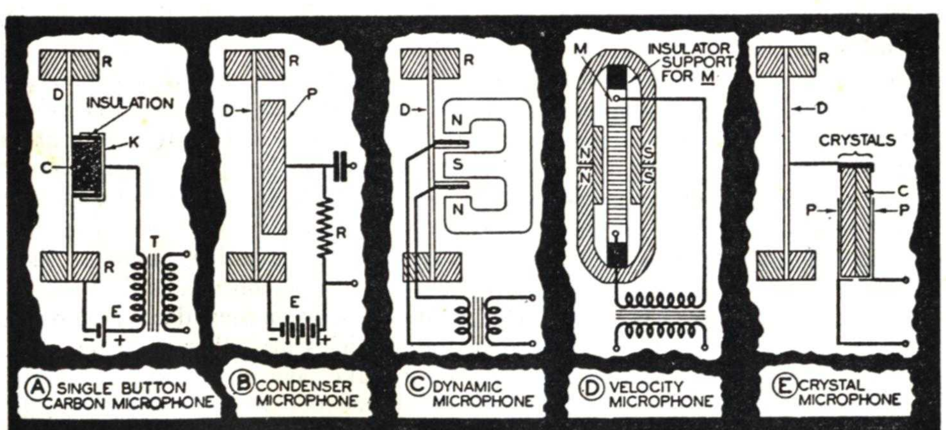


FIG. 12. Simplified diagrams illustrating the operating principles of five different types of microphones. Each converts changes in sound pressure into equivalent changes in electrical signals.

inches from the first stage of their amplifier. The capacity of long wires connecting the condenser microphones to the amplifier would completely blanket the capacity variations caused by the movement of the diaphragm.

The condenser microphone must work into a very high impedance, such as the grid circuit of an amplifier stage. A pre-amplifier is usually found next to the condenser microphone.

Dynamic Microphone. The dynamic microphone, shown in Fig. 12C has a thin diaphragm D on which is mounted a light-weight coil or wire. This coil moves between the poles of a permanent magnet when a sound moves the diaphragm. As a result, there is induced in the coil a varying voltage, the wave form of which is a reproduction of the wave form of the sound.

The dynamic microphone is also used for movie, radio studio pickups, and public address pickups.

The dynamic microphone is the most rugged and versatile of the high-fidelity type of microphones. It is obtainable in both expensive and inexpensive types. The expensive ones are more precise, and have a better frequency response than the inexpensive types.

Dynamic microphones have a low output of about the same order as the condenser microphones, but unlike the condenser microphones they are low impedance devices, and may be used with long leads. They are manufactured to operate into impedances of 50 to 5000 ohms. Because of their low impedances, dynamic microphones can be used some distance away from the amplifier without the need of a matching transformer at the microphone itself. The leads connecting the microphone to the amplifier may be as long as 100 feet before the capacity of the line impairs the high-frequency response of the microphone.

Velocity Microphone. The velocity microphone, illustrated in Fig. 12D, operates on a principle similar to that of the dynamic microphone, except that the sound waves move a thin, crimped metal ribbon M back and forth through the magnetic field of a permanent magnet. Thus the signal voltage is induced in this metal ribbon.

The uses and characteristics of a velocity microphone are similar to those of a dynamic microphone, except that the internal impedance of the velocity microphones is lowest of all microphones. A step-up transformer is used at the velocity microphone for impedance matching to the audio lines connecting it to the amplifier.

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 that exists between these charges is proportional to the strain. This principle is used in the crystal microphone as illustrated in Fig. 12E, where two square crystals are mounted back to back in order 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 the movements of the diaphragm, caused by sound waves, will bend the crystals. Electrons that flow to the flat surfaces of the crystals under this bending are collected by metal collector plates P.

Crystal microphones are inexpensive to make, but they have the disadvantage of instability caused by moisture entering the crystal. They are also affected by temperature. These microphones are used extensively in public address systems and by amateurs.

Crystal microphones have a high impedance, higher than carbon, velocity, and dynamic microphones. Because of this, crystal microphones cannot be connected directly to long audio lines without high-frequency attenuation.

► The audio signal that is generated by a microphone, in terms of voltage, is very small. To obtain enough power to modulate communications transmitters or to drive loudspeakers, vacuum tube voltage and power amplifiers, called "speech or program ampli-

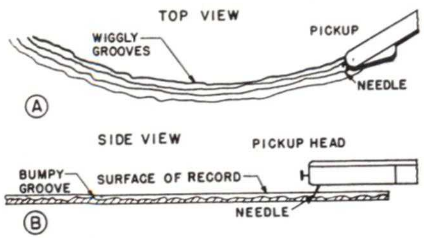


FIG. 13. A lateral-cut record, in A, is cut with a wiggly groove which the needle of the pickup follows. The needle is caused to move from side to side as the record rotates. A vertical-cut record, in B, has a bumpy groove. The bumps and hollows cause the needle to rise and fall as the record rotates.

fiers," are connected between the microphone and the output circuit.

SOUND RECORDING SYSTEMS

Broadcast stations transmit a large number of recorded or transcribed programs, and the radio operator or engineer is frequently assigned the job of making transcriptions, and of operating the record turntables. The sound-recording systems most generally used are disc recordings, tape recordings, wire recordings, and film recordings.

Disc Recordings. Disc recordings are of two main classifications, the "lateral" cut, and the "vertical" cut. The lateral-cut type is the popular phonograph "record" used by the general public. The bottom of the groove of the lateral-cut record is smooth, but the groove itself, instead of running in a true spiral, has minute wiggles in it. These wiggles correspond to the variations in amplitude and frequency of the audio signal, and cause the end of the needle to move from side to side when the record is rotated. This sideward motion is transmitted to the moving element of the pickup, and causes the generation of the audio signal in the form of a varying voltage. Fig. 13A illustrates the principle of the lateral-cut recording.

