SUBCOURSE
SS0524

EDITION
5

AUDIO PRODUCTION PRINCIPLES

THE ARMY INSTITUTE FOR PROFESSIONAL DEVELOPMENT

ARMY CORRESPONDENCE COURSE PROGRAM

AIPD

READINESS / PROFESSIONALISM THRU GROWTH

US ARMY AUDIO/TV PRODUCTION SPECIALIST
MOS 84F, Skill Level 1

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Audio Production Principles

SUBCOURSE NO. SS0524-5

US Army Signal Center
Fort Gordon, Georgia

Four Credit Hours

General

The Audio Production Principles subcourse part of the Audio/TV Production Specialist, MOS 84F Skill Level 1 Course, is designed to teach the knowledge necessary for performing tasks related to audio production. Information is provided on several tasks which are performed at increasing levels of difficulty at skill levels 1, 2, and 3. The subcourse is presented in four lessons, each lesson corresponding to a terminal objective as indicated below.

Lesson 1: DEFINE THEORY OF SOUND

TASK: Describe the theory and terminology of sound, sound measurement, and characteristics.

CONDITIONS: Given information about terms relating to audio production and information and diagrams describing the theory of sound.

STANDARDS: Demonstrate competency of the task skills and knowledges by responding to the multiple-choice test covering theory and terminology of sound and sound measurement.

(This objective supports STP Tasks listed at the end of this section).

Lesson 2: OPERATE AUDIO CONSOLES AND MICROPHONES

TASK: Describe the methods for operation of audio consoles and five types of microphones.

CONDITIONS: Given information on the function of audio mixers, characteristics of microphones, the five types of microphones and their use, and function of a patch panel.

STANDARDS: Demonstrate competency of the task skill and knowledge by responding to the multiple-choice test covering methods of operation of audio consoles and five types of microphones.

(This objective supports STP tasks listed at the end of this section).

Lesson 3: OPERATE TURNTABLES AND AUDIOTAPE RECORDER/REPRODUCERS

TASK: Describe the methods for operation of turntables and audiotape recorder/reproducers.

CONDITIONS: Given information about recorded sound, turntables, and audio recorders, (reel to reel, audio tape cartridges, and cassette units).

STANDARDS: Demonstrate competency of the task skills and knowledge by responding to the multiple-choice test covering operation of turntables and audiotape recorder/reproducers.

(This objective supports STP tasks listed at the end of this section).

Lesson 4: EDIT, SPLICE AND DUPLICATE AUDIOTAPE

TASK: Describe the characteristics, editing and duplication of audiotape.

CONDITIONS: Given information on the characteristics of audiotape and procedures of editing and duplicating audiotape, the limits of hearing and the storage and handling of records and tape.

STANDARDS: Demonstrate competency of the task skills and knowledge by responding to the multiple-choice test covering methods of editing and duplicating audiotape.

(The objectives for this subcourse support STP tasks:)

113-577-4001	Operate Turntable
113-577-4002	Operate Audiotape Cassette Recorder/Reproducer
113-577-4003	Operate Audiotape Tape Cartridge Recorder/Reproducer
113-577-4004	Operate Audiotape Reel-to-Reel Recorder/Reproducer
113-577-4006	Operate Studio Audio Console
113-577-5003	Edit Audiotape Manually
113-577-5004	Edit Audiotape Electronically
113-577-9008	Perform Operator's Maintenance of an Audio Recorder/Reproducer
113-577-9009	Perform Operator's Maintenance of a Turntable

TABLE OF CONTENTS

Whenever pronouns or other references denoting gender appear in this document, they are written to refer to either male or female unless otherwise indicated.

*** IMPORTANT NOTICE ***

THE PASSING SCORE FOR ALL ACCP MATERIAL IS NOW 70%.

PLEASE DISREGARD ALL REFERENCES TO THE 75% REQUIREMENT.

INTRODUCTION TO AUDIO PRODUCTION PRINCIPLES

An Audiovisual Production Specialist records sound, selects and places microphones, and operates an audio mixing console. This SSO discusses sound and some of its more important characteristics: microphones and their operation; audio consoles; operation of turntables and audiotape recorders/reproducers; and editing, splicing, and duplicating audiotape.

LESSON 1
DEFINE THEORY OF SOUND

TASK

Describe the theory and terminology of sound, sound measurement, and characteristics.

CONDITIONS

Given information about terms relating to audio production and information and diagrams describing the theory of sound.

STANDARDS

Demonstrate competency of the task skills and knowledge by responding to the multiple-choice test covering theory and terminology of sound, sound measurement, and terminology.

REFERENCES

None

Learning Event 1:
DESCRIBE THE THEORY OF SOUND

1. All liquids, solids and gases consist of very tiny bits of matter called molecules. Normally, these molecules remain in one area until they are disturbed. As these molecules collide, they transfer energy to each other through the medium (liquid, solid, or gas) which they make up.

2. The molecules continue to collide with each other and transfer energy until they make contact with the eardrum. The ear converts these vibrations into electrical signals, or pulses, which the brain processes. This conversion from eardrum vibrations to electrical pulses for the brain produces sound.

3. Molecules in any medium tend to stay in one spot and try to stay about the same distance apart from each other (fig 1-1).

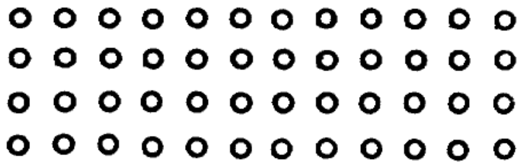

Figure 1-1. Undisturbed molecules

a. Any energy (disturbance) introduced into the medium causes the molecules closest to the disturbance (energy) to be pushed together. This we call compression (fig 1-2).

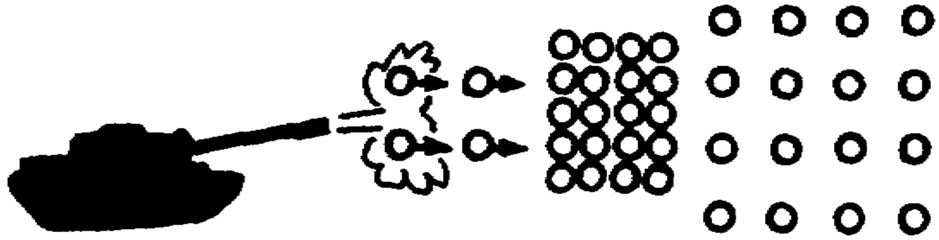

Figure 1-2. Compression

b. Because they always try to stay about the same distance from each other, the compressed molecules now attempt to return to their original state. Here they expand their distance between each other to what it was before the disturbance. This expansion we call rarefaction (fig 1-3).

Figure 1-3. Rarefaction

c. As the compressed molecules return to their original state, the molecules right next to them become compressed and then they expand. This occurs down the line in the medium until the collisions of the molecules from the compressions and rarefactions reach the eardrum (fig 1-4).

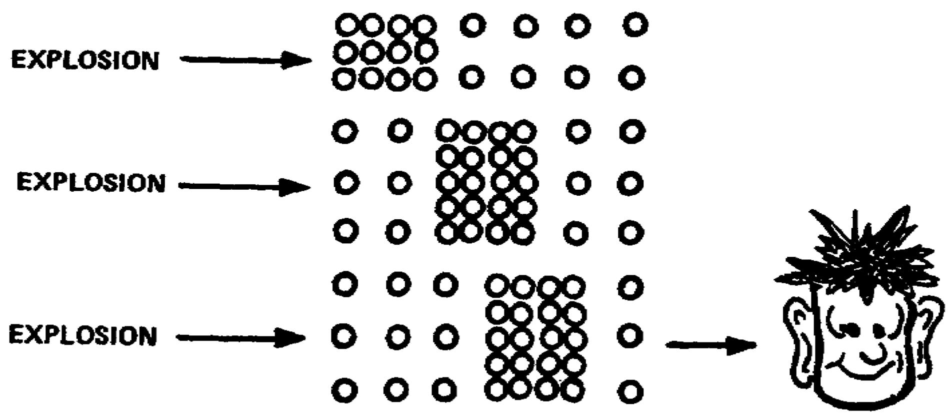

Figure 1-4. Rarefactions and compressions

d. The compressions and rarefactions occur quickly and simultaneously to produce a situation in which the colliding molecules react like a wave motion. That is, the energy transfer resulting from colliding molecules acts like a wave motion which can be bent, reflected, broken up, etc., just like light.

e. Because of this, the term "sound waves" is used to describe the complex combination process of compression and rarefaction between colliding molecules in a medium (fig 1-5).

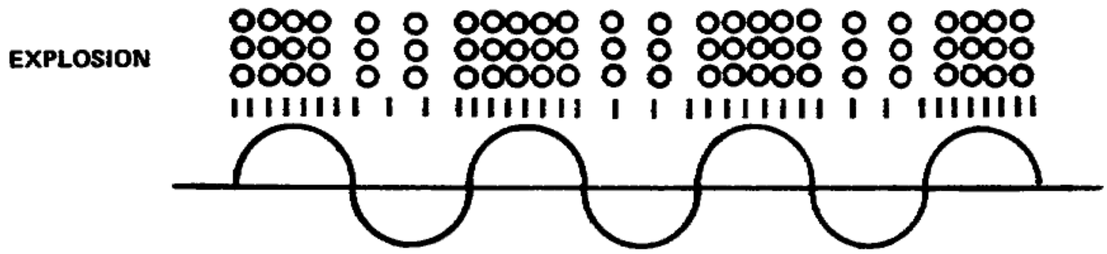

Figure 1-5. Rarefactions and compressions combine into a "sound wave"

4. Makeup of sound waves: Sometimes it may be necessary to see what a soundwave (compressions and rarefactions in a medium) looks like. To do this, an electronic instrument called an oscilloscope is used.

a. The oscilloscope provides an electronic translation of what the sound wave looks like. Should the sound be a pure tone, (a single frequency), then the oscilloscope will show an even pattern (fig 1-6).

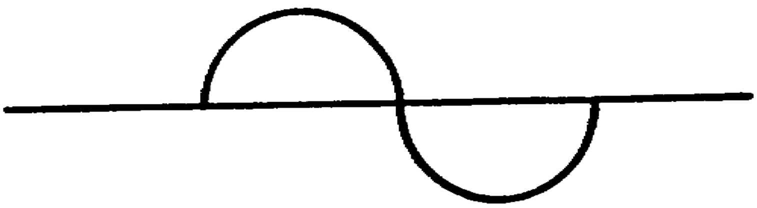

Figure 1-6. Sine wave

b. If the waveform consists of two or more single frequencies, or pure tones, then it appears as a complex wave (fig 1-7).

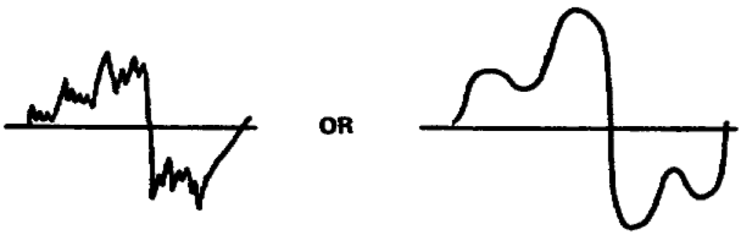

Figure 1-7. Complex wave

5. The oscilloscope shows electronic translations of the compressions and rarefactions of sound waves, and the various parts of these waves can be examined by using the sine wave (a representation of a pure tone) as an example. The following information will explain the characteristics of a sound wave which also constitutes the parts of a sine wave.

Learning Event 2:
MEASUREMENT CHARACTERISTICS OF SOUND WAVES

1. Definitions of Sound Wave Characteristics:

a. Frequency denotes the first characteristic of sound and a part of a sine wave. The frequency of a wave equals the complete number of cycles occuring in one second of time. One complete cycle starts at one point on a wave and traces to the next same point further on down, or from 0 degrees to 360 degrees (fig 1-8).

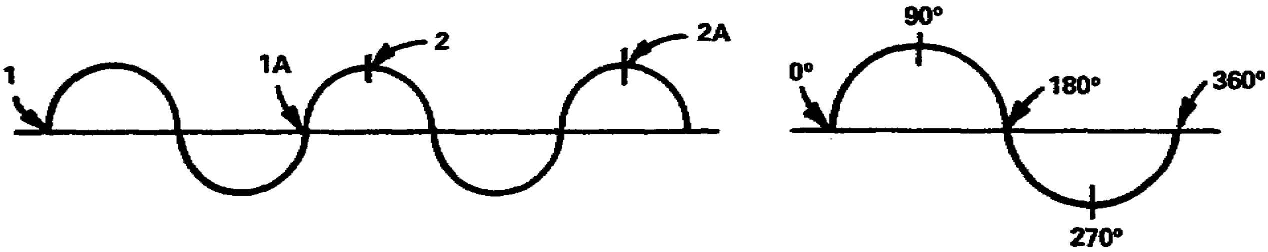

Figure 1-8. Sine waves

(1) For example, from 1 to 1a equals one complete cycle or from 2 to 2a equals one complete cycle. The frequency of a wave equals the number of complete cycles per second (CPS) or Hertz (Hz). Scientists derived the term "Hertz" in honor of a German scientist who worked in electromagnetic theory in the 18th century. When talking about the frequency of a wave, we use the terms Hertz (Hz) and cycles per second (cps) interchangeably.

(2) The audio spectrum (or the range of frequencies humans can hear) consists of frequencies from 15 Hz to 20,000 Hz.

b. Wavelength denotes the second characteristic of sound and another part of a sine wave. All sound waves, regardless of their length, go from 0 degrees to 360 degrees in one complete cycle. The amount of distance from 0 degrees to 360 degrees when plotted as a sine wave determines the length of the wave or wavelength. Wavelength depends on the frequency. The higher the frequency, the greater the number of cycles per second. The greater number of cycles per second requires a shorter distance from 0 degrees to 360 degrees for one complete cycle. This in turn produces a shorter wavelength (fig 1-9).

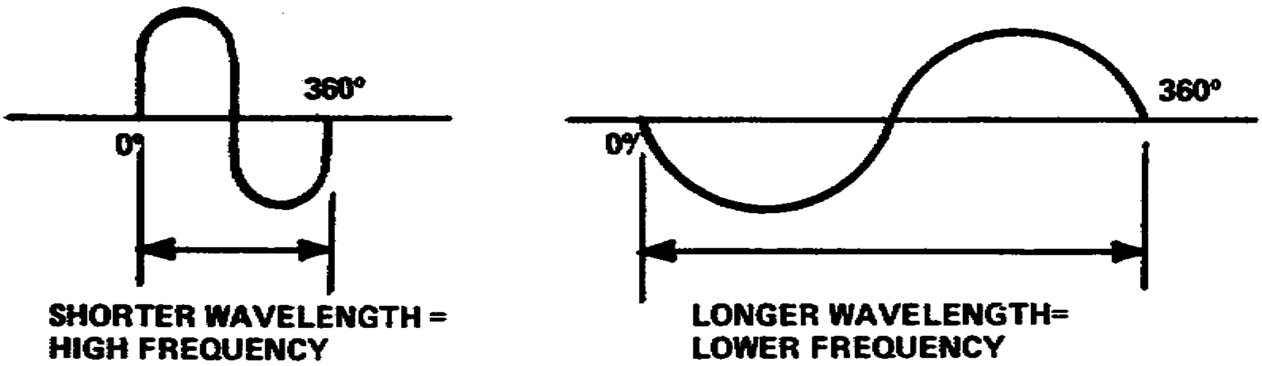

Figure 1-9. Wavelength

c. Amplitude is the third and last characteristic of sound and another part of the sine wave. Amplitude equals the height of a wave. The height of a wave can be measured in one of two ways.

(1) First, measure from the center reference line to either the positive peak or negative peak (fig 1-10). Measuring a wave this way is called the amplitude of a wave.

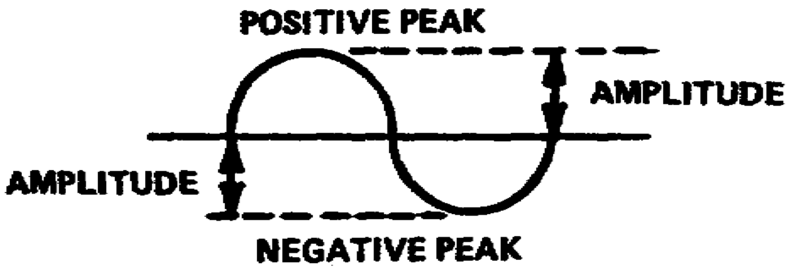

Figure 1-10. Amplitude of a wave

(2) Second, measure the height of a wave from its positive peak all the way to its negative peak. This type of wave measurement is peak-to-peak amplitude (fig 1-11).

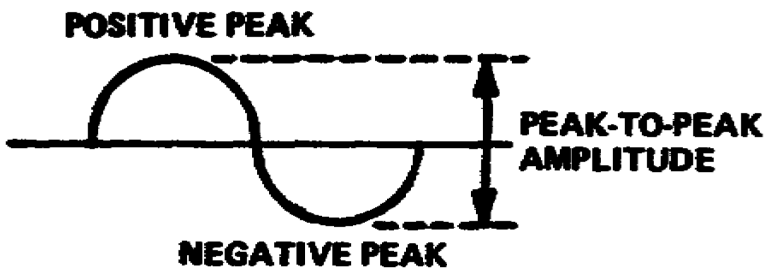

Figure 1-11. Peak-to-peak amplitude

d. The measurement characteristics of a sound wave, having been explained, need to be understood more clearly. Some of these characteristics are interrelated and affect hearing.

(1) An immediate relationship occurs between the number of cycles per second (frequency) of a sound wave and its physical length, or wavelength. One wavelength equals the length in space (or in any given medium) occupied by one cycle of the wave. If the frequency doubles, then one cycle or one wavelength occupies one half as much space. The higher the frequency, the shorter the wavelength; the lower the frequency, the longer the wavelength. Frequency also determines pitch. The subjective aspect of frequency, pitch, determines its position in the musical scale heard by the ear. The frequencies audible to human hearing ranges from approximately 15 Hz to 20,000 Hz, the audio spectrum.

(2) Amplitude determines the intensity or power of a sound. Increasing the amplitude or height of a wave increases its intensity or power and the violence by which the molecules collide with each other in the medium. The ear responds to this by “hearing” the sound as being louder.

e. Sound consists of more than what has been discussed so far. Sounds occur continuously every day: people talking, dogs barking, the sounds of cars passing by, music from radios, etc. All of these make up sound waves called complex waves. Complex waves consist of a multiple of frequencies which can be broken down into three components: a fundamental, harmonics, and overtones.

