SOUND TECHNICIAN

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INTRODUCTION TO SOUND

1.1 INTRODUCTION

In our daily life, we have conversation amongst ourselves. We hear the chirping of the birds or horn of the vehicles or mewing of the cat. They are of so many types, tones, and levels of sound. In fact, we can recognize a person by just his or her voice. We use sound waves (which are acoustic/ mechanical waves) in talking, while electromagnetic radio waves in sending voice through the radio set or telephone. In this lesson, you are going to study about the details of sound. Let us learn about the basics of sound, elements of sound, nature, characteristics of sound, principles of sound- propagation, refraction, reflection, transmission and absorption of sound.

1.2 OBJECTIVES

After reading this lesson, the learner will be able to :

- define sound
- discuss the basics, elements, nature and characteristics of sound
- recognize the principles, propagation, refraction and transmission of sound
- illustrate the absorption of sound

1.3 INTRODUCTION TO SOUND

Sound is a result of vibration. The vibration is produced by a source, travels in the medium, as a wave and is ultimately sensed through the ear - drum. Let's try to discuss it better by an activity. A simple activity can be done to show the association between vibration and sound.

Take a aluminum wire about 30 cm (in length) or simply a metallic hanger, such as of aluminum and bend it so as to shape it like a bow. Take a rubber band or an elastic string of sufficient length. You may also use small twig, tie a thread or elastic string such as rubber band to both the ends of bow, such that string remains under tension. Ask your colleague to record that :

(i) If you pluck the string, you can hear some sound. You may have to adjust the curve of the bow to be able to hear the sound. You'd notice that the sound vanishes if you hold the string after plucking. If you look carefully, you can realize that the sound comes only as long as the string vibrates.

(ii) You can check the vibrations. Take a small paper strip (about a cm in length and 2 to 3 mm in breadth), bend it in the middle to form a V and place it over the string. You may try the same with string instruments like guitar, sitar, and ektara or even use powder on percussion instruments like tabla, drum or dhol.

If you leave a little powder or dust on the tabla, and cause the membrane to vibrate, you may be able to 'see' the vibrations. A gentle touch with finger tips will also tell you that vibrations are associated with sound in all these cases. If you strike a steel tumbler with a spoon, hear the sound and then hold the tumbler with firm hand, the vibrations will cease and so will the sound.

Discuss the observations with your friends. Can you now conclude that the sound has an association with vibrations? These vibrations are transmitted in a medium mechanically and that is how sound travels. It travels like a wave. A medium is a must for mechanical waves like sound to travel. We speak and expect to be heard. But it will surprise you to learn that without some aid, we can't converse on moon, as we do here. This is because there is no air on moon (actually there is some but very little) and sound needs a medium to travel. In contrast, we can receive electromagnetic waves from distant stars and artificial satellites in space as electromagnetic waves need no medium to travel. A wave involves a periodic motion, movement that repeats itself. It also transports energy. Let us understand waves better.

What happens if you throw a stone in a pool of water? You will see a disturbance of a circular shape moving, from the point of fall of the stone, outwards. We also observe that the disturbance is made up of a raised ring in water, which seems to travel outward. Soon there is another similar circular feature originating at the same centre and moving outward. This goes on for quite some time. Even though there appears to be a movement of material, actually it is only the position of the disturbance that is changing. This is a wave and is made up of the raised part (crest) and low part (trough). So crest and trough are essential components of a wave. A wave transfers energy from one point to the other without the medium particles moving from one point to the other. Thus wave is clearly different from particle. Sound is a vibration that propagates, as a mechanical wave of pressure and displacement, through some medium. Sometimes sound refers to only those vibrations with frequencies that are within the range of hearing for human or for a particular animal.

Understanding the nature of sound requires observations. We observe a flute playercontinuously shifting fingers, over holes to produce different notes, while playing a tune. Similarly, a sitar player also keeps pressing the string at different points touching different frets (*parda* in Hindi). When you strike an empty and a water filled glass with a spoon and different notes are produced. The science of sound helps us in understanding the reasons behind such things. Besides, the understanding of sound has enabled scientists to devise gadgets which are very useful. These include hearing aids, sound instruments like speakers, sound recording and sound amplifying devices etc. We shall also learn about various technological tools that have been developed to improve communication. By improvement we mean, we can communicate to more people, at greater distances, and with more clarity. Therefore, in a nutshell it can be said that all sounds are vibrations travelling through the air as sound waves. Sound waves are caused by the vibrations of objects and radiate outward from their source in all directions. A vibrating object *compresses* the surrounding air molecules (squeezing them closer together) and then *rarefies* them (pulling them farther apart). Although the fluctuations in air pressure travel outward from the object, the air molecules themselves stay in the same average position. As sound travels, it reflects off objects, in its path, creating further disturbances in the surrounding air. When these changes in air pressure vibrate your eardrum, nerve signals are sent to your brain and are interpreted as sound.

1.4 NATURE OF SOUND

Sound is a longitudinal, mechanical wave. Sound can travel through any medium, but it cannot travel through a vacuum. There is no sound in outer space. Sound is a variation in pressure. A region of increased pressure on a sound wave is called compression (or condensation). A region of decreased pressure on a sound wave is called rarefaction (or dilation).

Representing a wave

We need to describe a friend by name, height, colour, gender for identifying. Similarly, we have to specify some qualities that we shall call parameters, for wave description. A wave is represented in terms of its wavelength, amplitude, frequency and time period.

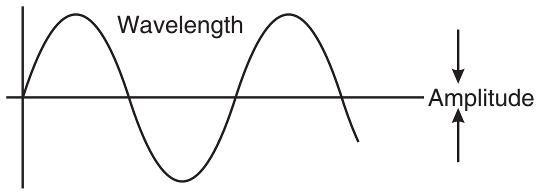

Fig. 1.1: A representation of wave

Consider a simple pendulum . It has a bob, a thread and a hinge to fasten it.

Pick up the bob to some height by pulling it tangentially and drop it . You can observe the bob falls down and rises up to the same height in the opposite direction and then reverses and traces the same path. This phenomenon repeats for long. This phenomenon is called as periodic wave , since a bob traces a point after a regular period of time . If we trace the distance for the lowest height of the bob with time, a periodic wave form will be drawn as follows :

1.4.1 Amplitude (Am)

It is basically a measurement of the vertical distance of the trough(C) or crest(A) of a wave from the average(B) as illustrated in diagram above.

1.4.2 Wavelength (λ)

The distance between adjacent troughs or adjacent crests, measured in unit of length such as meters and expressed by symbol λ (lambda). For longitudinal wave, it will be distance between two successive rarefactions or compressions.

1.4.3 Time Period (T)

This defines the time it takes for one complete wave to pass a given point, measured in seconds (s) (WL = T).

1.4.4 Frequency (n)

The number of complete waves that pass a point in one second , measured in Hertz (Hz). In simple words , the number of times the bob reaches appoint in (say A) in one second determines its frequency or repetitivity .

1.4.5 Speed or velocity (v)

The speed of the wave is proportional to the magnitude of the frequency . Wave speed is defined as the distance travelled by a wave disturbance in one second and is measured in meters/second ms^{-1} or m/s. Speed is scalar quantity while velocity is a vector quantity.

Not all of these properties are independent; one can relate some. Period is inversely related to the frequency. This means if the frequency is high, the period will be low. This is understandable because frequency is number of times a wave completes a set of up and down movements (or a set of crests and troughs) in 1 second. If these occur more frequently, it has to be done in very short time. Mathematically one may say period

$$T = 1/n$$

Where 'n' is frequency. We just said that wavelength is equal to the distance between two successive crusts or troughs. In one second this distance is covered a number of times given by frequency.

So, Velocity = frequency × wavelength

or $V = n \times \lambda$

The waves that produce a sense of sound for living beings are called sound waves or audible waves. Only those waves that have frequencies lying in the range of 16 Hz to 20,000 Hz are audible to human beings. However, this range is an average and will slightly vary from individual to individual. Sound waves with frequencies below 16 Hz are called infrasonic waves and those above 20 kHz are ultrasonic waves. Animals like bats are able to produce and sense waves beyond the range of human audibility and use it for 'seeing' in the dark.

INTEXT QUESTIONS 1.1

1. Which sound wave will have its crests farther apart from each other - a wave with frequency 100 Hz or a wave with frequency 500 Hz?

2. If the velocity of sound is 330 meters per second (ms^{-1}), what will be wavelength if the frequency is 1000 Hertz?

3. What is the approximate audible range of frequency for humans?

1.5 CHARACTERISTICS OF SOUND

A sound can be characterized by the following three quantities pitch , quality and loudness which are explained below:

Pitch is the frequency of a sound as perceived by human ear. A high frequency gives rise to a high pitch note and a low frequency produces a low pitch note. Fig. 1.2 shows the frequencies of some common sounds.

A pure tone is the sound of only one frequency, such as that given by a tuning fork or electronic signal generator. Generally the pure tone part of sound is called fundamental frequency (fo)

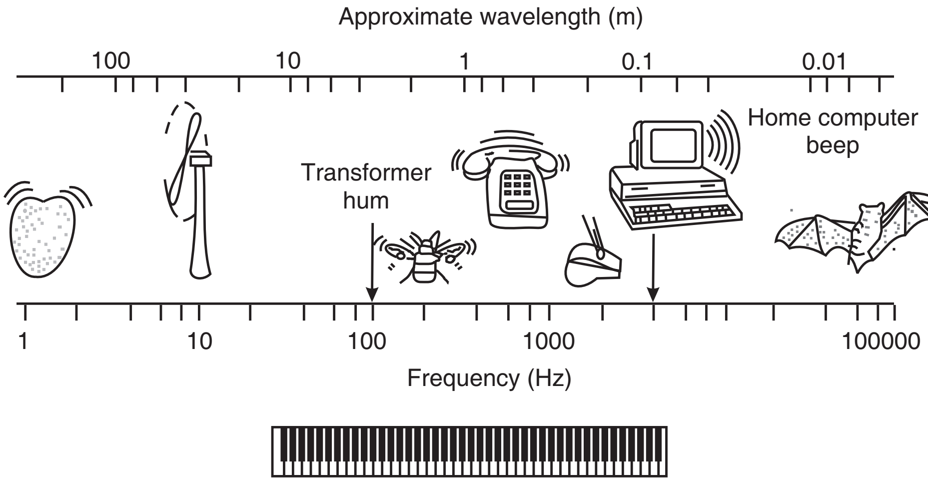

Fig. 1.2: Frequency and Wavelength of Everyday Sound

The fundamental note has the greatest amplitude and is heard predominantly because it has a larger intensity. The other frequencies such as 2fo, 3fo, 4fo, ... are called ***overtones*** or ***harmonics*** and they determine the ***quality*** of the sound.

Loudness is a physiological sensation. It depends mainly on sound pressure but also on the spectrum of the harmonics and the physical duration.

1.6 PRINCIPLES OF SOUND

1. Sound is produced by a vibrating body and travels in the form of a wave.
2. Sound wave travels through materials by vibrating the particles that makes up the materials.
3. The pitch of the sound is determined by the frequency of the wave and loudness by its amplitude.

1.7 PROPAGATION

Sound is a sequence of waves of pressure which propagates through compressible media such as air or water. (Sound can propagate through solids as well, but there are additional modes of propagation). During their propagation, waves can be reflected, refracted, or attenuated by the medium.

All media have three properties which affect the behaviour of sound propagation in them:

1. A relationship between density and pressure. This relationship, affected by temperature, determines the speed of sound within the medium.

2. The motion of the medium itself, e.g., wind. Independent of the motion of sound through the medium, if the medium is moving, the sound is further transported.

3. The viscosity of the medium. This determines the rate at which sound is attenuated. For many media, such as air or water, attenuation due to viscosity is negligible.

Sound waves travel in fluids and solids as longitudinal waves. A longitudinal wave is a wave in which vibration or the displacement takes place in the direction of the propagation of the wave. Sound moves due to difference in pressure. If a sound is produced in air, it compresses the adjacent molecules. Due to the compression, the air pressure increases. This causes the compressed molecules to move in the direction of the pressure that is the direction of the wave. But displacement of the molecules causes fall in pressure in the place they left. If the wave is continuing then another rush of molecules comes in, fills the empty or rarified space. This process is repeated and the disturbance propagates. Thus a chain of compressions and rarefactions is generated due to sound. They travel and transport energy. If there is no medium, then produced sound will not be able to push any medium-molecules and sound will not move. That is the reason why we can't hear on moon; there is no air in moon's atmosphere and sound can't travel.

Table 1.1: Velocity of sound in different materials

Medium	Velocity
Steel	5200 m/s
Water	1520 m/s
Air	330 m/s
Glass	4540 m/s
Silver	3650 m/s

Such difference in the velocities of light and sound means if there is an event in the sky, which produces light and sound both, we shall see the light almost instantly but it will be a while before we hear it. When there is a lightening in the sky, we see the light before we hear the sound. Mechanical wave can be either transverse or longitudinal while the electromagnetic wave is only transverse. The transverse wave is one in which the motion of wave and of the particles are perpendicular to each other. In a longitudinal wave, the motions are in the same direction. The sound wave can be of two types- transverse and longitudinal.

We can try to visualize transverse wave by tying one end of a rope to a hook or peg in a vertical wall (or to a door-handle) and holding the other end such that the rope remains loose. We can demonstrate a transverse wave travelling along the rope if we quickly give up- and down- jerk (or even in horizontal plane) to the rope at our end. We see the wave travelling between our hand and the peg while the points on the rope move perpendicular to the rope and wave. This is a transverse wave, as the particles of the medium move perpendicular to the direction of wave movement. In the example of wave when we throw a stone in stationary water in a pond, it is more complex but here we confine to what happens on the surface. We see that on water surface the wave moves from the centre to the shore. If we see a duck or a small paper boat there, it oscillates up and down with water that is goes up temporarily after which they come back to their mean positions without shifting the position horizontally. That makes it a transverse wave.

In a longitudinal wave, the displacement of the particles and propagation of the wave are in the same direction. For instance, if we blow a horn, speak, or quickly move an object in air we are pushing the air molecules. These molecules, in turn, push the adjacent molecules which impart their energy to the next ones. After losing energy in the interaction, the molecule is back to its original (mean) position. This results in formation of compressions and rarefactions. So it's the compression (or rarefaction) which is travelling and not the molecules. Just like the distance between two successive crests or troughs is a measure of wavelength for transverse waves, the distance between two successive compressions or rarefactions is termed wavelength of the longitudinal wave.

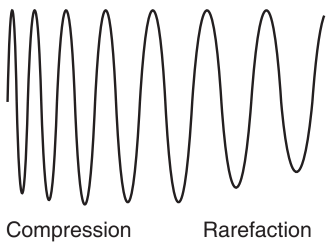

Fig. 1.3: Formation of rarefactions and compressions in air

While transverse waves form only in fluids (air and liquid), the longitudinal waves can form in all the three media viz. solid, liquid and gas. One way to visualize a longitudinal wave is to take a spring, fix it between two ends and then pull or press it on one end along the length. Compressions and rarefactions can be seen moving and rebounding along the axis of the spring.

1.8 REFRACTION OF SOUND

Refraction is the bending of waves when they enter a medium where their speed is different. Refraction is not so important a phenomenon with sound as it is with light where it is responsible for image formation by lenses, the eye, cameras, etc. But bending of sound waves does occur and is an interesting phenomenon in sound.

Notes

1.9 REFLECTION OF SOUND

The reflection of sound follows the law "angle of incidence equals angle of reflection", sometimes called the law of reflection. The same behaviour is observed with light and other waves, and by the bounce of a billiard ball off the bank of a table. The reflected waves can interfere with incident waves, producing patterns of constructive and destructive interference. This can lead to resonances called standing waves in rooms. It also means that the sound intensity near a hard surface is enhanced because the reflected wave adds to the incident wave, giving pressure amplitude that is twice as great in a thin "pressure zone" near the surface. This is used in pressure zone microphones to increase sensitivity. The doubling of pressure gives a 6 decibel increase in the signal picked up by the microphone. Reflection of waves in strings and air columns are essential to the production of resonant standing waves in those systems.

1.10 TRANSMISSION OF SOUND

As a sound wave travels across a room and touches a wall, a reflective wave is produced that will reintroduce a portion of that wave back into the room. The balance of the original sound will attempt to pass through the wall to the adjoining room. The energy that survives this transfer is called Sound Transmission.

1.11 SOUND ABSORPTION

When a sound wave strikes one of the surfaces of a room, some of the sound energy is reflected back into the room and some penetrates the surface. Parts of the sound wave energy are absorbed by conversion to heat energy in the material, while the rest is transmitted through. The level of energy converted to heat energy depends on the sound absorbing properties of the material.

A material's sound absorbing properties are expressed by the sound absorption coefficient, α, (alpha), as a function of the frequency. α ranges from 0 (total reflection) to 1.00 (total absorption) and is determined by :

1. Transmitted energy
2. Converted energy
3. Incident energy
4. Reflected energy

and is determined by :

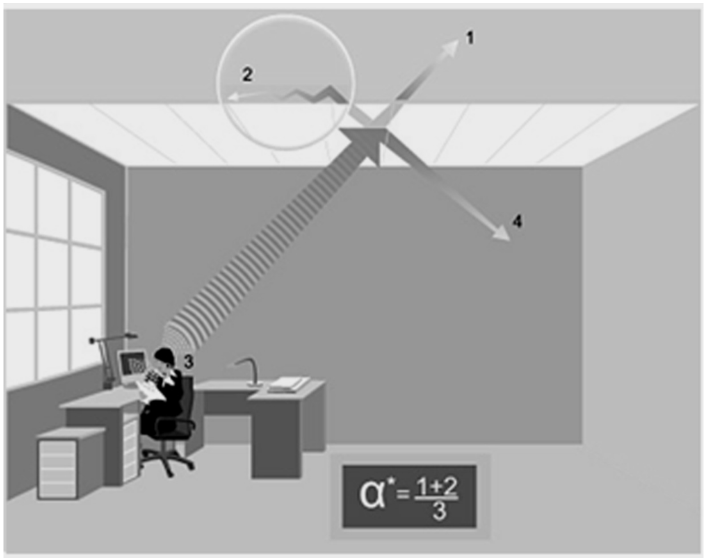

Fig. 1.4: Absorption coefficient

1.12 MUSIC AND NOISE

Music is a set of sound that is pleasing to hear and is not random. It refers to the quality of sound as well as the tune. Noise is random and irritates while music has periodicity whether in beats, or rhythm. For instance, in a song, you'd notice that the same tune is repeated after certain period. After a stanza, the singer comes back to the same tune (combination of notes). If we plot sound pressure with time, we'd notice that it is sweet if it changes in a regular fashion. Noise, in contrast, changes in an irregular fashion and irritates. Sound is evaluated by musicians in 3 terms: quality, pitch and loudness. Two sounds may have the same loudness, may be at the same pitch but can still have different quality/timbre. That is how we can distinguish the sounds from Sitar and guitar even when the loudness and the pitch are the same.

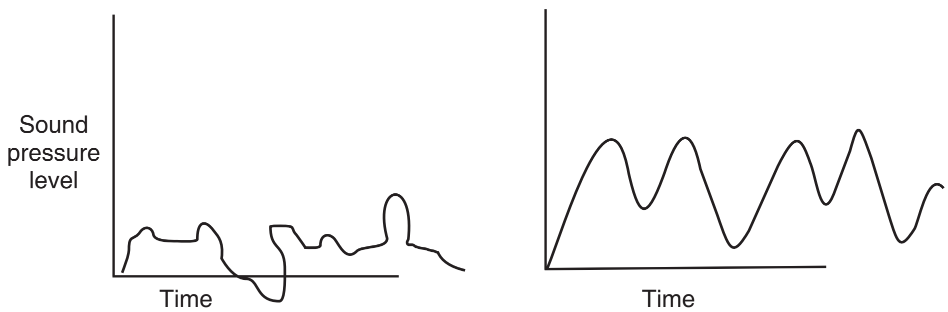

Fig. 1.5 : Graphical representation of changes in sound pressure with time in musical and noisy sound

Notes

1.13 WHAT YOU HAVE LEARNT

- Sound results from vibration and needs a medium to travel, be it gas (like air), solid or liquid. It is faster in solids than in liquids and is the slowest in the gases.
- A wave is described in terms of wavelength, frequency and amplitude. Velocity is equal to the product of wavelength and frequency.
- Noise is random while music is periodic. Music is pleasing to hear but it is also subjective. Sustained exposure to noise and even music at high decibel harms.
- The functioning of musical instruments like Tabla, Sitar and flute (Baansuri) can be understood as vibrations in membranes, strings and organ pipe.
- In telecommunicati/broadcasting they all work through the conversion of sound wave/text into electromagnetic waves at transmission end and reconversion to sound wave/text at the receiver's end.

1.14 TERMINAL QUESTIONS

1. Discuss the elements of sound.
2. Why cannot we hear each other on Moon?
3. Describe two experiments to show that sound has vibrations associated with it.
4. What is the relationship among velocity, wavelength and frequency?
5. What are the differences between longitudinal and transverse sound waves?
6. Will sound move faster in solid or air?
7. What is the basic difference between noise and music?
8. What are the features of sound?
9. Explain the 'transmission of sound'.
10. Illustrate sound absorption

1.15 ANSWERS TO INTEXT QUESTIONS

1.1

1. 500 2. 330 mm 3. 20 Hz to 20 kHz

1.16 REFERENCES

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2. Sound and recording: application and theory: Francis Rumsey and Tim McCommick

2

MEASUREMENT OF SOUND

2.1 INTRODUCTION

You have already learnt about the various elements and characteristics of sound in the previous lesson (1). In this lesson, you will be acquainted with decibels (dB) and hertz (Hz), and their usage, dynamic range and sound pressure level (SPL).