The vertical-cut records are used commercially for transcribed programs, and wired-music services. The vertical-cut records have grooves that are true spirals with smooth sides, but the bottoms of the grooves are bumpy. See Fig. 13B. The bumps correspond to the audio signal, and cause the pickup needle to move up and down instead of sideways. The up-and-down motion is transmitted to the moving element, and causes the audio signal voltage to be generated.

The vertical, or "hill-and-dale" recordings, have several advantages over the lateral-cut records. They are capable of higher fidelity and wider volume range, because one groove does not cut over into the next groove on passages of high amplitude as sometimes occurs in lateral-cut records. They are also capable of longer playing time because the grooves can be placed closer together.

Vertical-cut records are generally recorded, and reproduced at 33 1/3 r.p.m. instead of at 78 r.p.m., the rate at which popular-type records are recorded. At this slow speed a 15-minute program can be recorded on each side of a 16-inch disc. This type of record is sometimes called a "transcription."

Phonograph Pickups. Pickup heads are of two principal types, the magnetic and the crystal. The magnetic pickup is universally used by radio stations for transcription and record playing. The crystal pickup is used extensively, particularly in inexpensive phonograph pickups.

Fig. 14 illustrates the construction and operation of a magnetic phonograph pickup. This unit is designed for lateral-cut phonograph records, where the groove is always the same depth, but wiggles from side to side in accordance with the audio signal. In following the wavy path of a groove, the phonograph needle moves from side to side. This causes the pivoted iron armature to rock back and forth between the two U-shaped soft iron pieces, one of which is on the inside of each pole of the horseshoe-shaped permanent magnet. As shown in Fig. 14, the sideways

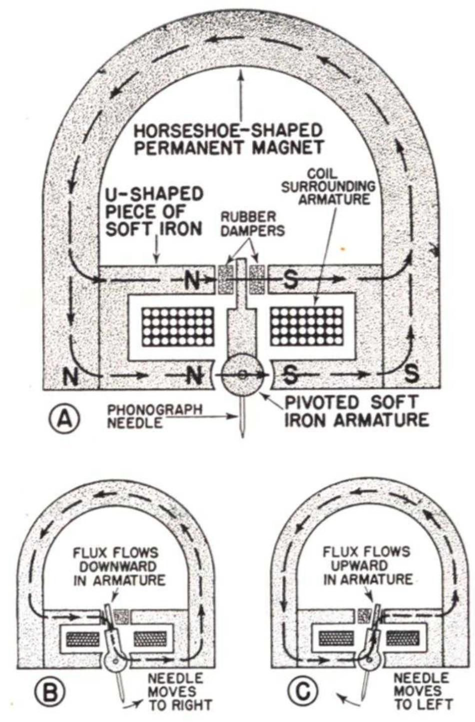


FIG. 14. Cross-sectional views illustrating the basic operating principles of a magnetic phonograph pick-up.

movement of the armature changes the magnetic field, and this causes a voltage corresponding to the movement of the needle to be induced in the pick-up coil.

Rubber dampers prevent the top of the armature from "sticking" in one position, and they also prevent it from vibrating greatly under certain conditions.

A dynamic pickup can be used for both lateral- and vertical-cut records. The needle is pivoted so that it can move both up and down as shown in the simplified view of Fig. 15, as well

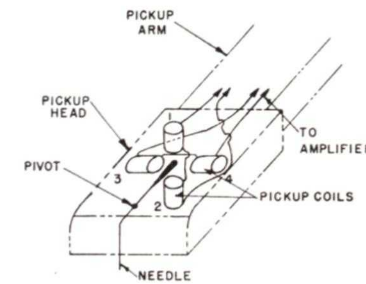


FIG. 15. Simplified view of a dynamic phonograph pickup that can be used for both lateral- and vertical-cut records.

as to the right and left, and two sets of pickup coils are used. When playing a lateral-cut record, the needle moves sideways, and the output voltage is taken from coils 3 and 4; when playing vertical-cut records the needle moves up and down, and the output is taken from coils 1 and 2. A switch is used to connect either the vertical or the lateral reproducing coils to the amplifier.

► A crystal pickup is similar to the crystal microphone shown in Fig. 12E. The only essential difference is that the mechanical motion comes from the needle riding in the groove instead of from the diaphragm D.

► In addition to disc recordings where a needle, or stylus, makes physical contact with the record, there are wire-recording and film-recording systems that use magnetic fields and light beams to reproduce the audio signals.

Wire Recordings. In wire recordings, sound is recorded as a varying magnetization along a fine steel wire.

Fig. 16 shows, basically, how wire recording and reproducing is accomplished. The fine, steel wire is held by the two reels shown in Fig. 16A. The reels are arranged so that the wire may be run from one reel to the other at a constant speed. Between the two reels is located the recording and pickup "shoe" (so called because of its shape) that may be used to record the sound on the wire, or to reproduce the sound after it has been recorded. The soft iron pickup shoe is shown in detail in Fig. 16B. In the recording procedure, the audio signal from an amplifier is connected to the coil around the shoe. The audio signal causes an alternating flux to pass through the shoe pole pieces and the short section of wire between them. The steel wire retains a part of the imparted magnetism in the form of a permanently magnetized section of wire. As the wire passes the shoe, it is magnetized in polarity and amplitude, in accordance with the audio signal.

In the reproducing procedure, the magnetized wire is again passed across the pole pieces. This time, however, it is the wire that supplies the magnetic flux. This flux flows from the wire through the pickup shoe, and back to the wire as shown by the arrows. This flux variation, as the wire moves along, causes an audio voltage to be induced in the pickup coil.

The advantages claimed for wire recording are the elimination of background noise, and a practically unlimited recording time.

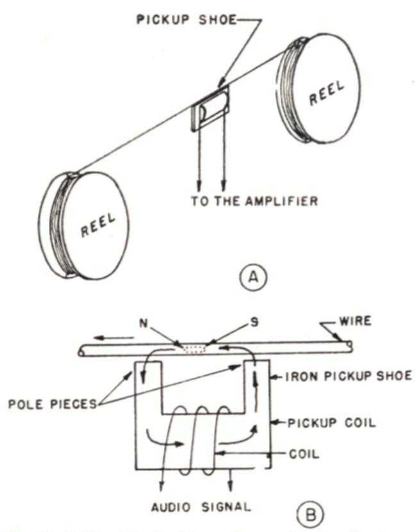


FIG. 16. This illustration shows basically how a wire recorder works. The same machine may be used for recording or reproducing.