(1) The fundamental frequency, the lowest frequency of the complex wave, determines the pitch of a musical note.

(2) Harmonics. Any frequency that is an odd multiple of the fundamental frequency, such as 1, 3, or 5 times the fundamental frequency.

(3) Overtones. A harmonic of the fundamental frequency of a complex waveform.

Learning Event 3:
DESCRIBE FACTORS AFFECTING SOUND QUALITY

1. When a sound wave travels through a medium such as air and encounters another medium such as water, colder air, or a solid object, part of the sound is reflected back from the object in a manner similar to a beam of light. The balance of the sound passes into and is transmitted by the second medium.

a. Reflection. If the sound wave strikes a hard surface at an angle, a large part will bounce off and be reflected at an angle which is exactly equal to the angle of incidence (fig 1-12). Reflected sound increases the intensity, causing a type distortion by adding an echo (a reflected sound) or reverberation (a multiple reflection of sound). Large areas with hard surfaces, such as auditoriums, produce a lot of reverberation and echo. The shape and size of the area also determines the reflection of sound. Large flat areas reflect more sound and produce echos and reverberation, whereas irregular shaped areas produce less.

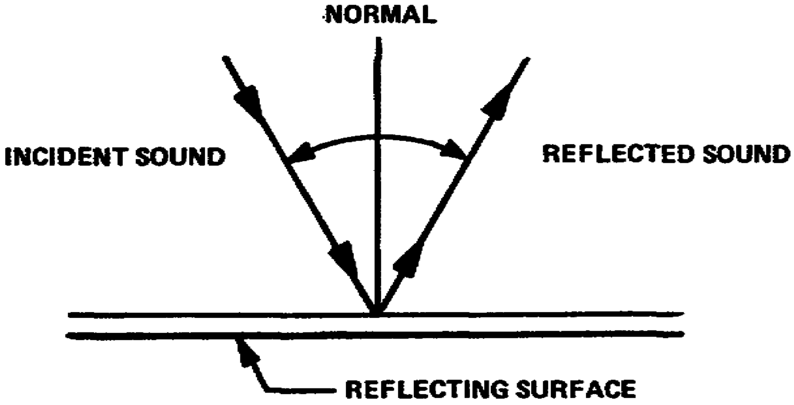

Figure 1-12. Reflection of sound wave

b. Absorption. When sound strikes a surface, it is both reflected and absorbed. Depending upon the construction of the surface, the amount of reflected and absorbed sound differs; soft porous surfaces like drapes, curtains, or acoustical tile, absorb more sound than hard surfaces. The ideal location to record absorbs sound very well and is acoustically dead and echo free (fig 1-13).

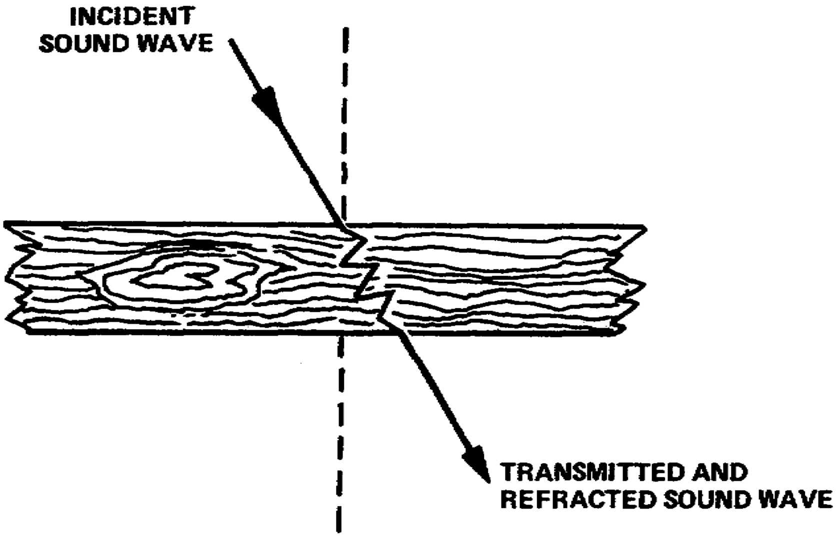

Figure 1-13. Absorption of a sound wave

c. Refraction. As it passes from one medium into another, sound changes speed and direction. As sound travels from air into glass, the speed increases and its direction changes. The denser the medium, the faster the speed of sound. Sound travels about 1127 feet per second, or 343 meters a second (fig 1-14 and 1-15).

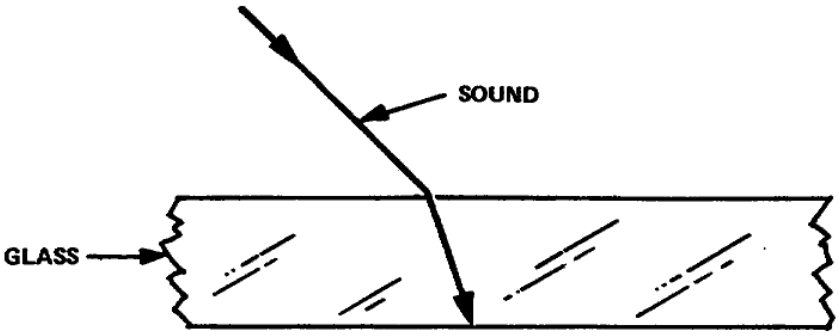

Figure 1-14. Sound traveling from air to glass

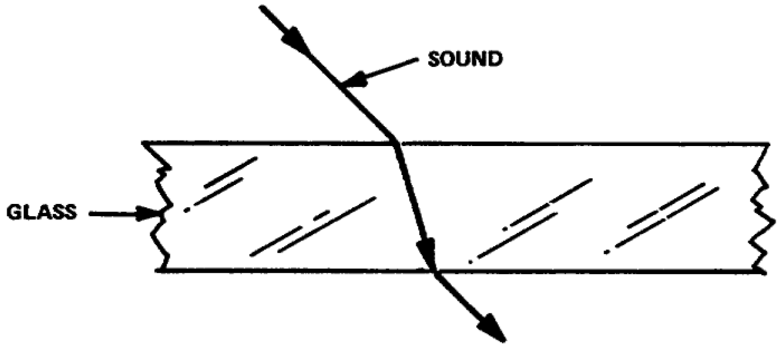

Figure 1-15. Sound traveling from air, to glass, to air

(1) As the sound passes from the solid back to the air, it changes its speed and direction again.

(2) In refraction, the angle of change is always perpendicular to its original path.

d. Diffraction. When sound encounters an object in its normal path of travel, it bends around that object and continues back to its normal path, (fig 1-16). Example: You are placed in a room with an open window and in the distance you can hear the sounds of a marching band. The sounds most readily heard are those of the bass drum and other instruments which produce the lower frequencies of the audio spectrum. The lower frequencies, due to their longer wavelengths, diffract more readily than the higher frequencies. As sound from instruments that produce the lower frequencies enter through the window, this opening or aperture causes these low frequencies to scatter in all directions.

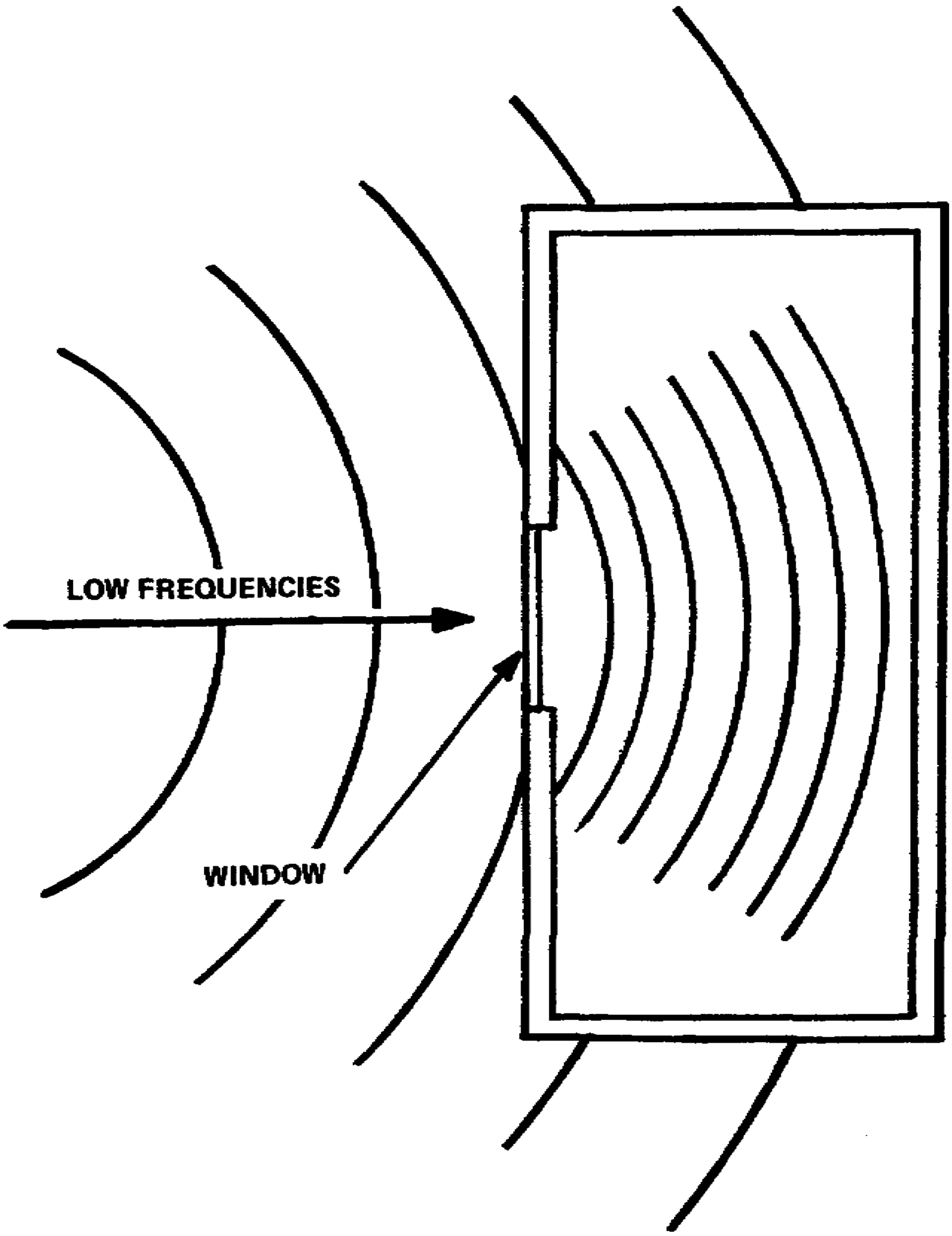

Figure 1-16. Diffraction of lower frequencies

2. Phase Relatively of Sound. The relationship of the position of two identical complex waveforms, at any given time, is phase relationship. Phase relationships break down into five areas: resonance, interference, beat frequencies, echo, and reverberation.

a. Resonance. Resonance occurs when two like waves unite to reinforce or aid each other. This occurs when compressions from the sources and rarefactions occur at the same instant in time, with respect to each other. Resonance causes a larger volume of air to move; therefore, a louder sound. For example:

(1) People phase stereo speakers to get full volume. Both speakers push or pull (compress or rarefy) air at the same time.

(2) Rear and front speakers in a car face each other. For full volume the rear speaker pushes while the front speaker pulls. Both speakers are said to be "in phase", with each other.

b. Interference. In interference, there is no complete union of waves because the compressions and rarefactions occur at different times (fig 1-17). They are occurring off from each other, causing interference. It is said that these sound waves are "out of phase." The effect of interference is a smaller volume of air moving, hence a loss of overall volume of sound. Sound waves "out of phase" 180°, result in complete cancellation of sound (fig 1-18).

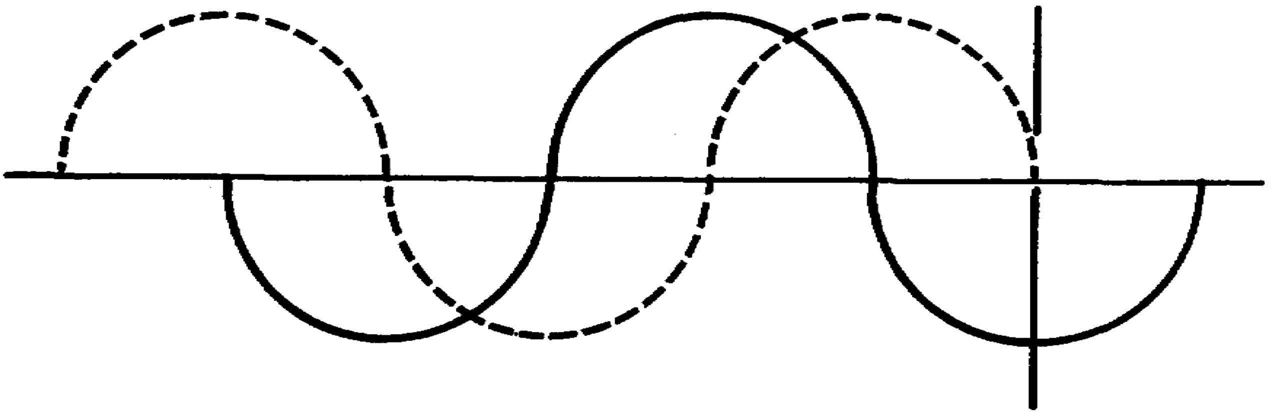

Figure 1-17. Sound waves partially "out of phase"

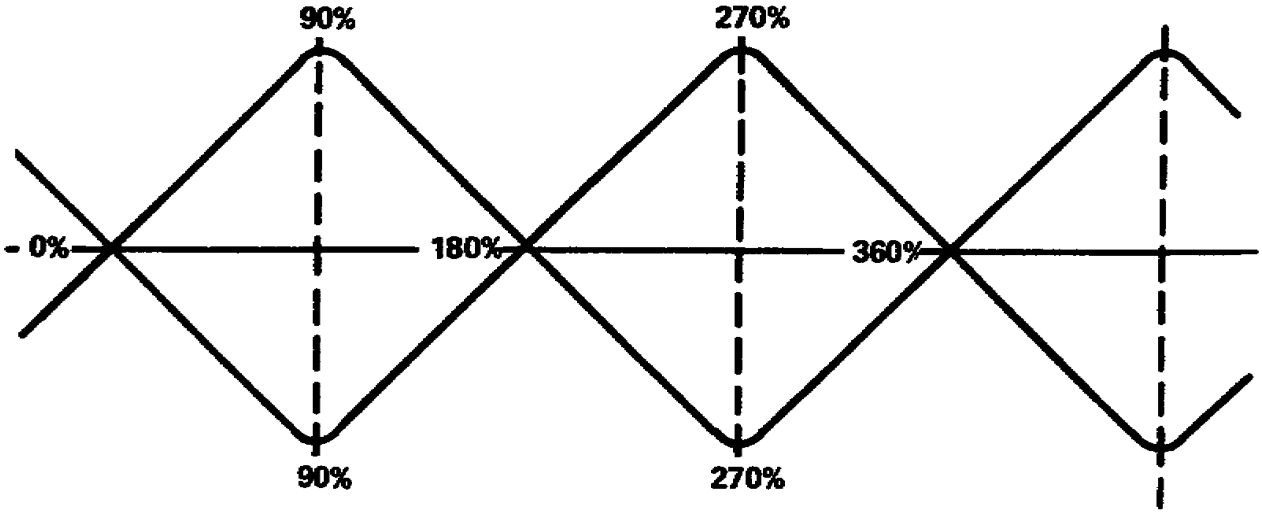

Figure 1-18. Sound wave 180° “out of phase”

c. Beat Frequencies. A beat frequency results from the partial union of like or unlike frequencies. At any instant of time the waves may be in phase with each other, out of phase, or in some degree of phase relationship between each other. This continual shifting back and forth of phase relationship results in a beating or pulsating sound and is referred to as beat frequency (fig 1-19).

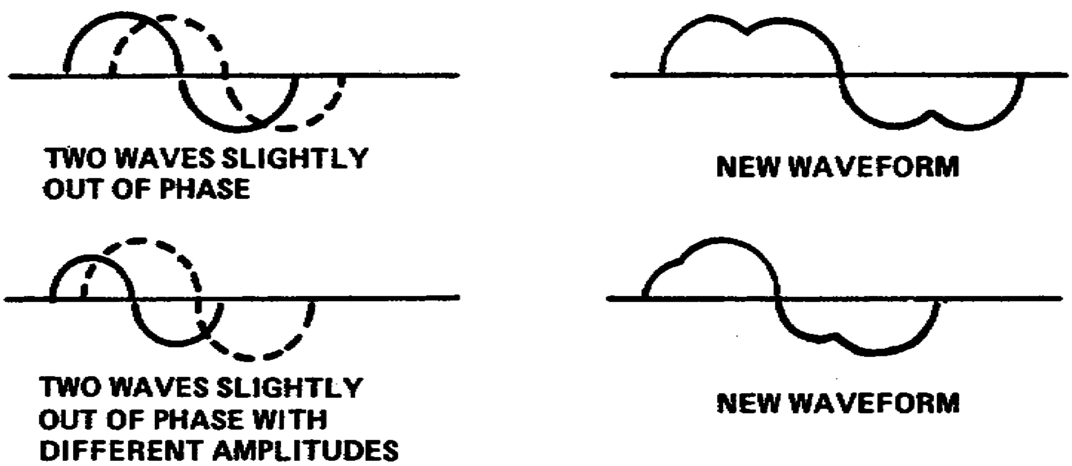

Figure 1-19. Examples of beat frequencies

d. Echo. An echo is the repetition of a sound caused by reflection from a surface. To be an echo, the reflected sound must be 1/20 of a second or longer behind the original sound.

e. Reverberation. A reverberation is the persistence of sound within an enclosure after the original sound has ceased. Reverberation may also be considered as a series of multiple echos that merge into a single continuous sound.

Learning Event 4:
USE SOUND MEASUREMENT

1. The decibel (db). Even though the ear "hears" a range of frequencies from 15 to 20,000 Hz (the audio spectrum), it hears the range of frequencies from 500 to 6,000 Hz better. Also, the intensity or power of a sound must be increased 10 times before the ear notices a doubling of the sound loudness. Because of these two situations, scientists developed a logarithmic measurement based on tens called a db (decibel).

a. Engineers developed the db or decibel to measure frequencies (especially single frequencies) where the signal does not fluctuate, but remains constant. The sounds heard each day consist of complex waves (two or more single frequencies). Therefore, technicians developed another unit of sound measurement called the VU or volume unit.

b. The volume unit equals the db in that the mathematics to figure both the VU and db remains the same. The VU measures complex waves with their continuous fluctuations and not the single constant frequencies as measured by the decibel.

c. Because a volume unit approximately equals a db, people often use the two terms interchangeably.