Sound is simply a wave, and its perception by our ears is of some mechanically created waves resulting from the vibration of air molecules. We experience sound, when our ears are excited by vibrations in the air that surrounds us. Sounds occur continuously every day: children crying, people talking, singing or shouting, dogs barking, the sounds of vehicles/traffic, music from radios or from television programmes. These all make up various sound waves.

Since the range of intensities that the human ear can detect is so large, the scale that is frequently used by physicists to measure intensity is a scale based on multiples of 10. This type of scale is sometimes referred to as a logarithmic scale. The scale for measuring intensity is the **decibel scale**. The threshold of hearing is assigned a sound level of 0 decibels (abbreviated as dB); this sound corresponds to an intensity of $1*10^{-12}$ W/m^2. A sound that is 10 times more intense ($1*10^{-11}$ W/m^2) is assigned a sound level of 10 dB. A sound that is 10*10 or 100 times more intense ($1*10^{-10}$ W/m^2) is assigned a sound level of 20 db. A sound that is 10*10*10 or 1000 times more intense ($1*10^{-9}$ W/m^2) is assigned a sound level of 30 db. A sound that is 10*10*10*10 or 10000 times more intense ($1*10^{-8}$ W/m^2) is assigned a sound level of 40 db.

Notes

2.2 OBJECTIVES

After reading this lesson, the learner will be able to :

- explain how sound is measured
- describe loudness of sound and its measurement
- recognise concepts such as Frequency, Dynamic range and Sound Pressure Level (SPL)

2.3 DEFINITION OF DECIBEL

Loudness or volume (also called amplitude) is an important characteristic of sound. It is a perception that is based on a scale between quietness to anything that is not. The explosion of a plastic cover in a closed room might be heard loudly, but the same noise may not even be felt, on a noisy road, which means that loudness is relative.

Loudness of sound is measured in decibels.

The decibel (dB) is a unit, used to measure sound intensity and other physical quantities. A decibel is one tenth of a bel (B), a unit named after Graham Bell, the inventor of the telephone. Its logarithmic scale is convenient to represent the entire range of human hearing. Decibels (dB) is an intricate measurement of sound levels. A sound wave‘s actual amplitude is measureable, however, loudness is subjective concept. So, decibel is not a unit in the sense that a ‘foot’ is. Feet are defined quantities and distance. A decibel is a relationship between two values of power.

What is loud to one person (announcer/anchor) may not be sound loud to another person (announcer/anchor). Sound amplitude is measured in **decibels** (abbreviated dB). It is the volume of the sound. In technical jargon/ parlance, the amplitude is the height of the sine wave (i.e. the visual representation of the characteristics of sound).The higher the amplitude, the louder the sound. Loudness is the human perception of the intensity of sound waves. Our standard conversation has an intensity of around 50 dB, whereas 20 dB is the sound level of a whisper, 75 dB is the noise level in the city. These measurements are expressed in a specific form called dBSPL i.e. decibel sound pressure level.

The decibel of sound pressure level (dB SPL) is taken as a reference for the minimum sound pressure level that the average human ear can detect. The smallest audible sound to humans, is typically 0 dB SPL (hearing threshold). In practice, “dB” often stands for “dB SPL”.

Since the decibel scale is logarithmic, a three-decibel increase in the sound level represents a doubling of [sound] intensity. For example, a normal conversation may be about 65 dB and someone shouting can typically be around 80dB. The difference is only 15 dB but the shouting is 30 times more intensive.

The formula for comparing two values using decibels is as follows:

$$dB = 20 \times \text{Log } B/A$$

where dB is the answer in decibels, A is the reference value to be measured and B is the value that is being compared against the reference.

For example, let 0.0002 microbar (the threshold of hearing) be the reference level. How many decibels above the threshold of hearing is normal speech at a distance of three feet? I have already mentioned that speech at three feet is about 0.3 microbars.

We can use the formula as follows:

$$\begin{aligned} dB &= 20 \text{ Log } (0.3/0.0002) \\ &= 20 \times 3.176 \\ &= \text{approx. } 64 \text{ dB} \end{aligned}$$

The units of decibels can be used to compare many different quantities. In this case, it's comparing the relative loudness of "normal" speech to the threshold of hearing. This is in units called "dBspl", and is written "64 dBspl".

Please note that perception of loudness, is not exactly the same as sound pressure level. To account for the fact that particularly low and high-pitched sounds appear less loud to the human ear, noise is usually measured in A-weighted decibels (dB(A)).When sound waves reach the ear or measuring instrument the resulting change of pressure can be measured. Sound intensity is usually expressed in decibels of sound pressure level (dB SPL).

The table 2.1 lists some common sounds with an estimate of their intensity and decibel level.

Table 2.1

Source	Intensity	Intensity Level	# of Times Greater Than TOH
Threshold of Hearing (TOH)	$1{*}10^{-12}$ W/m^2	0 dB	10^0
Rustling Leaves	$1{*}10^{-11}$ W/m^2	10 dB	10^1
Whisper	$1{*}10^{-10}$ W/m^2	20 dB	10^2
Normal Conversation	$1{*}10^{-6}$ W/m^2	60 dB	10^6

Busy Street Traffic	$1*10^{-5}$ W/m^2	70 dB	10^7
Vacuum Cleaner	$1*10^{-4}$ W/m^2	80 dB	10^8
Large Orchestra	$6.3*10^{-3}$ W/m^2	98 dB	10^9
Walkman at Maximum Level	$1*10^{-2}$ W/m^2	100 dB	10^{10}
Front Rows of Rock Concert	$1*10^{-1}$ W/m^2	110 dB	10^{11}
Threshold of Pain	$1*10^{1}$ W/m^2	130 dB	10^{13}
Military Jet Takeoff	1*102 W/m^2	140 dB	10^{14}
Instant Perforation of Eardrum	$1*10^{4}$ W/m^2	160 dB	10^{16}

Recording level and decibels

Meters measuring recording or output level on audio electronic gear (mixing consoles etc) are almost always recording the AC root mean square voltage (see links to find out about AC and rms). For a given resistor R, the power P is V^2/R, so

$$\text{difference in voltage level} = 20 \log (V_2/V_1)\ \text{dB}$$

$$= 10 \log (V_{22}/V_{12})\ \text{dB}$$

$$= 10 \log (P_2/P_1)\ \text{dB}$$

or $$\text{absolute voltage level} = 20 \log (V/V_{ref})$$

where V_{ref} is a reference voltage.

How to convert dBV or dBm into dB of sound level?

There is no simple way. It depends on how you convert the electrical power into sound power. Even if your electrical signal is connected directly to a loudspeaker, the conversion will depend on the efficiency and impedance of your loudspeaker. And of course there may be a power amplifier, and various acoustic complications between where you measure the dBV on the mixing desk and where your ears are in the sound field.

Table 2.2: Sound pressure and level, together with examples

Sound pressure (micro Pascals)	**Sound level (dB)**	**Example**
200,000,000	140	threshold of pain
	130	riveting on steel plate
20,000,000	120	pneumatic drill
	110	loud car horn at 1m

2,000,000	100	alarm clock at 1m
	90	inside underground train
200,000	80	inside bus
	70	street-corner traffic
20,000	60	conversational speech
	50	business office
2000	40	living room
	30	bedroom at night
200	20	broadcasting studio
	10	normal breathing
20	0	threshold of hearing (@ 1kHz)

The human ear can hear very low frequencies from about 35 Hz to very high frequency sound up to 20,000 Hz. Our ear is very sensitive and can hear a tremendous range of sound amplitudes so the decibel scale is logarithmic. Near total silence is noted as 0 dB, a sound 10 times louder than this is 10 dB, a sound 100 times more powerful (louder) is 20 dB, and so on. Decibels represent the ratio of different audio levels and measure the relative intensity of sound.

Sound, in the ranges of 0 dB (the threshold of hearing) to 120 dB (the threshold of pain), are detected by the human ear, but those sounds which are near and exceeding the high end, can be painful and can damage your hearing. Any sound above 85 dB can cause hearing loss, but it depends on how close the listener is to the sound and how long he or she is exposed to it. The sound at many rock concerts has been measured around the 120 dB range, which explains why your ears often ring for day or two after the show. Table 2.3 gives an idea of the range of loudness measurements encountered in everyday life.

Table 2.3: Range of loudness measurements encountered in everyday life

Power (Watts)	**Power Level (db) Re:10^{-12} watt**	**Source**
20-40 million	195	Saturn rocket
100,000	170	Jet afterburner
10,000	160	Jet engine
1000	150	
100	140	Propeller airplane

10	130	Orchestra, rock band
1	120	Small plane
0.1	110	Blaring radio
0.01	100	Auto on highway
0.001	90	Voice shouting
0.0001	80	
0.000 01	70	Average conversation
0.000 001	60	
0.000 000 1	50	
0.000 000 01	40	Average house
0.000 000 001	30	Very soft whisper

2.4 MEASUREMENTOF FREQUENCY

Formerly, frequency was measured by cycles per second (cps), now, in SI units, the result is measured in **Hertz (Hz),** named after the German physicist, Heinrich Rudolf Hertz. He was one of the pioneers of the science of sound/acoustics and demonstrated first the existence of radio waves.1 Hz means one cycle (or wave) per second.

A sound wave that vibrates at twenty thousand cycles per second is said to have a frequency of 20,000 hertz, the term **kilohertz (kHz)** is often used. It denotes 1,000 cycles per second, so 20,000 hertz could also be called 20 kilohertz. For, example, the fundamental frequency of an average male voice is around 300 Hz. Human speech can range as higher as 9,000 Hz.

Bandwidth: Bandwidth is the range of frequencies; for example bandwith of audible frequencies is 20-20 kHz

2.5 MEASUREMENT OF LEVEL OF SOUND

VU Meter / Meter Gauge (Potentiometer)

VU meter or volume unit indicator (Fig.-) is a metering device that enables the operator to determine what level of sounds going out of the line, relatively. One common type VU meter has moving needle on a graduated scale. Usually the upper position of the scale is calibrated in disables (dB), and the lower portion of the scale is calibrated in percentage.

In audio engineering, a reading of 0 db is 00 percent volumes, or the loudest you want the signal to go. The VU meter is important for consistent audio production work. As discussed earlier, the level of sound or how loud something sounds is very subjective. What is loud to one announcer/ presenter/anchor may not be deemed loud by another. It also depends on how differently their monitor speaker volume has been set in. The VU meter gives an electronically precise reading of volume and it is not subjective.

The accuracy of VU meter sometimes debated /questioned. Occasionally VU meter have drawback indicating about the abrupt increases in volume of the sound signal. Most VU meters are designed to specify an average volume level and overlook the occasional sound boosts. VU meter also tend to exaggerate to the bass (low frequency) portion of the sound. When a sound signal is intense in the bass frequencies, it will perhaps show a higher VU reading than it would if the total sound signal were being correctly read. In spite of these shortcomings, the VU meter remains the best display of volume levels in sound design and audio / radio production. Usually a sound recordist or assistant/ operator should control the signal –when it ranges between 80 percent and 100 percent, approximately. When the needle goes/cross above 100 percent, the signal is designated as in the level of red, because that portion of the VU meter scale is usually indicated with a red line. This is a caution to the sound operator to lower the gain of the console /fader or pot.

Sound signal above 100 percent cause over modulated of sound transmitter or saturation of amplifiers and can cause a distorted sound. Unlike analog equipment, professionals, of digital production equipment, will instantly find that the digital equipment is very risky to record "in the red" level.

Digital equipment are not endured to record at any level above 100 percent and will create distortion in the recorded signal, that exceeds it. Professional and good production practices would be to record the sound in the range around -10 dB, when using digital equipment.

2.6 DYNAMIC RANGE

Dynamic range is a measure of the ratio of the largest signal in a circuit that is capable of handling to the highest amplitude frequency component present in the system. It describes the range of the input signal levels that can be reliably measured simultaneously, in particular the ability to accurately measure small signals in the presence of the large signals. According to this definition the dynamic range may be a useful parameter of any system and is the very important parameter of the measurement system. The dynamic range is mostly given in dB. For example, an 8-bit converter can achieve 60 dB or more dynamic range.

Here is a table of dynamic ranges for some frequently used audio equipment:

Notes

Table 2.4

Audio cassette (standard tape)	35 dB
Standard VHS VCR audio tracks	45 dB
Professional Beta video deck (BVW-35)	50 dB
Sony TCD-D5 cassette, Chrome tape	58 dB
w/ Dolby C on	68 dB
BVW-35 w/metal tape, regular tracks	72 dB
Nagra tape recorders	73 dB
BVW-35 AFM tracks	85 dB
Sony TCD-D10 portable DAT recorder	85 dB
Best quality CD players	92 dB
Hafler P-500 power amp (225 watts)	95 dB
Human ear	120 dB

Sound professionals often use dynamic range to describe the ratio of the amplitude of the loudest possible undistorted sine wave to the root mean square (rms) noise amplitude, say of a microphone or loudspeaker.

The dynamic range of human hearing is roughly 140 dB. The dynamic range of music as normally perceived in a concert hall, doesn't exceed 80 dB, and human speech is normally perceived over a range of about 40 dB.

The dynamic range differs from the ratio of the maximum to minimum amplitude, a given device can record, as a properly dithered recording device can record signals well below the rms noise amplitude (noise floor).

For example, if the ceiling of a device is 5V (rms) and the noise floor is 10μV (rms) then the dynamic range is 500000:1, or 114 dB:

$$20 \times \log_{10}\left(\frac{5\text{V}}{10\mu\text{V}}\right) = 20 \times \log_{10}(500000) = 20 \times 5.7 = 114 \text{ dB}$$

INTEXT QUESTIONS 2.1

(i) Threshhold of human hearing is ....

(ii) Frequency spectrum available for a human ear is ....

2.7 SOUND PRESSURE LEVEL

In the field of acoustic /sound measurements, the sound pressure level SPL is measured in decibels. SPL is the logarithmic ratio of sound pressure (p) referred to a sound pressure $p_0 = 20\ \mu Pa$ (micro Pascal's). Sound pressure p_0 is the lower limit of the pressure, which the human ear can perceive in its most sensitive frequency range (around 3 kHz). This pressure level is known as the threshold of hearing.

When stimulus strength is expressed in terms of sound pressure, the following relationship is used, where P_1 and P_2 are two sound pressures. For studies of hearing, P_2 is taken as the sound pressure at the threshold hearing of a normal listener.

$$dB = 20 \log10 (P_1/P_2)$$

If, for example, the sound pressure from one source (P_1) is is ten times greater than that from a second source (P_2), the difference is 20 dB.

$$dB = 20 \log10 (10/1) = 20 \times 1 = 20$$

The sound pressure of a very loud sound, such as a jet plane, may be one million times (10^6) the pressure of the weakest sound that can be detected by someone with normal hearing; these two sounds differ by 120 dB:

$$20 \log10 (P_1/P_2) = 20 \log_{10} 10^6 = 20 \times 6 = 120 \text{ dB}$$

Human ear' audible sound pressure levels range from 20 µPa (hearing threshold) till 20 Pa (pain threshold), resulting in the scale 1:10,000,000.

Since, using such a large scale is not practical, a logarithmic scale in decibels (dB) was introduced which is also in agreement with physiological and psychological hearing sensations.

Decibel (dB) of sound pressure level (dB SPL) is defined as: 20 log10 p_1/p_0 where p_1 is actually measured sound pressure level of a given sound, and p_0 is a reference value of 20µPa, which corresponds to the lowest hearing threshold of the young, healthy ear. In the logarithmic scale the range of human ear's audible sounds is from 0 dB SPL (hearing threshold) to 120-140 dB SPL (pain threshold) (see 2.5 table).

Notes

Table 2.5

Source/observing situation	Typical sound pressure level (db SPL)
Hearing threshold	0 dB
Leaves fluttering	20 dB
Whisper in an ear	30 dB
Normal speech conversation for a participant	60 dB
Cars/vehicles for a close observer	60-100 dB
Airplane taking-off for a close observer	120 dB
Pain threshold	120-140 dB

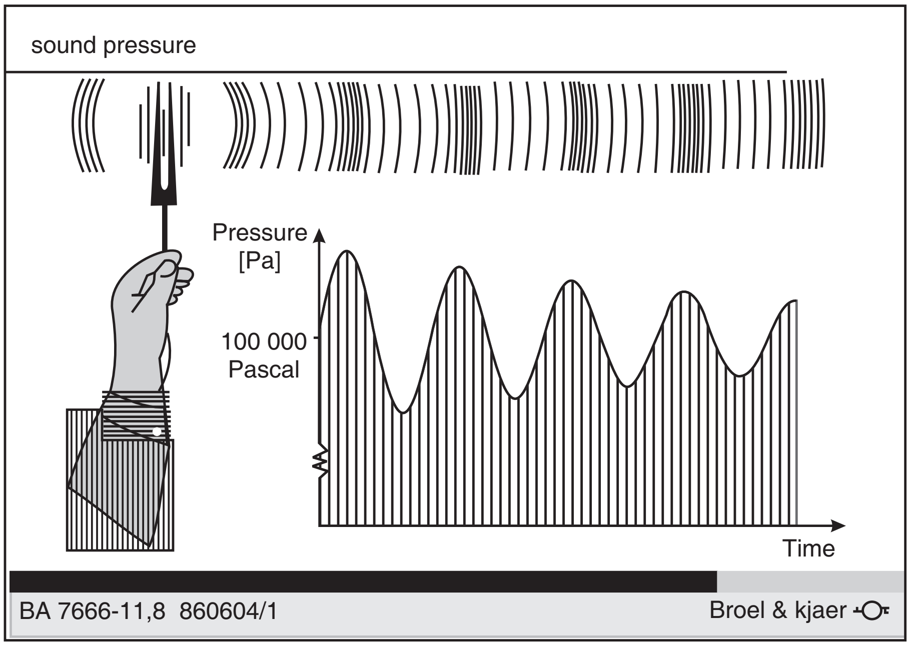


Fig. 2.1

When a sound source such as a tuning fork, vibrates it sets up pressure variations in the surrounding air. The emission of the pressure variations can be compared to the ripples in a pond caused by a stone thrown in the water. The ripples spread out from the point where the stone entered. However the water itself does not move away from the center. The water stays where it is, moving up and down to produce the circular ripples on the surface. In the case of Sound, the stone is the source, the pond is the air, and the ripples are the resulting sound wave.

The acoustic pressure vibrations are superimposed on the surrounding static air pressure which has a value of 10^5 Pascal.

2.7 WHAT HAVE YOU LEARNT

We have dealt with various aspects of units and measurements of sound in this chapter [**2**]. A methodical knowledge about sound waves- it's units & measurements, dynamic range and SPL and its various facets and safety would certainly play a significant role for the understanding of the science and technique of sound design, for our professional as well as for our personal life.

2.8 TERMINAL QUESTIONS

1. Define frequency.
2. Explain the concept of dynamic range discussing its utility also.
3. Define Decibel.
4. Explain sound pressure level.

2.9 ANSWERS TO INTEXT QUESTIONS

2.1

(i) 20 spl

(ii) 35 Hz to 20 kHz

Notes

3

FUNDAMENTALS OF SOUND TECHNOLOGY

3.1 INTRODUCTION

In this lesson, you will learn about different technologies i.e. analog and digital used for processing of sound (audio) signals, and differentiate between them. You will learn about how the respective technology is used, to capture, process and reproduce the signals, i.e. about the analog to digital conversion and digital to analog conversion.

We will also discuss why, today, digital technology is more widely used than analog technology.

3.2 OBJECTIVES

After reading this lesson you will able to

- explain the structure of analog and digital signal;
- explain the process of capturing sound both in analog mode and digital mode;
- explain why digital technology is used more widely than analog technology;
- recognize the process of analog to digital conversion and vice versa.

3.3 ANALOG SIGNAL

In order to record, reproduce, or transmit sound, it first needs to be converted into an electrical signal. The beginning of this process requires a microphone. A microphone has a thin diaphragm that is suspended in or attached (depending on the type of microphone) to a magnetic field. The diaphragm moves back and forth in reaction to the sound waves that pass through it, and that movement within the

Notes

magnetic field creates a small electrical signal, which is an electrical representation of the compressions and rarefactions of the sound wave. The signal is transmitted from the microphone along its cable to be amplified. Microphones generate only a small amount of signal (measured in volts), which is further amplified by using an amplifier (which you will learn in further chapters)

3.4 DIGITAL SIGNAL

A digital signal refers to an electrical signal that is converted into a pattern of bits. Unlike an analog signal, which is a continuous signal that contains time-varying quantities, a digital signal has a discrete value at each sampling point. The precision of the signal is determined by how many samples are recorded per unit of time. For example, the figure 3.1 below shows an analog pattern (represented as the curve) alongside a digital pattern (represented as the discrete lines).