Film Recording. In film recording, the audio signals are recorded on photographic film in the same manner as they are recorded in the sound track of sound movies.

In Fig. 17A are shown the two methods of sound-on-film recording. In the variable density method, the variations in amplitude of the sound to be recorded produce variations in the opaqueness of the film, and therefore, consist of a series of narrow lines, varying from almost pure white to almost jet black.

In the variable area method, the amplitude of the sound determines the ratio of the black-to-white portion of the sound track. The sound film then has a hill-and-dale appearance.

The reproduction method of Fig. 17B can be used for each type, for each method varies the amount of light reaching the photoelectric cell. A light aperture restricts the amount of total light passing through the sound track. When the film moves at a constant speed through this light beam, the amount of light reaching the photoelectric cell varies in accordance with the sound track of the film, and the output current of the photoelectric cell is a reproduction of the original recorded sound. The movie industry uses film recording exclusively.

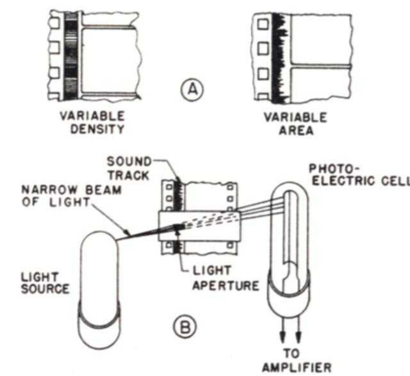


FIG. 17. In A, is a section of film illustrating variable density and variable area sound-on-film recording. In B, the basic reproducing mechanism is shown.

SOUND REPRODUCERS

After we have converted sound into audio signals, and transmitted the audio signals to their reception point, it is necessary to reconvert them into sound. The radio operator, the aircraft pilot, the police officer, and the broadcast listener, must have a device that will convert the electrical signals, varying at an audio rate, back into audible sound waves.

Headphones and loudspeakers are the devices used for sound reproduction. There are a number of different types, but they all work on the principle that the electric signals cause a diaphragm to vibrate, and this vibration re-creates the sound. The process is the reverse of that which takes place in the microphone.

Headphones. In the common headphone unit, illustrated in Fig. 18A, a thin flexible steel diaphragm is placed over the two poles of a horseshoe magnet, and a coil, having many turns of insulated wire, is placed around each leg of the magnet. The audio-signal current, flowing through the two coils, alternately increases and decreases the attraction that the permanent magnet has on the diaphragm, and this causes the diaphragm to move in and out, producing sound waves.

Other types of headphones use Rochelle-salt crystals, and operate in the reverse procedure of a crystal microphone. The audio signal voltages are impressed upon the crystals, and this causes them to vibrate. This vibration is transmitted through a mechanical lever to the headphone diaphragm.

Magnetic Loudspeaker. If a high sound output is desired, the reproducer must be able to displace a large amount of air, and so it must have a large diaphragm, or cone, that can be moved appreciable distances. The balanced armature electromagnetic reproducer, shown in Fig. 18B, has been used widely for this purpose. In this loudspeaker,

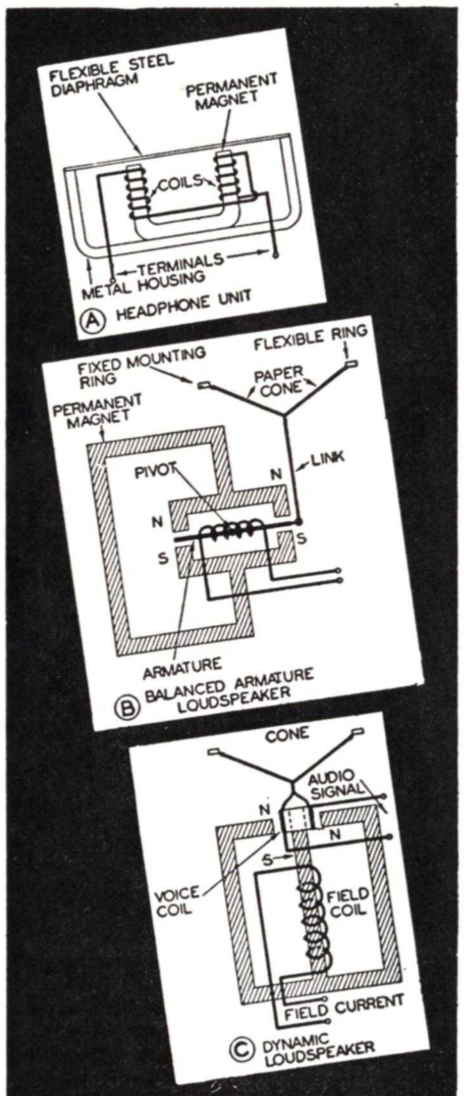


FIG. 18. Simplified diagrams illustrating the operating principles of three different types of sound reproducers. Each converts electrical energy into mechanical energy (motion), which in turn is used to produce sound waves.

a soft steel armature is pivoted between two sets of N and S poles that are parts of a powerful permanent magnet. Surrounding the armature is a solenoid, or coil, that carries the audio current. This coil makes the ends of the armature alternately of opposite polarity. The ends of the armature, therefore, alternately move toward 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, thus setting the surrounding air into vibration, and producing sounds that are reproductions of the original wave forms. 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 freedom of movement of the armature.

Dynamic Loudspeaker. The dynamic loudspeaker, shown in Fig. 18C, is the most widely used sound reproducer, for it is capable of delivering high sound outputs when the cone is used with a baffle.* A coil of wire, called a "voice" coil, is wound on a thin bakelite or paper tube, and attached directly to a paper cone (as shown), or to a metal diaphragm (not shown). The audio signal passes through this coil, and causes a varying magnetic field. This field interacts with a fixed field that is produced either by a permanent magnet, or by an electromagnet. The result is that the coil is forced to move in and out of the fixed field, and drives the cone, thus moving the air ahead of it.