2. Percentage of Modulation. Equipment manufacturers have established a maximum power or signal amplitude which may go through the equipment without being affected.

a. One hundred percent modulation of a recording system equals this maximum power or signal amplitude which may be applied without being distorted by the audio equipment. Any percent (%) below 100% modulation equals a percentage of the total or maximum signal amplitude which can be applied to the equipment. When a VU meter reads 40% modulation, then only 40% of the normal maximum signal amplitude travels through the equipment. If the VU meter reads 50% modulation, then 50% of the maximum signal amplitude travels through the equipment, and so on.

b. Since 100 percent modulation equals the maximum power or signal amplitude to be applied to the equipment, the 100% modulation becomes a reference by which db reference levels are established. Zero (0) db equals 100% modulation or the maximum signal amplitude (power) which may be applied to the equipment.

c. Anything above 100% modulation or the 0 db reference level, consists of plus (+) dbs (i.e., +1 db, +2 db, etc.). Anything below 100% modulation or the 0 db reference level, consists of negative (-) dbs (i.e., -1 db, -2 db, etc.) or percentage of modulation which are less than 100%. (i.e., 10%, 30%, 60%, etc.).

PRACTICE EXERCISE

1. What is a single frequency sound wave called?

a.	Pure tone	c.	Complex wave
b.	Compression	d.	Rarefaction wave

2. The higher the frequency of a sound, the greater the number of:

a.	Cycles per second	c.	Electromagnetic sounds
b.	Sine wave	d.	Amplitude

3. Which characteristic determines the pitch of sound?

a.	Harmonics	c.	Frequency
b.	Reverberation	d.	Cycles per second

4. What are the frequencies that are audible to the normal human learning range?

a.	5 Hz to 15,000	c.	20 Hz to 30,000 Hz
b.	15 Hz to 20,000 Hz	d.	25 Hz to 35,000 Hz

5. Which of the following occurs when there is no complete union of waves because of compressions and rarefactions?

a.	Interference	c.	Absorption
b.	Reverberation	d.	Complex tone

6. Which measurement is used to measure a signal that does not fluctuate?

a.	Volume unit	c.	Pure tone
b.	Decibel	d.	Single wave

7. The ideal location to record should absorb sound and is acoustically dead.

a.	True	b.	False

8. What is the lowest frequency of the complex wave called?

a.	Amplitude	c.	Fundamental frequency
b.	Rarefaction	d.	Cycles per second

9. Which factor affects sound quality from the partial union of like or unlike frequencies?

a.	Resonance	c.	Beat frequencies
b.	Deflected	d.	Spectrum

ANSWERS TO PRACTICE EXERCISE

1. A
2. A
3. C
4. B
5. A
6. B
7. A
8. C
9. C

LESSON 2
OPERATE AUDIO CONSOLES AND MICROPHONES

TASK

Describe the methods for operation of audio consoles and five types of microphones

CONDITIONS

Given information on the function of audio mixers, characteristics of microphones, the five types of microphones and their use, and the function of a patch panel.

STANDARDS

Demonstrate competency of the task skill and knowledge by responding to the multiple-choice test covering methods of operation of audio consoles and five types of microphones.

REFERENCES

None

Learning Event 1:
DESCRIBE AUDIO MIXING

1. Definition of Audio Mixing. Audio mixing is defined as the technical and aesthetic blending of sound from two or more sound sources. Each sound source is called a channel. In this definition, there are two key words, "technical" and "aesthetic". The term technical refers to the actual physical operation and manipulation of equipment and how well it is operated. Aesthetic is used to describe the listening portion of the mixing process. How well does it sound? Is it pleasing to the ear?

2. Three Areas of Audio Mixing. The three areas of audio mixing are really an extension of the definition of audio mixing. The three areas are: sound levels, balance, and operating techniques. The first two terms, sound levels and balance, have to do more with aesthetics while the last term, operating techniques, deals with the technical aspects.

a. Sound Levels. A sound level is the amount of sound loudness heard in the medium. Volume controls on the loudspeakers control sound. Signal levels are found within the equipment in the form of electrical signals and come from mics, turntables, or recorders, etc. Signal levels are controlled by attenuators or pots on the audio console; 100 percent modulation or 0 db is read on the VU meter. Sound levels are what is heard, but signal levels are what is actually being recorded or going through the equipment.

(1) With the use of audio equipment, a correct level is established when the signal peaks at zero db or the 100 percent modulation point on the VU meter. Any continuous modulation which goes over zero db or 100 percent modulation mark is called “over modulation” or “riding in the mud.” The end result of over modulation is distortion, because the signal amplitude is too high for the equipment to handle.

(2) Just as over modulation produces distortion, under modulation or “riding in the mud”, will produce a low volume and a low signal-to-noise ratio. Because both over modulation and under modulation are undesirable situations, they should and can be avoided by “riding the gain.” Riding the gain is the process of constantly readjusting the signal levels during a recording to maintain a proper level.

(3) Occasional peaking into the red area of the VU meter, because of abrupt changes in volume level in music selections or emphasis in voice tones, are acceptable as long as they do not happen too often. When levels must be changed during a recording, the changes should be done gradually so they are unnoticed. This may present a real challenge when using a musical selection which has many abrupt changes in levels as do marches or some classical presentations.

b. Balance. Balance is the difference in sound loudness between two or more sound sources. It is dependent on the audio engineer's hearing and aesthetic sense or balance (how the various sound sources sound together). Studio speakers must all be the same volume of loudness to achieve a true balance. A good example of this is the use of music under an announcer's voice. The audio engineer must determine if the music is too loud and drowning out the announcer's voice or if the music is too low and cannot be heard.

(1) An ear for balance is something which most audio people develop after working in the field for awhile. In determining the balance of various sound levels a variety of things can be done. These may vary from just increasing or decreasing the signal gain on a particular source to moving a microphone closer to a speaker.

(2) The balance of many sounds should be done prior to recording. Balance is achieved mostly through testing and trials.

c. Operating Techniques. Operating techniques refers only to the technical aspect of audio mixing and good techniques depend upon how well the operator handles the equipment and how well he mixes or blends his sources. The operating techniques should be smooth so that any changes made will be unnoticed by the listener. Some of the terms used to describe basic operating techniques are: fading, crossfade, segue, down and under, and up and under.

(1) Fading is used at the closing and opening of gain control. It may be fast or slow with sound being clipped in or out prior to fading in or out.

(2) Crossfade is the simultaneous fading out of one signal and fading in of another.

(3) Segue is the beginning of one musical section; or source, immediately following another with no fading of announcement between.

(4) Down and under is the fading down of music to a low level for the entrance of a voice, holding momentarily, and then fading the music out.

(5) Up and under is the “sneaking up” of music to a low level while an announcement is being made, usually being followed by “music up full” after the announcement is concluded.

> NOTE: Avoid crossfading with music vocals where the voice will cut in either direction. Avoid using down and under, up and under, etc., with vocals unless as required for emphasis.

Learning Event 2:
USE AUDIO CONSOLES

1. Audio consoles basically consist of the same functions and associated equipment. However, the following equipment and associated equipment may <u>not</u> be included on all audio consoles.

2. Subassemblies. The major subassemblies for audio consoles usually consist of a power supply, a program amplifier, a monitor amplifier, cue amplifiers, a pre-amplifier, and a relay system. Each of these subassemblies have a special purpose in the functioning of the audio console.

a. Power Supply. The power supply is the section of the audio console which contains the master power ON-OFF switch and fuses for the console. Its primary purpose is to furnish all the power to the console's components.

b. Program Amplifier. The program amplifier uses a standard 600 ohm impedance and its output level is controlled by the master attenuator. Its primary purpose is to amplify console output.

c. Monitor Amplifier. The monitor amplifier's output level is controlled by the monitor attenuator. Its primary purpose is to amplify monitoring signals, audition signals, and talkback facilities.

d. Cue Amplifiers. The cue amplifier can be used for talkback. Its primary purpose is to amplify signals from inputs for cueing before programming.

e. Pre-amplifiers. The pre-amplifier is used with signals from microphones or phono cartridges. Its primary purpose is to amplify signals from a low level input.

f. Relay System. The relay system can sometimes be DC operated and operates the warning lights. Its primary purpose is to prevent feedback by muting loudspeakers in the presence of live microphones.

3. Controls. The controls for an audio console usually consist of master pots, channel pots, program selector switches, and VU meters.

a. Master Pots. The master pots are used to regulate the overall output of the board. Once the master pots are set up using a 1 KHz tone, they remain as they are until the entire board is neutralized.

b. Channel Pots. The channel pots are used to regulate the output of the individual channels. Channel pots must be adjusted to their individual sources.

c. Program Selector Switches. Program selector switches usually consist of keys or interlocking pushbuttons which are used for rapid selection of inputs to the console. Program selector switches are used to program sound source, monitor a sound source for auditioning or to neutralize a sound source.

d. VU Meters. VU meters are used to measure the average of the complex waves which come from the console. There are usually two meters.

e. Remote Channel. These pots connect the remote lines to TB (talk-back), CUE (monitor output), or MIX (over the air) (Pot 8).

f. Network Channel. Pot 9 is the network channel and can be used as input for a 600 ohm network input such as an off-the-air tuner, or cart tape machine.

g. Nemo Channel. The nemo channel as referred to here; is not used as a remote input. The use of the term here refers simply to a high level channel which has been provided for auxiliary use (Pot 10).

h. Line Amp Feed Switches. The line amp feed switches are the tab key selector switches located above Pot 10. Line amp feed, key AL3, may be amplified for simultaneous feed from both channels.

i. VU Meters. There are two VU meters located in the center of the audio console. The meter on the left is the program VU meter while the one on the right is used for addition.

j. VU Selector Switch. The VU selector switch allows for monitoring of AL1, AL3, and network on the audition VU meter. AL2 and utility are not used.

k. Monitor Bank. The monitor bank is the area directly below the two VU meters. It consists of six smaller dials, a talk button and a small speaker.

l. Volume Control-Intercom. This dial controls the volume for the intercom/cueing speaker located in the top center of the monitor bank.

m. Intercom Selector Switch. The intercom selector switch determines the booth or remote that is run through the intercom facilities.

n. Monitor Selector Volume. This switch controls the volume for the monitor speaker found in the studio.

o. Monitor Selector Switch. This provides for monitoring of either the audition or program channel or external source over the monitor speaker.

p. Phones Switch. The phones switch allows you to monitor any circuit over the headphones. Only lines AL1 and AL3 are used.

3. Gates Diplomat Audio Console

NOTE: For the purpose of this subcourse, the Gates Diplomat Audio Console, will be used as the example.

a. The Gates Diplomat is a fully transistorized monaural console. It is completely dual channel, with ten mixing channels, cue-intercom and 28 upper level tab keys. It is a low impedance type console, with ladder attenuators which operate in 2 db steps (fig 2-20).

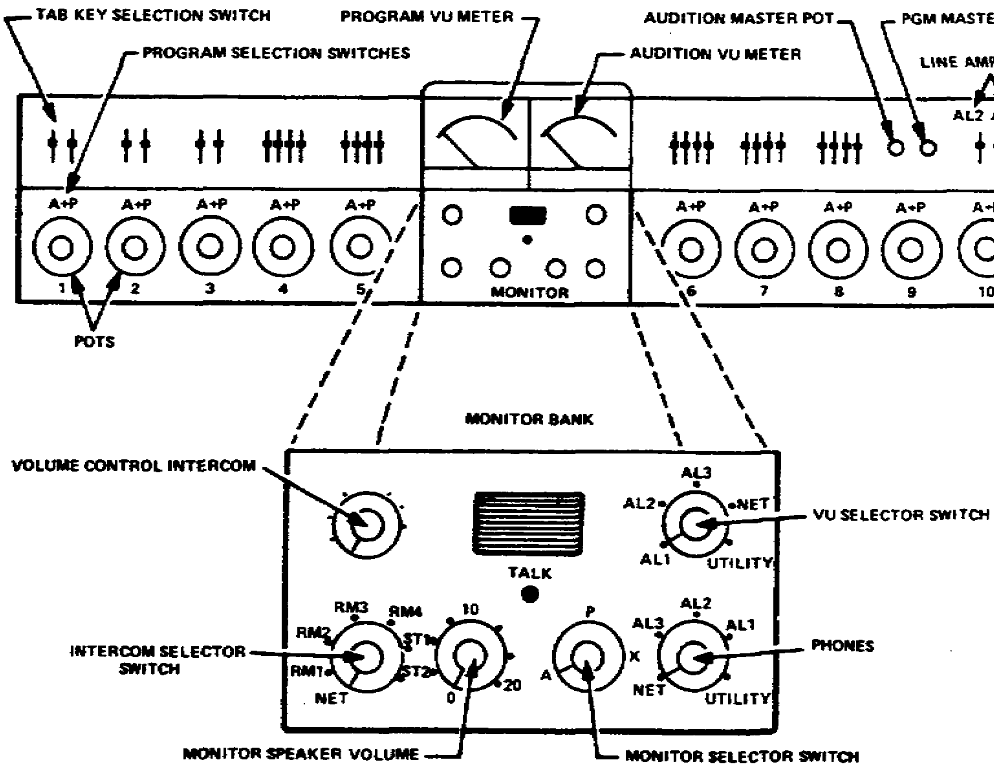

Figure 2-20. Audio console, Gates Diplomat

b. Control Locations.

(1) Program Selector Switches. The program selector switches are located directly above each pot and regulate the direction of the channel into either the audition or program amplifier. There are three positions for the program selector switch and they are: left for audition, center for neutral, and right for program.

(2) Tab Key Selector Switches. The tab key selector switches, which are located above each pot and program selector switch, are used to simply select a particular input to the pot.

(3) Microphone Channels. The microphone channels are made up of Pots 1, 2, and 3. Through the use of the tab key selector switches up to 12 microphones can be put into the three pots at various times.

(4) Turntable Channels. The turntable channels are made up of Pots 4 and 5. Four upper tab key selector switches above each pot select the turntable input to be used.

> NOTE: On pots four through 10 (fig 2-20) there is an extra position on the pot itself, below the zero mark, which is called the cue or detent position. This cue position connects any input to the pot to the cue amplifier and allows for the cueing up of the source.

(5) Tape Channels. The tape channels are made up of Pots 6 and 7. The tab key selector switches above each pot are used to select the tape channel input as with the turntable channels.

Learning Event 3:
DESCRIBE MICROPHONE CHARACTERISTICS

1. Introduction. Up to this point, you have learned about sound as it is heard in the form of acoustical (sound) energy. With the use of a microphone, we convert acoustical energy into electrical energy. The microphone becomes the first link between sound as it is heard in varying forms of air pressure, and its permanently recorded forms of tape, disc, and film. Almost any recorded material, such as voice, instrument or special effects, must first pass through a microphone. In the following text, the basic construction uses of microphones, the five types of microphones, their general and specific characteristics, and their uses will be discussed.

2. Principles of Operation. A microphone (electracoustic transducer) converts sound energy into electrical energy. Physical structure for all microphones includes a housing, diaphragm, magnetic field, and a moving part within that field according to their principles of operation. Microphones break down into two categories.

a. Pressure-Operated Microphones. Pressure-operated microphones include carbon, crystal, dynamic, and condenser (capacitor) microphones. In a pressure-operated type microphone, the electrical output results from the motion of a conductor, usually a small coil moving back and forth, which moves with the varying sound pressures in a magnetic field. In a pressure-operated microphone, sound waves strike only one side of the diaphragm.

b. Velocity Microphone. The velocity, or ribbon microphone, uses the moving conductor principle. A ribbon is suspended so that it vibrates freely in a magnetic field. In this case, the ribbon is the diaphragm, since it is not housed in any closed type of case and is exposed to the air on both sides. Sound waves strike the sides of the ribbon. The output is proportional to instantaneous sound pressure from the velocity component of the sound wave.

3. Primary Microphone Characteristics. All microphones share common characteristics called “primary microphone characteristics.” These consist of frequency response, sensitivity, directivity, impedance values, noise and distortion.

a. Frequency Response. The frequency response is the part of the audio spectrum that the microphone picks up.

b. Sensitivity. A microphone's sensitivity determines its ability to pick up various sounds. Sensitivity, measured in decibels, starts from the input of a signal to the output of a signal for a given pressure at the diaphragm.

c. Directivity. All microphones do not pick up sound equally from all directions. The directivity or pickup pattern of a microphone tells how to position a sound source at the microphone for optimum sound pickup. Four types of pickup patterns describe a microphone's directionality. They are:

(1) The Omnidirectional Pickup Pattern. Microphones with this pattern configuration accept sound equally from any direction without loss of characteristics. There are no variations of this pattern (fig 2-21).

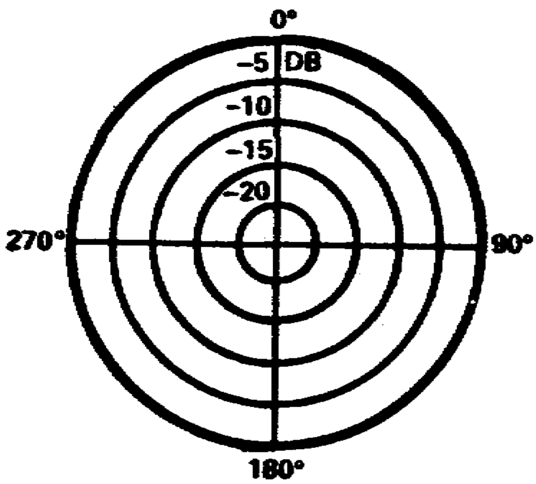

Figure 2-21. Omnidirectional pickup pattern

(2) The Bidirectional or Figure-eight Pattern. Microphones with this configuration accept sound best at the 0 and 180° direction points with least acceptance at the 90° points (fig 2-22). They have varying degrees of acceptance at the in-between points on the graph. This pattern varies as the size of the figure eight is changed in relationship to each other. If either the upper or lower lobe is made larger or smaller, then the pattern size changes.