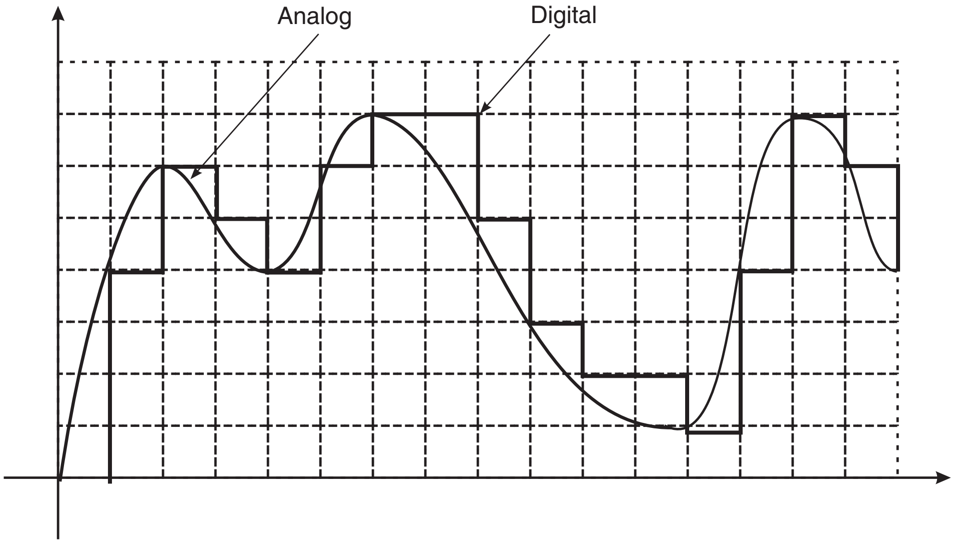


Fig. 3.1: Analog signal and coresponding digital level at different points

A digital signal is easily represented by a computer because each sample can be defined with a series of bits that are either in the state 1 (on) or 0 (off).

Digital signals can be compressed and can include additional information for error correction.

3.5 ANALOG V/S DIGITAL SIGNAL

Sound is recorded by converting continuous variations in sound pressure into corresponding variations in electrical voltage using microphone, this varying voltage is then converted into varying pattern of magnetization (by recording head) on tape or alternatively into a pattern of light and dark areas on an optical soundtrack on film.

Notes

In case of digital recording it converts the electrical wave form from a microphone into a series of binary numbers, each of which represents the amplitude of the signal at a sampling time.

Digital audio has advantages of benefitting from the developments in the computer industry, and is particularly beneficial because the size of that industry results in scope for mass production (and therefore cost savings). Now-a-days it is common for sound to be recorded, processed and edited on relatively low cost desktop equipment.

Any analog signal like sound electrical signal temperature, pressure etc., has infinite values between two limits.

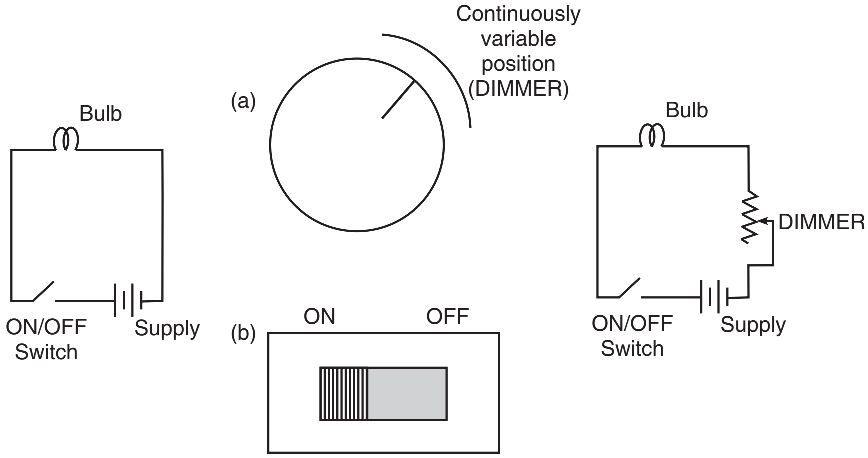


Fig. 3.2: Analog and digital signal examples

Where as in digital, signals have only two distinct values or state i.e. either on/off or 0/1, whenever we light up on electrical lamp (Fig. 3.2 L.H.S), through a simple electrical switch it has only two positions if ON/OFF assign for light and dark, if the same bulb is connected through a dimmer in the place of switch Fig. 3.3 R.H.S), then the fall/rise of brightness seen by human eyes smoothly because the brightness has several intermediate values, which is continuous in nature.

Electrically the analog signal is represented as a varying voltage or current like a sine wave below (a).

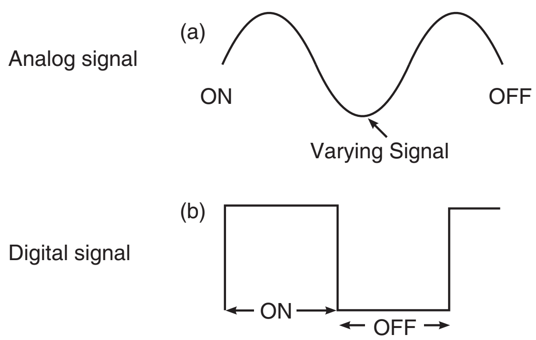


Fig. 3.3

Where as a digital signal is represented by a square wave shown in above figure (b) above. You can see it has two distinct values only either '0' or '1'.

In any system, if the output is similar to the input that means the system is analogue system.

Where as in digital system, the output wave is not similar to the input.

3.6 ADVANTAGES OF DIGITAL SIGNAL OVER ANALOG SIGNAL

(i) Analog signals are prone to be affected by noise, whereas the noise added to digital signal doesn't matter. Because state '1' will be detected as state '1' and state '0' will be detected as state '0', despite addition of noise.

(ii) Analog signal can be recorded and played back but recording and playback process deteriorate the signal in terms of addition of noise, distortion, and change of frequency response.

(iii) In digital signal, the recording and play back process does not add up noise, distortion and change of frequency response.

(iv) With falling rates of computer, components for storage of digital signal has become quite cheap. Whereas analog signals are still being recorded on expensive magnetic medium or optical medium.

(v) Signals once converted into digital form, are much easier to store, manipulate and transport.

3.7 ADVANTAGES OF DIGITAL TECHNOLOGY OVER ANALOG TECHNOLOGY

The quality of digital audio is independent from the medium and depends only on the conversion process. The conversion of audio from analog to digital domain, provides the following advantages.

(i) In Analog system the errors caused by noise, distortion and jitter (due to long cables) cannot be removed fully. Whereas in digital system, these errors can be removed by easier and cheaper means.

(ii) Deterioration due to flutter, print through, drop out noise, alignment errors (change in angle of head) do not occur in digital systems.

(iii) In Digital technology, data can be copied infinitely without generation loss.

(iv) In Digital technology, the data can be accessed instantly whereas Analog technology makes this process complicated and lengthy. Hence Digital technology borrowed RAM (Random Access Memory) and Hard disk technology from computer industry to make DDS (Digital Data Storage).

(v) Editing in digital domain is easier than tape edit.

(vi) Digital audio broadcasting can be carried out in the digital domain with less interference, fading and multipath reception problem as compared to Analog broadcast. Hence the allotted bandwidth can be used more efficiently.

(vii) Maintenance is easier in Digital systems because digital equipment can have self-diagnosis program, built in the system, to point out its own failure or error.

(viii) In Analog – recording, editing and playback is linear in nature.

(ix) In Digital – recording, editing and playback is non-linear in nature. This saves time and enhances creativity.

(x) The cost of Digital equipment is much less than that of analog equipment and the size is also smaller due to integrated circuits & cheap technology.

INTEXT QUESTIONS 3.1

1. Sound signal is first converted into .... signal before processing
   (a) Voltage　　(b) Electrical
   (c) Magnetic　　(d) None of the above
2. Digital signal is stored in which form?
3. Digital recording converts the electrical wave into a series of decimal numbers. True or False.
4. Digital audio has advantages for mass production. True or False.
5. Is it true that an analog signal has infinite values between two limits?

3.8 ANALOG TO DIGITALCONVERSION (A/D CONVERSION)

Analog to digital conversion is a process by which an analog signal is converted to a series of binary digits which represents its value at different sampled point.

First the analog audio signal, which is a time varying continuous electrical voltage or current is passed through an A/D converter (i.e. Analog to Digital Converter). In this process the audio signal is sampled many thousands time per second and converted into a series of samples, which are the snapshots of audio signal taken at the time of each sampling and each such sample is represented by a number. See figure 3.4 below.

A sample and hold circuit can be used for obtaining these samples

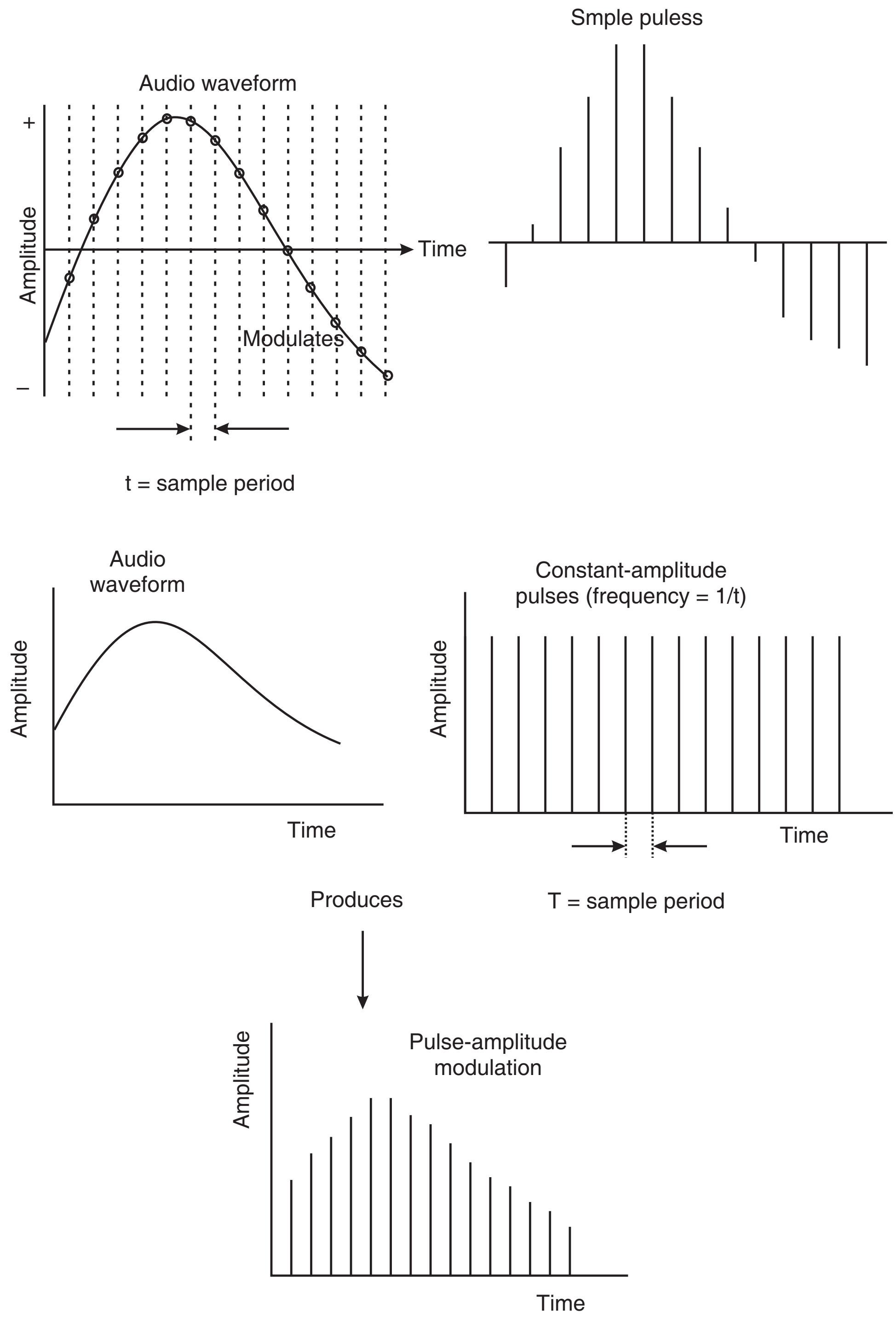


Fig. 3.4: Sampling of analog signal and corresponding pulse levels

The sample pulses represents the instantaneous amplitude signals at each point in time interval, the samples can be considered as "still frames" or "snap shots" of the continuous audio signal, which when put together serially in a sequence for a continuous pulse forms of the audio signal. In order to represent the details of the audio signal it is necessary to take large number of samples per second.

In order to convert analog signal into digital signal it is necessary to measure its amplitude at specific point in time, is called "Sampling" the process of assigning a binary digital value to each measurement called quantization.

Do you know?

The mathematical theory by "Shannon" says at least two samples per audio cycle must be taken to convey necessary information about the signal or in the other words sampling frequency must be at least twice as high as the highest audible sequences in practice.

The quality of A/D conversion is determined by

1. Sampling rate
2. Quantization
3. Dynamic range/Bit depth
4. Dithering

3.8.1 Sampling Rate/Frequency

The choice of sampling frequency determines the maximum audio bandwidth available. Sampling frequency is at least twice the highest audio frequency to be sampled. Since, the audio frequency band extends up to 20Khz, implying the need for a sampling frequency of just over 40Khz for high quality audio work. There are in fact two standard sampling frequency between 40 and 50 kHz, the compact disc rate is 44.1khz and so called "professional" rate is of 48 kHz. These are both allowed in original AES standard (Audio Engineering Society) which sets sampling frequencies for digital audio equipment.

Audio Sampling frequencies

Frequency (kHz)	Application
8	Telephone (speech quality) standard
32	Used some broadcast coding system e.g. NICAM, DAT Long Play mode
44.1	CD Sampling frequency, AES secondary rate.
48	AES Primary rate for professional application basic rate for blue-ray disk (which no longer specifies 44.1Khz as an option)

96	This is optional for DVD-Video, DVD Audio & Blue ray disks for AES secondary rate for high bandwidth application
192	Four times the basic standard rates. Optional in DVD-Audio. 192Khz is the highest sampling frequency allowed on Blu-Ray Audio disks.

3.8.2 Quantization

After sampling, the the sampled/ modulated pulse chain is quantized. In quantizing the range of sample amplitude is mapped into a scale of stepped binary values.

The quantization determines which of a fixed number of quantizing intervals (size of Q) represented. This is done so that each sample amplitude can be represented by a unique binary number in pulse code modulation (PCM).

PCM: It is the form of modulation in which signals are represented as a sequence of sampled and quantized binary data words.

In linear quantizing, each quantizing step represents an equal increment of a signal voltage. Most high quality audio systems use linear quantizing.

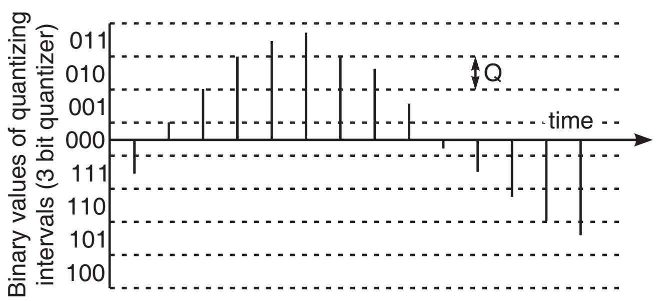
Quantizing error: it is an inevitable side effect in the process of A/D conversion and the degree of error depends on the quantizing scale, higher the scale lower the quantizing error.

- 4 Bit scale offers 16 possible steps (more error)
- 8 Bit scale offers 256 possible steps
- 16 bit scale offers 65536 possible steps. (lesser error)

The quantized output of an A/D convertor, can be represented in either serial or parallel form

3.8.2.1 Quantization Resolution

The quantizing error may be considered as unwanted signal added to the wanted signal as shown in Fig. 3.5 and Fig 3.6 below.



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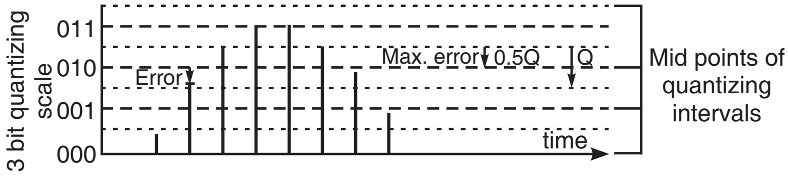


Fig. 3.5 : Quantization errors

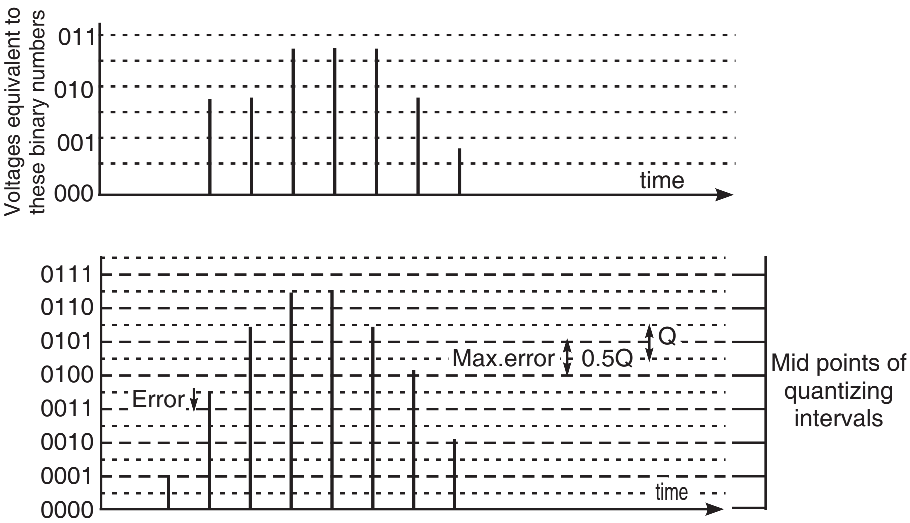


Fig. 3.6: Higher level Quantization and lesser error

The error is classified either as distortion or noise, depending on their characteristics and the nature of the quantizing error signal.

The bit rate of the digital signal is therefore is the multiplication of sampling frequency with the quantization bits used. For example if a audio signal is sampled at 48 kHz and 16 bit quantization is used, then the bit rate of digital signal will be 48*16= 768 Kbps(kilo bits per second). If compressed by four it will become 192 Kbps for a mono(single channel signal) and 192*2= 384Kbps for stereo signal.

3.8.3 Dynamic Range/Bit Depth

The human hearing capabilities should be regarded as the standard against which the quality of digital systems is measured, since it could be argued that the only distortion and noise that matter are those that can be heard, Louis Fielder and Elizabeth Cohen work out to establish dynamic range requirements for high quality digital audio systems, Fielder was able to show, what was likely to be heard at different frequencies in terms of noise and distortion and where the limiting elements might be in a typical recording chain, they determined a dynamic range requirement of 122 dB for natural reproduction, taking into account microphone

Notes

performance and the limitations of consumer loudspeakers. This requirement dropped to 115dB for consumer systems.

When there are no more bits available to represent a higher level signal at this point the waveform will be hard clipped and will become highly distorted. The effect too is very different from that encountered in analog tape recorders which tend to produce gradually more distortion as the recording level increases. Digital recorders remain relatively undistorted as the recording level rises until the overload point is reached at which point very bad distortion occurs. The number of bits per sample therefore dictates the signal to noise ratio of a linear PCM digital audio system 16 Bit Linear PCM was considered as normal for high quality audio application for many years. This is the CD standard and is capable of offering a good S/N ratio range over 90db, but it fails to reach the psychoacoustic idea of 122db for subjectively noise free reproduction in professional systems.

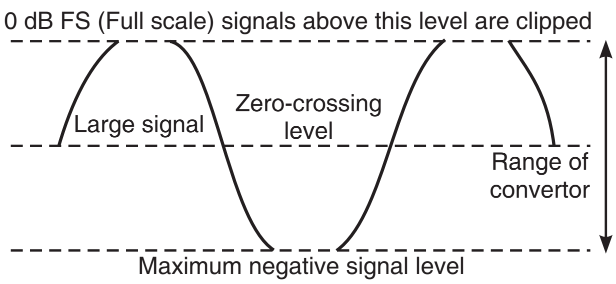


Fig. 3.7: Clipping of signal

For professional recording purpose and avoidinf clippin (Fig. 3.7) we may need a certain amount of head room for distortion free recording/reproduction.

Headroom

Unused dynamic range above the normal peak recording level which can be used in unforeseen circumstances such as when a signal overshoot its expected level this can be particularly necessary in live recording conditions where one is never quite sure what is going to happen with recording level.

This is another reason why many professionals feel that a resolution of greater than 16 Bit is desirable for original recording. Twenty and 24 bit format are become very popular for this reason.