The amount of movement depends on the strength of the voice-coil magnetic

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

field and the strength of the fixed magnetic field. Hence, the voice coil is fed with considerable audio power, and the fixed field is made strong by making the separation between the N and S poles of the magnet as small as possible.

A dynamic loudspeaker can handle from 3 to 200 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 about 50 milliwatts.

The Fundamentals of Television

There are two kinds of visual signals—the television signal and the facsimile signal. Although both transmit visual intelligence, they are different in characteristics and complexity.

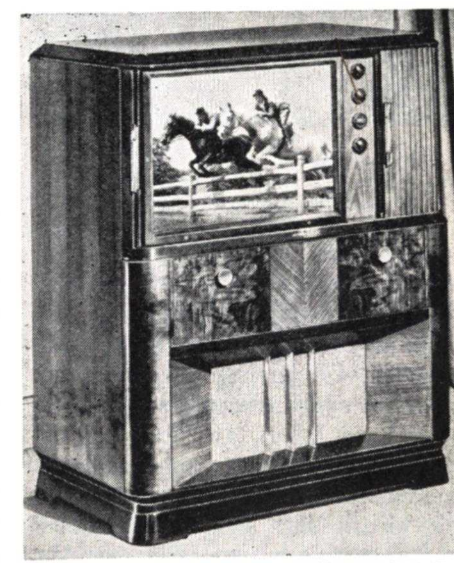
Television is the transmission of "live" or action pictures. Facsimile is the transmission by wire or radio of photographs and other printed matter. It is used by newspapers and press associations for the transmission of news pictures, and by telegraph companies to transmit messages. Radio facsimile can also be used to transmit newspapers, market reports, weather reports, and other vital data to interested subscribers who need a faster delivery of news than the newspapers can provide. Facsimile transmission has the advantage of providing a permanent copy of information, and it does not require an attendant to receive it.

A television system must transmit 30 complete pictures or "frames" per second, while a facsimile system may take 7 minutes to transmit one picture. A television system is more complex than a facsimile system, for it transmits 12,600 pictures while a facsimile system is transmitting one.

Television and facsimile have different methods of reproduction. In television the transmitted picture is viewed by the observer as it is received. In facsimile the received picture is printed by the receiving apparatus before it is viewed.

TELEVISION SIGNALS

The television systems that are in use today do not attempt to pick up a complete scene, and transmit it to a receiver all at one time. Instead, television takes advantage of an eye characteristic known as *persistence of*



Courtesy RCA-Victor

This is an example of a modern television receiver.

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 part of it.

Scanning. To transmit a scene, it

is first broken up into elements by a process that is very like the way you are reading this printed page. You read the first line from left to right, swing quickly back to the left side of the second line, read the second line, go back to the left side of the third line, and repeat the process until you have taken

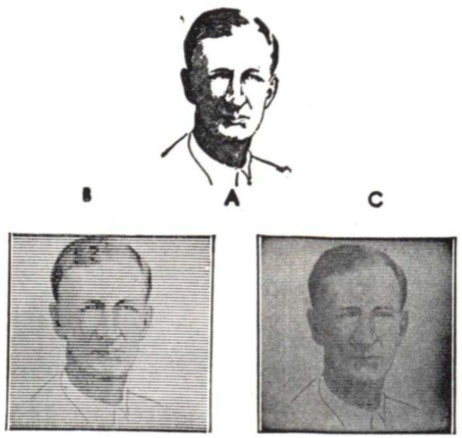


FIG. 19. The drawing at A is reproduced as a series of lines at B and C. Greater detail is obtained by using more lines, as at C.

in every word by itself. This process is what we technically call *scanning*.

The television "camera" works in a similar way. (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 side, moves down slightly, travels horizontally over the scene again, and repeats the process until the whole scene has been scanned. This eye is an electron beam that strikes a light-sensitive surface in the camera (called the photosensitive plate) and converts the light that it has received from the scene into an electric current.

As the electronic eye scans across the picture, light portions of the picture cause an increase in current in the pickup circuit, while dark portions of the picture cause a decrease in the pickup current. It is these variations in the pickup current that constitute the television video signal which is transmitted by the television transmitter to the television receiver. At the receiver, the process is reversed, and the original picture is traced out line by line.

The electronic eye scans the picture so fast that it takes only 1/15,750 of a second to scan across one line. It divides the picture into 525 separate horizontal lines, and scans a complete picture in 1/30 of a second.

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

Fig. 19 illustrates the general effect that is produced when a scene is scanned. Suppose that we wish to televise a picture like the one shown in Fig. 19A. After it has been scanned by the camera, transmitted to the receiver, and reproduced on the receiver screen, it will appear as shown in Figs. 19B or 19C, that is, it will consist of a series of lines that vary in brightness along their length, and so make up the picture. The more lines we have in a given area, the greater will be the detail of the final picture. Fig. 19C has 120 lines, and exhibits more detail than Fig. 19B, which has only 60. Standard television transmitters divide a picture into 525 lines.

► Note that as you move Figs. 19B and 19C away from you, a point is reached at which the details of each illustration 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

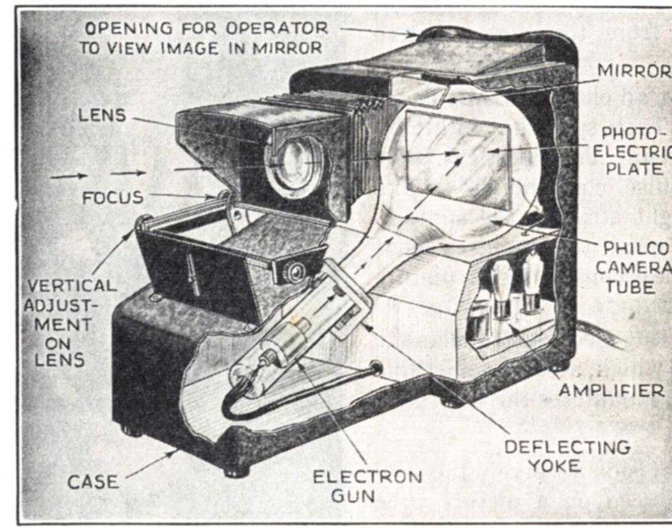
Before considering the technical details of breaking up a scene into a number of lines, it will be valuable to get a clear idea of how a scene is electronically scanned and reproduced.