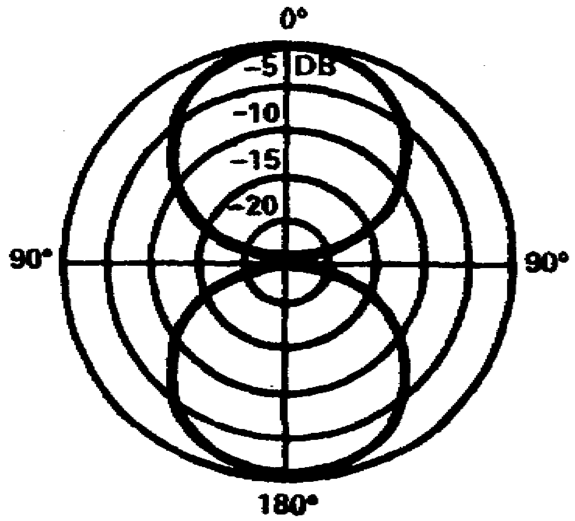

Figure 2-22. Bidirectional pickup pattern

(3) The Cardoid Pickup Pattern. Microphones with this polar pattern accept sound best at 0° point, with minimal response at the 180 degree point and with varying degrees of response at the in-between points (fig 2-23). This pattern varies at the size and shape of the lobes below the 90 degree points on the graph change.

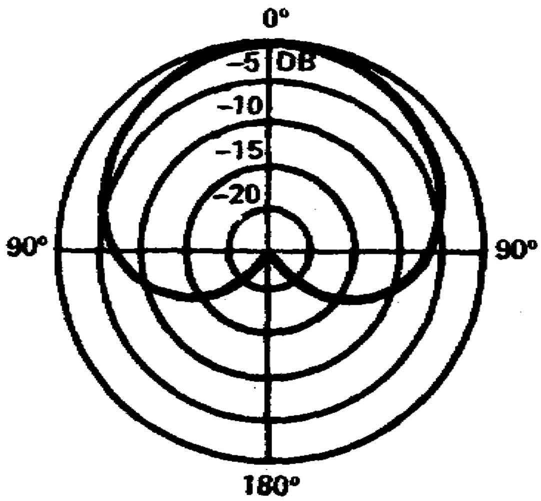

Figure 2-23. Cardoid pickup pattern

(4) The Unidirectional Pickup Pattern. The pickup pattern for this microphone looks like a candle. It accepts sound only from the 0° point with the better microphones. Poor quality microphones may pick up slightly from the 90° and 180° points, but this sound quality will have a loss of frequency on the lower end of the scale.

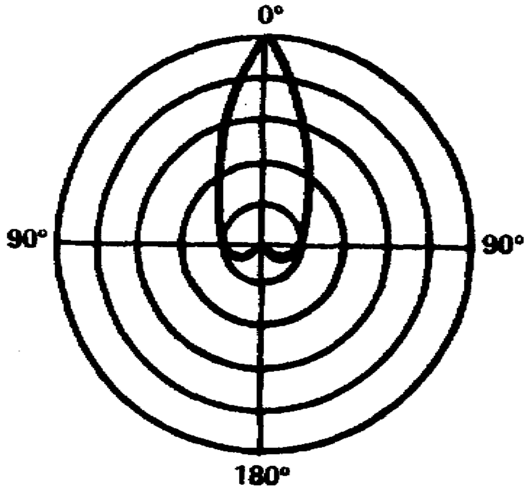

Figure 2-24. Unidirectional pickup pattern

(5) In both the bidirectional and unidirectional type microphones, the lobe shapes result from the design characteristics developed by certain manufacturers.

(6) Impedance Values. Impedance values are as follows:

(a) 600 ohms or less - Low impedance.

(b) 1000 ohms or higher - High impedance.

Impedance values of microphones determine microphone cable length. High impedance microphones generally develop poor signal-to-noise ratios, low or insufficient input signals, or loose frequency response, if the cable is longer than 25 feet (7.62m). Low impedance microphones may use cables of unlimited length.

(7) Noise. Noise is inherent in some types of microphones, particularly in carbon microphones. Most manufacturers list noise in their specifications rated in dbms.

(8) Distortion. Distortion equals the amount of wave form change present in a signal for a specific microphone.

Learning Event 4:
USE BASIC AND SPECIAL TYPES OF MICROPHONES

1. To further narrow a microphone's use for specific job applications, study the specific characteristics of each type. These specific characteristics must be considered to best select a microphone to perform a particular job. The five types of microphones presented here begin with the least effective and progress to the most efficient.

2. Carbon Microphone. This microphone is activated by carbon granules, and is held in a container (a brass cup) attached to a metallic diaphragm (fig 2-25). Sound waves striking the diaphragm cause a change in contact resistance among the granules. This change in contact causes a current from a battery, connected in series with the carbon button (brass cup), and the input of a transformer to vary in amplitude. The result is a current waveform similar to an acoustic waveform striking the diaphragm. After leaving the output of the transformer, the sound becomes amplified and reproduced. This principle of operation causes a high internal noise making the carbon microphone limited in its usage. All telephone receivers use carbon microphones because telephone transmissions require only intelligibility and voice clarity.

a. Frequency Response: 200-2000 Hz with much distortion present.

b. Sensitivity: Low or poor, must be close to sound source.

c. Impedance: Low.

d. Internal Noise: Very high due to the friction of the carbon granules.

e. Other characteristics: Inexpensive and rugged. Primarily used for communications (telephone).

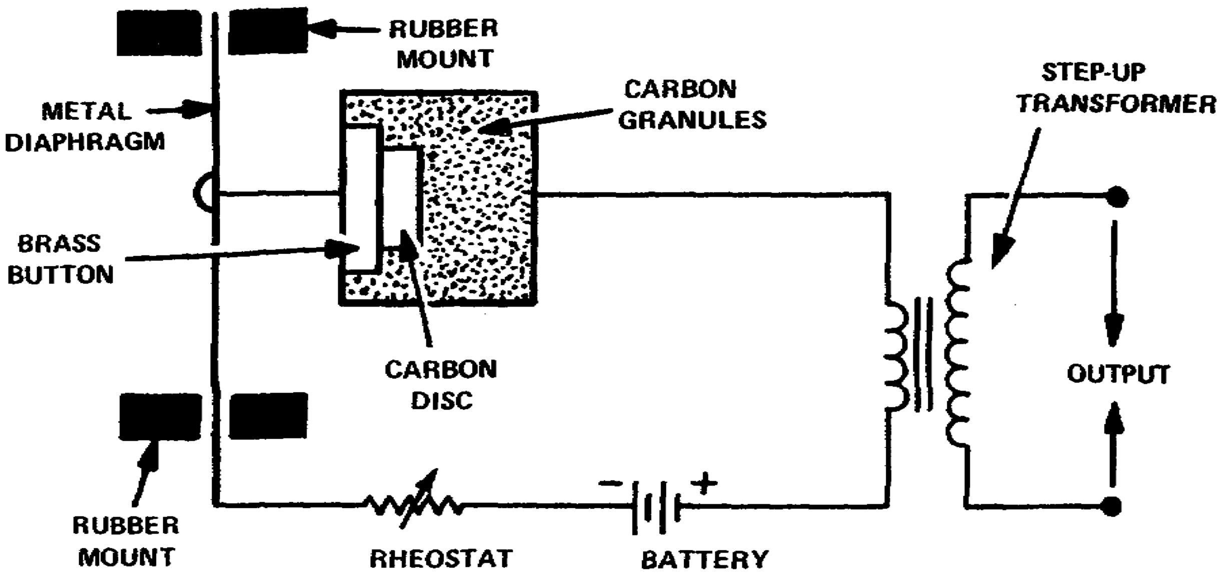

Figure 2-25. Carbon microphone

3. Crystal Microphone. The crystal microphone uses one or more Rochelle salts placed in such a way that when a sound wave strikes them they bend or twist. This action produces an electrical current called the piezoelectric effect (pressure electricity).

a. When exposing the crystal to a mechanical stress such as a sound wave striking a diaphragm (bimorph) or the crystal itself (sound cell), a minute current develops which is directly proportional to the mechanical pressure.

b. The Bimorph and Sound Cell (trade names) make up the two most popular types of designs for crystal microphones (fig 2-26).

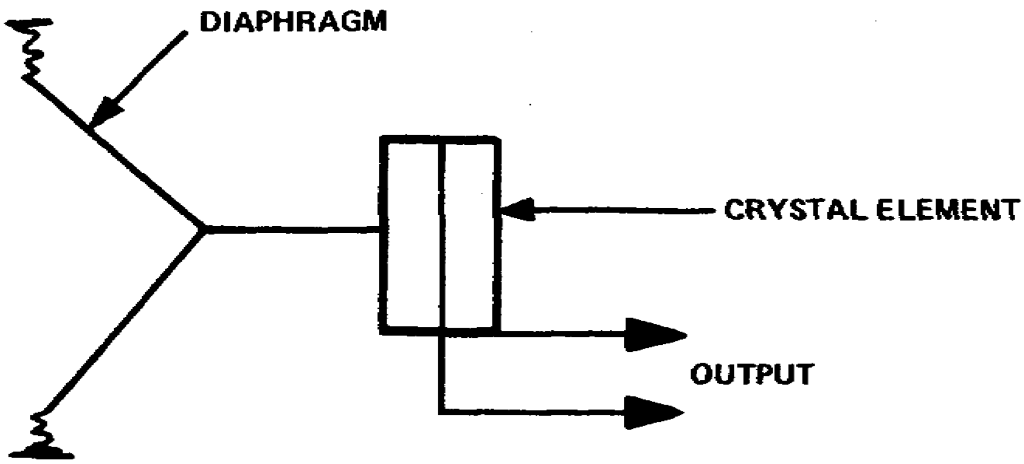

Figure 2-26. Bimorph microphone

c. In the Bimorph, sound waves strike the surface of the diaphragm, creating pressure. The crystals consist of two slabs, separated by a thin piece of foil, which are connected to one side of the external circuit. The outer surfaces of the crystal slabs covered with foil connect to the other side of the external circuit.

(1) Frequency Response: 80 - 6590 Hz

(2) Directivity: Nondirectional

(3) Impedance: High

(4) Other Characteristics: Pressure-operated, inexpensive and small in size

d. Sound Cell Characteristics. Lacking a diaphragm, the sound cell is directly activated. Sound waves strike the crystal elements which are stacked in a pile. This bends and twists the crystals, creating an output voltage.

(1) Frequency Response: 20 - 16,000 Hz

(2) Directivity: Nondirectional

(3) Impedance: High

(4) Other Characteristics: Used for home tape recorders and radios

e. Dynamic Microphones. Known as the most ruggedly constructed of all broadcast microphones, the dynamic microphone may be found in sports, remote broadcasts, and studio recording situations. Because “wind pickup” noise does not affect the microphone, it is used outdoors.

(1) The dynamic microphone favors high frequencies over low. In the studio, this aids the speaker with a deep or bass voice to achieve a higher pitch. Due to its sensitivity to higher frequencies, it accentuates sibilance in a person's voice. (Some people produce a hissing sound while pronouncing s's and ch's. This hissing sound is sibilance.)

(2) Dynamic Microphone Characteristics. This microphone operates on the moving coil generator principle (fig 2-27). It uses a diaphragm and coil which move the field of a permanent magnet. Sound waves striking the diaphragm cause the coil to be moved. This movement in a magnetic field generates an output voltage.

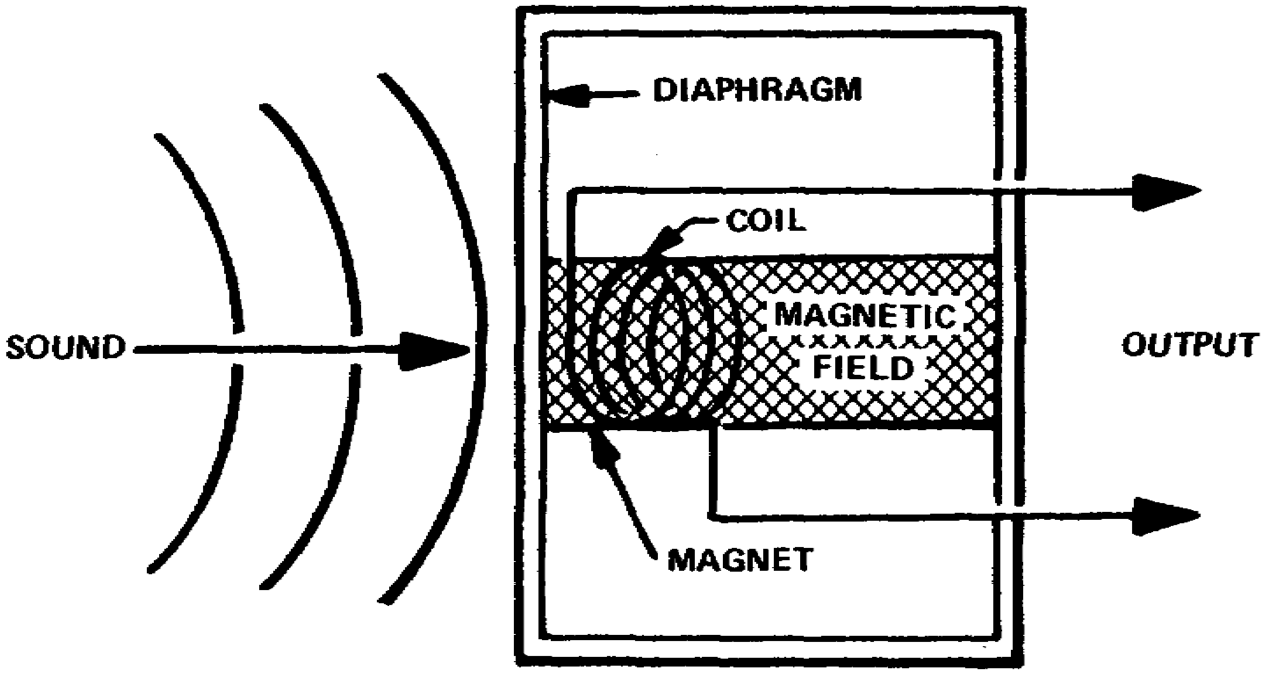

Figure 2-27. Dynamic microphone

(a) Frequency Response: 20 - 15,000 Hz which is nearly all of the audio spectrum.

(b) Directivity: Omnidirectional

(c) Sensitivity: Very high

(d) Impedance: Low

(e) Other Characteristics: Pressure-operated, rugged and sturdy

f. Velocity Microphone. Because of its pickup element, manufacturers and studio broadcoasters also call the velocity microphone a ribbon microphone. Do not use the velocity microphone outdoors or in a studio recording situation where it will have to be moved a great deal. Strong winds or rough handling could cause damage to the ribbon (fig 2-28).

(1) The velocity microphone favors low over high frequency sounds and consequently may be used to deepen a voice which is too high in pitch. The closer a performer is to this microphone the deeper his voice will sound.

(2) When used in its normal mounted position, the velocity microphone possesses two live and two dead sides. Its pickup pattern is bidirectional. A performer may talk into either of the live sides of the microphone, or two performers may use both live sides alternately or simultaneously.

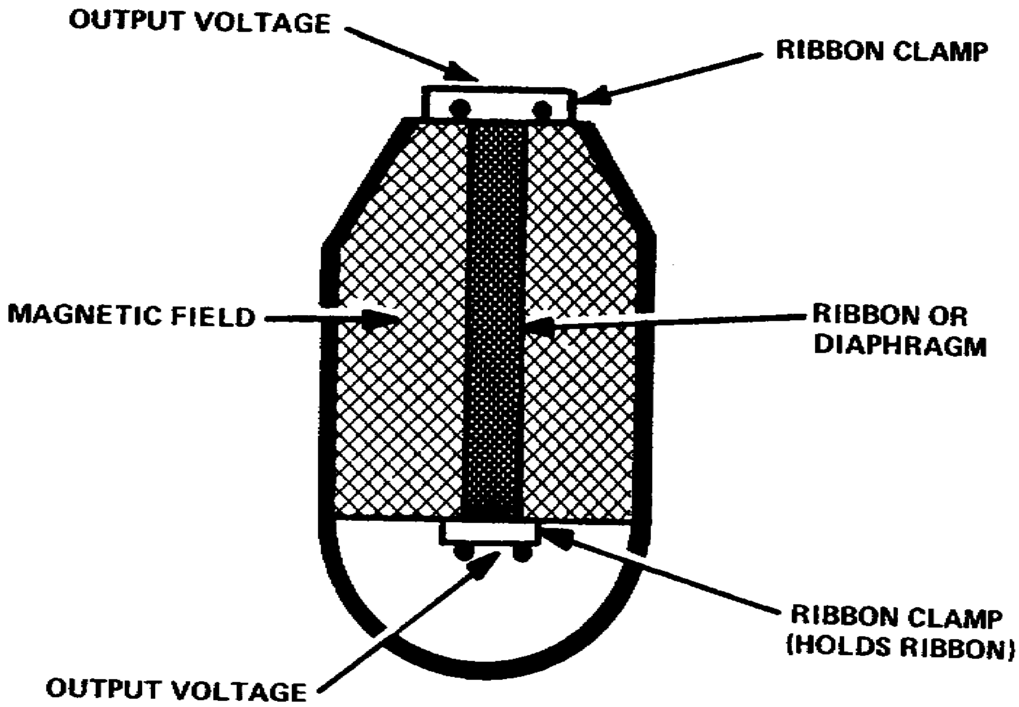

Figure 2-28. Velocity (ribbon) microphone

(3) Principles of operation. A metallic ribbon is suspended between the poles of a permanent magnet which vibrates freely in a magnetic field. This ribbon, constructed of a thin aluminum strip, serves as the diaphragm. Movement of this ribbon by a sound wave causes the magnetic lines of force to be cut crosswise, creating a corresponding voltage between the two ends of the ribbon.

(a) Frequency Response: 20 - 15,000 Hz

(b) Directivity: Basically bidirectional

(c) Sensitivity: Excellent

(d) Impedance: Low

(e) Other Characteristics: Since the ribbon is easily damaged, never use it outdoors

g. Capacitor (Condensor) Microphone. These microphones get extensive use in recording studios because of the broad frequency range, low distortion, little internal noise, and excellent sensitivity (fig 2-29). But, due to their principle of operation, they require a preamplifier as an integral part of the housing, plus a power supply for the preamplifier. They also require an output transformer that converts the extremely high impedance of the microphone capacitor head to a low impedance for unlimited cable length. Their quality and principle of operation combine to make them the most expensive professional microphones.