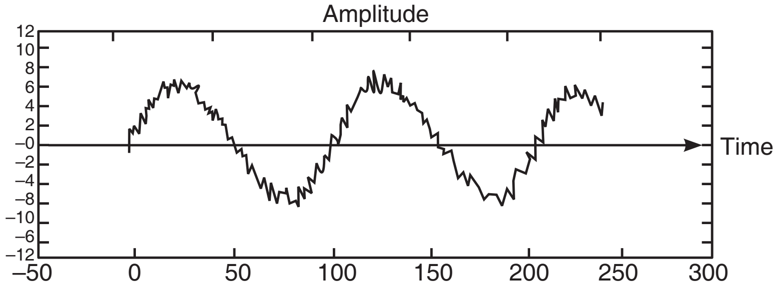
Quantizing Resolution

The table shows some commonly encountered quantizing resolutions and then application

Bits per sample	Approximate dynamic range with dither (db)	Application
8	44	Low moderate quality for older PC Internal sound generation and some older multimedia application.
12	80	Original EIAJ format PCCM adaptors such as Sony PCM-100
16	92	CD-standard, DAT standard commonly used high quality resolution for consumer media, some professional recorders and multimedia pcs. Usually twos compliment binary numbers.
20	116	High quality professional audio recording and mastering applications
24	146	Maximum resolution of most recent professional recording system also of AES 3 digital interface, dynamic range exceeds psycho-acoustic requirements

3.8.4 Dither

Dither in A/D conversion has the effect of linearzing a normal convertor (in other words it effectively makes each quantizing interval the same size) and turns quantizing distortion into a random, noise line signal at all times. (Fig. 3.8 below)



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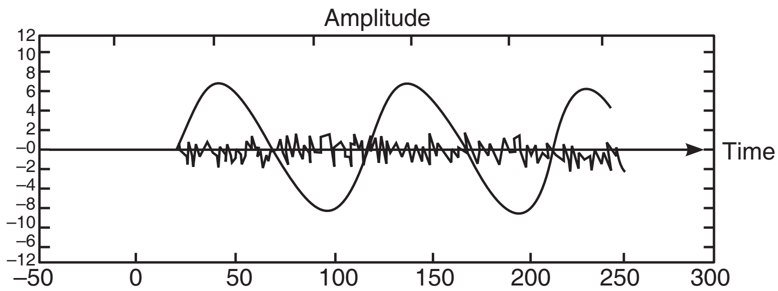

Fig. 3.8: Dithering or linearization

This is used for number of reasons

1. It allows signal to be faded smoothly down without the sudden disappearance.
2. It allows the signal to be reconstructed even when their level is below the noise floor of the system.
3. The white noise at very low level is less subjectively annoying than distortion.

Undithered audio signal begin to sound "grainy" and distorted as the signal level falls. Quiescent hiss will disappear if dither is switched off making a system seem quieter but a small amount of continuous hiss is considered preferable to low level distortion. The resolution of modern high resolution convertors is such that the noise floor is normally inaudible in any case.

Quiescent Hiss – The hiss generated from the machine during standby mode.

White Noise – This is a noise in which all frequencies have same amplitude or in same level.

3.9 DIGITAL TO ANALOG CONVERSION (D/A CONVERSION)

In D/A conversion, the audio sample words are converted back into stair case like chain of voltage levels corresponding to the sample values. This is achieved simple D to A convertors, by using the states of bits to turn current sources on or off, making up the required pulse amplitude by the combination of outputs of each of these sources.

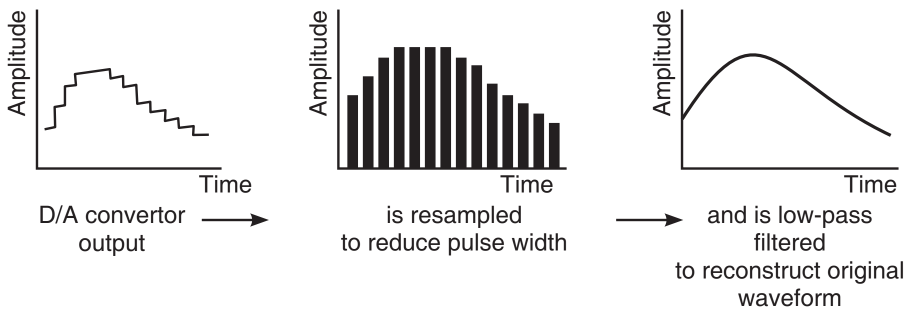

Fig. 3.9: Digital to analog conversion

These staircases are then resampled to reduce the width of the pulses before they are passed through a low pass reconstruction filter whose cut off frequency is half the sampling frequency. The effect of the reconstruction filter is to join up the sample points to make a smooth wave form as shown in figure 3.9 above.

Re sampling is necessary to avoid any discontinuities in signal amplitude at the sample boundaries, otherwise the averaging effect of the filter would result in a reduction, in the amplitude of high frequency audio signal.

INTEXT QUESTIONS 3.2

Write down the correct option from the ones given below:

1. Converting analog signal into digital signal by measuring amplitude at specific points is called

   (a) Quantization (b) Dithering

   (c) Sampling

2. The process of assigning a binary digital value of each sampled measurement is called

   (a) Sampling (b) Quantization

   (c) Dithering (d) Dynamic range

3. Sampling frequency determines at least .... the highest audio frequency.

   (a) Once (b) Twice

   (c) Thrice

4. What is the sampling rate for a Compact Disc?

5. Match the following:

Sample rate	Application
48	Telephone
32	DVD – Audio
96	CD – Audio
8	Broadcast coding
44.1	Basic Blu ray disk

6. According to Fielder what is the dynamic range for natural reproduction?
7. Dither allows a signal to be faded smoothly without the sudden disappearance .True or false
8. What is quiescent hiss?
9. What is white noise?

3.10 WHAT HAVE YOU LEARNT

After reading the lesson, we have learnt about the different types of signals— analog and digital. We saw, how sound is first converted into an electrical signal when it enters a microphone.

We then proceeded to learn about the advantages of digital signal over analog signal. How digital signal lends itself to compression and therefore occupies less space.

The process of analog to digital conversion and vice-versa were two other points covered in this lesson. :

3.11 TERMINAL QUESTIONS

1. Briefly define analog and digital signal? What are the main differences between them?
2. What are the advantages of digital signal over analog signal?
3. What are the advantages of digital technology over analog technology?
4. Explain analog to digital (A/D) conversion.
5. Briefly describe the following terms.
   (a) Sampling Rate

(b) Quantizing

(c) Dithering

6. Explain dynamic range and list a few applications?

7. Explain digital to analog conversion.

3.12 ANSWERS TO INTEXT QUESTIONS

3.1

1. Electrical
2. Binary
3. False
4. True
5. Yes, true.

3.2

1. Sampling
2. Quantization
3. Twice
4. 44,100 Hz (or 44.1kHz)

5.

Sample rate	Application
48	Basic Blu ray disk
32	Broadcast Coding
96	DVD - Audio
8	Telephone
44.1	CD - Audio

6. 122 dB

7. True.

8. The hiss generated from the machine during standby or sleep mode.

9. The noise in which all frequencies have same amplitude or in other wards in same level

4

STUDIO ACOUSTICS

4.1 INTRODUCTION

Previously you have studied about the fundamentals of sound technology. Now, let us learn about the studio acoustics.

Sound production not only depends upon the quality(notes) of the sound generated from a source like human vocal-chord or an electro-mechanical speaker, but it depends upon the medium through or from which is comes after reflection, refraction, diffraction, absorption or interference. Thus, acoustics plays a vital role in sound production.

In this lesson, you will learn about the basic acoustic principles and construction techniques for consideration, while designing sound production facility. Acoustics pertains to the art of designing that necessitates intuition, experience and common sense. This will include understanding reflection, refraction, reverberation, noise level, materials used and transmission techniques.

4.2 OBJECTIVES

After reading this lesson, the learner will be able to :

- explain reflection;
- identify laws of reflection;
- analyze reverberation, sound isolation and noise level;
- enumerate basics of psychoacoustics.

4.3 MEANING OF ACOUSTICS

Acoustics can be defined as a science dealing with the production, effects and transmission of sound waves through different media including reflection, refraction, diffraction, absorption and interference. It also includes consideration about characteristics of auditoriums, theatre and studios along with their design.

4.3.1 Identifying need of Acoustic treatments

There are various situations needing enhacing the quality of sound in an enclosed space, tackling the interference problems etc. which require acoustic treatment to be done. Important role of studio acoustics (SA) is to identify such problems first. Like:

(a) While clapping your hands if a fluttering sound is heard because of sounds bouncing back and forth from hard parallel walls then this needs acoustic treatment.

(b) If the studio is made up like concrete block basement or garage, so you hear too much room reverberation.

(c) If the studio is too small

(d) If you hear, outside voices in the recordings.

(e) If bass guitar amps & monitor speakers sound booms

(f) If you hear a lot of leakage in mic signals.

All these would need attention to be paid to the acoustic aspects to improve the quality of sound

4.3.2 Factors governing studio and control room acoustics

Studio and control room acoustics, play a very important role while recording any musical program. Irrespective of which type of studio facility is being designed, built and being used, there are various factors, which should be addressed, in order to achieve the best possible acoustic results. Such as:

1. **Acoustic isolation**- appropriate isolation techniques are necessary to incorporate into their design, in order to keep external noises to the minimum. Whether that noise is transmitted through the medium of air or though solids special construction techniques are often required to dampen these external sounds. The goal here is to build a studio wall, floor, ceiling, window or door out of thickest and most dense sound absorbant material available to improve sound isolation

2. **Symmetry in control room and monitoring design**-There should be symmetrical reflections on all axes within the design of a control room or single room project of a studio. The center and acoustic imaging is best when the listener, speakers, walls and other acoustical boundaries are symmetrically centered around the listener's position . In a rectangular room, the best low end response can be obtained by orienting the console and loudspeakers into the room's long dimension. Placing the listening environment symmetrically in a corner, is another example of how the left/right imagery can be improved over an off center placement.

3. **Frequency balance:**The frequency components of a room shouldn't adversely affect the acoustic balance of instruments and speakers. The acoustic environment shouldn't alter the sound quality of the original or recorded performance**.** Room should exhibit a relatively flat frequency response, over the entire audio range, without adding its own sound coloration. The most common way to control the tonal character of a room, is to use materials and design techniques that govern the acoustical reflection and absorption factors.

4. **Reflection:** Reflection is the change in direction of a wave front at an interface between two different media so that the wave front returns into the medium from which it originated. Common examples include the reflection of light, sound and water waves.

DO YOU KNOW?

The *law of reflection* says that, for spectacular reflection, the angle at which the wave is incident on the surface equals the angle at which it is reflected (both values measured from the normal N). Light is known to behave in a very predictable manner. If a ray of light could be observed approaching and reflecting off a flat mirror, then the behavior of the light as it reflects would follow a predictable *law* known as the **law of reflection**. The diagram below (Fig. 4.1) illustrates the law of reflection.

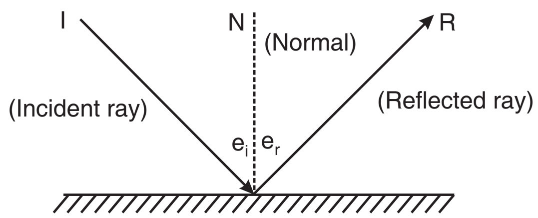

Fig. 4.1: Law of reflection

Reflection of sound – just as a mirror reflects, light, when sound waves, radiating out from a source, strike a rigid obstacle, the angle of reflection of the sound waves is same as the angle of incident, if no absorption occurs)

5. **Absorption:** Absorption occurs, when only a portion of the incident acoustic energy is reflected back from a material's surface. The absorption of acoustical energy is inverse of reflection. Whenever sound strikes a material, the amount of acoustic energy that absorbed, relative to the amount that's reflected, can be expressed as a simple ratio known as the material's absorption coefficient. You have learnt about absorption coefficient in first chapter on Introduction to sound.

6. **Reverberation:** A sound created in a big hall will persist by repeated reflection from the walls until it is reduced to a value where it is no longer audible. The repeated reflection that results in this persistence of sound is called reverberation. Reflection of sound waves of the surfaces can lead to **reverberation**. A reverberation often occurs in a small room with height, width, and length dimensions of approximately 17 meters or less. Reflection of sound waves also leads to **echoes**.

   Like any wave, a sound wave doesn't just *stop* when it reaches the end of the medium or when it encounters an obstacle in its path. Rather, a sound wave will undergo certain behaviors when it encounters the end of the medium or an obstacle. Possible behaviors include reflection off the obstacle, diffraction around the obstacle, and transmission (accompanied by refraction) into the obstacle or new media.

DO YOU KNOW?

Why the magical 17 meters? The affect of a particular sound wave upon the brain endures for more than a tiny fraction of a second; the human brain keeps a sound in memory for up to 0.1 seconds. If a reflected sound wave reaches the ear within 0.1 seconds of the initial sound, then it seems to the person that the sound is *prolonged*. The reception of multiple reflections off the walls and ceilings, within 0.1 seconds of each other, causes reverberations - the prolonging of a sound. Since sound waves travel at about 340 m/s at room temperature, it will take approximately 0.1 s for a sound to travel the length of a 17 meter room and back, thus causing a reverberation ($t = v/d = (340\ \text{m/s})/(34\ \text{m}) = 0.1\text{s}$). This is why reverberations are common in rooms with dimensions of approximately 17 meters or less.

7. **Cost factors:** Designing acoustical studios can be a costly affair. Studio designers and construction teams spend a lot to create plush décor that has been acoustically tuned to fit the needs of both owners and budget minded production facilities , however the same can be done in cost effective ways also keeping in view the production needs and future prospects of growth.

In addition to above some related factors are explained hereunder.

4.4 PROPAGATION OF SOUND

Propagation of sound through the air is in the form of longitudinal waves. The speed of sound depend on the properties of air and not on the frequency or amplitude of sound waves. As in case of light same lwaw of reflection applies to sound also i.e., angle of incident is equal to the angle of reflection where measured about the normal.

INTEXT QUESTIONS 4.1

1. Reflection is ....?

   (a) Change in the direction of a wave front

   (b) Change in sight

   (c) both (a) and (b)

   (d) none of these

2. In acoustics, reflection causes .... ?

   (a) echoes (b) reflection

   (c) wave (d) light

3. The human brain keeps a sound in memory for up to .... seconds.

   (a) 0.2 Seconds (b) 0.1 Seconds

   (c) 1 Seconds (d) None of these

4. Reflection of sound waves also leads to ....

   (a) Echoes (b) Clear Sound

   (c) Bad Sound (d) A and B

5. Light is known to behave in a very .... manner

   (a) predictable (b) unpredictable

   (c) hypothetical (d) straight

4.5 REFRACTION

The turning or bending of any wave, such as a light or sound wave, when it passes from one medium into another of different optical density, is called Refraction.

Notes

Refraction is the change in direction of a wave due to a change in the medium of transmission. The phenomena is governed by the laws of conservation of energy and momentum. When medium changes, the phase velocity of the sound also changes but frequency remains unaffected, especially at incident angles other than 90° or 0°. This phenomena is observed in all the waves like sound and water. As per Snell's law for a given pair of media and a wave with a single frequency, the ratio of sine's of the angle of incident Q_1 and the angle of refraction Q_2 is equal to the ratio of phase velocities V_1/V_2 in the two media to the opposite ratio of the indices of refraction (n_2/n_1)

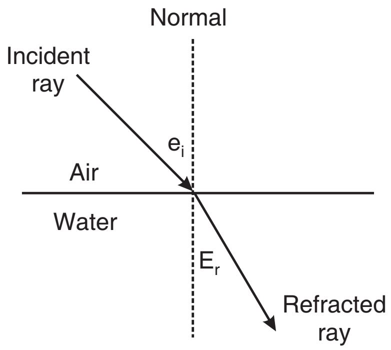

Fig. 4.2: Law of Refraction

As shown in the diagram (Fig 4.2) above, a normal line is drawn to the surface at the point of incidence. This line is always drawn perpendicular to the boundary. The angle that the incident ray makes with the normal line is referred to as the **angle of incidence**. Similarly, the angle that the refracted ray makes with the normal line is referred to as the **angle of refraction**. The angle of incidence and angle of refraction is denoted by the following symbols:

θ_1 = angle of incidence

θ_2 = angle of refraction

The amount of bending that a light ray experience, can be expressed in terms of the angle of refraction. A ray of light may approach the boundary of a medium at an angle of incidence of 45-degrees and bend towards the normal. If the medium into which it enters causes a small amount of refraction, then the angle of refraction might be of a value of about 42-degrees. On the other hand if the medium into which the light enters causes a large amount of refraction, the angle of refraction might be 22-degrees. (These values are merely arbitrarily chosen values to illustrate a point.) The diagram below depicts a ray of light approaching three different boundaries, at an angle of incidence of 45-degrees. The refractive medium is different in each case, causing different amounts of refraction. The angles of refraction are shown on the diagram. (Fig. 4.3) below.

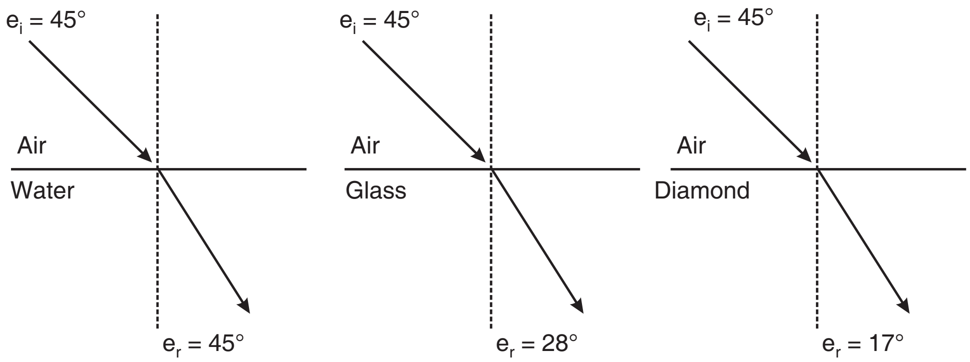

Fig. 4.3: Different angles of Refraction for different mediam

Of the three boundaries in the diagram, the light ray refracts the most at the air-diamond boundary. This is evident by the fact that the difference between the angle of incidence and the angle of refraction is greatest for the air-diamond boundary.

INTEXT QUESTIONS 4.2

1. Refraction is the ....?
   (a) change in direction of a wave
   (b) change in direction
   (c) change in nature of work
   (d) all of these
2. Refraction is essentially a .... phenomenon
   (a) surface (b) normal
   (c) usual (d) transmission
3. In Refraction its frequency remains ....
   (a) constant (b) normal
   (c) abnormal (d) speedy
4. The speed of a sound wave in air depends on the ....
   (a) Temperature (b) clouds
   (c) rains (d) b and c
5. The angle of refraction is dependent upon the ....
   (a) Speeds of light (b) speed of air
   (c) speed of volume (d) speed

4.6 NOISE LEVEL

Noise is the unwanted sound and in many applications noise reduction is a must. Noise can cause of loss of hearing power, interfere with sleep, speech, cause discomfort.. The importance of noise issue could be well understood by looking at regulations that have been passed by governments to restrict noise production or control noise pollution in society.

Active Noise control

Passive noise control refers to those methods that aim to suppress the sound by modifying the environment closer to the source. Since no input power is required in such methods, Passive noise control is often cheaper than active control, however the performance is limited to mid and high frequencies. Active control works well for low frequencies hence, the combination of two methods may be utilized for broadband noise reduction as shown below Fig. 4.4.

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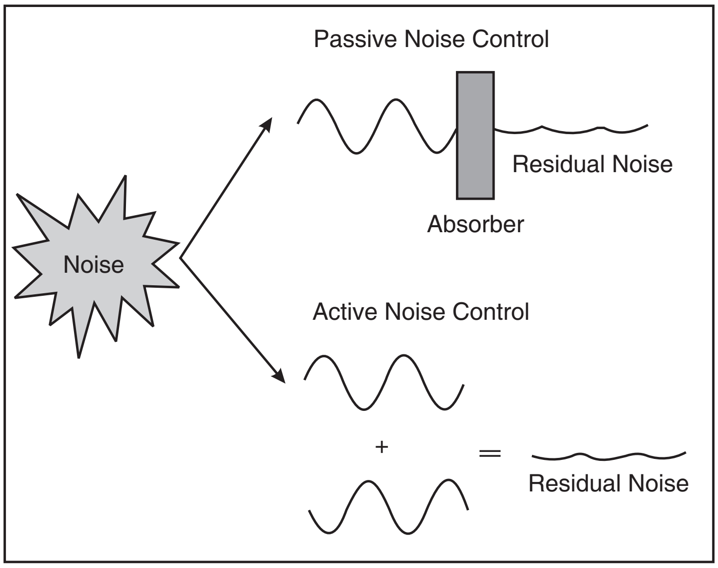


Fig. 4.4: Passive Noise Control and Active Noise Control

Sound waves striking an arbitrary surface are reflected, transmitted or absorbed; the amount of energy going into reflection, transmission or absorption depends on acoustic properties of the surface. The reflected sound may be almost completely redirected by large flat surfaces or scattered by a diffused surface. When a considerable amount of the reflected sound is spatially and temporally scattered, this status is called a diffuse reflection, and the surface involved is often termed a diffuser. The absorbed sound may either be transmitted or dissipated. A simple schematic of surface-wave interactions is shown in Fig. 4.5.

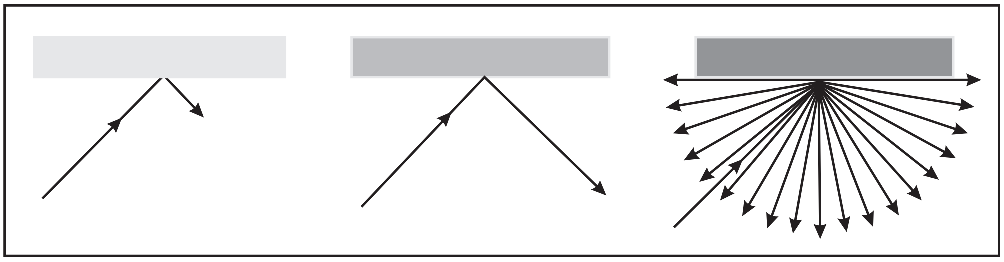


Fig. 4.5: Surface wave interaction

Sound energy is dissipated by simultaneous actions of viscous and thermal mechanisms. Sound absorbers are used to dissipate sound energy and to minimize its reflection.