Electronic Picture Scanning. Fig. 20 shows how an "iconoscope" tube is used in a television camera for electronic scanning. The camera is directed toward the scene to be televised, and the image of the scene is projected, by the lens system, onto the photosensitive plate of the iconoscope. This 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 though it were covered with grains of sand.

When a scene is projected by the lens onto the photosensitive plate in the iconoscope, the light drives out the 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 photosensitive plate elements more or less positively charged (because they have lost electrons).

The amount of electrons lost from any given section of the photosensitive plate depends upon the amount of light reaching that particular section. Thus, some spots on the plate are more positively charged than others, and we have an electronic image of the scene.

An electron gun now shoots a fine stream of electrons at the photosensitive plate. This electron "beam" is adjusted so that it takes 525 of them to make the height of the picture. The electron beam is the eye that scans the image on the photosensitive plate. The beam starts at the top left side of the image, scans across the top, and then returns to the left side again. As it returns to the left, it drops down to the



Courtesy Radio-Craft

FIG. 20. A cut-away sketch showing the arrangement of parts inside one type of electronic television camera.

level of the next picture line so that the second horizontal line may be scanned. Then the beam is moved across the image on the second scanning line and so on.

When the beam has completed the scanning of the picture, it is returned to the upper left corner to begin scanning the complete image again.

When the electron stream strikes a positively charged surface, the positive 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 photosensitive plate.

In this way, an electronic impulse is relayed from each spot that is hit by the electron beam. The electrical value of each impulse corresponds to the amount of light striking the spot, so that the sum of all the impulses (sent one at a time) constitutes a picture signal.

The supporting electrode collects the picture signal which, after considerable amplification, modulates the television transmitter.

► At the television receiver, the picture is reproduced on a picture tube which is a cathode-ray tube, often called a "kinescope" or "picture tube." Here an electron beam moves across the end of the tube in synchronism with the electron beam of the iconoscope, and reproduces the picture on the fluorescent screen of the tube, line by line, and element by element, as it was transmitted.

In the cathode-ray receiving tube, the screen of the tube gives off a fluorescent glow when it is bombarded by an electron stream. The intensity of the glow is proportional to the number of electrons striking the screen. Fig. 21 shows the basic construction of the receiving tube. The electron gun forms the slender electron beam which, under the influence of the deflection coil magnetic field, is moved across and down the picture screen. The strength of the electron beam is varied by the picture signals that are sent from the transmitter, and picked up by the receiver. As the beam travels across the screen, it will become more intense when the light parts of the original picture are being passed over, and less intense when the darker portions of the original picture are being passed over. The varying intensity of the electron beam produces a corresponding variation in the strength of fluorescence on the screen, and the televised picture is reconstructed.

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.

Synchronism. The electron beam of the kinescope and the iconoscope must operate in synchronism to reproduce accurately the televised picture

In order to synchronize the sweep circuits of the kinescope to the sweep circuits of the iconoscope, additional signals, called synchronizing pulses are added to the television picture signals. These pulses perform the following functions in the television receiver:

1. They start the horizontal sweep at the proper moment (horizontal synchronization).
2. They start the vertical sweep at the proper moment (vertical synchronization).
3. They cause the electron beam to be extinguished during its return trip from the right-hand side of the picture to the left-hand side (horizontal blanking).
4. They cause the electron beam to be extinguished during its trip from the bottom of the picture back to the top after each picture has been completed (vertical blanking).
5. They cause an equalization of timing between the two fields of a single frame for interlaced scanning which we will shortly study, (equalizing pulses).

Fig. 22 shows the wave form of the composite television signal that contains the picture, plus the synchronizing pulses. This figure shows the picture signals for the last few lines of a frame, or picture, then the synchronizing signals for starting the scanning of the next frame, and a few lines of the picture signal for the next frame.

VIDEO FREQUENCIES

The frequencies that constitute the picture signal are called "video" frequencies. The word *video* is from the Latin verb *videre* meaning "to see." To amplify the picture signal, a very special kind of amplifier is required. It must amplify a wide range of frequencies (from at least 10 cycles to 4 megacycles), and this amplification must be accomplished not only with a minimum amount of amplitude and frequency distortion, but also with a minimum amount of phase shift (phase distortion) throughout the band of frequencies to be amplified.

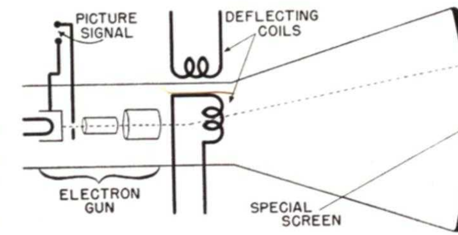


FIG. 21. A simplified diagram of a television cathode-ray tube or kinescope used in television receivers.