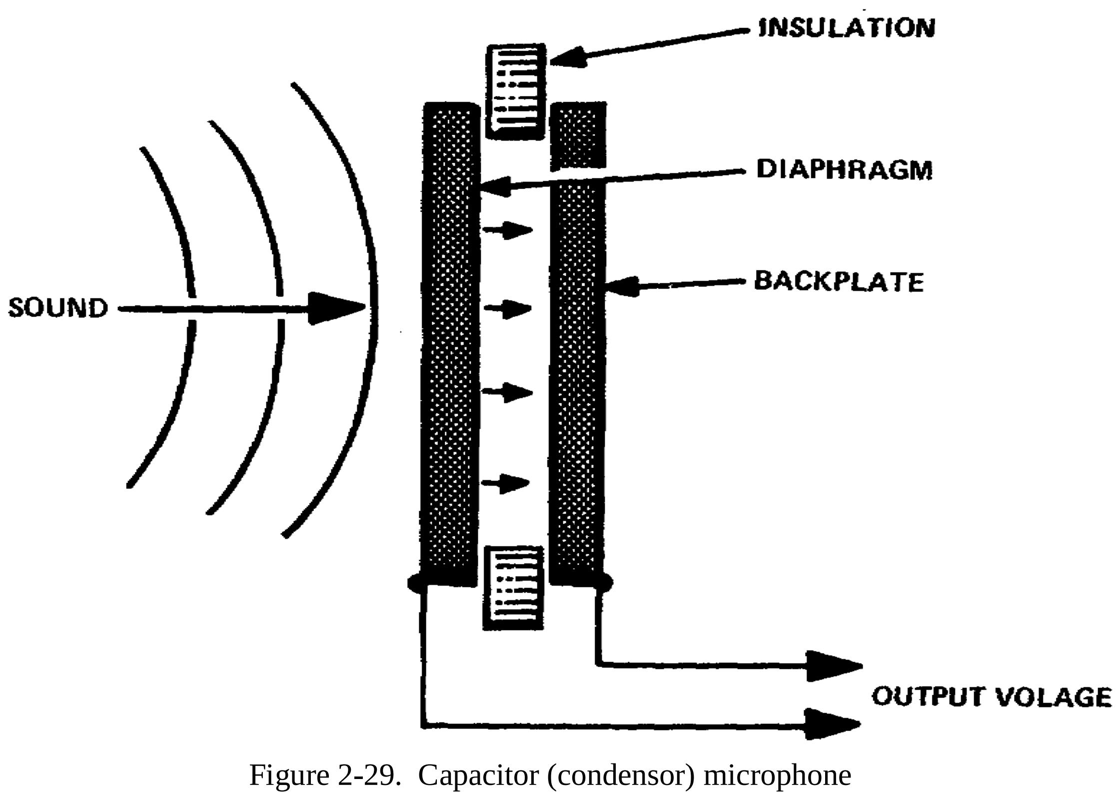

Figure 2-29. Capacitor (condensor) microphone

(1) Capacitor Microphone Characteristics. The capacitor microphone operates on the storage of an electrical charge. The head of the microphone contains two plates. One is a stretched diaphragm, the other is heavy backplate. The backplate is insulated from the diaphragm and spaced parallel to the rear surface of the diaphragm.

(2) As sound waves enter the microphone, the pressure causes a change in the spacing of two plates thereby varying the internal capacitance. This produces a signal voltage proportional to the pressure component of the wave. It then gets amplified and passed through the output transformer so conversion from high impedance to low impedance may occur.

(a) Frequency Response: Widest of all microphones; nearly covers the complete audio spectrum

(b) Directivity: Earlier models all omnidirectional, may now be obtained with omni-, bi-, and unidirectional characteristics

(c) Sensitivity: Exceptional

(d) Impedance: High impedance. The output transformer converts the high impedance of the microphone capacitor head to a low impedance

(e) Other Characteristics: Has very faithful signal reproduction. The disadvantages of this microphone are its fragility, and cost. Its extreme sensitivity may prove a disadvantage, depending upon the recording requirements.

h. Special Purpose Microphones.

(1) Wireless or FM Microphones. All wireless microphones work according to a basic FM transmission principle. The performer wears or holds a medium sized microphone which is connected to a small pocket transmitter. The sending antenna of this transmitter is either worn around the waist or pinned along the trousers or skirt of the performer. The most flexible wireless microphones are entirely self-contained. For example, the “Vega Microphone” has its transmitter built into the microphone itself. The sending antenna either sticks out of the microphone or serves as the neck cord by which the microphone is worn. A special receiving station, with one or several antennas, can be located as far as 1,000 feet from the microphone. This receiving station amplifies the signal and sends it to the master audio mixer.

(2) Despite the obvious advantages of wireless microphones, their operation has been restricted to specific production tasks. Most often wireless microphones are used in restricted remote locations. Less frequently they are used in studio productions because the audio quality transmitted by the wireless system is somewhat below that of the regular cable method.

(3) Shotgun Microphones. These special purpose type microphones, sometimes referred to as “machine-gun” microphones, are highly directional and designed to pick up sounds over relatively great distances. The shotgun microphone is a rugged, omnidirectional microphone with tubes that range from 2 inches to 5 feet in length. These tubes make the omnidirectional microphone highly directional. The long barreled shotgun microphone, aimed at a distant sound source, follows that sound source much as a gun follows a moving target. Unfortunately, this microphone is quite heavy and must rest on a special pedestal which permits simultaneous tilting and panning. Despite its highly directional feature, the audio pickup is adequate, at best, and is generally used when quality is the least important factor in the sound transmission.

(4) Parabolic Reflector Microphones. The parabolic reflector is one of the earliest devices developed for picking up sound over relatively great distances. Instead of the long barrel of the shotgun microphone, a parabolic disc catches and focuses the distant sound waves and directs them into an ordinary microphone which is placed in the focal point of the parabolic disc. Depending on the size and material of the disc, only certain frequencies can be picked up. Since high and low frequencies react quite differently, the overall quality is not satisfactory. So far, parabolic pickups have been used for broadcasting crowd reactions, band music originating across the stadium, and other sounds that provide atmosphere rather than specific information. Of course, there is ample room for experimentation in long distance sound pickup.

(5) Contact Microphones. A device used to pick up sound vibrations from a solid material such as an electrical guitar is called a contact microphone. For the changing of acoustical energy to electrical energy any microphone could be used, but a contact microphone is supported only by the surface, thus eliminating the problem of microphone placement.

(6) Variable Polar Pattern. These are microphones in which the operator changes the polar pattern manually from omnidirectional to bidirectional, to unidirectional (cardoid), or to a combination of all three. Microphones of this type, offering adjustable pickup patterns, are termed polydirectional. An example is RCA 77DX velocity microphone.

(7) Differential Microphone. Noise-cancelling microphones, basically designed for use in automobiles, aircraft, boats, tanks, public address systems, and industrial plants are used to advantage where the ambient noise level is greater than 100 db. They discriminate against all sounds originating more than 1/4 inch from the front of the microphone. This microphone is of the dynamic type and the pickup pattern is unidirectional.

(8) Impedance. The output of a microphone will either be high impedance working directly into the circuitry of the input amplifier stages, or low impedance using a 30, 150, or 250 ohm transformer. This low impedance works into the input transformer of an amplifier. Some microphones built for various applications have a tapped output transformer with a screwdriver adjusted switch to select either high or low impedance outputs.

NOTE: In general, high impedance microphones are used for public address installations and consumer type tape recorders.

(9) Electret Condenser Microphone (Lavalier). The microphone element is attached to the perfomer utilizing the supplied tie clasp. The microphone is comprised of two elements, the microphone head and the battery/transformer housing, interconnected by 6 feet of miniature cable. The battery/transformer housing is clipped to the performer's belt. This microphone is placed at hands' length from the performer's mouth to obtain optimum sound quality.

NOTE: Cable Length. The effect of microphone cable length directly relates to the impedance and the application of the sound system. The metallic shielding in a microphone cable forms a capacitance with the inner conducting wires which is added to the capacity between the wires themselves. This is the same as placing a capacitor across the line; the higher the impedance used, the greater will be the shunting effect across the line which will cause a loss of high frequency response. Cable lengths must be considered when using high impedance microphones. The effect of any cable length is negligible for low impedance microphones. A general rule is to use a low impedance microphone for any cable run over 25 feet. Low impedance must be used in any form of communications where the operator works at a point remote from a microphone amplifier, since medium and high frequencies are very important for crispness and intelligibility of speech.

i. Operator Maintenance of Microphones, Microphones Cables and Connectors.

(1) Inspect the microphone, microphone cables and connectors to insure

(a) Wind screens are in place and serviceable.

(b) Pickup element is serviceable.

(c) Microphone housing is not bent or broken.

(d) Connector pins are not bent, broken or corrected. If connector pins are broken, replace using slides, soldering silver, and flux.

(e) Cables are clean and wire shielding is unbroken. (Clean with water, mild soap and a sponge or cloth.)

(2) Clean and service microphones, microphone cables and connectors in accordance with the manufacturer's manual.

(3) Identify defects that cannot be repaired by operator and turn in to repair personnel.

Learning Event 5:
USE PATCH PANELS

1. The patch panel contains receptacles for all inputs and outputs of the audio console except for microphone inputs, intercom wiring, and monitor speaker outputs. Inputs normally used with specific mixers are internally interconnected to eliminate complicated patching.

2. Patch panels provide for quick interconnection of equipment, standardization of plugs and connectors and the reduction of control room wiring. A complete patching facility will greatly extend the flexibility of the audio console.

3. The patch panel has a series of lug receptacles (in pairs) for both inputs and outputs. The top row of receptacles are for the outputs of various pieces of equipment and sources while the bottom is for inputs to different pots on the audio console and certain pieces of equipment.

a. Above each pair of plug receptacles are labels describing what the various outputs and inputs are (fig 2-30).

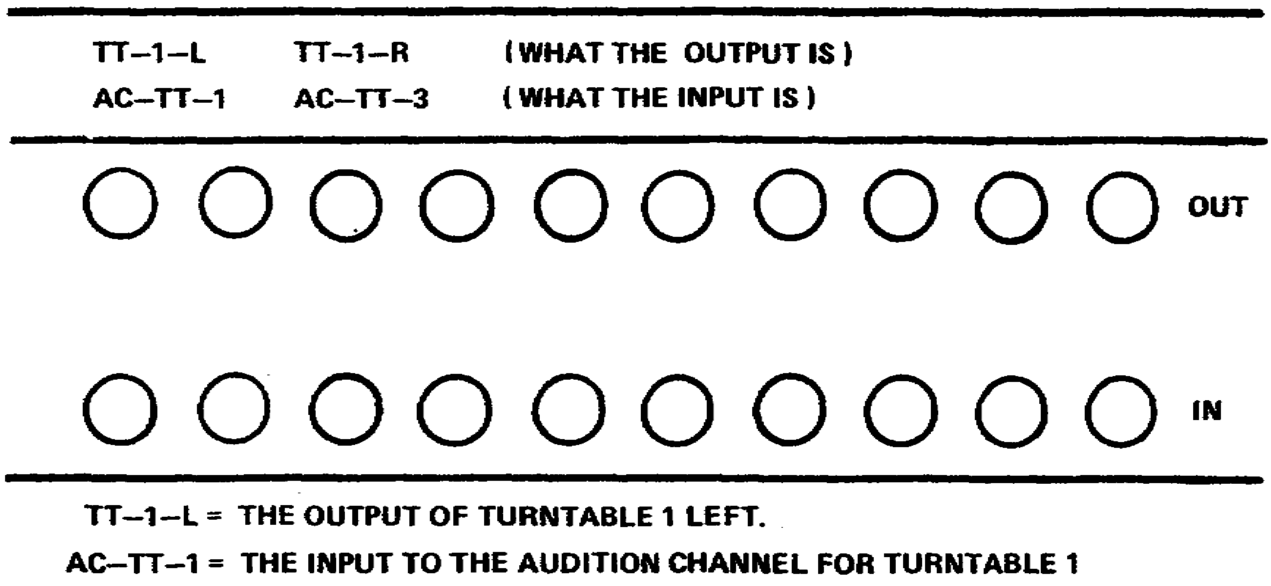

Figure 2-30. Patch panel

b. Any time a patch panel cable is put into one of the plug receptacles, the input which normally goes to the board is cancelled. For instance, if the input plug receptacle for Pot 9 (which is normally used for cart machine playback), has a patch cable in it from the output of a tape machine, then the cart machine playback function will be cancelled out.

c. The plugs on the patch cables have a set of “ribs” on one side. Make sure the ribs on both plugs of the patch cable go in the same direction which will ensure no problems with “phasing”. A good thing to remember is “Keep the Ribs to the Right”.

4. Safety Precautions and Practices.

a. Remove all jewelry around electrical equipment.

b. Plug patch cable only into the patch panel.

c. Pull plugs out of the patch panel by pulling on the plug, <u>not</u> the cable.

PRACTICE EXERCISE

1. Occasional peaking into the red is acceptable when recording

A. True B. False

2. Which of the following is the difference in sound loudness between audio sources?

a. Sound level
c. Balance
b. Modulation
d. Volume

3. Which of the following is a subassembly of the audio console?

a. Network channel
c. Program amplifier
b. Line AMP feed switches
d. Master pots

4. Which of the following is a fading out of one signal and fading in of another?

a. Segue
c. Down and under
b. Cross fade
d. Up and under

5. In which type of microphone do sound waves strike onto one side of the diaphragm?

a. Pressure electricity microphone
b. Pressure operated microphone
c. Velocity microphone
d. Ribbon microphone

6. What is the impedance value for a microphone to be classed as low impedance?

a. 600 ohms or less
b. 700 ohms or less
c. 800 ohms or less
d. 900 ohms or less

7. The velocity microphone is also called a _______________ microphone.

a. Ribbon
b. Parabolic
c. Wireless
d. Condenser

8. Which microphone should not be used outdoors but excells in sensitivity and frequency response?

a. Capacitor microphone
b. Dynamic microphone
c. Velocity microphone
d. Contact microphone

9. Which microphone is the most ruggedly constructed of the broadcast microphones?

a. Capacitor microphone
b. Dynamic microphone
c. Contact microphone
d. Cardiod microphone

ANSWERS TO PRACTICE EXERCISE

1. A
2. C
3. C
4. B
5. B
6. A
7. A
8. C
9. B

LESSON 3
OPERATE TURNTABLES AND AUDIO
TAPE RECORDER/REPRODUCER

TASK

Describe the methods for operation of turntables and audiotape recorder/reproducers.

CONDITIONS

Given information about recorded sound, turntables, and audio recorders (reel-to-reel, audiotape cartridges, and cassette units).

STANDARDS

Demonstrate competency of the task skills and knowledge by responding to the multiple-choice test covering operation of turntables and audiotape recorder/reproducers.

REFERENCES

None

Learning Event 1:
USE AUDIO TURNTABLES

1. Recorded sound is used in television and motion picture operations in a great variety of ways. Background music for films or video productions, live performances, and recorded sound effects are the most commonly used recorded sound. Recorded sound can be reproduced from live sources: (1) records, (2) electrical transcriptions (records intended for broadcast use only), (3) audiotape, (4) videotape, and (5) film. Records and audiotape are only indirectly coupled with the picture portion: that is, they are not mechanically synchronized with the picture. The audio portion of videotape and film is directly recorded on these media. The sound is thus mechanically synchronized with the picture. Records are still the most frequently used prerecorded sound in both television and motion picture operations. For short segments, the standard records are being replaced with the more efficient cartridge-tape operations. However, we can safely assume that standard records will be used for sometime to come and that audio control rooms will still have to be equipped with turntables.

2. Turntables have provisions to play all speeds - 78, 45, 33 1/3 rpm (revolutions per minute), and all record sizes - 6 1/2, 7, 10, 12 and 16 inches in diameter. A special attachment built into the turntable is necessary for the

wide-hole 45 rpm commercial records. Different types of records also require different styli. Turntable tone arms are equipped with interchangeable cartridges. Although a 78 rpm record is seldom, if ever, used, a cartridge with a 78 record needle should be available. Be sure to use the 78 cartridge on 78 records only, since its larger stylus would ruin any long playing record immediately. Different cartridges are used for mono and stereo records. With a stereo cartridge, stereo as well as mono long playing records can be played. Stereo records should not be played with a mono cartridge. Most tone arms now have a stereo cartridge which can handle any record except 78s.

3. Turntable Controls:

a. Stylus. Located on the end of the tone arm. They must be gently cleaned and kept free of dust buildup.

b. Tone arm. Located on a rotating support and balanced for the styli being used. Care must be used when playing a record; do not hit the tone arm or the styli will be forced across the record and scratch the grooves.

c. Speed selector lever. Located in the lower left corner of the turntable. Lever has four positions: 33 1/3 rpm, 45 rpm, 78 rpm, and neutral (located at the top between 33 1/3 and 45 rpm).

d. ON/OFF Button. Located in the lower right corner. Used to start and stop the turntable.

e. Plater. Located in the center of the turntable and covered with a felt pad. The pad is used to protect the records.

f. Spindle. There are two spindles that come with turntable platers. Both are located in the center of the plater. The small spindle is for 33 1/3 and 78 rpm records. The larger spindle is spring loaded and locks down out of the way until needed for 45 rpm records. To raise the 45 rpm spindle, push down and turn it clockwise and it will come up into position.

4. Operation of a Turntable:

a. Conduct preoperative inspection turntable and related equipment in accordance with the manufacturer's manual.

(1) Check output/input connectors and ensure equipment is grounded.

(2) Check stylus/cartridge mounting.

(3) Check stylus pressure.

(4) Check speed control switch.

(5) Check function of on/off switch, kick pedal and cue bar.

(6) Clean equipment and record disc.

b. Position disc on turntable.

c. Turn power on and engage speed control.

d. Position tone arm at the outermost groove of the starting point.

> NOTE: Operator is permitted to "slip-cue" the disc, but should not stop or reverse the revolving plate (turntable).

e. Place the turntable input key on the audio console to CUE (monitor cue start point).

f. Rotate turntable counterclockwise to allow lead-in time for turntable to attain full speed. A failure to allow sufficient lead time could result in a "wow" sound.

Approximate lead-in times are:

33 1/3 1/4 to 1/2 turn

45 1/3 to 3/4 turn

78 3/4 to 1 1/2 turn

g. At the end of the audio selection, lift the tone arm to a neutral or standing position (either manually or mechanically).

h. Place speed control in neutral.

i. Turn power off.

j. Remove record disc from turntable.

5. Operator's maintenance of a turntable involves the following steps:

a. Disconnect and inspect the AC power cord for serviceability.

b. Clean all exterior surfaces.

c. Check drive belts (if belt driven) for cracks and proper tension. Replace belt if broken, cracked, or stretched, using associated tools.

d. Clean the drive system using denatured alcohol and cotton swabs.

e. Check and lubricate the disc center guide/spindle using a lubricant.

f. Visually inspect the tone arm for damage. Replace if damaged.

g. Remove and inspect the cartridge. Replace if necessary.

h. Remove and inspect the stylus for damage if bent, crimped, or broken.