Furnishing curtain against the walls, absorb high frequency better than lower areas. When curtains are placed further away from walls, lower frequency is also absorbed. The amount of sound absorbed will depend upon the type of the material used for certain fiberglass (Rockwool) has the highest absorbing capacity. Fiberglass is used with brick, stone, concrete etc. due to its sharp edges. Timber, steel reflects higher frequencies but absorbs certain percentage of lower frequencies, balance of the lower frequency pass through the wall. Bass frequencies are the most difficult to absorb.

The 1/4 wavelength rule: Absorbent material (soft or fiberglass) must be kept at 1/4 wavelength of the lowest frequency to be absorbed. This helps to absorb all higher frequencies. This may make the room seem smaller but it will be calmer and relaxing. Ref. Fig 4.6 below.

Notes

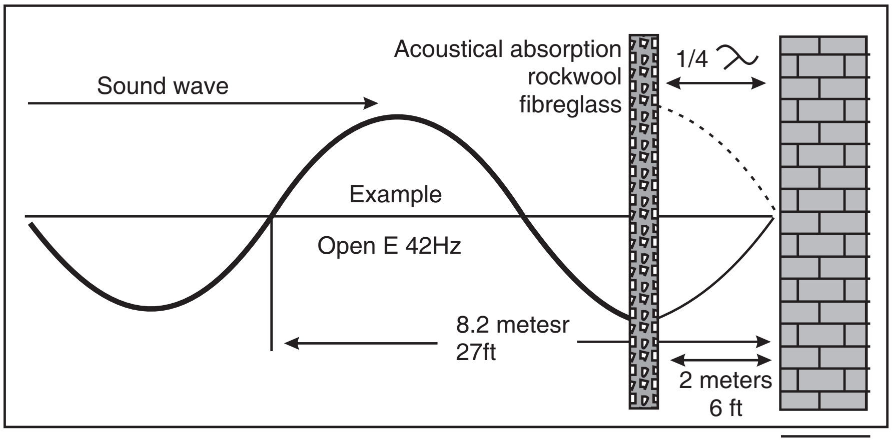


Fig. 4.6: Placement of sound absorbent material

4.6.1 Making a quieter studio

An ideal noise level in a studio should be 23-28 dB.As measured on SPL meter. Following tips will be useful to keep away the noise from studio recordings.

1. All appliances, air conditions and telephones etc. should be switched off while recording.
2. All doors and windows of the studio, if any should be closed or cover them with thick plywood.
3. Objects that can rattle or buzz should be removed.
4. There should be carpet or several layers of plywood on the floor above studio. Moreover, there should be an insulation in the air space between studio ceiling and the floor above.
5. Microphones should be placed, close to the instruments and use directional microphones. This will reduce noise picked up by the microphones from other directions/ sources.

INTEXT QUESTIONS 4.3

1. Modifying and canceling sound field of electro-acoustical approaches are called ....

   (a) active noise control (b) passive noise control

   (c) negative voice control (d) mix noise control

2. Wool (fiberglass) has the .... absorption capacity

   (a) highest (b) lowest

   (c) medium (d) none of these

3. Passive noise control is often .... than active control

   (a) cheaper (b) higher

   (c) medium (d) within cost

4. The amount of sound energy absorbed depends on type of .... weight and pleating width

   (a) material (b) glass

   (c) energy (d) none of these

4.7.1 Sound Transmission

Sound is a travelling pressure wave that can be transmitted through air, liquids, or solids and is sensed by the human ear. The wave is a vibration or fluctuation in pressure and can vary in amplitude (i.e. Loudness or volume) and frequency (i.e. pitch). Sound volume is measured indecibels, while the frequency of sound is measured in hertz. The frequency and amplitude of sound can change as the wave travels through different mediums. For example, the loudness of a sound is reduced as it travels from the air through a wall of concrete

Acoustic transmission in building design refers to a number of processes/ways by which sound can be transferred from one part of a building to another. Typically these are:

1. **Airborne transmission** - To isolate one room from the other, so that sound from other rooms is not heard in the adjacent room, structural isolation is considered at the time of design stage of the building, especially in the recording studios or the highly sensitive areas. A tighten sealed door also helps in reduction of such air borne sound. Heavy dividing walls, help in effective reduction of airborne sound transmission better than a light one.

2. **Impact transmission** - A noise source in one room, results from an impact of an object onto a separating surface, such as a floor and transmits the sound to an adjacent room. A typical example would be the sound of footsteps in a room being heard in a room below. Acoustic control measures usually include attempts to isolate the source of the impact, or cushioning it. For example, carpets will perform significantly better than hard floors.

3. **Flanking transmission** - A more complex form of noise transmission, where the resultant vibrations from a noise source are transmitted to other rooms of the building usually by elements of structure within the area.

4. **E building**. For example, in a steel framed building, once the frame itself is set into motion the effective transmission by vibration can be pronounced.

Sound Absorption

Sound absorption is the process by which we can reduce the reflection of the sound energy by the surfaces. As we have already seen the absorption phenomenon of sound wave from which it is clear that when sound wave strikes the acoustically treated surface some of the sound wave penetrates the acoustic material covering the wall and portion of that sound energy is retained by the absorbing material. This absorbed sound energy is converted into heat energy thereby preventing any retransmission or reflection of sound wave from the surface. The absorbing material will be selected on the basis of frequency distribution of noise and application of the studio. Different absorbers show different absorption characteristics which are non uniform over the complete frequency spectrum.

For achieving optimum R/T characteristics combination of acoustic absorbers is used in the studio. Every material has some absorptive qualities. This is described by its coefficient of absorption, a number between 0 and 1, the value 0 corresponds to totally reflective and 1 corresponds to an open window . These numbers can be used to compare materials and to predict the results of treatment. Some of the commonly used absorbers are:

(i) ***Porous Materials:*** Porous materials are used for the absorption of Mid and High Frequencies. Mineral wool, glass wool are members of this class. These materials are very good absorber and are most effective in *Mid and High Frequencies.* These absorbers are used with the covering material which acts as a face of such absorbers. Fabric used as a Carpet and Curtain also act as absorber for Mid and High Frequencies.

(ii) ***Fibrous Materials:*** Insulation boards, perforated tiles fall in fibrous material category. The tiny holes in the fibrous material acts as a trap which are responsible for the absorption of sound and dissipation of the sound energy. The Absorption of these materials increases with increase in the softness of the material. These materials have very poor absorption on low frequencies.

(iii) ***Panel/ Resonant Absorbers:*** Panel absorbers are thin wooden ply/ veneers with an air cavity behind. This is generally used as *Low Frequency Absorber (LFA).*

Notes

4.8 BASICS OF PSYCHOACOUSTICS

4.8.1 Psychoacoustics – Threshold of hearing and pain

The intensity level of a loud sound, which gives pain to the ear, is usually between 115 and 140 dB (see graph). For some listeners, with hyperacusis, these levels may be much lower.

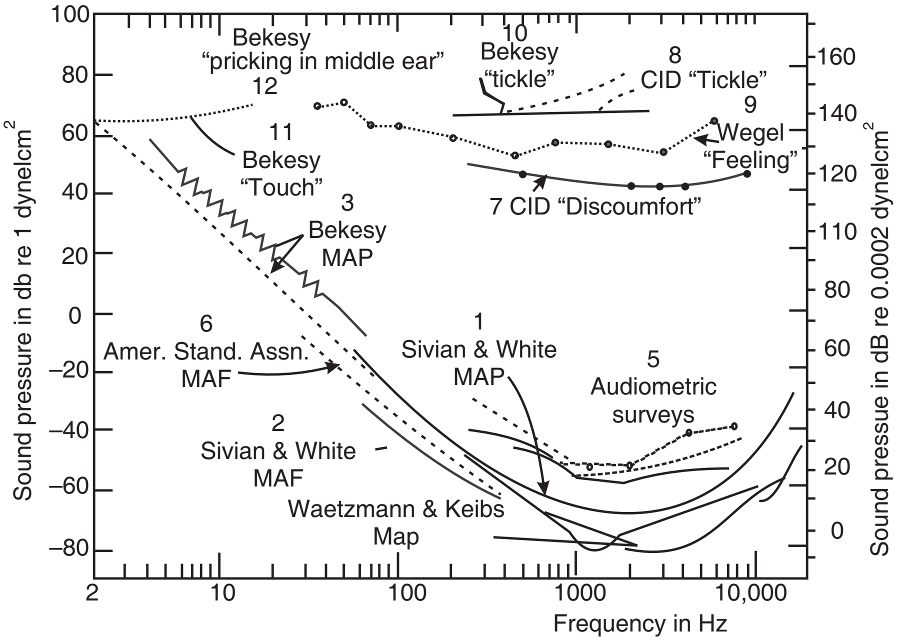


Fig. 4.7

The **threshold of hearing** is the Sound pressure level SPL of 20 μPa (micropascals) = 2×10^{-5} pascals (Pa). This low threshold of amplitude (strength or sound pressure level) is frequency dependent. See the frequency curve in Fig. 2 below

The **absolute threshold of hearing** (ATH) is the minimum amplitude (level or strength) of a pure tone that the average ear with normal hearing can hear in a noiseless environment.

The **threshold of pain** is the SPL beyond which sound becomes unbearable for a human listener. This threshold varies only slightly with frequency. Prolonged exposure to sound pressure levels in excess of the threshold of pain can cause physical damage, potentially leading to hearing impairment.

Different values for the threshold of pain:

Notes

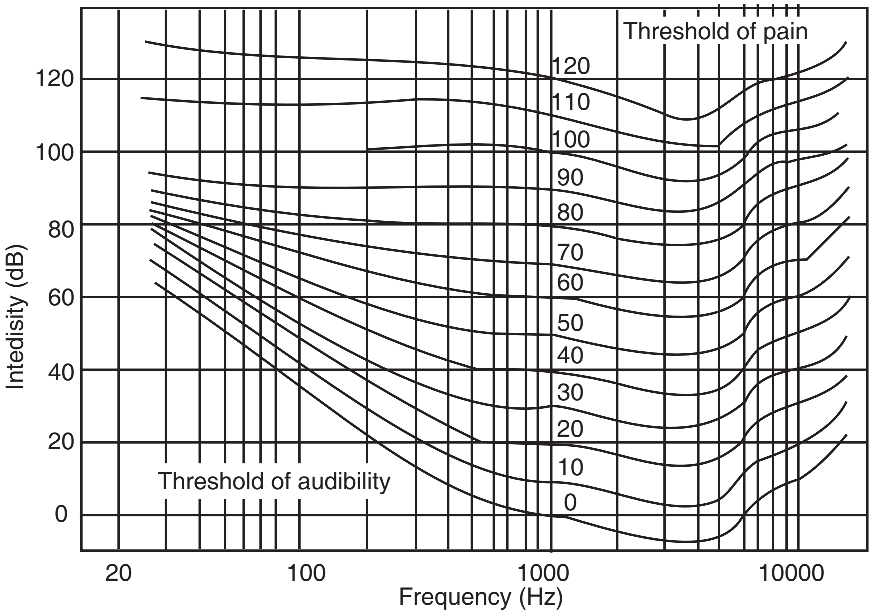


Fig. 4.7: Threshold of audibility

Threshold of pain

SPL	Sound pressure
120 dBSPL	20 Pa
130 dBSPL	63 Pa
134 dBSPL	100 Pa
137.5 dBSPL	150 Pa
140 dBSPL	200 Pa

The threshold of haring is the minimum sensitivity of the ear and lies with 1 kHz to 5 kHz. It is frequency dependent (see curve in Fig. 4.5) where lowest curve is the absolute threshold of hearing and the highest curve in the threshold of pain. Any part of the audio signal whose amplitude is below the ATM level, can be removed without affecting any change to the signal.

INTEXT QUESTIONS 4.4

1. Sound is a travelling pressure wave that can be transmitted through air, liquids, or solids and is sensed by the human ....?

   (a) ear (b) eyes

   (c) hands (d) tongue

2. The lntensity level of a loud sound which gives pain to the ear, usually between ....

   (a) 115 and 140 dB (b) 110 & 120 dB

   (c) 90 & 95 dB (d) none of these

3. The threshold of hearing is the .... level SPL

   (a) sound pressure (b) water Pressure

   (c) air Pressure (d) wall Pressure

4. The threshold of pain is the SPL beyond which sound becomes .... for a human listener

   (a) unbearable (b) easily bearable

   (c) adjustable (d) bearable

4.8 WHAT HAVE YOU LEARNT

In this lesson you learnt about the meaning of acoustics and the factors governing studio and control room acoustics factors which govern the studio acoustics include acoustic isolation symmetry in control room and monitoring design, frequency balance, reflection absorption, reverberation and cost implications etc. In addition to this, noise control and psych acoustics were also discussed.

4.9 TERMINAL QUESTIONS

1. Define acoustics. Discuss why on acoustic treatment is required in a studio setup?

2. Explain the various factors governing studio and control room acoustics.

3. Write short note on

   (i) Psychoacoustics

   (ii) Refraction

4.10 ANSWERS TO INTEXT QUESTIONS

4.1

1. (a) 2. (a) 3. (b) 4. (a) 5. (a)

4.2

1. (a) 2. (a) 3. (a) 4. (a) 5. (a)

4.3

1. (a) 2. (a) 3. (a) 4. (a)

4.4.

1. (a) 2. (a) 3. (a) 4. (a)

Notes

5

AUDIO ELECTRONICS

5.1 INTRODUCTION

In the preceding lesson, you have studied the fundamentals of studio acoustics that include reverberation, sound isolation, noise level, basics of psychoacoustics. This lesson deals in basic electrical and electronic elements, used in sound equipments which include various types of electrical and electronic components and circuits commonly used in sound systems.

Emphasis will be given on resistors, thermistors and transistors.

5.2 OBJECTIVES

After going through this lesson, you should be able to:

- learn the basic concepts of voltage and current as applicable to recording equipments;
- identify various types of components used in electronic circuits;
- understand the functions of such components;
- identify the type, value and rating of components;
- learn some of the important points to be remembered while replacing these components;
- learn about the interference and methods to eliminate such problems;
- understand the importance of earthing and the problems that are likely to occur due to bad earthing;
- learn procedure to carry out adjustments of sound levels while using the equipments.

5.3 CONCEPT OF VOLTAGE AND CURRENT

Consider a Water tank at the top of a multi storey building. It supplies water to all the floors of the building. You would have observed that the pressure with which water is supplied at the ground floor is more than at the top floor or terrace. This is because that water pressure is proportional to height difference between top level of water in tank and tap from which water is drawn. Further, water flow rate at the outlet (i.e., tap) is proportional to diameter of the pipe and tap.

Analogously, in electrical systems, CURRENT flow (a quantity similar to water flow rate) is proportional to pressure of the Direct Current (DC) source (e.g., 1.5 Volts Cell, mobile battery 4.5 VDC or car battery 12 VDC) or Alternating Current (AC) source (e.g., AC power supply 220VAC). Similarly, VOLTAGE is analogous to height difference of top of water level in tank and water tap.

Voltage

Before learning about voltage, let us learn Electric Field and Electric Potential. Electric Field is the region around a point charge, within which its effect can be experienced. Electric Potential of a point (say P) is the work done in bringing, unit positive charge, from infinity (beyond the electric field of P) to that point (P). Consider two points P1 and P2 with potentials V1 and V2, respectively. If there is potential difference between P1 and P2, this is termed as Potential Difference (PD), defined as $V = V1 - V2$. Another term for PD is Voltage. For the flow of electric current through an electrical element, voltage should exist across the terminals of the element. In simple words, voltage in electrical system is the ability of the energy source (cell, battery, generator, etc.) to produce a current. Voltage is measured in Volts and is represented by letter 'V' in electrical circuits and calculations.

Most of the modern day sound/recording equipments operate on DC voltage sources, obtained either from dry cells or by converting from AC mains voltage. Emphasis, in this lesson, has been limited to DC voltage sources for easy understanding.

Current

Current is analogous to water flow in the example. Before defining current let us know its genesis. We experience existence of electric charge many times in our daily life. If you rub surfaces of ebonite, glass or even a comb, it acquires electric charge. Try rubbing a comb for some time against your hair, the comb can pick up small pieces of paper. If you watch an aircraft landing in dark, a flash will be observed near the wheels. All these are examples of charge acquired on surfaces due to friction. The flash observed is due to the flow of charge. Thus, we can define the conventional Current as the rate of flow of charge (positive) per unit of time. 1 Ampere of current is flow of 1 Coulomb of charge per second. Unit of current is Ampere and is represented by letter 'I' in electrical circuits and calculations. Smaller units such as mA (1milli Ampere =1/1000 A) are most commonly used in electronic circuits.

The magnitude of the electric current depends not only upon the electromotive force but also upon the nature and dimensions of the path through which it circulates. The magnitude of current flowing through a simple circuit can be determined by use of a most important and basic law called as OHM'S LAW.

Ohm's law states that the current in a DC circuit varies in direct proportion to the voltage and is inversely proportional to the resistance of the circuit. (The term resistance is analogous to the opposition offered to the movement of water flow by pipes, bends etc.) Resistance is represented by a letter 'R' and its unit of measurement in electrical/electronic circuits is Ohm (Ω).

Mathematically, this law can be expressed as

$$\text{Current} = \text{Electromotive force/Resistance}$$

Using the symbols I, V and R to represent current, voltage and the resistance respectively, Ohm's law can be written as:

$$I = V/R \text{ or } V = I \times R$$

Example: If 1volt is applied across a resistor of 1 ohm, a current of 1 Ampere will flow through this resistor.

Note that this law not only holds for a complete circuit, but can be applied for any part of a circuit provided care is taken to use the correct values for that part of the circuit.

Note:

1. Conventional current is considered to be in a direction opposite to the flow of electrons.
2. Electric current flows through a path if path is part of a closed loop with an independent supply source.

INTEXT QUESTION 5.1

1. A recording equipment operates on 12 V Battery. If the net resistance offered by the recording equipment is 300 ohms (resistive), how much current is likely to flow through the equipment. Answer briefly in the space provided for the purpose.

5.4 COMPONENTS USED IN ELECTRONIC CIRCUITS

Components used in electronic circuits can be broadly categorized in following two ways:

(a) Passive components – Resistors, Inductors, capacitors, diodes, thermistors, varistors and transformers are examples of passive components. Passive components are those which do not produce or amplify A.C. signals.

(b) Active Components – Active components are those components which can generate or amplify A.C. signals. It is important to note that power of amplified AC Signal at the output of an electronic amplifier is not generated in the device but is drawn from D.C. power supply. The example of active components are Bipolar Junction Transistors (BJTs), Field Effect Transistors (FETs) , Operational Amplifiers, Analogue ICs and Digital ICs. Integrated Circuits (ICs) are devices which integrate thousands of resistor, capacitors and transistors etc. to perform a desired function.

5.4.1 Passive Components

1. Resistors

Resistors provide means of controlling voltage and / or current in a circuit.

Resistors are typically used in electronic circuits to;

1. Establish bias potential and current for proper operation of transistors circuits.
2. Convert collector or emitter current of a transistor into corresponding output voltage.
3. To provide a preset level of attenuation.

Electrical characteristics of a resistor are determined largely by material used and its construction. While selecting a particular type of resistor following parameters need to be considered:

(i) Denomination of a resistor in terms of Ω (Ohm), kΩ (Kilo-ohms), or MΩ (Megaohms) etc.

(ii) Its desired accuracy or tolerance (i.e., maximum permissible percentage deviation from circuit design value). Resistors with tolerance of +/- 2% or less are termed as close tolerance resistors.

(iii) Its power rating (which must be equal to or greater than the maximum expected power dissipation)

(iv) Its temperature coefficient (expressed in ppm (parts per million) per unit change in temperature.)

(v) Its stability (expressed in terms of long or short term percentage variation of resistance value under specified physical and electrical conditions). Manufacturers usually specify it to be of "High Stability" if resistance is stable.

(vi) The noise performance (expressed as equivalent noise voltage generated by resistor under specified physical and electrical conditions). Manufacturers usually specify it to be of "Low Noise" if it is so.

High stability, low noise and close tolerance resistors are required in critical applications e.g. initial stages of amplifier dealing with very small level signals, input stage of Test and Measuring Instruments (TMIs) etc.

Though high performance (i.e. high stability, low noise, close tolerance) may be used in less or not so critical applications but it would be uneconomical to use, since such resistors are expensive.

Aluminium clad wire wound resistors rated for 25 Watts and above should be mounted on suitable heat sink. Its power rating should be de-rated by more than 50% if it is mounted in free air.