A picture with the greatest possible detail is desirable, for familiarity with

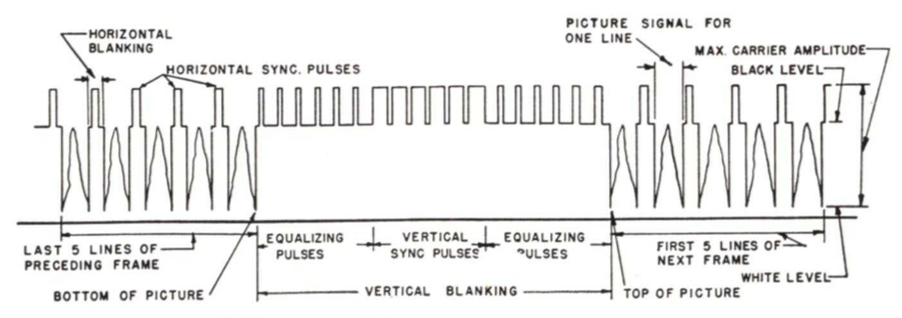


FIG. 22. The complete television signal including all synchronizing, or "sync," pulses. The illustration shows the signal for the last 5 lines of a field, the transition from the bottom of the picture back to the top, and the first 5 lines of the next field.

the details of photographs and movies, naturally makes us expect television to be equally good.

► Before we consider why the wide range of frequencies given above is necessary for high picture definition, let us consider the problem of *flicker*. To give the sensation of motion, the scene must be scanned over and over again, one scanning rapidly replacing the other. Each scanning is called a "frame," and when the number of frames per second is more than about 20, the eye does not see them separately, for they are blended into a continuous picture. If the frame rate is lower than this, a flickering of the image becomes noticeable. The television frame frequency has been standardized at 30 per second.

► 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 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 that can be handled.

Picture Elements. The maximum frequency of the picture signal current is of interest to every television engineer and operator, for all television equipment must be designed to handle this frequency. In order 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. These squares are called "picture elements," because each one of them is the smallest part, or element, of a scene that the camera can "see."

The signal current goes through one cycle, that is, it reaches a maximum and a minimum value, each time the electron-scanning beam passes over one light and one dark picture element For our checker-board scene, the picture signal current, as shown in Fig. 23, 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, we 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 to see how high this maximum frequency can be.

If each picture element is as high as it is wide, it is easy to compute the number of elements in one complete picture. Let us assume that we have a square picture with N lines, then there will be N picture elements per line, or N x N picture elements in the complete square picture, 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 x 525 or 276,625 picture elements. For ordinary calculations, 276,000 elements will be sufficiently accurate.

Aspect Ratio. The pictures commonly involved in television are not square; 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 frame has been increased by the aspect ratio, which we designate as a. The number of picture elements per frame, or picture, will be N x N x a. For the example just considered, the total number of elements will, therefore, be 276,000 x 4/3, that is 368,000.

Frame Frequency. The number of pictures sent per second is the *frame* (or picture) frequency, which we designate 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 x N x a x 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. Since the standard frame frequency is 30, the maximum frequency is obtained by multiplying 368,000 by 30, and then dividing by 2; the result is 5,520,000 cycles per second.*

Factors Affecting Frequency. In television practice, the elements of a horizontal line are scanned in about 85% of the time; the remainder of the time is used for sending the line synchronizing impulse. This increases our maximum picture frequency because we must crowd our picture elements into 85% of a second. We must, therefore, multiply our computed value by 1.17, that is, $\frac{100}{85}$, 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, one dark and the next light. Of course, this is not true in actual practice, for no scene is made up of perfectly arranged checker-board 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.

This is quite fortunate, for it reduces the maximum frequency required. Tests and experience have shown that an apparatus capable of sending about 60% of the maximum frequency is fairly satisfactory. Thus we obtain 3.9 megacycles (6,458,400 x .6) 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 in a television band, gives a definite improvement in picture fidelity.

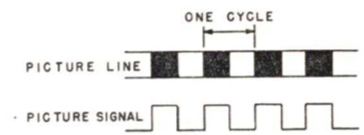


FIG. 23. Two picture elements, one black and one white, make a cycle.

The Minimum Frequency. The upper part of the average outdoor scene (usually the sky) is bright, and the lower part is considerably darker. In scanning such a scene, the picture elements vary 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 consider 10 cycles as the minimum frequency for a practical high-fidelity television system.

Compare this television frequency range of 10 cycles to 3.9 megacycles with the frequency range of high fidel-

*The final formula is: Maximum theoretical picture frequency $f_p = 1/2\ N \times N \times a \times F$.

ity sound, which extends from about 30 to 15,000 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, if the frame frequency of a standard television signal is increased to 60 frames per second, the maximum frequency becomes about 7.8 megacycles, which is beyond the present ability of television apparatus.

To obtain an *effective* frame frequency of 60, and thus reduce the flicker, and yet not increase the band required, a scanning procedure known as interlaced scanning is used. Interlaced scanning (illustrated in Fig. 24) is the process of scanning only every other line the first time down the picture, and then filling in the missed lines on the second time down the picture. For instance, on the first vertical scan the solid lines 1, 3, 5, etc. are scanned. The next vertical scan catches lines 2, 4, 6, etc.

Since only one-half of the lines of the picture are sent on each of these scannings, the number of scannings per second is *doubled* without changing the maximum signal frequency. Each scanning is 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.

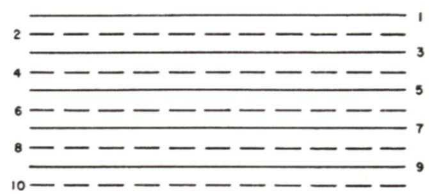


FIG. 24. Example of interlaced scanning. In one field, lines 1, 3, 5, 7, and 9 are scanned, then in the next field, the lines 2, 4, 6, 8, and 10 are scanned. The two fields form one frame of 10 lines.

By using an odd number of scanning lines per picture, the picture automatically interlaces, that is, the lines of one field will be between the lines of the other field of the frame. (How this automatic interlacing occurs, and why it simplifies television systems will be discussed in detail in the Television Lessons.) That is why 525 lines is the present standard, rather than 500 lines or 520 lines, or any *even* number.

► Just as the operators of broadcast transmitters monitor and control the input to the transmitter, so also must television operators monitor the television signal and the transmitted picture. It is impossible to use a simple meter to adjust the level of television signals as is done with audio signals, so cathode-ray oscilloscopes are used to adjust the level, and a kinescope is used in the operating room to reproduce the picture program as the television audience will receive it.