NOTE: The stylus should be at approximately an angle of 45° when mounted on the cartridge.

i. Check for correct speed control using a strobe disc.

j. Check input and output connectors.

Learning Event 2:
IDENTIFY THE PARTS OF AUDIOTAPE RECORDER/REPRODUCERS

1. Audiotape recorders/reproducers as they apply to an audio or TV production are discussed here. The discussion is in three main sections: transport mechanism, head assembly, and electronics. There are variations in the functions of each of these main sections.

a. Transport mechanism. All of the tape machines require some type of mechanism to move the tape past the record and playback heads. Such mechanisms have been given various names, but "tape transport" is generally accepted as standard.

(1) Features of the tape transport (fig 3-31) include the tape supply reel, which has either a friction brake or an active back-torque. The back-torque is supplied by the drive system or a torque motor. Back tension (torque) keeps the tape from becoming tangled due to the inertia of the tape reel. The tension idler holds a certain amount of tape in its loop; this spare tape is temporarily let out during quick starts. A slight delay is allowed for the supply reel, which has appreciable inertia, to start turning at operating speed. The tension idler and back-torque work together to smooth out irregularities caused by the rubbing of the tape against the supply reel sides, sticking together of tape layers, or other causes.

(2) The tape is drawn from the tension idler, across the rolling tape guide, erase head, tape guide, record head, tape guide, and reproduce head. The force which draws the tape across the heads at a constant speed is provided by the capstan and the capstan pressure roller. The combination of the capstan, the tension idler, and the reverse torque of the supply reel keeps the tape under constant tension. There is friction between the tape and the stationary heads. This friction is a source of vibration. To help eliminate this vibration the transport mechanism components are mounted on a rigid base. Other causes of vibration are: the amount of wrap around the head, smoothness of head faces, tape tensions, tape condition, tape composition, temperature, and humidity. The capstan may be either the shaft of the drive motor or a shaft driven through a speed-reducing mechanism. The capstan drive motor, and any associated mechanism must be made with precision, and drive the tape at a constant speed, or they will cause problems during both record and playback.

(3) Immediately following the capstan and the capstan pressure roller is another tape guide to keep the tape in alignment with the heads. If the tape guides permit any vertical variation of the tape, the recorded signal may be attenuated during reproduction. In extreme cases the signal can be

lost entirely, or the erase head will either fail to erase, or improperly erase when a recording is made. The tension idler near the takeup reel serves the same purpose as the other tension idler. The torque on the takeup reel changes according to the amount of tape on the reel.

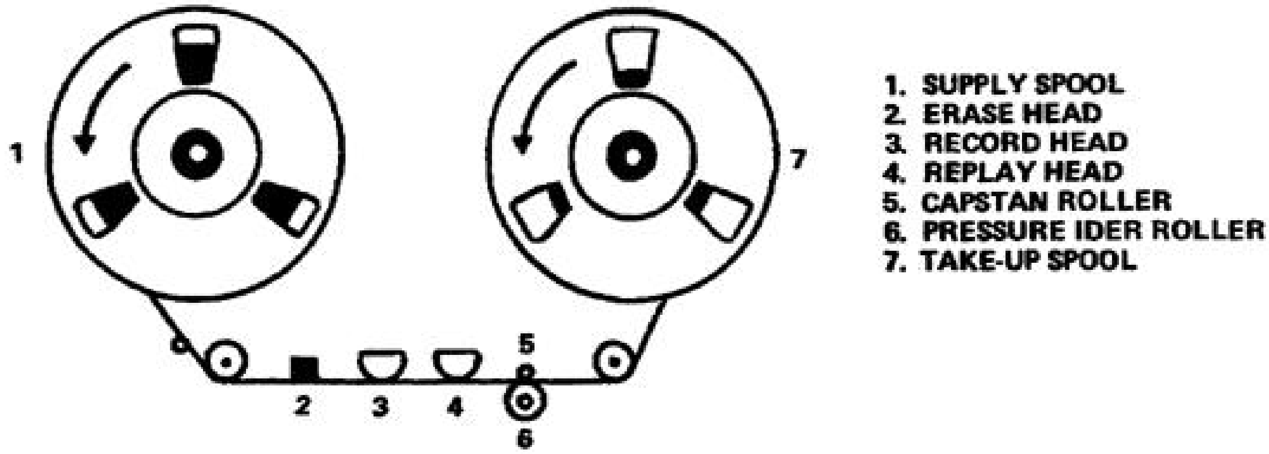

Figure 3-31. Tape transport

b. Functions of the tape transport are:

(1) Reproducing. With the recorder/reproducer turned on and the tape properly threaded, selecting the reproduce mode causes the control circuitry to release the main-reel brakes and starts the reel motors, the capstan motor, and the capstan idler. As a result, the recorded tape is pulled past the heads at a constant speed. The signal sensed by the record or reproduce head is equalized and amplified.

(2) Recording. In the record mode, the tape is moved as in the play mode. During recording, an erase signal from an internal oscillator is fed to the erase head, which clears any previously recorded signals from the tape before it reaches the record head. Information to be recorded is amplified, mixed with a bias signal, and applied to the record head. The information is recorded on the tape as it is pulled past the record head.

(3) Fast Forward/Rewind. With the recorder/reproducer turned on and the tape properly threaded, selecting the fast forward mode locally or remotely causes the control circuitry to release the main-reel brakes and apply full power to the takeup reel motor and hold back power to the supply reel motor. The control circuitry also causes the tape to be lifted away from the heads. Tape is then rapidly wound onto the takeup reel. The rewind mode is similar to the fast forward mode, except that full power is applied to the supply reel motor, and the tape is wound rapidly onto the supply reel.

(4) Edit. Three edit modes are selectable at the front panel of the

recorder/reproducer: (1) stop/edit, (2) fast wind/edit, and (3) play/edit. Selecting the stop/edit mode sets only the edit brakes of the tape reel motors, thus facilitating manual cueing and threading of the tape. Selecting the play/edit mode causes the tape to be pulled past the heads and spilled off the right side of the transport. This mode is typically used when unwanted tape is to be cut off. The fast wind/edit mode brings the tape into contact with the heads while the tape is being moved in the fast forward or rewind modes, making the recorded portions audible for high-speed search.

c. Head assembly. Professional tape recorders employ three separate heads (fig 3-31). In their order of head placement they are the erase head, the record head, and the reproduce head. The first to be discussed will be the erase head.

(1) Previously recorded signals are erased from a magnetic tape by passing a high frequency current through the erase head (fig 3-32). The tape passes over the erase head before it arrives at the record head. To erase in the record mode, a high frequency signal supplied by the recorder's erase oscillator, feeds current to the erase head. The current causes a strong alternating magnetic field to be developed. This magnetic field returns a previously recorded tape to its original state before being rerecorded.

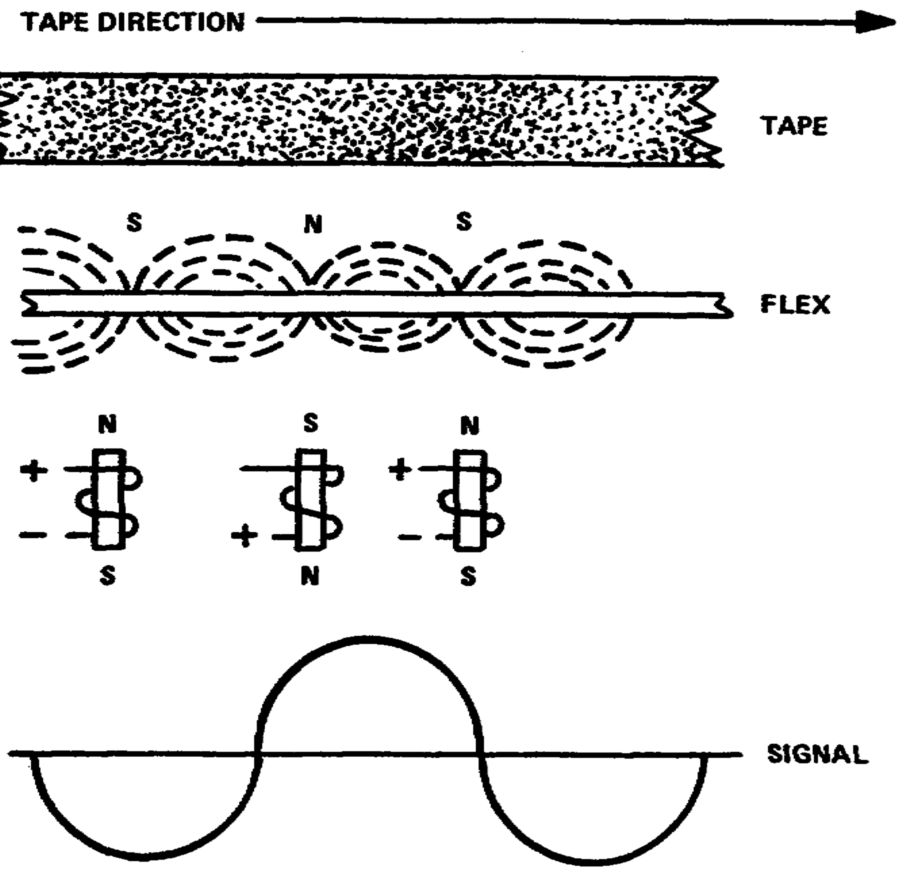

Figure 3-32. Basic operation of recording or reproducing head

(2) As audiotape moves across the recordhead, molecules on the tape orient themselves with the direction of a magnetic field. The magnetic field is directly proportional to the audio signal being fed to the record head. This audio signal reaches the tape from the record head with unnecessary distortion or quality loss by a carrier signal commonly referred to as the bias.

d. Electronics. Because nonlinearity is a normal characteristic of magnetic tape used for recording sound, the transfer of the audio signal from the record head onto the magnetic tape is not in direct proportion and results in severe distortion of the signal (fig 3-33).

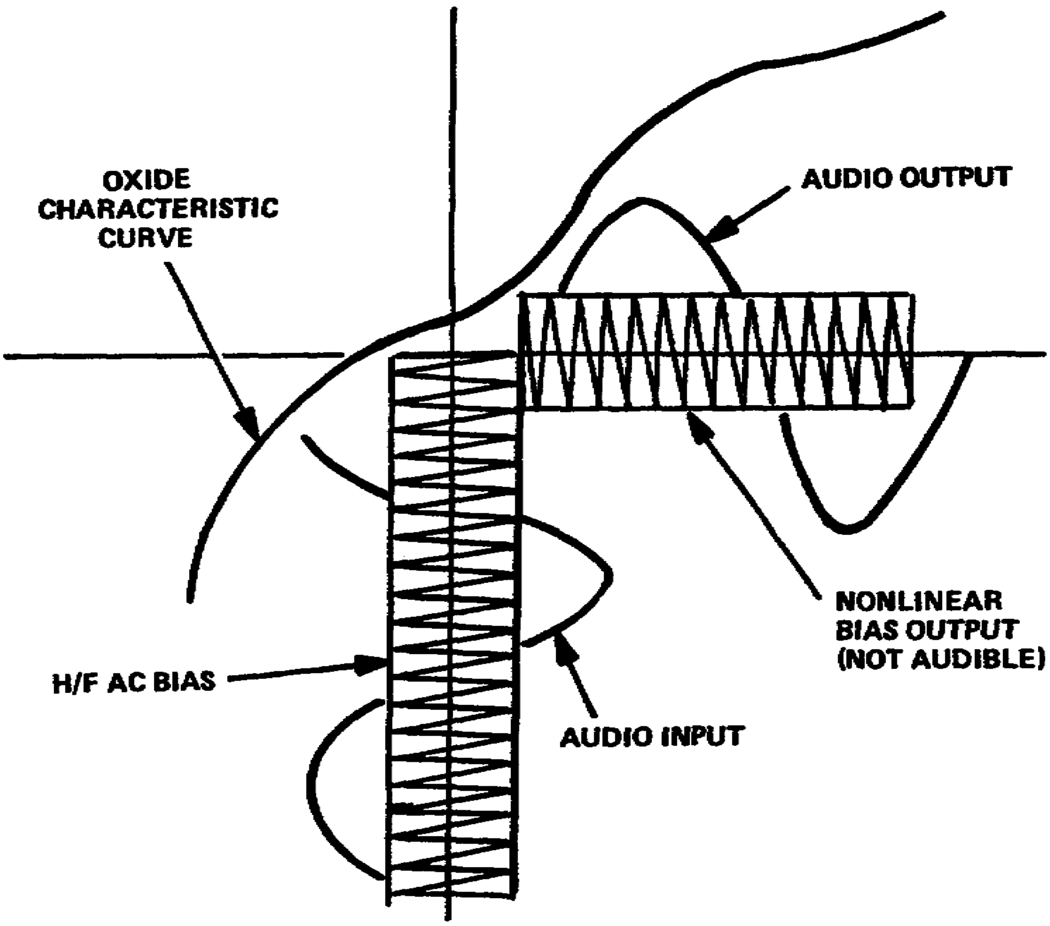

Figure 3-33. Bias current applied to audio signal

(1) A high frequency current several times that of the highest frequency to be recorded, is applied to a tape, along with the audio signal. This results in a signal or low distortion. The high frequency current, referred to as bias, also allows a higher signal-to-noise ratio.

(2) Bias combined with the audio signal and applied to the record head (fig 3-34), moves the audio signal in direct proportion onto the magnetic tape.

(3) The amount of critical bias current varies with different types of tape recorders. Usually the bias is adjusted to five times that of the highest audio frequency to be recorded, plus 5 KHz. Thus a tape recorder with a frequency response up to 15,000 cps requires a bias frequency (current) of at least 80 KHz. Higher frequencies of bias give better signal-to-noise ratios, on the average, than do lower bias frequencies. Too little bias results in high noise and distortion levels. Too much bias partially erases the high frequencies as they are recorded.

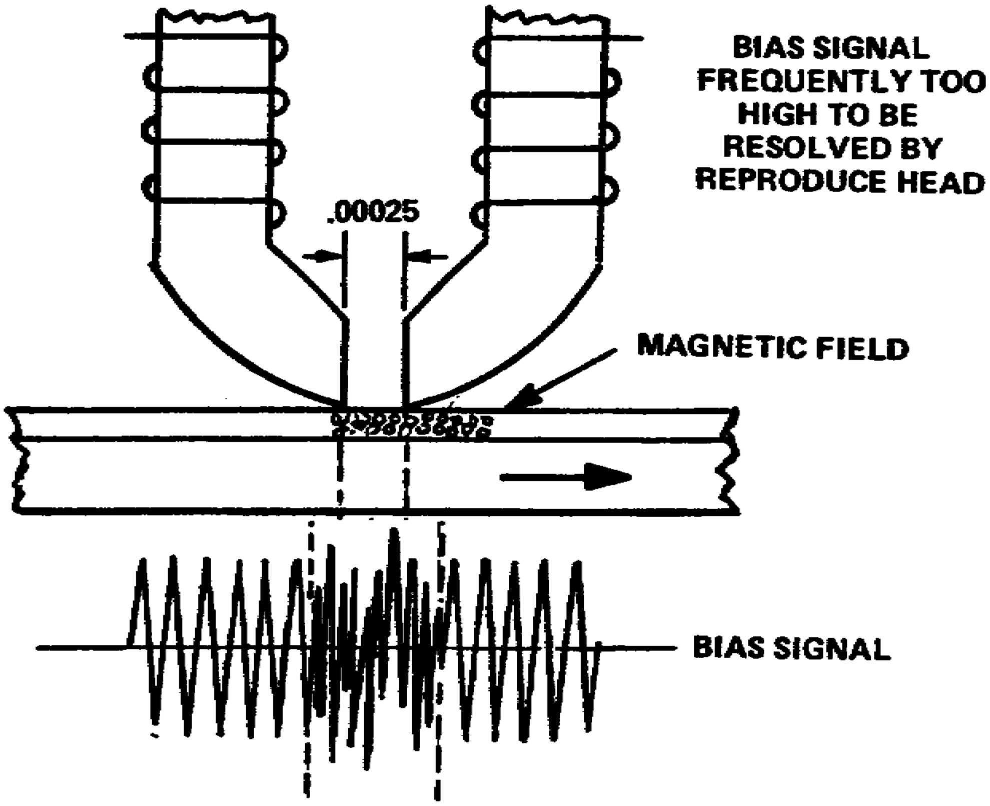

Figure 3-34. Reproduce head gap showing an audio signal and bias current combined on the tape

(4) The application of bias current is not an amplitude modulation process as the bias frequency does not enter the recording or playback process. Since the wavelengths of the bias frequencies are small, they are not picked up by the playback head.

(5) During playback, the magnetized surface of the tape passes over the gap of the reproduce head which causes magnetic lines of force to be induced into the core of the head, thus generating a voltage. This playback voltage is proportional to the rate of change and is therefore, dependent upon frequency.

(6) Equalization maintains a uniform frequency response in both recording and reproducing circuits. The playback circuit takes care of low frequency equalization and the record circuits take care of high frequency equalization. The recording equalization compensates for record head core losses and self demagnetization of the short wavelength approaching the head gap.

(7) High frequency equalization, sometimes known as preequalization, increases the amplitude of frequencies above 1000 Hz and obtains a greater signal-to-noise ratio during reproduction. Low frequency equalization, sometimes known as post-equalization, has an inverse frequency characteristic to that of the preequalization.

Learning Event 3:
OPERATE REEL-TO-REEL TAPE RECORDER

1. To operate as reel-to-reel tape recorder follow the following steps:

a. (1) Check output/input connectors.

(2) Check external sound source.

(3) Check power cord for defects.

b. Put on and thread tape through record/playback heads.

CAUTION: DO NOT STRETCH TAPE BY PULLING.

c. Select tape speed that optimizes tape to be used.

d. Check equalization position if not automatically selected.

e. Check and position the record/safety override, (if applicable).

f. Check record channels or format (mono or stereo).

g. Input external sound sources (audio console, microphone, or turntable).

h. Set record level at 0db or 100% VU meter from external sound source.

i. Press PLAY and RECORD buttons and record sound sources.

j. Press STOP button and stop tape.

k. Press REWIND button and rewind tape to original start point.

l. Check output selector.

m. Adjust console board so that tape can be monitored.

n. Press PLAY button and check the following:

(1) VU meter for playback levels.

(2) Any distortion or “drop outs” (intermittent loss of audio).

o. Press STOP at end of recording.

p. Press REWIND until tape is off take-up reel, remove tape from machine.

q. Label tape reel and container with production data, and store tape.

r. Turn off equipment.