Value of Resistors

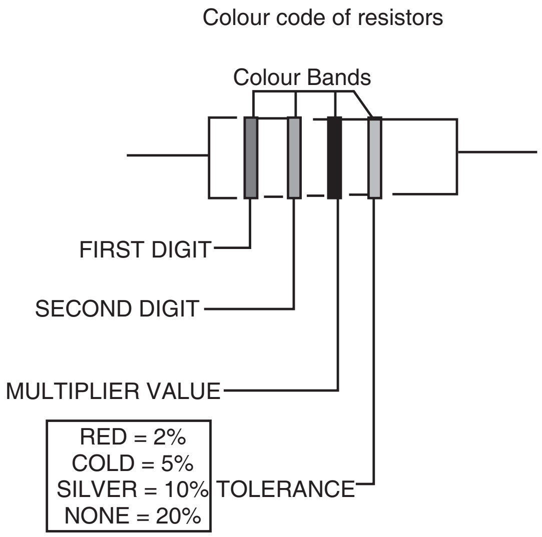
Resistance values marked on resistors are merely a guide to its actual value. A resistor marked 220 Ω with a tolerance of +/- 10% will have a value falling in range of 198 Ω to 242 Ω. In non critical application, where a resistor of say 230 Ω is needed, a 220 Ω with tolerance of +/- 10% will be satisfactory. However in critical applications, a close tolerance resistor of 220 Ω with 1% tolerance will be needed.

Resistors are also available in multiple series of fixed decade values (Decadic means in ratio of 1:10:100: 1000 etc.). However, each fixed decade value is governed by the tolerances involved.

Resistors are usually colour coded. These codes are available in terms of number of colour bands provided on the body of the resistor. Decoding the colour code helps in knowing the value of a resistor. In some of the specific cases the value of a resistor are directly marked on its body.

See Box: 5.1 below for details of the code along with a specific example to use the code.

Box 5.1: Colour code of resistors



DIGIT	COLOUR	MULTIPLIER
	Silver	0.01
	GOLD	0.1
0	BLACK	1
1	BROWN	10
2	RED	100
3	ORANGE	1k
4	YELLOW	10K
5	GREEN	100K
6	BLUE	1M
7	VIOLET	10M
8	GREY	
9	WHITE	

EXAMPLE:

BAND	COLOUR	VALUE
FIRST BAND	RED	2
SECOND BAND	GREEN	4
THIRD BAND	B;ACL	X1
FOURTH BAND	SILVER	10%

HENCE THE VALUE OF

INTEXT QUESTION 5.2

1. In a recording equipment, it is observed that one of the resistor is burnt. The colour bands provided on the burnt resistor are as given below:

   First band = Yellow,

   Second Band = Violet

   Third Band = Yellow

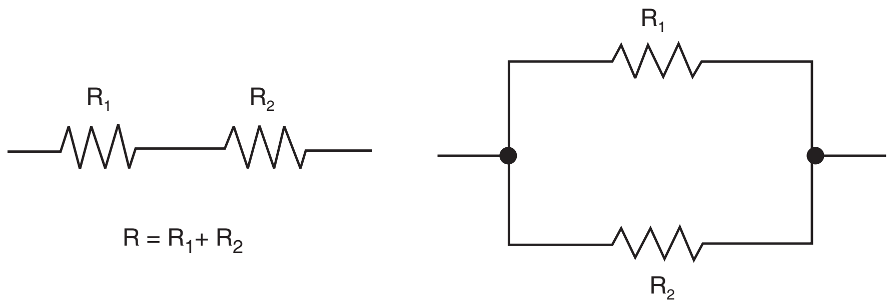
   Fourth Band = Red

   Using the Colour Code provided in the Box 1 above, find the value and tolerance of the resistor to be replaced.

Series and Parallel combination of resistors

In some of the cases, it is not possible to have an exact value of resistor that is required in the electronic circuit. Exact value is then obtained by use of series or parallel combination of resistors as shown in Fig. 5.1.

Notes



$1/R = 1/R_1 + 1/R_2$

Or

$R = R_1R_2/(R_1+R_2)$

A. Series Combination B. Parallel combination

Fig. 5.1: Series and Parallel combination of resistors

In series combination (Fig. 5.1A), the resultant value of resistance is equal to the sum of value of individual resistors. Here $R = R_1 + R_2$

In parallel Combination (Fig. 5.1B), The resultant value of resistance is given as $1/R = 1/R_1 + 1/R_2$

Or $R = R_1R_2/ (R_1 + R_2)$.

Example: A resistance of 4.9 KΩ is needed. What series combination will make it 4.9 KΩ.

Solution: Any of the a series combination of (2.2 KΩ & 2.7 KΩ) or (3.9 KΩ, 1 KΩ) or (4.7 KΩ & 0.2 KΩ) or or (3.3 KΩ & 1.6 KΩ) etc can be made. However, tolerance percentage has to be taken into account provided precise value is required.

Point To Remember

Care should be taken to ensure that power dissipated in each resistor does not exceed individual power rating of resistors.

Voltage Divider

A very common use of resistor is to provide fixed potential division of voltage as shown in Fig. 5.2.

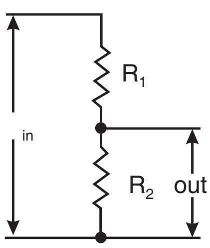


Fig. 5.2: Voltage Divider Circuit

Notes

Here

$$V_{out} = (R_2/R_1 + R_2) \times V_{in}$$

Points to Note

1. Close tolerance resistors (e.g. +/– 1%) should be used to obtain accurate value of voltage division.
2. A major disadvantage of such a simple voltage division is that output voltage will fall in case more current is drawn from the arrangement. The load resistance appears in parallel with R_2 and hence effective resistance of R_2 (i.e. R_2 in parallel with RL) decreases. This disturbs the voltage division ratio Vout/Vin. To ensure that voltage division ratio remain same, RL should be at least 10 times of the value of R_2. In precision applications, RL should at least 100 times of R_2.
3. It is important to note that current drawn by R_1 & R_2 in series from Vinput source is not excessive. In electronic circuit such input current will be of the order 1 to 10mA.

Current Divider

Another important application of resistors that is commonly used for measurement of high value of current is called current divider circuit. Here, a parallel combination of resistors is used to divert a portion of current to another branch of circuit. Figure 5.3 shows one such circuit.

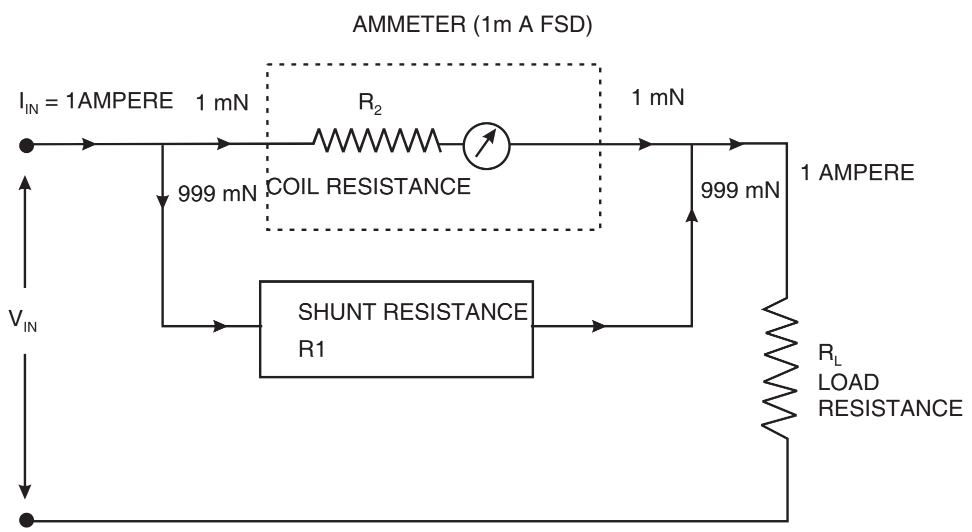


Fig. 5.3: Current Divider Circuit

Here, current through the Ammeter is given as

$$I_{out} = (R_1/R_1+R_2) \times I_{in}$$

Shunt provided in parallel with Ammeter, to provide higher measuring capacity, is an example of parallel combination of resistors.

Points to Remember:

1. Close tolerance resistance (+/- 1%) should be used to obtain accurate value of current division.
2. For precision applications, R2 should be at least 100 times value of R1.

Preset Resistors

Preset resistors enable adjustment of total resistance value without need to change the resistance which requires soldering & de-soldering. Preset resistors of following type are commonly available:

- Open track presets (for horizontal & vertical mounting on PCB)
- Encapsulated carbon & Multi turn cermet type

Variable Resistors

Variable resistors, like presets are commonly available in variety of forms.

1. Carbon track potentiometers.
2. Wire wound potentiometers.

Potentiometers are 3 terminal variable resistors. Carbon potentiometers are available in linear or semi-logarithmic law tracks. They may be in rotary or linear/ slider form.

Sometimes, control of such potentiometers are mechanically linked together to make identical movement of rotary shaft or slider movement. Such resistors are called Ganged, tandem or stereo potentiometers. These are extensively used in Stereo Amplifiers.

Points to Remember

1. Open carrier track presets are relatively more noisy & unreliable. So cermet components should preferably be used.
2. Carbon track variable resistors suffer from track wear & noise. So these should not be used for critical application like low noise amplifier & for instruments amplifiers.

3. Logarithmic law potentiometers should be used in audio equipment for better controls. This is needed as our hearing is also logarithmic.

4. Carbon track controls should never be used for controlling regulated power supply voltage because intermittent contact on carbon slider may result in appearance of full input voltage at the output side of voltage regulator.

Thermistors

Thermistors are those resistors whose resistance changes considerably more with temperature than normal resistors. Thermistors are therefore, employed as temperature sensing & temperature compensating elements in electronic circuits. Thermistors are of following two types.

N.T.C. (Negative Temperature Coefficient) Thermistor

The resistance of Negative temperature Coefficient resistor decreases rapidly as temperature increases as shown in Fig. 5.4.

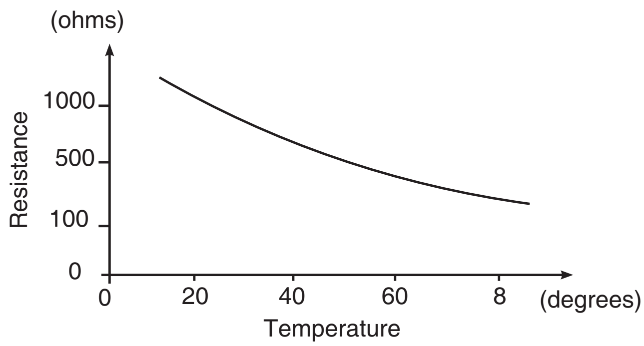


Fig. 5.4: Graph showing the characteristics of NTC Thermistor

P.T.C. (Positive Temperature Coefficient) Thermistors

PTC thermistors, on the other hand, exhibit a typically flat value say 100 Ohms over a range from low temperatures to say 80 deg C. As temperature rises further, their resistance increases rapidly to values more than 10k Ohms as shown in Fig. 5.5.

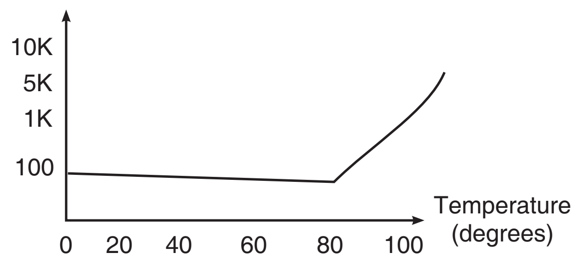


Fig. 5.5: Graph showing the characteristics of a PTC Thermistor

Voltage Dependent Resistors (Varistors)

Varistors are those resistors whose resistance decreases on applying increasing voltage. Fig. 5.6 shows typical characteristics of a varistor.

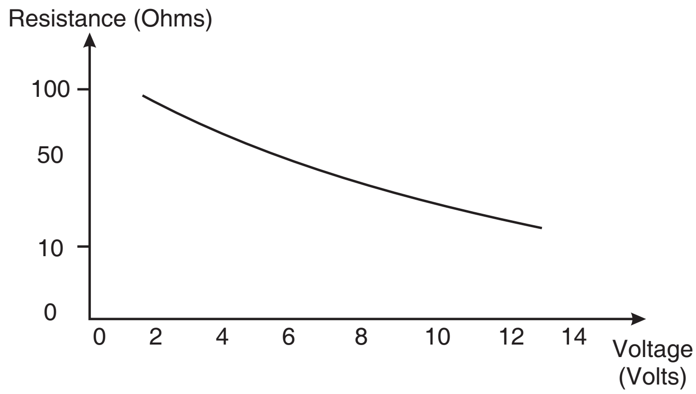


Fig. 5.6: Graph showing the characteristics of a Varistor

Varistors are used to suppress high voltage transients borne in AC mains due to switching On/OFF of inductive loads like motors ,transformers etc. VDRs are also used in D.C. Power supply to protect against application of high voltage to sensitive electronic circuits.

Capacitors

Capacitors are passive components which store energy in electric field (1/2 CV^2). These components do not dissipate energy like resistors. Typical applications of capacitors include:

1. Working as energy reservoir & smoothing circuits in DC power supply
2. Coupling AC signals between different stages of amplifier
3. Decoupling power supply from one stage of amplifier to another.

Electrical characteristics of capacitors are determined by physical dimension and type of dielectric material used in the capacitors. The capacitors are therefore, classified according to the type of dielectric used such as paper capacitors or ceramic capacitors. Both fixed and variable capacitors are commonly used in electronic circuits.

The capacitors are represented by a letter ‘C’ in circuit drawings. The actual measurement of capacity of a capacitor is termed as its ‘capacitance’. The unit of capacitance is the ‘Farad’ and is expressed as $C = Q/V$, where C is in farads, Q is in the charge of coulombs and V is in volts. The farad is a rather large unit, so we use smaller units such as microfarad (=10^{-6} of a farad) or picofarad (=10^{-12} of a farad).

Series and Parallel combination of capacitors

In order to get a required value of a capacitor as per circuit design, both series and parallel combinations are commonly used.

A. CAPACITORS IN SERIES
1/C = 1/C1 + 1/C2
or
C = C1 C2/C(C1 + C2)

B. CAPACITORS IN PARALLEL
C = C1 + C2

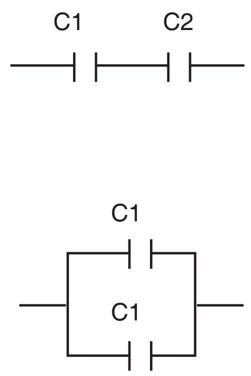


Fig. 5.7: Series and parallel combination of capacitors

As may be seen in Figure 5.7, the resultant capacitance of two capacitors C1and C2 in series is expressed as

$$1/C = 1/C1 + 1/C2 \quad \text{or } C = C1\ C2 / (C1 + C2)$$

In case of parallel combination, it is expressed as $C = C1 + C2$

Following factors should be considered, while selecting a capacitor, for a particular application:

(a) Required value of capacitor (in farad, micro farad, nanofarad, picofarad etc.)

(b) Required voltage rating (i.e. maximum voltage that can be applied without failing it)

(c) Accuracy or tolerance of capacitor (expressed as max. percentage variation from its specified value)

(d) Stability of capacitor (expressed in long term or short term %age variation of capacitive value under specified physical & electrical operating conditions)

(e) Temperature coefficient (expressed as variation in ppm per unit temp change)

(f) Leakage current (flowing in dielectric at rated DC voltage at a given temperature). Alternatively an insulation resistance (ideally infinite) is specified. This is resistance measured between capacitor plates under given set of physical & electrical conditions.

Points to Remember

Please keep in mind following points while selecting/using capacitors for different applications.

1. It is important that capacitor is operated at voltage well below their nominal maximum working voltage.
2. Since working voltage is a related to operating temperature, capacitor should be de-rated for working at higher temperatures.

5.4.2 ACTIVE COMPONENTS

1. Transistors

Transistors fall in two main categories i.e.

1. BJTs – Bipolar Junction Transistors
2. FETs – Field Effect transistors

They are also classified according to semiconductor material used i.e.

Silicon transistors and Germanium transistors

Transistors are also classified according to their field of application;

1. General purpose application
2. Audio frequency application
3. Radio frequency application
4. Switching application

Box: 5.2: Glossary used with transistors

Glossary generally used while referring to transistors:

Low Frequency	: for audio frequency applications typically below 100 kHz.
High frequency	: for RF applications (typically 100 kHz or higher)
Power	: Transistors operating at significant power levels. Such transistors are further sub divided into **AF power type** or **RF power type.**
Switching	: Transistors designed for switching applications.
Low noise	: Transistors which have low noise characteristics & are intended for low level signal applications.
High voltage	: High voltage transistors are designed for use with high voltage DC.
Drivers	: Transistors operating at medium power & voltage levels as penultimate stage of a power amplifier.

Transistor Codes

Transistor codes have either

(a) 2 letters & 3 figures for general purpose transistors OR

(b) 3 letters & 2 figures for special purpose transistors

First letter denotes type of semiconductor material used

A – indicates Germanium semiconductor material

B – indicates Silicon semiconductor material

Second letter denotes application i.e.

C – for Low power, low frequency applications

D – for High power, low frequency applications

F – for Low power, high frequency applications

L – for High power, high frequency applications

The third letter does not have a particular significance in case of transistors for special applications. For example

Transistor AF 115 is a general purpose Germanium transistor for low power & high frequency applications.

Transistor BC 108 is a general purpose Silicon transistor for low power & low frequency applications

Transistor BFY 50 is a special purpose Silicon transistor for low power high frequency applications.

Bipolar Junction Transistors (BJTs)

Bipolar Junction Transistors (BJTs) comprise of PNP or NPN junctions made up of either doped Germanium or doped Silicon material. In both cases, electrodes are labelled as collector, base & emitter. Silicon transistors are much more widely used than Germanium transistors in majority of applications. This is because Silicon transistors do not suffer from the problem of "thermal run away" (see note1) as much as Germanium transistors. Therefore you may find very few Germanium transistors being used now-a-days.

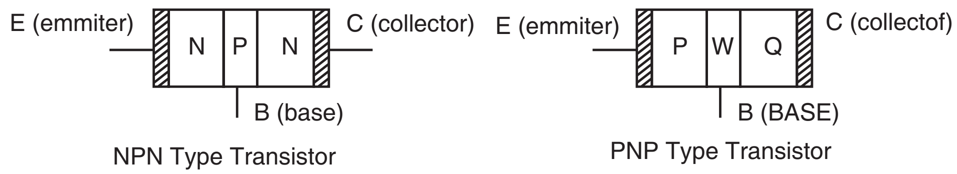


Fig. 5.8: Basic junction design configuration of NPN and PNP type of Transistors

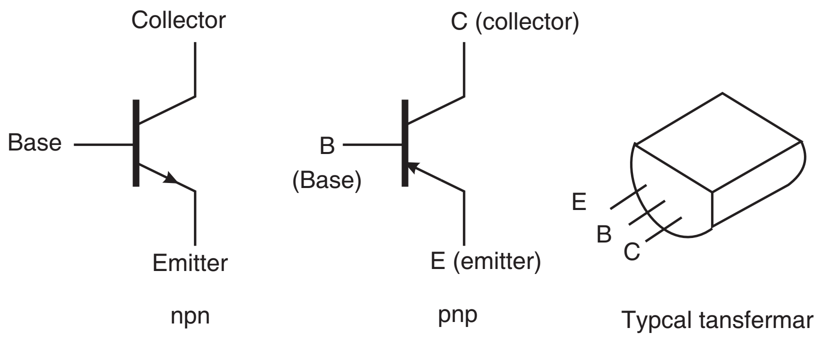


Fig. 5.9: Symbolic representation of NPN and PNP Transistors

Figures 5.8 and 5.9 show basic construction & symbolic representation of NPN & PNP transistors. Base region of both types of transistors (i.e. NPN & PNP) is made very thin so that electrons or holes are swept away across the base region resulting in very small base current. Therefore base current is very small (say 1/ 100 of emitter current). Thus small base current controls 100 times more current in emitter or collector circuit. This forms the basis of control or amplification of a larger current (emitter current or collector current) by a much smaller base current (say 1/100 of emitter current).

Please note that flow of current (conventional) is from collector to emitter in cased of NPN transistor & from emitter to collector in case of PNP transistors. This is in opposition to the flow of electrons

Note 1: In transistors reverse leakage current increases as temperature rises. The increased reverse leakage current results in more heat dissipation & still higher temperature. This cycle continues & results in phenomenon known as "thermal run away". This results in improper operation of transistors circuit & may result in damage of the transistor device itself.

2. Field Effect Transistors

Field effect transistors (FET) may be divided into two main categories, namely junction and insulated gate. Basically a junction FET is a slice of silicon whose conductance is controlled by an electric field acting perpendicularly to current path. This electric field results from a reversed-bias pn junction and because of the importance of this transverse field the device is named as Field Effect Transistor. FETs are basically of two types namely MOSFET (Metal Oxide Semiconductor Field Effect Transistor) and JFET (Junction Field Effect Transistors). Because of difference in their performance characteristics they are used for different specific circuits.

One of the main differences between a junction FET and a conventional transistor is that for the former current is carried by only one type of carrier, the majority carriers. In the case of a latter both majority and minority carriers are involved. Hence the FET is sometimes referred to as a Unipolar Transistor while the conventional type is called a Bipolar Transistor.

Another important difference is that the FET has high input impedance, while ordinary transistors have low input impedance. Because of this, FETs are voltage operated as opposed to current operated bipolar transistor.

The electrodes of FET are named as 'Source', 'Gate' and 'Drain'. The symbolic circuit representation of FET is shown in Figure 5.10 below.