SUMMARY

We must remember these things about the television signal:

1. It is composed of the picture signal plus the synchronizing signals.
2. The picture signal band width for high-fidelity picture reproduction is from 10 cycles to about 4 megacycles.
3. The frequency range of the television signal is directly related to the number of picture elements desired.

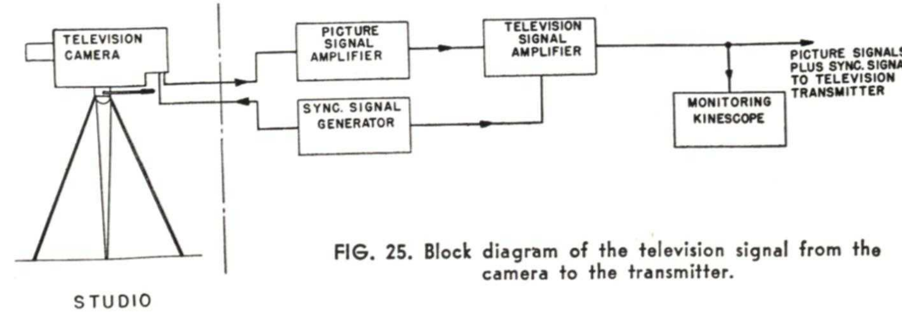


FIG. 25. Block diagram of the television signal from the camera to the transmitter.

A block diagram of the television signal from camera to transmitter is illustrated in Fig. 25. This shows how a common synchronizing signal generator controls the iconoscope of the transmitter as well as the kinescope at the receiver.

Facsimile Transmission

A facsimile system, like a television system, scans the image and simultaneously reproduces a picture at the receiver. Here the similarities end. Both are widely different in the speed of scanning, the method of transmission, and the transmission frequency band required. Facsimile requires a long time to reproduce one picture, generally seven minutes, but sometimes it takes as long as twenty minutes. On the other hand, facsimile requires only a very narrow band of frequencies for transmission.

There are several methods of facsimile transmission. In the system used by *Wirephoto,* the facsimile photograph-transmission service used by newspapers and periodicals, shown basically in Fig. 26, the standard picture size is 5 x 7 inches. The photograph (Fig. 26A) is wrapped around the drum, 2 inches in diameter, with its 7-inch side in the direction of the axis of the drum.

The drum and picture rotate at 100 revolutions per minute. (More recent developments use higher drum speeds, and thereby reduce the transmission time.)

A small beam of light, supplied by a lamp, scans the picture as it rotates. The beam of light moves along the length of the drum as the drum rotates. The size of the light beam is 1/100 inch square, and its lateral movement is 1/100 of an inch for each revolution of the drum. Therefore, it moves along the drum, scanning the picture at the rate of one inch a minute. When the picture is 7 inches long, the total scanning time will be 7 minutes.

Light reflected from the photograph is directed into a photoelectric cell. When the beam passes over a dark area of the photograph, little light is reflected into the photocell, and its response is low. When the beam passes over a light area, considerable light is reflected into the photocell, and its response is high. The output of the photocell consists of a pulsating d.c. current that corresponds in intensity to the shading of the picture. Inserted in the path of the light beam to the photocell is an oscillating mirror, not shown in the figure, that oscillates at 1800 cycles per second. This breaks up the output of the phototube into little pulses of varying amplitude that constitute the facsimile signal.

The facsimile signal is then amplified, and placed on telephone lines for transmission to the receiving point. If the picture is to be transmitted by radio, it is used to modulate a carrier.

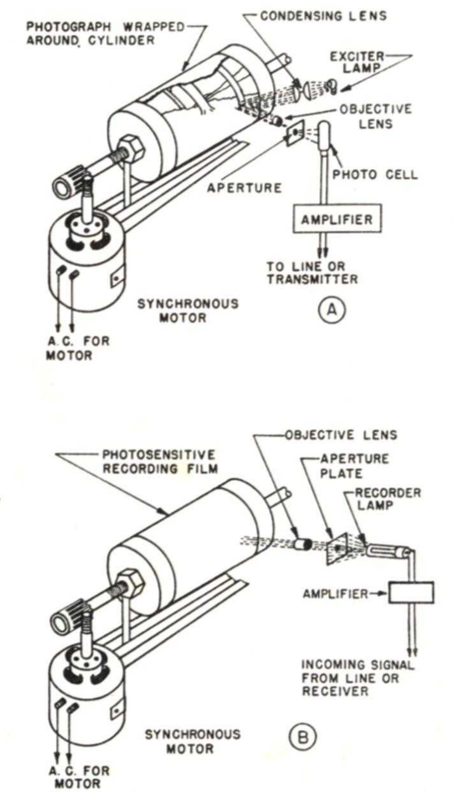


FIG. 26. The essential components of facsimile transmitters and receivers.

The receiving equipment in this system of facsimile is shown in Fig. 26B. A piece of photographic film of the proper size is placed on the rotating drum, similar to the one used in the transmitter. The received facsimile signals control the swing of the oscillating mirror so that it acts as a light valve, and regulates the intensity of the light beam which exposes the photographic film. In this system the photograph must be developed before the received image is visible. However, in some facsimile systems, the image is printed directly on the paper, and is visible immediately.

Several methods of synchronizing the receiving drum to the transmitting drum are used in various facsimile systems. Pictures transmitted by this process cannot be distinguished from the original picture except by experts.

FACSIMILE FREQUENCY BAND

The maximum and minimum frequencies of the facsimile signal are calculated in the same way the frequencies for television are calculated. Let us assume that we have a checkerboard arrangement of black and white squares of the same size as the light beam, and let us calculate the frequency of the signal generated.

We calculate the maximum frequency of the facsimile signal with a 2-inch diameter drum, rotating 100 revolutions per minute, and black and white spots 1/100 inch square. The circumference of the cylinder is 2 x 3.14 or 6.28 inches. If the dots are 1/100 inch long, there can be 628 dots on the photograph wrapped around the drum. When the dots alternate white and black, the frequency is 314 per revolution. Since the drum turns at a rate of 100 revolutions per minute, the number of cycles per *minute* is 31,400. This is approximately 530 cycles per second. The minimum frequency is almost zero cycles.