2. Operator's maintenance of reel-to-reel recorder/reproducer includes the following:

a. Disconnect and inspect the power cord for serviceability.

b. Dust all exterior surfaces with a soft cloth.

c. Replace any damaged outer parts, with spare parts.

d. Clean the drive system, including the capstan and pinch rollers, using denatured alcohol and cotton swabs.

e. Clean the recording and reproducing heads and tape guides, using denatured alcohol and cotton swabs.

f. Visually inspect the tape tension arm. If it is damaged, replace it in accordance with the manufacturer's manual.

g. Visually inspect the tape tension mechanism. If it is damaged or not functioning properly, report it to the proper personnel.

h. Check the input and output connectors.

i. Perform an operational check of the audio recorder/reproducer, in accordance with the manufacturer's manual, using a prerecorded audiotape.

Learning Event 4:
OPERATE AUDIOTAPE CARTRIDGE AND CASSETTE UNITS

1. Audio production work consists of short announcements, news inserts, and other types of brief information that may accompany a slide, film or videotape insert.

a. Tape cartridge units can hold and play back several cartridges individually or simultaneously. A cartridge, which contains an endless tape loop (continuous loop), is inserted into a unit. The cartridge tape provides an immediate playback without wow and flutter.

b. To record on a cartridge unit, the following steps should be followed:

(1) Conduct preoperational check of cartridge machine in accordance with the manufacturer's manual.

(a) Check output/input connectors.

(b) Check external sound sources.

(c) Check power cord for defects.

(2) Connect cartridge recorder/reproducer to power source and turn-on (audio console, microphone, or turntable).

(3) Insert blank cartridge into cartridge slot.

NOTE: Lamp lights under STOP button to indicate proper position.

(4) Press the RECORD switch momentarily.

(5) Provide input from external sound sources (i.e.; test tone, music, or voice).

(6) Set input control or program level to 0db or 100% on the VU meter.

(7) Press the START button while observing performance measured.

(8) Press the STOP button.

NOTE: To record more than one element on the same cartridge, press the START button, again observing performance measure; otherwise the tape will continue to run until it is stopped by the automatic cue tone.

(9) Place audio console in PLAY BACK and review material for compliance to local broadcast standards.

(10) Let tape run, after review, to cut point.

NOTE: Tape will automatically stop at cue point if not interrupted.

(11) Remove the tape from recorder/reproducer.

(12) Turn power off.

c. Cassette units are similar to the cartridge system except that the tape is contained in one inclosure that has two reels (one supply and one take up).

d. All professional recorder/reproducer units have five controls that regulate tape motion. They are:

(1) Play - moves the tape at a designated speed.

(2) Record - activates the erase and record heads.

(3) Fast forward - advances the tape at high speed.

(4) Reverse - rewinds the tape at high speed.

(5) Stop - brakes the reels to a stop.

e. Functional operating procedures for the audiotape cassette recorder/reproducer are:

(1) Connect and install all accessories required in accordance with manufacturer's manual.

(2) Turn power switch to ON.

(3) Check operating indicators and battery levels.

(4) Set playback volume controls.

(5) Activate the PLAY mode 10 seconds.

(6) Set input levels to 0db, (100%) by depressing the PAUSE, PLAY and RECORD buttons.

(7) Reset the counter to 000.

(8) Activate Record mode by depressing the PAUSE button and record input source.

(9) Activate STOP mode.

(10) Activate REWIND mode until the counter reads 000.

(11) Activate PLAYBACK mode to check recorded input.

(12) Activate STOP mode and rewind.

(13) Activate EJECT mode and remove cassette.

(14) Turn power off.

f. Operator's maintenance of audiotape cartridge and cassette units

(1) Disconnect and inspect the power cord for serviceability.

(2) Dust all exterior surfaces with a soft cloth.

(3) Replace any damaged outer parts, with spare parts.

(4) Clean the drive system, including the capstan and pinch rollers, using denatured alcohol and cotton swabs.

(5) Clean the recording and reproducing heads and tape guides, using denatured alcohol and cotton swabs.

(6) Visually inspect the tape tension arm. If it is damaged, replace it in accordance with the manufacturer's manual.

(7) Visually inspect the tape tension mechanism. If it is damaged or not functioning properly, report it to the proper personnel.

(8) Check the input and output connectors.

(9) Perform an operational check of the audio recorder/reproducer, in accordance with the manufacturer's manual, using a prerecorded audiotape.

PRACTICE EXERCISE

1. The most frequently used prerecorded sound is which one of the following?

    a. Records
    b. Audiotape
    c. Videotape
    d. Laser disc

2. To play a 45 RPM record, broadcast turntables need no modification.

    a. True
    b. False

3. How far counterclockwise would you turn a platter to allow sufficient lead in for a 45 RPM record?

    a. 1/4 to 1/2 turn
    b. 1/3 to 3/4 turn
    c. 3/4 to 1 1/2 turn
    d. 1 1/2 to 1 3/4 turn

4. The operator can "slip cue" a record but should not stop or reverse the revolving plate.

    a. True
    b. False

5. Which function of the recorder causes an erase signal to be fed to the erase head and clears any previously recorded signals?

    a. Reproducing
    b. Editing
    c. Recording
    d. Playback

6. What type of cleaning fluid is used to clean the reproducing heads on reel-to-reel tape machines?

    a. SW 40
    b. Warm soapy water
    c. Denatured alcohol
    d. Solvent

7. High frequency equalization is also known as ______________________________.

    a. Pre-equalization
    b. Balance
    c. Cover modulation
    d. Volume control

8. A tape recorder that has a frequency response up to 15,000 cps requires a bias frequency of how many KHz?

    a. 60 KHz
    b. 80 KHz
    c. 100 KHz
    d. 120 KHz

9. Which tape unit is automatically stopped by a cue tone?

a. Cartridge
b. Cassette
c. Reel-to-reel

10. Which tape unit uses the endless loop tape principle?

a. Reel-to-reel
b. Cassette
c. Cartridge

11. To conduct a preoperational check of a cartridge machine, check power cord for defects, external sound source, and what else?

a. Check recording mode functions
b. Check input/output connectors
c. Check playback controls
d. Check rollers for cleanliness

ANSWERS TO PRACTICE EXERCISE

1. A
2. B
3. B
4. A
5. C
6. C
7. A
8. B
9. A
10. C
11. B

LESSON 4
EDIT, SPLICE, AND DUPLICATE AUDIOTAPE

TASK

Describe the characteristics, editing and duplication of audiotape.

CONDITION

Given information on the characteristics of audiotape and procedures of editing and duplicating audiotape, the limits of hearing and the storage and handling of records and tape.

STANDARDS

Demonstrate competency of the task skills and knowledges by responding to the multiple-choice test covering methods of editing and duplicating audiotape.

REFERENCES

None

Learning Event 1:
DESCRIBE AUDIOTAPE

1. Magnetic tape characteristics. All magnetic tapes consist of a coating, or emulsion, permanently bonded to a plastic base. The coating contains microscopic oxide particles that can be magnetized by the electromagnetic recording head. The base film, which determines the mechanical properties of the tape, is the physical support for the coating. Holding the particles to the base material is a binder composed of a highly complex blend of resins and plasticizers. It binds all the oxide particles together into a strong but flexible coating and plastic base.

a. In most cases, it is easy to distinguish the shiny base side from the dull coated side of the tape. However, some tapes, particularly cassettes, have a polished coating that is nearly as shiny as the base.

b. Open reel tapes are always wound with the coated sides facing the hub. In cassette and cartridges, the coating faces out because of the tape path arrangements.

2. Tape qualities. There are three main qualities that magnetic tape should possess:

a. Permeability. The ability of a material to attract a magnetic force.

b. Retentivity. The ability of a substance to retain a magnetic force.

c. Coercivity. The ability of a material to resist demagnetization. With plastic-backed iron oxide tape, permeability and retentivity are both present in more than ample amounts. However, this kind of tape lacks coercivity when recorded at lower speeds. Chromium dioxide tape is a good example of a highly coercive magnetic tape. At lower speeds this tape produces a much higher quality signal than iron oxide tape.

3. Frequency response. In addition to magnetic and mechanical properties, magnetic tape must have electroacoustic properties. It is important that there be low noise, and that frequencies on the tape be reproduced without variation.

a. Head to tape relationship: The materials used in making tape heads differ from those used in making magnetic recording tape. The magnetic recording head must concentrate its magnetic field effectively while the iron oxide tape must be able to absorb and retain the magnetic influences subjected to it.

b. Contact between the sensitive (dull) side of the tape and the tape must be as close as possible. Unsteady contact caused by "lumpy" oxide, dirt, or dust causes a loss of frequency response.

c. In recording, the iron oxide particles on the plastic backing arrange themselves in a pattern of vertical stripes. The distance between each stripe is determined by the frequency of the magnetic vibration, influenced by the sound signal.

4. Tape uniformity. Uniform thickness of the magnetic layer is a primary requirement for a good sound tape; any unevenness results in the recording and reproduction of a poor quality signal. The iron oxide must be spread evenly over the entire surface of the tape and be thin enough to allow for equal sensitivity to high and low frequencies. The iron oxide particles must be very fine if rapid changes in frequency are to be recorded.

5. Advantages and disadvantages.

a. Advantages. Magnetic recording tape provides the audio industry with several positive features:

(1) Easy to use: Bulky recording equipment is not needed. Recorders no larger than the palm of ones hand permit recording at locations where other methods cannot.

(2) Information storage: Several tracks of information can be recorded on magnetic tape and be easily retrieved.

(3) Immediate playback: Recorded information can be played back immediately after it has been recorded without special processing.

(4) Physical makeup: Magnetic tape can be cut, spliced, and easily repaired if broken.

b. Disadvantages. There are a few drawbacks of magnetic recording tape:

(1) Effects of external magnetism: Magnetic recording tape is greatly affected by external magnetic fields from motors and electromagnets which can cause partial or even total erasure of the recorded information.

(2) Temperature and humidity variations: Magnetic recording tape expands and contracts with extreme variations in temperature. The plastic backing becomes brittle when cold, and stretchable when hot. Moisture retained within the iron oxide particles causes some of the particles to fall off, resulting in tape head clogging.

(3) Dirt, dust, and oil: Dust or dirt settling on the tape can cause loss of frequency response or loss of the entire signal. Introduction of body oils from the fingers through frequent handling can also cause loss of frequency response.

Learning Event 2:
EDIT AND SPLICE FORMATS

1. Simply stated, editing is removing or adding segments to a tape recording. It may consist of a simple removal or adding of complete sections, combining sections, or cutting out and replacing a single word. In a well-edited tape, there should not be the slightest hint that editing has been performed. No pops, clicks or extraneous sound, should be heard. There should only be the smoothly played back recording.

a. Types of formats. The only kind of format that may be edited is the single track format. This is to include any number of tracks that are recorded in the same direction. Therefore, tape recorded on one track of a two-or four-track recorder, may be edited.

b. Sound recognition. Sound on any magnetic medium is made up of variations recorded at a certain speed; for example, when $7^{1}/_{2}$ ips is played back at the 7 1/2 ips speed, the reproduced sound is almost exactly the same as the original. If sound recorded at one speed is reproduced at a slower speed, the pitch of the sound is lower and the sound is dragged out in articulation. For example, if a tone frequency of 1,000 Hz was recorded at the speed of 30 ips and played back at 15 ips, it would be heard at a frequency of 500 Hz (at 7 1/2 ips the frequency would be 250 Hz and at 3 3/4 ips it would be 125 Hz).

(1) The ability to recognize sounds recorded at slower speeds should begin with those easiest to recognize; the fricative sounds of f, v, s and z, and with the hard sound of t, b, p and d. At very low speeds there is a similarity between the sound of t and s. The beginning tongue click constitutes almost the only differentiating sound.

(2) After acquiring the ability to recognize the hard consonant

sounds, practice on the vowel and dipthong sounds; those of a, e, i, o, v, and their various combinations.

(3) Learn the difference between sh and z sounds and similar sounds. After a while, recognition of sounds will become automatic and there will be little difficulty in recognizing them at lower editing speeds.

c. Editing connected speech. After learning to recognize and edit speech sounds of all kinds in every conceivable combination, the task of editing connected speech can be undertaken.

(1) One of the more fascinating aspects of tape editing deals with spoken human reactions and sensibilities. A person learns to sense the “mood” in a series of words. One senses all the emotions in the way words are spoken. It is true the human voice can be the most superb musical instrument ever known. However, it is very easy to ruin a fine expression by careless editing; the mood of the edited version must match that of the original unedited expression. In condensing a speech by editing, don't jump from a sentence that is spoken in one mood to another sentence--or part of one--that is in an entirely different mood. Even if the edited version makes perfect sense in all other respects, if the moods differ the whole thing appears ridiculous. Very few people remain in exactly the same mood for any length of time.

(2) Moods change, perhaps ever so slightly but still perceptibly, and although these changes are not striking in the original speech, they become immediately noticeable in a crudely edited version where the mood of spoken sounds has not been taken into account while editing. Since the primary reason for tape editing is to improve on the original while maintaining its sense in all respects, the editor must learn to match the mood of the edited to the original speech, to preserve the real meaning and color of the speech.

d. Tempo or pace. Speakers have their own pace in speaking. It may change according to mood but within each mood, generally, there will be a corresponding pace or tempo. Pace includes the spacing between words or speech sounds as well as that between the words themselves. Allowance must be made for breathing sounds, for “ers” and “ahems” and all the countless vocal tricks that people play with their speech apparatus. There is a single way to maintain a speaker's pace which is also an efficient way to edit speech; imitate the speaker's tempo and then say the desired edited version out loud, before cutting the tape. If the speech can be said easily and in the proper tempo and rhythm, it can then be cut to sound the same way. Always cut from sound to sound. Do not cut in the middle of a so-called “quiet” tape unless it simply cannot be avoided. This rule is based; upon the fact that when cutting from sound to sound, the speakers natural pace can be more easily maintained. A tape editor must retain meaning and at the same time create a natural sounding tape. If it sounds natural, it is right.

e. Continuity of background sounds. When there is a repetitive sound in the background of the speech being edited, treat it the same as a musical accompaniment and keep it in the same rhythm in the edited version as in the original. If not kept in the same rhythm, the listener becomes immediately aware of a discontinuity of rhythm and the impact of the piece is lost.

f. Word inflection problems. When pronouncing words, inflection always gives them definite meanings which apply only in the context in which those words are used. Inflection poses a problem in editing that can be almost insurmountable. It is very difficult to use a heavily inflected word or phrase except in its natural context. Speech may be cut abruptly after a heavily inflected word by these rules of exception:

(1) A sudden noise (like a cough or click).

(2) Sudden applause.

(3) Another speaker interrupting.

(4) The safest way to edit speech complicated by inflections is to avoid cutting after an inflection unless the next word can be voiced naturally. If it cannot be voiced naturally, interrupt by the insertion of some short extraneous, but believable sound.

g. Edit from sound to sound. When editing, cut from the beginning of one word to the beginning sound of the next word to be retained. For example, in this sentence: "Editing according to the rules we are following is not difficult", the obvious ways to eliminate the qualifying phrase "according to the rules we are following" would be to cut from just before "according" to just before "is". There is another way to edit this sentence; that is to cut in the middle of the "ng" in "following". This cutting within sound is a professional refinement in editing both speech and music. The technique is used frequently in many cases where a speaker garbles a word and abruptly corrects himself. It can also be used to correct a bar or two of music by inserting the corrected piece, recorded after the original was completed.

(1) In using normal editing technique to correct a mispronounced or garbled word that is immediately corrected by the speaker, one cannot produce a smooth edit. It will sound edited because the piece is left with a heavily accented word (the correction), but with no indication why. By cutting within sound edit from the good part of the mispronounced word to the accented part of the corrected word.

(2) An example of this method: "The President returned to Wash lig -Washington - by train." If one cuts from the middle of the "sh" sound in "washlig" to the same point in "Washington", there will be a perfectly natural "Washington" with normal accent and emphasis. By cutting within sound, the possible addition of editing noise is avoided. Any noise created by the splice is drowned out in the sound.

Learning Event 3:
USE THE LIMITS OF HEARING IN EDITING

1. The human ear, when it has not been desensitized by noise or extreme age, is an extremely sensitive apparatus. Most people can never appreciate how sensitive it is. At its point of greatest sensitivity, approximately 3,000 cycles per second (cps), it is almost sensitive enough to detect noises caused by the collision of particles of air as they bump in space. Hearing declines in sensitivity at both the low and high frequency ends of the audible spectrum, estimated to extend from 15 cps to upward of 20,000 cps.

2. When editing recordings made against changing background sounds, make sure that everything recorded on the tape is heard clearly. An editor tires easily during a long spell of editing and, once he has become affected by hearing fatigue, it can cause him to make errors he would not have made under normal conditions. There are two good ways to avoid this fatigue; stop and rest frequently, and adapt the speaker amplifier to reproduce sound in a particular way.

3. Learn to recognize the differences between sounds heard at low frequencies. Should both the loudspeaker and one's hearing be inefficient at low frequencies, as they generally are at low volume, one will have to strain to hear, or else turn up the volume or sound considerably. Either way, a person will rapidly become fatigued. When the intensity of sound is raised almost to the point of hurting, the recognition of pitch changes. To cure this, incorporate into the playback system a low frequency boost. Highs must be heard clearly at normal volume in order to tell the difference between a “t”, “d”, or “s” or similar sounds. This necessitates a slight high boost. Thus, the hearing (if room acoustics are average), is furnished with sound that is high at both ends and average in the middle.

a. In order to make hearing less fatiguing, there must be compensations for deficiencies in both hearing and loudspeaking. It is best to have variable control so there can be the exact amount of boost to prevent hearing fatigue.

b. Remember that this boost is intended for use only in editing--do not employ it for any other purpose. Remember also that excessive bass boost can induce fatigue as quickly as lack of bass boost. If the sound balances so that it makes comfortable listening for long periods of time, then it is just right.

4. Changing noise to music. One useful quirk of hearing has to do with what is heard as noise and what is heard as musical sound. Sounds that rise gradually and fall away gradually are musical sounds and are not disturbing. Sounds that begin or end too abruptly disturb hearing and the beginnings and ending are heard as noise. It takes a definite time, varying with individual perception, of course, in which to recognize a sound as musical in nature. If the time is too short for recognition to take place, noise is heard. If the time is much shorter nothing is heard. To apply this observation in tape editing, use a diagonal cut in splicing tape in order to make sound start and stop gradually. For example, in splicing tape a speed of 15 ips, a 45-degree

angle cut makes the splice 1/60 second long. If sound is cut at this same speed of 15 ips at a 90-degree angle, there would be two possible reasons to hear noise: as explained previously, the possible accumulation of iron dust in the space between the tape ends being read as noise by the play gap or the difference in voltage between the bias voltage on each tape end; and the too rapid onset or cessation of sound being heard as noise by the hearing system.