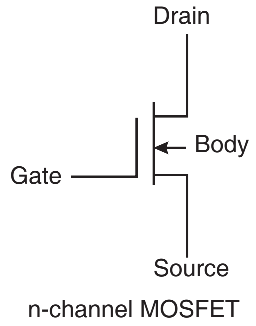


Fig. 5.10: Symbolic representation of a Field Effect Transistor (FET)

INTEXT QUESTION 5.3

State whether the following statements are True or False. Write T or F in the box provided against each statement.

(a) Resistance of a material always decreases if the temperature of the material is decreased. [ ]

(b) In an electric circuit electrons flow from a point of lower potential to a point of higher potential. [ ]

(c) Any amount of current can be passed through a fuse wire provided that the heat is dissipated before the wire melts. [ ]

(d) In both NPN and PNP transistors, current flowing through the base is very small as compared to Collector or Emitter. [ ]

(e) FET is a Unipolar device. [ ]

5.5 INTERFERENCE

While listening to radio programmes, you might have observed that the quality of the recorded message/song you are listening may not be good. It may be noisy or may even be suffering from unwanted interfering signals. Sometimes it may be so annoying that you may be forced to change the radio channel. Any interfering signal or stray pick up constitutes a form of noise. Here the spectrum and amplitude characteristics depend on the interfering signal. For example, 50 Hz pickup from power supply circuit has a sharp spectrum and constant amplitude where as car ignition noise, lightning, and other impulsive interferences are broad in spectrum and spiky in amplitude. Other sources of interference are radio and television stations, electrical equipment, motors, fans and switching regulators etc. Many circuits, as well as detectors and even cables, are sensitive to vibration and sound. These are called microphonic noises.

Sources of Interference

- In electronic circuits, the noise can be due to thermal noise produced by resistors, transistors and other components.
- Carbon potentiometers after a long use may give crackling noise.
- The transistors used in amplifiers may not be of low noise characteristics.
- Interfering signals can enter an electronic instrument (for example, a tape recorder) through the power- line inputs or through signal input and output lines.
- Bad or loose connections, especially at the connector points usually pick up the radio frequencies.
- Use of unshielded cables and unbalanced circuits are common sources, for picking up interfering signals.
- Sometimes, even the crosstalk, between Left and Right channels of the stereo system, is also a source of interference.

Eliminating interference

Numerous effective tricks have been evolved to handle most of these commonly occurring interference problems. Many of these noise sources can be controlled by careful design, selecting of proper components, shielding and filtering.

5.6 EARTHING PROBLEMS

In the preceding section, you have learnt that the interfering signals, 50 Hz pickup (power supply hum), and signal coupling via power supplies and ground paths can turn out to be far greater practical importance than the noise sources generated by discrete components like resistors and transistors. All these problems are due to bad earthing and are commonly referred as ‘earthing’ problems.

Eliminating of Earthing Problems

- These interfering signals can all be reduced to an insignificant level with proper layout and construction, earthing and extensive electrostatic and magnetic shielding.
- A good low resistance separate earth electrode (Earth Pit) may be provided for Audio equipments as close to the building as possible.
- All the audio equipments should be connected to this earth pit by copper straps of sufficient thickness to have low resistance path.
- The lengths of these copper straps should be as small as possible.
- Separate earth straps should be used for each equipment.
- Ensure good and permanent connection at all the junction points.
- Audio earths should not be looped with power and RF earths.
- Use good quality shielded cables for audio interconnection between various equipments.
- Ensure the connections at the connectors are well soldered and not loose.

5.7 CARRY OUT ADJUSTMENTS

Recording of programmes by use of recording equipments is a highly skilled job which is mastered only by practice. Controls provided on various types of equipments are to be adjusted for proper alignment of audio signals. Manufacturers of equipments usually specify certain controls and commands for initial alignment and calibration. Such instructions and guidelines must be followed. For example, a digital recorder may require a low level input signal than an analogue recorder. Every equipment is specified for its minimum and maximum input level. Deviations from these minimum and maximum input levels may deteriorate the quality of programmes.

The feeding of audio signals from one equipment to the next is usually controlled by volume controls (faders commonly called in audio consoles). This arrangement is shown in Fig. 5.11.

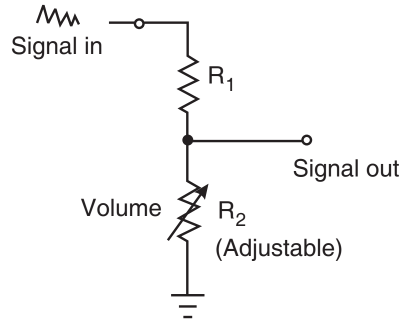


Fig. 5.11: Schematic drawing for audio signal level control

As may be seen in Figure 5.11, output signal to next stage is controlled by variable resistor R2. In case of stereo recordings, volume controls of both left and right channels are ganged to have equal alignment levels of both lines.

Similarly VU (volume Units) meters or PPM (Peak Programme Meters) are provided on consoles and recorders to know the exact input/output levels.

TERMINAL QUESTIONS

1. A moving coil meter, with 1 mA FSD (Full Scale Deflection) and 200 Ω coil resistance is required to read 1A. Determine the value of shunt resistance required. (Refer Fig. 5.3 for circuit drawing)

2. With reference to Figure 5.12, calculate the following values:

   (i) Resistance between terminals B& C

   (ii) Resistance between terminals A&C

   (iii) Total current flowing through the circuit

   (iv) Voltage drop from A to B point

   (v) Voltage drop across B & C

   (vi) Current in 3 Ohm resistor

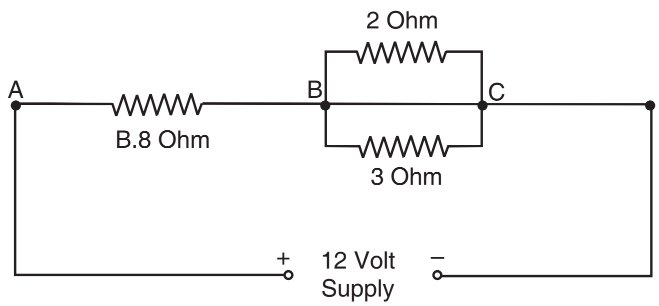


Fig. 5.12

5.8 WHAT HAVE YOU LEARNT

In this lesson, on basic electronics, you have learnt the basic concepts of voltages and currents as applicable to recording equipment. Modern day recording equipments operate on DC voltage sources, obtained either from dry cells or DC supplies obtained from AC mains voltage (220 VAC). Rate of flow of positive charge per unit of time is called conventional current. Magnitude of current through a circuit or part of it, can be determined by use of Ohm's law. According to this law, current in a DC circuit is directly proportional to its applied voltage and inversely proportional to the resistance of the circuit.

All electronic circuits are made up of passive components (resistors, capacitors, diodes and thermistors etc.) and active components (transistors, ICs etc.). These circuits can be analysed if the properties and functions of components are understood. Method of series and parallel combination of resistors and capacitors helps us in finding a suitable combination if the exact replacement of a desired value is not available. By use of colour code, the value of a resistor can be determined easily. Some circuits operate on different supplies. Voltage divider circuits help in providing the desired voltages from the available DC supply source.

Any unwanted signal, getting mixed with wanted signal, is termed as interfering signal. This interfering signal can be due to circuit noise generated by components, RF pickup from a radio transmitter or power hum from AC mains. Many of these noise sources can be controlled by careful design, selecting of proper components, shielding and filtering. By taking necessary steps, like earthing of equipments by providing low resistance paths to nearest earth electrode and use of good quality shielded cables, the problems due to bad earthing can be solved. Necessary precautions and steps to be taken while carrying out adjustments and alignments have also been described.

5.9 ANSWERS TO IN-TEXT QUESTIONS

5.1

According to Ohm's law current flowing through the circuit = Voltage Applied / Resistance of the circuit. Therefore, Current I = 12/300 Ampere = 12X1000/300 m A = **40 m A**

5.2

Using the colour code we can find its value as follows:

First band = Yellow = 4

Second Band =Violet = 7

Third Band (multiplier) = Yellow = X 10 K

Fourth Band (Tolerance) = Red = 2%

Hence the Value of Resistor is = 47 × 10K= **470 K with tolerance of +/- 2%**

5.3

(a) [F] (b) [T] (c) [T] (d) [T] (e) [T]

5.10 ANSWERS TO TERMINAL QUESTIONS

1. (Refer Figure 5.3 for circuit drawing)

Let shunt resistance value be R1Ω. Since coil resistance is 200 Ω. Therefore R2 is 200 Ω. Since Full Scale Deflection of meter is 1mA, therefore Iout = 1mA.

Now, Since

Iout= (R_1/R_1+R_2) x I_{in}

Therefore, 1mA = $(R_1/R_1+200) \times 1000$mA

Therefore, R1+200 = 1000 R1

Therefore 999R1 = 200

Therefore R1 (Shunt Resistance) = 200/999 = **0.2 Ω** (approx).

You must note that, R2 is coil resistance and not some individual discrete component.

2.

(i) Resistance between B& C. Using the formula 1/R = !/R1 +1/R2, we get 1/R = ½ +1/3 = (2+3)/6 = 5/6 or R = 6/5 = **1.2 Ohm**

(ii) Resistance between A& C = Resistance between A&B +Resistance between B&C = 3.8 +1.2 = **5 Ohm**

(iii) Total current in circuit = Voltage between A &C / Resistance between A &C = 12V/5 Ohm = **2.4 Ampere**

(iv) Voltage drop from A to B = Current flowing between A&B x Resistance between A&B = 2.4x 3.8 = **9.12 V**

(v) Voltage drop across B&C = Voltage across A&C – Voltage across A& B = 12V-9.12V = **2.88 V**

(vi) Current in 3 Ohm resistor = Voltage across B&C /Resistance =2.88/ 3 **=0.96 Ampere**

Notes

6

MICROPHONES

6.1 INTRODUCTION

In the previous lessons you were acquainted with principles of sound, sound technology, digital vs analog etc. You learnt about the importance of acoustics.

In this lesson, you will learn about the microphones, their classification etc. Also you shall learn about the major differences between dynamic and condenser microphones along with key considerations while selecting a particular type of microphone.

6.2 OBJECTIVES

After studying this lesson you will be able to:

- Defines microphone and explain the functioning and working of microphones.
- Identify and explain the different types of microphones.
- Categorize the microphones based on their polar patterns.

6.3 CONCEPT

The microphones (mic or mike in short) and the speakers are very common audio equipment. You see them not only in public meetings and conferences, you come across them even when you use your phone. The work of a microphone and a speaker are opposite of each other. A microphone converts sound vibrations into electrical entity (voltage/current) while a speaker converts the voltage/ current into sound vibrations by moving the diaphragm of the speaker and producing vibrations in the air. Basically a microphone has a diaphragm which moves when sound pressure pushes it. This movement can be converted into proportional voltage using several possible transducers. Here, a transducer is a device which receives electrical, mechanical or acoustic waves from one medium and converts them into related waves for a similar or different medium. Thus, it can be said that a microphone (mic) is a transducer that converts acoustical sound energy into electrical energy. Its basic function is therefore to convert sound energy into electrical audio signals which can be used for further processing.

6.4 CLASSIFICATION

We can classify the microphones based on contruction/ directivity as shown below:

Microphones

↓

On the basis of

↓

Construction/type of transducer used

(a) Condenser Microphone
(also capicitor/electrostatic microphone)

(b) Dynamic microphone

(c) Ribbon microphone

(d) Carbon microphone

(e) Piezoelectric microphone

(f) Fiber optic microphone

(g) Laser microphone

(h) MEMS (Micro electrical mechanical sysstem

Pick up or directionality properties

(a) Omni-directional

(b) Unidirectional

(c) Bidirectional

6.4.1 Microphones based on type of transducer /construction

1. **Condenser Microphone** called as **Capacitor Microphone or Electrostatic Microphone** also, is made up of two parallel very thin plates, positively and negatively charged respectively. The diagram 6.1 below shows a condenser microphone.

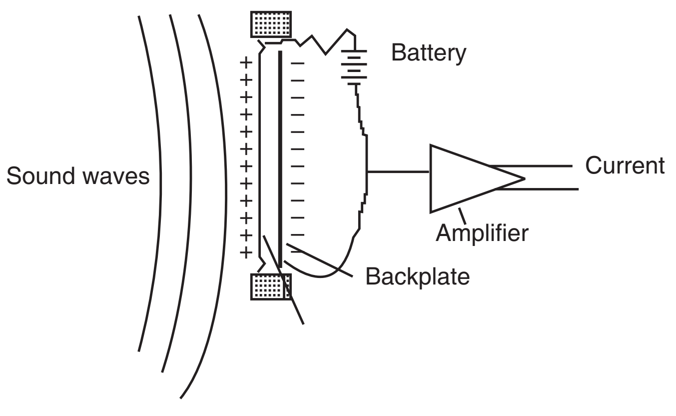


Fig: 6.1: Condenser microphone

It has a very thin diaphragm of thickness 1 to 10 micrometers. One micrometer (or micron) is one millionth of a meter or one thousandth of a millimeter. Close to this plate (metallic or metalised plastic) stands another metallic plate with holes. These 2 plates act as electrodes and are kept at opposite polarities by supplying D C to behave as a condenser, they should be insulated from each other. When sound wave pushes the diaphragm, it vibrates and the capacitance of the condenser (or capacitor) changes. This is because the capacitance is proportional to the potential difference and inversely proportional to the separation between the plates. Any change in the separation changes the capacitance. The capacitance is also dependent upon the medium but as the medium here remains the same, so we ignore this parameter. The values of the resistance and the capacitance are chosen such that the change in voltage is immediately reflected in the voltage across the resistance in series. Any change in sound leads to change of the capacitance and leads to voltage change. The voltage is fed to an amplifier to amplify the level of the signal. Condenser microphones were invented in Bell Labs in 1916.

2. **Dynamic Microphone:** works on the principle of electro-mechanical induction. This type of microphone is called moving coil microphone also. Here a very small coil is used which is attached to a diaphragm and suspended in a magnetic field of a magnet as shown in the diagram: 6.2 below. When sound waves impinge on the diaphragm it vibrates and attached coil moves. This movement of the coil inside the magnetic field produces a emf across the terminals of the coil. The current so produced in the coil is in proportion to the sound.

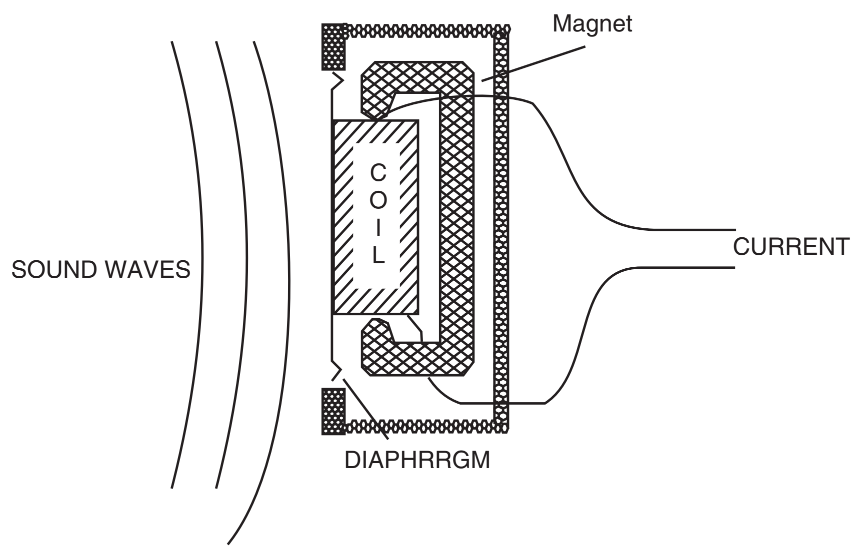


Fig. 6.2: Dynamic microphone

3. **Ribbon Microphone:** A ribbon microphone uses a corrugated ribbon made of a metal is suspended in a magnetic field as shown in the diagram: 6.3 below. Sound causes the ribbon to vibrate. This means change in magnetic flux through the ribbon. This induces an electric current which drives a speaker. When this current is flown through a coil attaches to diaphragm of the microphone, the diaphragm vibrates and produces sound. Special materials developed using nano technologies are being used to make ribbons that will be light but strong. Being light improves the response to sound. The ribbon microphone senses pressure-gradient and not just pressure. Therefore, it detects sound from both sides.

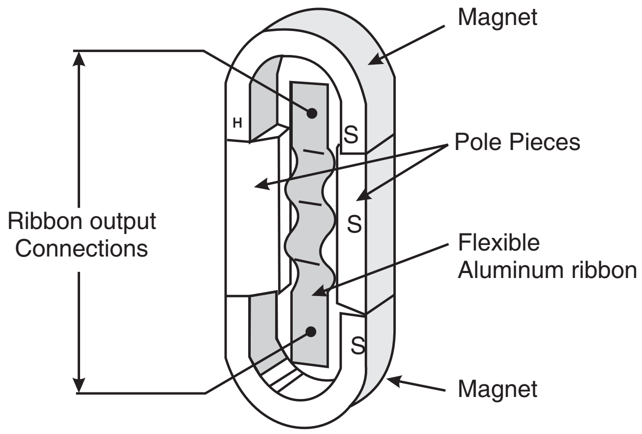


Fig. 6.3: Ribbon microphone

4. **Carbon microphone is** also known as a carbon button microphone (or sometimes just a button microphone), use a capsule or button containing carbon granules pressed between two metal plates.

5. **Piezoelectric microphone** uses the phenomenon of piezoelectricity—the ability of some materials to produce a voltage when subjected to pressure—to convert vibrations into an electrical signal.

6. **Fiber optic microphone** converts acoustic waves into electrical signals by sensing changes in light intensity, instead of sensing changes in capacitance or magnetic fields as with conventional microphones

7. **Laser microphones** are often portrayed in movies as spy gadgets, because they can be used to pick up sound at a distance from the microphone equipment.

8. **MEMS** (Micro Electrical-Mechanical System) microphone is also called a microphone chip or silicon microphone. The pressure-sensitive diaphragm is etched directly into a silicon chip by MEMS techniques, and is usually accompanied with integrated preamplifier.

6.4.2 Microphone Classification based on Polar Patterns

A microphone's directionality or polar pattern indicates how sensitive it is to sounds arriving at different angles about its central axis. Some microphone designs combine several principles in creating the desired polar pattern. Generally the pick up pattern / Polar pattern of microphones fall in following three categories:

(i) Omni-directional

(ii) Uni-directional

(iii) Bi-directional

(i) **Omni directional**

An Omni directional (or non-directional) microphone's response is generally considered to be a perfect sphere in three dimensions as shown in the diagram: 6.4 below. In the real world, this is not the case. As with directional microphones, the polar pattern for an "omnidirectional" microphone is a function of frequency.

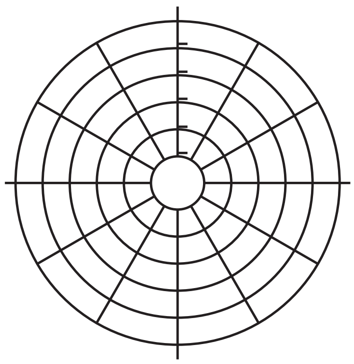


Fig. 6.4: Omni directional

(ii) Unidirectional

A unidirectional microphone is sensitive to sounds from only one direction. The diagram 6.5 below shows a photo and Diagrams 6.7 to 6.11 illustrates a number of these patterns

Fig. 6.5: University Sound US664A dynamic supercardioid microphone

The most common unidirectional microphone is a cardioid microphone, so named because the sensitivity pattern is a cardioid. A cardioid microphone is effectively a superposition of an Omni directional and a Fig. 6.8 microphone; for sound waves coming from the back, the negative signal from the Fig. 6.8 cancels the positive signal from the omnidirectional element, whereas for sound waves coming from the front, the two add to each other.