Since the photoelectric cell output is divided into pulses at the rate of 1800 per second by the oscillating mirror, the 1800-cycle signal "beats" with the picture signal to produce a band of frequencies equal to the 1800 cycles plus and minus 530 cycles, that is, from 1270 to 2330 cycles. This band of frequencies is narrower than one required for speech transmission, so that facsimile may be transmitted over telephone lines or telegraph circuits that have comparatively poor frequency transmission characteristics.

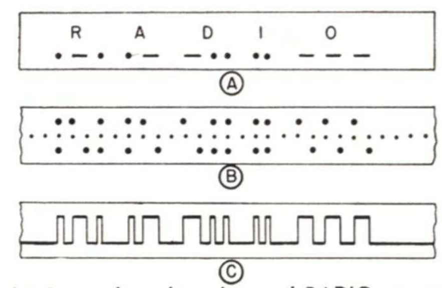
The Code Signal

Code signals are generated by interrupting the constant frequency carrier of a radio transmitter to form the dots and dashes of the Morse code. At the receiving station the code signals are converted back into the original signal.

minute. Since one usually talks at a rate of 100 words per minute, it is possible to transmit, in code, several times faster than a person can talk.

► In hand-keying a transmitter, the telegraph key controls, directly or through relays, the operation of one of the transmitter stages. In automatic code transmission, the message is transferred to a paper tape which is perforated with the characters to be sent. As the tape passes through a pickup device, the perforations close the transmitter circuit in the same way a hand key allows contact to be made through the perforations.

In receiving automatic high-speed code signals, the received signals are recorded in ink on a continuous piece of narrow tape by an "ink recorder," and they are then transcribed by the receiving operator.



This figure shows how the word RADIO appears in code signals. In A, we see it in dots and dashes, in B, it is punched on a tape by a Wheatstone Perforator for automatic code transmission, and in C we see the ink-recorded signal which the receiving operator reads.

Transmission and Reception of Code Signals. Coding and decoding the dot-dash signals may be accomplished manually or automatically. Commercial radio message services use automatic transmission systems in order to speed up the handling of traffic and to reduce errors. Amateurs, and some commercial services such as ship stations use manual transmission and reception.

Hand-keyed code signals are usually transmitted at a maximum speed of about 50 words per minute, but automatic code transmission systems operate at rates as high as 500 words per

Code Spacing. A dot is formed by closing the key momentarily, a dash is three times as long as a dot. The spacing between the dots and dashes which form a letter is equal to one dot. The spacing between two letters is equal to three dots, and the spacing between two words is equal to five dots.

Sometimes a dot cycle, which is the time for a dot plus the space following it, is used in referring to the proportions of code signals.

Frequency Band. The frequency band width required for code communication at various speeds can be calculated when we know that the average word or character group consists of five characters, and the average length of each character is five dot cycles. Therefore, for each word or character group there are 25 dot cycles. At keying speeds of 100 words per minute the fundamental dot frequency is 2500 cycles per minute or 41 cycles per second.

For proper dot formation, that is, a rapid rise to the full output level of the transmitter, at least the third harmonic of the dot frequency must be transmitted so that a frequency range of 123 cycles must be considered. At keying speeds of 500 words per minute, a range of 615 cycles is required. At 25 words per minute this frequency range is only 31 cycles per second.

From the standpoint of the efficiency of transmission, high speed code signals allow faster transmission of a given amount of intelligence with a narrower band of frequencies than audio signals.

Band of Frequencies Required for Various Services

Voice (standard broadcast)	30—15,000 c.p.s.
Television	10—4,500,000 c.p.s.
Facsimile	1270—2330 c.p.s.
Manual code (50 w.p.m.)	0—62 c.p.s.

A brief summary of the band of frequencies necessary for the various types of radio signals.

Lesson Questions

Be sure to number your Answer Sheet 13RC.

Place your Student Number on *every* Answer Sheet.

Most students want to know their grade as soon as possible, so they mail their set of answers immediately. Others, knowing they will finish the next Lesson within a few days, send in two sets of answers at a time. Either practice is acceptable to us. However, don't hold your answers too long; you may lose them. Don't hold answers to send in more than two sets at a time or you may run out of Lessons before new ones can reach you.

1. What are the two important characteristics of a sound that has the form of a simple sine wave?
2. What three things can happen to sound waves when they strike the walls of a studio?
3. How can the reverberation time of a room be reduced?
4. How much change in db of a complex signal is necessary for the average human ear to recognize a noticeable change?
5. Why are single-button carbon microphones used in aircraft service?
6. To connect an amplifier through a 100-ft. cable without a matching transformer at the microphone, and still get fair high-frequency response, would you use: 1, a dynamic microphone; or 2, a crystal microphone?
7. Of the carbon, velocity, crystal, and dynamic microphones, which has the highest impedance? Which has the lowest impedance?
8. In television communications, how many picture elements make a cycle?
9. Why is an odd number of lines used in television?
10. Of the television, facsimile, sound broadcast, and manual code services, which produces the widest band of frequencies? Which produces the narrowest band of frequencies?

FIRST IMPRESSIONS

First impressions mean a lot in this busy world. An applicant for a job has a pretty tough time making the grade if his appearance and first few words do not make a favorable impression on the employment manager. A salesman likewise gets the "cold shoulder" if there is anything about him that annoys the prospect.

With technical material of any kind, however, first impressions can be very treacherous. Oftentimes a simple technical book will contain a number of apparently complicated diagrams, charts, graphs, sketches or tables. Since we glance mostly at illustrations when inspecting a book, we are likely to get a misleading impression. Paragraphs, pages, or entire lessons may also seem difficult during the first reading, but become almost magically clear during the second or third reading.

If first impressions of a required task are favorable, fine and dandy; if unfavorable, don't be discouraged, but wade right into the work, and give it a chance to prove that *first impressions don't always count.*

J. E. SMITH.