5. Persistence of hearing. The persistence of hearing is another of the facts of hearing that has led to the development of a very interesting technique in tape editing. Persistence of hearing is akin to persistence of vision, which has made motion pictures possible, but the time it persists is not nearly as long. It takes a definite time to hear sound as musical and that fact must be used in editing tape. Next it will be explained how to use persistence of hearing in tape editing.

a. With normal hearing, at sound intensities commonly encountered, the persistence effect lasts but a short time. Experience indicates it cannot be depended on for much longer than .03 to .06 seconds, which is perfectly normal and, in fact, desirable. Music, as known, could not exist if persistence time were much longer. How difficult it would be to understand anyone if the sounds of his speech persisted in one's hearing and blended together in an incomprehensible mixture. Persistence time varies with loudness and with the pitch of the sound. If the sound is more intense, persistence will be longer. The effect on hearing of persistence may be used in editing, sometimes to avoid rerecording and still get a natural sounding transition from one tape to another. For example, a person may wish to eliminate commercial or other messages from a recorded program. In these programs, there is usually a managed round of applause, followed by the message, which in turn is followed by the resuming program. If the tape is cut immediately and spliced to the first sound of the resuming program materials, leaving no space between, the applause will seem to blend into the following sound. In fact, when this operation is properly performed, it will sound as though the applause continued for a split second underneath the following sound. This can be called a “persistence blend.” Of course, there are many conditions where it may be difficult to create a natural sounding persistence blend.

b. All the factors discussed previously--mood, pace, level, background--effect the creation of this kind of edited effect. It would not do, obviously, to cut from a laughing to a serious voice, or from an extremely rapid pace to a very slow one of the same speaker. Moods must agree. For instance, if laughter is to be persistence blended to a following voice, the latter must be in a jovial mood or be saying something amusing. A good deal of practice in making these blends will show exactly what can be done by means of persistence blending and what cannot be done.

6. The editing tick. The editing tick is in the same category as persistence blending. It is useful when there is either no time for dubbing (rerecording) or no equipment available for that purpose, and when there is a noticeable difference between background sounds on two sections of tape that have to be joined together. The insertion between two tapes of any slightly shocking sound that is in character--a cough, a mike noise, a door slamming--provided it has been recorded in the same background as either of the two

tapes, will cause the listener to forget the difference in backgrounds and accept the sequence as perfectly natural. The intervening noise has shocked the hearing system into forgetting temporarily everything before the shocking noise, so he does not notice a background change at all.

7. In utilizing any of these techniques, do not consider the job well done unless the sequence can be listened to critically without noticing the transition. It must sound natural--as if it happened that way. An imaginative editor, for example, in joining heavy applause to a following voice in a persistence blend, might cut part way into the first following word, on the assumption that if the sequence had actually happened the way it was edited, part of the first word would have been drowned out by the applause.

8. Effects of editing too closely. When different sounds are spliced too closely together, a third sound will be heard which may be a puzzle. It may be caused partly by transient vibrations in the physical ear, partly by hearing persistence, and partly by lack of recognition time. In any event, this garbled sound can always be eliminated by the insertion, at the splice, of a minimum of 1/60th of a second of tape from the same background of either of the two tape segments spliced together. If a person gets into the habit of saying the portion to be edited, the inserting operation to remove a third sound will not be necessary.

9. Program timing. Timing of tapes may be important before duplicating. It may be necessary for a program to last an exact amount of time. Sometimes, this means that more program time should be added or time subtracted. The method for doing this is to start by timing the original tape. Decide which areas are to be eliminated and subtract them from the total. Exact timing can be achieved by fadeouts at opportune moments, addition of narration; and by using themes or sound effects as fillers.

a. Timing tape may be used to insert exact pauses of time.

b. To use this timing tape, determine the length of pause needed. Use the markings on the timing tape to cut the tape at the desired time. Insert by splicing into the sound.

Learning Event 4:
USE EDITING, SPLICING, AND DUPLICATION TOOLS AND PROCEDURES

1. Editing and splicing tools.

a. An "editing block" is simply a block (metal or plastic) with a channel cut in it to hold the tape and cutting grooves at 45 and 90 degrees. The editing block can be fixed firmly to the table by using bolts or screws (fig 4-35).

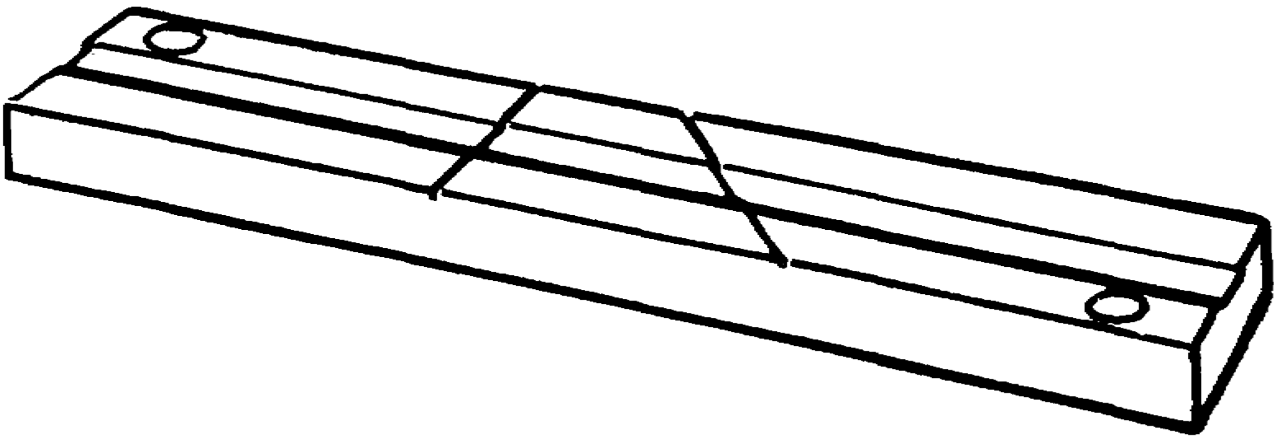

Figure 4-35. The EdiTall tape editor

b. A single-edged razor blade is the safest tool, although a double edge can be used with a holder.

c. A soft wax or grease pencil for marking editing points. Yellow or white will show up best on the tape.

d. Leader or timing tape (a nonmagnetic paper or plastic), uncoated and marked in intervals of 3 1/4, 7 1/2, and 15 inches. This tape (made of plastic or paper) is used for timed pauses, spacing, visual identification, and as leader at the beginning and end of a recording segment.

e. Splicing tape is supplied in varying widths. The width selected should be based on the splicing requirements. For use with the editing block, a tape slightly narrower than the magnetic tape is performed. This makes trimming unnecessary.

2. Editing your material. Locate the area of tape to be removed. Monitor a broad portion of the area. Hand wind tape across the area until exact location is found.

a. Since sound is represented at the gap of the playback head, once a sound is located on the tape, it must be cut exactly at the playback head gap.

b. On most professional tape recorders, the distance from the playback head to the point where the tape comes out of the headgate is 1 1/2 inches. This is also the distance from the diagonal slot to the straight slot on the editing block.

c. To check this, record a 1000 cycle tone on a tape and splice a length of leader tape to it. Cue the tape to the point where the tone is first heard. The splice should be directly over the playback head gap at this point.

d. Mark the leader where it comes out of the headgate, and place it in the editing block so that the splice lines up with the diagonal slot. The mark should line up with the straight slot. If not, mark the editing block to match the mark made on the leader tape.

e. The diagonal cut across the editing block is designed to cut with a single-edged razor blade. With the tape in the groove ready for cutting, splice the tape by pulling the blade back while holding it firmly down in the slot.

f. Make certain that the razor blade is not magnetized. If it is magnetized, it will magnetize the tape; and a “click” will be heard when that part of the tape is played back.

g. Mark the back of the tape with a grease pencil for cutting. Cut out desired portion and splice the tape together.

3. Splicing material. Magnetic tape can be cut and spliced very easily with present day equipment.

a. Should the tape break, there is a method of repairing it without affecting or losing much of the signal recorded on the tape. Use a splicing block to hold the tape and position it for proper butting. Broken or damaged tape will need to be cut to remove the ragged tape edges. Editing blocks have a diagonal slot at 45 degrees used for splicing tape. This 45-degree angle is necessary to prevent popping noises that would be heard at normal tape speeds. The ends of the tape must butt together perfectly, so there will be no exposed adhesive to accumulate dust, dirt, and oxide particles.

b. More important, the adhesive used on the splice must not “bleed”, or flow from the repair to adjacent turns of tape on the real so that the tape sticks as it unwinds during use. Adhesive that bleeds to the oxide coating of the tape will cause a sound dropout as the bleed passes over the heads, and will gum up the heads. Bleeding and protruding splices are easily avoided by using standard splicing tapes which have a special adhesive that will not run or bleed under conditions normally encountered in use or in storage. Never use cellophane tape, as the adhesive used on cellophane tape runs like a leaky faucet.

c. Very few editors can work accurately without moving the tape reels by hand. Practice doing this until a word recorded on the tape at normal speed can be recognized. Learn to recognize sounds when the tape is moved at less than normal speeds. The ability to distinguish one sound from another at low tape speed will come in time, but <u>only</u> after much practice.

4. Improper edits. The angle at which the recording tape is cut is not critical as far as strength is concerned as long as the tape is cut on a diagonal and both ends are cut at the same angle. Tape cut with a square end invariably makes a noisy splice. It also takes more of a beating as it passes overheads and idlers, and so it is more likely to fail. Devices with an adjustable cutting angle gives no advantage and may mismatch the tape ends.

a. Splicing devices. As long as the tape ends are cut properly and barely touch, most splicing devices will keep the tape from slipping to one side or falling at an angle. Some really cheap splicers that use a pressure-formed tape channel take on an angle where the tape cutting slot is sliced through the channel. If the joined tape ends are not absolutely flush, an accumulation of oxide flakes and other foreign matter in the gap will make a noisy, weak splice. In editing blocks particularly, carelessness in positioning tape ends can easily produce an overlay ready to snag on just about anything.

b. Ragged edges. In extreme cases, poor splicer design can even damage the tape. Ragged edges in the channel of very cheap blocks can snag and nick the tape. With thin polyester-base tapes, too, there is the danger that a device with too good a grip--plus a careless user--can yank the tape until it stretches or its edges curl (fig 4-36).

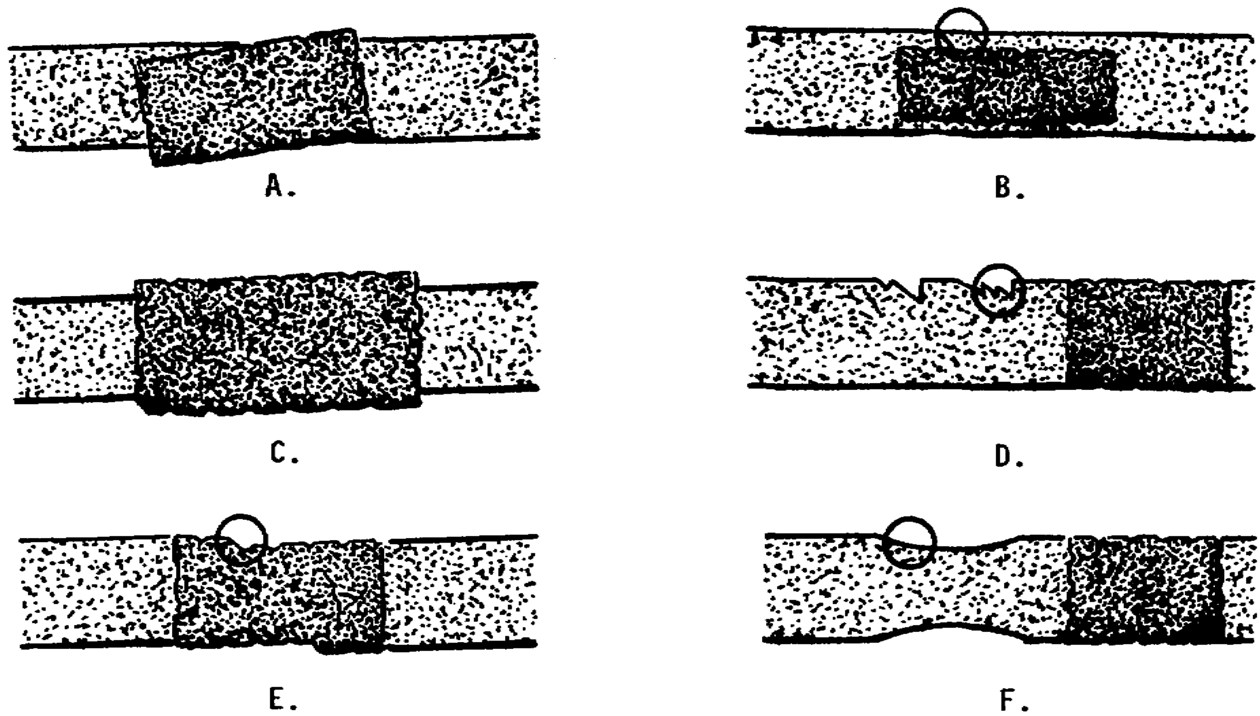

Figure 4-36. Examples of how NOT to splice tape

5. Electronic editing. This type of editing requires the use of two tape recorders/reproducers. Refer to Lesson 3 for functions of a reel-to-reel recorder/reproducer. Use the procedure below to make electronic edits.

a. Make -a copy of the master audio tape to be edited.

b. Locate edit material on the copy tape.

(1) Find edit start point, mark with grease pencil.

(2) Time audio to be edited with a stop watch or clock watch with second hand.

(3) Find edit end point.

(4) Recue tape to edit start point.

c. Locate insert material on tape with editing material.

(1) Find insert start point.

(2) Time audio to be edited.

(3) Find edit end point.

(4) Reconcile times so that time of edit and time of insert are equal.

(5) Recue tape to insert start point.

d. Record silence to edit space.

e. Simultaneously play back insert while recording is in edit space.

f. Play back edited material to check edit and time.

6. Duplication. One way of duplicating tape is to use at least two tape recorders. One plays the original, and the other plays the copy. The basic hookup is from output of playback deck to the input of the recording deck. To interconnect two or more tape machines, it is necessary to impedance match the systems. If impedance matching is not done, there will be a loss of frequency response (a db loss) and possibly unwanted distortion.

a. Duplicating procedures are relatively simple. First interconnect the equipment observing impedances, then set the playback level of the original and the record level for the duplicates. After making sure the decks are in the correct mode, start the tape machines. Start the playback deck by pressing the play button and start the duplicating decks with play and record buttons.

b. Speed copying. Copies may be made at speeds other than the original. For example, the original copy may be 7 1/2 ips. The tape to be duplicated can be made in half the time if both the decks are set on 15 ips. The recording comes out the same.

c. The main disadvantage of speed copying is that it cannot be monitored. When performing speed copying, the levels must be set at the higher speed to prevent over modulation.

7. Modification of Playback Speed. The playback speed can be changed from the original if desired.

a. For example, all that is available for playback is a recorder with speed for tapes a 7 1/2 ips. The original tape is a 3 3/4 ips. The original has to be made into a 7 1/2 ips. To do this, play back the original at twice its original speed. Which is 7 1/2 ips. The recorded copy would be set at 15 ips which is twice the speed of 7 1/2 ips.

b. If the original is at 7 1/2 ips and a copy that will play back at 3 3/4 ips desired, start by playing the original at 15 ips. If recording a copy at 7 1/2 ips, it will be a 3 3/4 ips copy.

Learning Event 5:
STORE AND HANDLE RECORDS AND TAPE

1. Audio transcriptions (record). When handling a record, touch it only by the outermost edges or by the center label. Avoid touching the grooves. Oil and grease on the fingers transfer to the record and attracts dust. Above all, avoid dust! Over a period of time, dust becomes a record's worst enemy.

a. To remove dust, wash the record with lukewarm water. In some cases, use a very mild detergent. After washing, wipe the record clean with a lint free cloth. Because the record may become gummy and collect dust, avoid record preservative and lubricants.

b. Remember to store records on their edge in a vertical fashion. Storing them horizontally causes them to warp over a period of time. Store records in dust-free, moderately dry areas with normal room temperatures.

2. Audio tape/video tape. Audiotape and videotape consist of the same materials; therefore, the storage and handling procedures for audiotape also apply to videotape.

a. When handling tapes, keep these things in mind. First, excessive heat dries out tapes. Second, any excessive contact with a strong magnetic field erases the tape. Third, avoid the rapid rewind or fast forward of audiotape as this causes tape stretching and wearing while damaging the tape heads. Fourth, try to keep fingers off the emulsion side of the audiotape.

b. Use metal cans or cabinets for the storage of tapes. Metal tape cans or cabinets protect tapes from magnetic fields and keep out dust. Avoid extreme variations in temperature and humidity. For long periods of storage, wind audiotape tails out and rewind for playing. Storing audiotape tails out prevents some signal transfer.

PRACTICE EXERCISE

1. The ability to attract a magnetic force in audiotape is a quality in which of the following?

   a. Coercivity
   b. Retentivity
   c. Permeability
   d. Uniformity

2. A primary requirement for good audiotape is which of the following?

   a. Mechanical properties
   b. Physical makeup
   c. Uniformity
   d. Recording/adaptability

3. How would the rules of exception apply to editing a heavily inflected word?

   a. Use of repetitive sounds
   b. Another speaker interrupting
   c. Editing from sound to sound

4. Which track format is the only kind of format that can be edited?

   a. Single
   b. Double
   c. Three
   d. Four

5. If sound is recorded at one speed and played back at a slower speed, the pitch of the sound is higher.

   a. True
   b. False

6. Persistence time varies with the pitch of sound and what other condition?

   a. Balance
   b. Loudness
   c. Articulation
   d. Phonetics

7. Timing tape is a non-magnetic paper or plastic that is used in editing.

   a. True
   b. False

8. What is the distance from the playback head to the point where the tape comes out of the headgate on most professional tape recorders?

   a. 3/4 inches
   b. 1 inch
   c. 1 1/2 inches
   d. 1 3/4 inches

9. Audiotape is cut at what angle to prevent popping noises?

a. 45° c. 75°

b. 60° d. 90°

10. When performing speed copying, the levels must be set at what speed to prevent over modulations?

a. Same speed c. Higher speed

b. Lower speed

ANSWERS TO PRACTICE EXERCISE

1. C
2. C
3. B
4. A
5. B
6. B
7. A
8. C
9. A
10. C