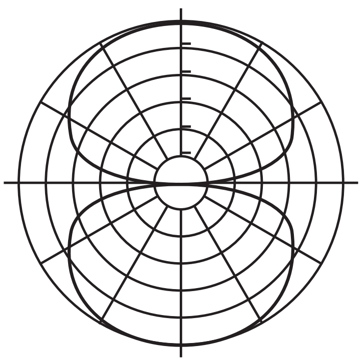


Fig. 6.6: Bi-directional

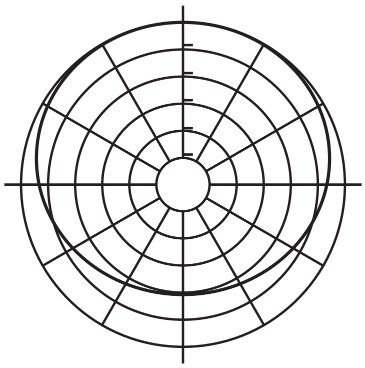


Fig. 6.7: Subcardioid

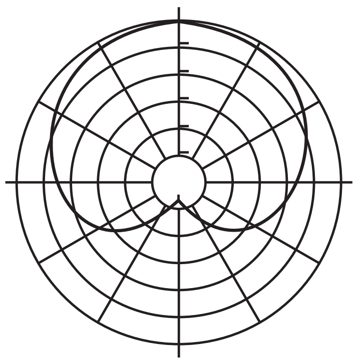


Fig. 6.8: Cardioid

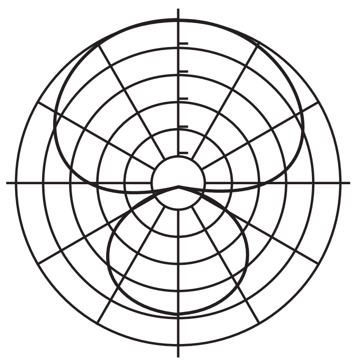


Fig. 6.9: Hypercardioid

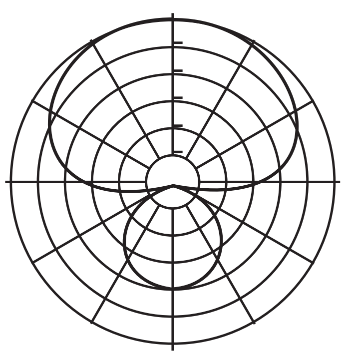


Fig. 6.10: Supercardioid

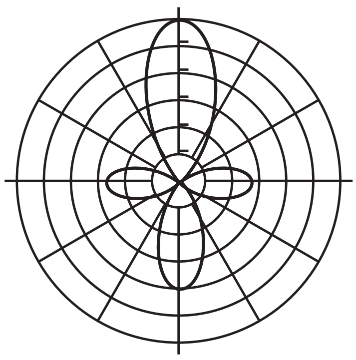


Fig. 6.11: Shotgun

Notes

A **hyper-cardioid (Fig6.9)** microphone is similar, but with a slightly larger figure-8 contribution leading to a tighter area of front sensitivity and a smaller lobe of rear sensitivity. A **super-cardioid** microphone is similar to a hyper-cardioid, except there is more front pickup and less rear pickup. While any pattern between omni and Fig. 6.8 is possible by adjusting their mix, common definitions state that a hypercardioid is produced by combining them at a 3:1 ratio, while supercardioid is produced with a 5:3 ratio **Shotgun**

Fig. 6.12: An Audio-Technica shotgun microphone

Shotgun microphones (Fig 6.12 above) are the most highly directional. They have small lobes of sensitivity to the left, right, and rear but are significantly less sensitive to the side and rear than other directional microphones. This results from placing the element at the back end of a tube with slots cut along the side; wave cancellation eliminates much of the off-axis sound. Due to the narrowness of their sensitivity area, shotgun microphones are commonly used on television and film sets, in stadiums, and for field recording of wildlife etc.

(iii) **Bi-directional**

"Figure 6.8" or bi-directional microphones receive sound equally from both the front and back of the element. Most ribbon microphones are of this pattern. In principle they do not respond to sound pressure at all, only to the *change* in pressure between front and back; since sound arriving from the side reaches front and back equally there is no difference in pressure and therefore no sensitivity to sound from that direction.

INTEXT QUESTIONS 6.1

1. A microphone (mic) is a transducer that converts acoustical …….energy into electrical energy

(a) sound (b) noise (c) mechanical. (d) Kinetic

2. Condenser microphone was invented at Bell Labs in ……

(a) 1917 (b) 1916 (c) 1918 (d) 1919

3. A microphone's directionality or polar pattern indicates how sensitive it is to

   (a) light (b) wind (c) sound (d) energy

4. A unidirectional microphone is sensitive to sounds from ……..direction/directions

   (a) one (b) bi (c) multi (d) corner

5. ……….. microphones are the most highly directional

   (a) Ribbon (b) Carbon (c) Cardiod (d) Shotgun

6.5 FACTORS TO BE CONSIDERED WHILE SELECTING MICROPHONES

Selecting on appropriate type of microphone is an important factor to control noise and produce quality sound. Factors which should be considered before selecting a microphone are:

(i) **Impedance:** Generally, resistance is considered for circuits width dc source. In case ac supply is used and circuit contains inductor and capacitor in addition to resistors, impedance is considered. While selecting microphones impedance has involvement of frequency factor. The length of the cable used for microphone varies as per the impedance needed for a particular circuit or load. In most of the cases the wire impedance is kept low to increase the length of the cable for mic system.

(ii) **Frequency response:** Frequency response is a microphone's capability to service high and low frequency sounds, Ideally, quality microphone can receive frequencies ranging from 20 to 25,000 Hz.

(iii) **Pick up pattern:** This refers to choice of sound from one or all directions. Different microphones are used to cater such varied requirement.

(iv) **Balanced and unbalanced microphone:** Balanced microphones are used for professional recording purposes and they carry three wires while unbalanced connection use 2 wires i.e. central conductor carrying audio signal while shield/ basket carrying ground wire.

(v) **Sensitivity:** it is the ability of microphone to pick up faint sounds and thus termed as sensitivity Highly sensitive microphones are used in studio while in field, low sensitivity based microphones are used as the chances of microphone being affected by strong wind are higher.

Notes

6.6 WHAT HAVE YOU LEARNT

In this lesson you have learnt about microphones and fundamental principle of working of microphone. Different types of mics based up on their working principle were discussed to familiarize with the basic differences of microphones. Later pickup patterns of microphones were discussed based up on their applications. Different mics and their applications were also discussed.

6.7 TERMINAL QUESTIONS

1. What is a microphone?
2. What do you understand by ribbon microphone?
3. Explain a bi-directional microphone.
4. Differentiate between Omni and Uni-directional microphones
5. What are the precautions to be taken while handing microphones.

6.8 ANSWER TO IN-TEXT QUESTIONS

1. A 2. B 3. C 4. A 5. D

6.9 REFERENCES

1. Audio Production/Module4/Unit 13 Certificate in Community Radio Technology CEMCA

Notes

7

LOUDSPEAKERS

7.1 INTRODUCTION

In the previous lesson, you have studied about microphones, their classification along with working principle. As a microphone converts sound vibrations into electrical entity (voltage/current), a speaker converts the voltage/current into vibrations in the air.

In this lesson, you will learn about the various categories of loudspeakers, their working, classification and specifications etc.

7.2 OBJECTIVES

After studying this lesson you will be able to:

- define a loudspeaker.
- identify what a loudspeaker is.
- explain the working procedure of loudspeakers
- classify the types of loudspeakers.

7.3 MEANING OF LOUDSPEAKER

Loudspeaker is an equipment that converts electrical signals/ impulses into sound. The term “loudspeaker” may refer to individual transducers, which are popularly known as “drivers” or to complete speaker systems consisting of an enclosure containing one or more drivers. In technical terms, a loudspeaker (Fig. 7.1) is an electro acoustic transducer that produces sound in response to an electrical audio signal input.

Therefore, it can be said that the loudspeaker is an equipment where the sound chain begining with the microphone, whose signal is sent to a console or mixer, hardware or software based for routing and then processed, recorded and is heard through a loudspeaker.

7.4 USES OF LOUDSPEAKERS

Primarily loudspeakers are used for following four purposes:

(a) ***Aural Communication*** : Loudspeakers have been an integral part of our day to day life, serving the purpose of aural communication systems, conferencing systems etc., therefore, they perform the function of information dissemination through sufficient sound level to a large audience / group in a convenient and safer way.

(b) ***Sound reinforcement :*** In various locations such as in auditoriums, amphitheatres etc., the sound created by the voice is not of sufficient loudness to be heard or understood satisfactorily, thus in those locations sound reinforcement system can provide the acoustic gain to overcome such problems and produce sound at sufficient level.

(c) ***Sound production :*** In various stages of musical production, a loudspeaker can supplement recording the sounds of various types of musical instruments such as guitars, bass and keyboards etc.

(d) ***Sound reproduction :*** Sound reproduction system is required where the recorded sound or music has to played again and again. For example, in movie theatres, announcement systems at railway or metro stations, recording studios, etc.

REMEMBER

There are many factors deciding which loudspeaker is the "best". Among them are: *frequency response, linearity, amplifier power, distortion, dynamic range, sensitivity, polar response, and polarity*; the interaction between the room's acoustics and loudspeaker; the loudspeaker's placement within the room; and how the loudspeaker is mounted.

Fig. 7.1: A horn loudspeaker

DO YOU KNOW?

- Johann Philipp Reis installed an electric loudspeaker in his telephone in 1861.
- Alexander Graham Bell patented his first electric loudspeaker (capable of reproducing intelligible speech) as part of his telephone in 1876, which was followed in 1877 by an improved version from Ernst Siemens. The modern design of moving-coil (also called 'dynamic') drivers was established by Oliver Lodge in 1898. The first practical application of moving-coil loudspeakers was established by Danish engineer Peter L. Jensen and Edwin Pridham, in Napa, California. Jensen was denied patents, for years after the invention of the loudspeaker.
- The moving-coil principle commonly used today in direct radiators was patented in 1924 by Chester W. Rice and Edward W. Kellogg.
- About the same period, Walter H. Schottky invented the first ribbon loudspeaker together with Dr. Erwin Gerlach.

7.5 COMPONENTS OF LOUDSPEAKERS

A loudspeaker comprises of various parts . These are transducer, radiator, enclosure and crossover

(a) Transducer: An electro-mechanical transducer contains three elements: coil, diaphragm and suspension. The coil converts electrical energy into mechanical energy and the diaphragm converts mechanical energy into acoustic energy. A suspension supports the diaphragm, allows it to move in an appropriately constrained fashion , exerts a restoring force proportional to displacement from its equilibrium position and provides a dampening force proportional to the velocity of motion that serves to prevent the diaphragm from oscillating in an undesired manner.

(b) Radiator: A Cone type structure that help radiate sound energy at a wide angle.

(c) Enclosure : To house the loudspeaker body and reinforce the sound signal as well as provide front to back isolation.

(d) Crossover: Multi way loudspeakers incorporate crossover network which is a collection of electrical filters each of which allows a specific portion of the frequency spectrum to pass through it. The filtered signal is then applied to one of the bands in the loudspeakers. The types of electrical filters used to execute the crossover function are low pass, high pass, and band pass to reproduce the sound in respective bands. Accordingly, using the speakers of smaller size cone reproduce higher frequencies and are called 'treble loudspeakers' While speakers with larger size cone reproduce lower frequencies and called 'bass loud speakers'. Rest are called 'midband loudspeakers'.

INTEXT QUESTIONS 7.1

Choose any one of the following:

1. A loudspeaker is the equipment that converts .... into sound

(a) electrical impulses (b) magnetic impulses

(c) ultrasonic impulses (d) electro-magnetic impulses

2. The term " Bass loudspeaker" may refer to a type of loudspeaker which enhances .... frequencies

(a) Tracks (b) Low

(c) high (d) mid

3. .... installed an electric loudspeaker in his telephone in 1861

(a) Albert Einstein (b) Walter H. Schottky

(c) Alexander G. Bele (d) Johann Philipp Reis

4. The .... commonly used today in direct radiators was patented in 1924.

(a) moving-coil principle (b) Moving-coil galvanometer

(c) Moving-coil aerostat (d) Moving-coil magnetism

5. Walter H. Schottky invented the first .... together with Dr. Erwin Gerlach

(a) ribbon loudspeaker (b) Carbon loudspeaker

(c) silica loudspeaker (d) magnetic loudspeaker

7.6 CONSTRUCTION OF LOUDSPEAKERS

The cone of a loudspeaker, usually made of paper, plastic or metal, is attached on the wide end to the suspension. The suspension, or surround, is a rim of flexible material that allows the cone to move, and is attached to the driver's metal frame, called the basket. The narrow end of the cone is connected to the voice coil. The coil is attached to the basket by the spider, a ring of flexible material. The spider holds the coil in position, but allows it to move freely back and forth (as shown in Fig. 7.2 and 7.3) and opened out details in the Fig. 7.4 and Fig. 7.5 show you the different parts/ components of loudspeaker, their mounting and working

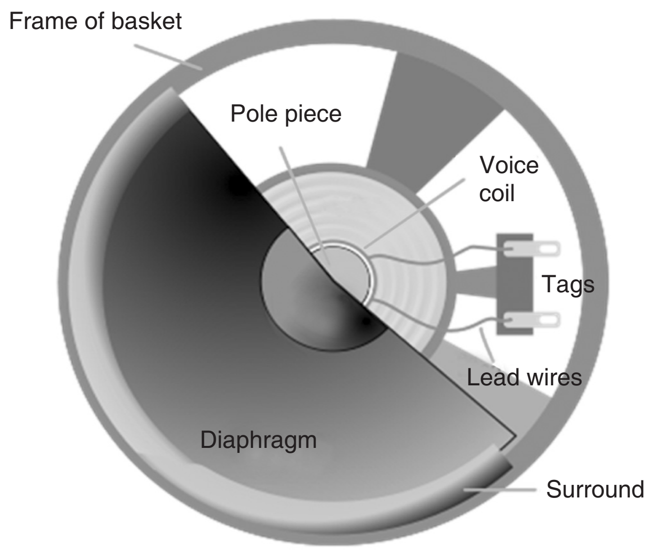


Fig. 7.2 : Constructional details of Loudspeaker

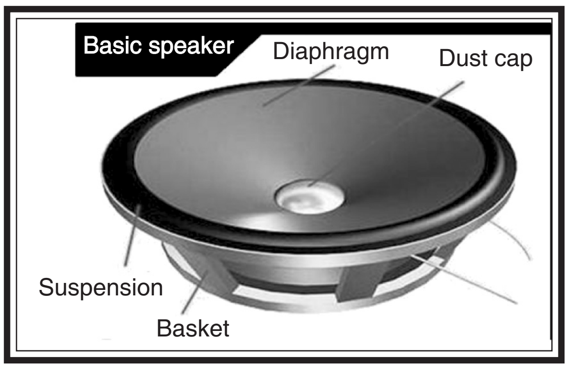


Fig. 7.3: Parts of a loudspeaker

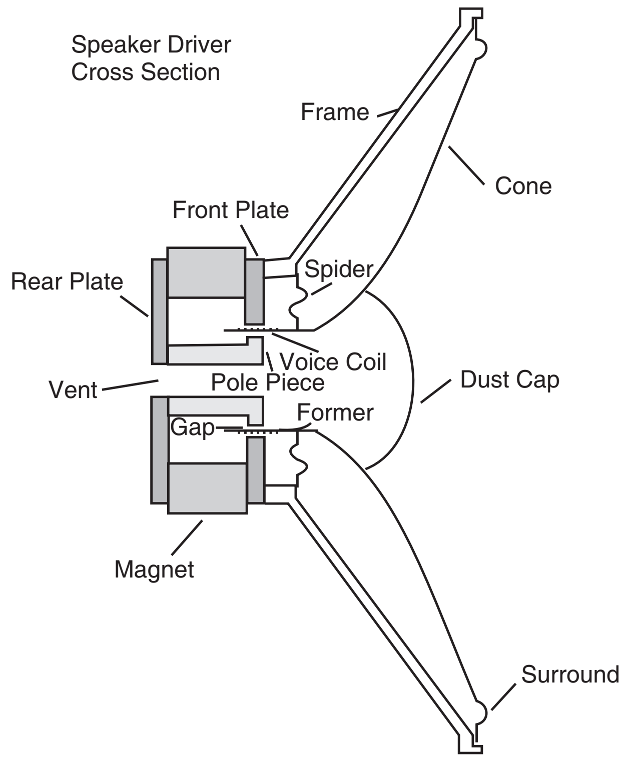


Fig. 7.4: Inside components of a loudspeaker

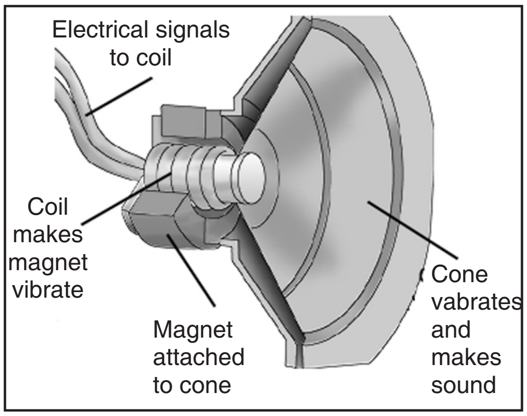


Fig. 7.5: How loudspeaker works

7.7 WORKING OF A LOUDSPEAKER

Whenever an electrical audio frequency signal is applied to the terminals/ coil of the loudspeaker, Fig. 7.5 above, the interaction between the force produced by the flow of current in the coil and the magnetic force of the magnet in the loudspeaker makes the suspended body of the coil to exhibit displacement motion or vibrates in proportional to the strength and frequency of the audio signal. This motion of the coil through a diaphgram and the attached cone produces vibrations in the surrounding air converting the electrical energy into acoustical energy thus producing high level sound waves, The loud speaker thus reproduces the sound signal in correspondence to the audio signal applied which was in turn the replica of original low level sound signal.

7.8 SPECIFICATIONS OF SPEAKERS

Speaker specifications depend on -

- Speaker or driver type
- Size
- Rated Power
- Impedance
- Baffle or enclosure type
- Number of drivers
- Crossover frequency
- Frequency response
- Sensitivity
- Maximum sound pressure level

7.9 LOUDSPEAKER

Classification based on mounting arrangement:

1. Floor standing Speakers

Floor standing or tower loudspeakers are a set of loudspeakers which have been mounted in a vertical enclosures/boxes for floor stand as shown in Fig. 7.6.

Advantages

An extremely wide frequency response and dynamic range make floor-standers the choice where performance is the primary purchasing criteria. And while they tend to be large, many current models feature slender cabinets with small footprints, minimizing placement difficulties and visual impact.

Disadvantages

When space is at a premium towers simply might not fit. Floor-standers should be located 2-3 feet from nearby walls for best performance.

Fig. 7.6: Floor standing speaker

2. Bookshelf Speakers

Bookshelf speakers (Fig. 7.7 below) work where towers do not. These speakers are not only more placement friendly but, since small enclosures are more rigid, they produce less sonically degrading box resonance than all but the best towers.

Advantages

Usually modest in price as well as size, bookshelf speakers fit rooms and budgets that cannot accommodate a pair of towers. The small, solid cabinets are both versatile and able to excel in bookcases, atop shelves or hung on walls-and feature excellent midrange clarity.

Disadvantages

Reduced cabinet volume and driver surface area limit the dynamic and bass frequency range of bookshelf speakers, and can also compromise power handling and efficiency. Fortunately, the addition of a subwoofer can overcome these problems.

Fig. 7.7: Bookshelf speaker

Notes

3. Subwoofer/Satellite Systems

When even the smallest bookshelf speakers are too visible to fit one's lifestyle, a subwoofer/satellite (sub/sat) system is required. By combining palm sized satellites with a subwoofer designed specifically to work with them, sub/sat systems have become one of the most popular categories in home audio. See fig 7.8 below.

Advantages

The big advantages here are size, placement flexibility and cosmetics. The satellites can be placed just about anywhere, on a shelf, on the wall, in a cabinet or on a table. Most are small enough to fit anywhere and are hard to spot when placed alongside books and bric-a-brac.

Disadvantages

Those little satellites cannot reproduce bass of their own, making it tough to achieve a seamless blend between satellite and sub. A sub/sat system may not be right for a very large room to fill with sound. The other issue is bass response. Some of the so-called subwoofers in these systems are passive.

Fig. 7.8: Subwoofer/Satellite System

4. In-Wall Speakers

For environments where box-type (tower or bookshelf) loudspeakers are unacceptable, in-wall speakers' flush-mount in holes cut into the walls are used (Fig. 7.9 below). Even the area of bass reproduction can be addressed with in-wall, in-floor and in-ceiling subwoofers. A completely invisible, no compromise sound system can be fairly easily obtained with this type of speaker.

Advantages

Since they consume no floor or bookshelf space and can be easily concealed, in-walls work when and where other speakers will not. They are also useful as rear surround speakers when the room configuration makes it impossible to properly place box speakers.

Disadvantages

These days there aren't very many since this speaker category has made significant strides in the past few years. They can produce the same level of high performance sound quality as any other type of speaker.

Fig. 7.9: In-wall Speaker

7.10 TYPES OF LOUDSPEAKERS

1. **Horn loudspeakers** are the oldest form of loudspeaker system. The use of horns as voice-amplifying megaphones dates at least to the 17th century, and horns were used in mechanical gramophones as early as 1857.

2. **Piezoelectric speakers** are frequently used as beepers in watches and other electronic devices, and are sometimes used as tweeters in less-expensive speaker systems, such as computer speakers and portable radios.

3. **Magnetostrictive transducers**, based on magnetostriction, have been predominantly used as sonar ultrasonic sound wave radiators, but their usage has spread also to audio speaker systems.

4. **Electrostatic loudspeakers** use a high voltage electric field (rather than a magnetic field) to drive a thin statically charged membrane they are driven over the entire membrane surface rather than from a small voice coil.

5. **Ribbon speaker** consists of a thin metal-film ribbon suspended in a magnetic field. The electrical signal is applied to the ribbon, which moves with it to create the sound.

6. **Bending wave transducers** use a diaphragm that is intentionally flexible. The rigidity of the material increases from the center to the outside. Short wavelengths radiate primarily from the inner area, while longer waves reach the edge of the speaker.

7. **Flat panel loudspeakers,** most accurately called exciter/panel drivers can be made in a neutral color and hung on walls where they are less noticeable than many speakers, or can be deliberately painted with patterns, in which case they can function decoratively.

8. **Heil Air Motion Transducer**, a pleated diaphragm is mounted in a magnetic field and forced to close and open under control of a music signal. Air is forced from between the pleats in accordance with the imposed signal, generating sound.

9. **Plasma arc loudspeakers** use electrical plasma as a radiating element. Since plasma has minimal mass, but is charged and therefore can be manipulated by an electric field, the result is a very linear output at frequencies far higher than the audible range.

10. **Digital speakers** have been the subject of experiments performed by Bell Labs as far back as the 1920s.The design is simple; each bit controls a driver, which is either fully 'on' or 'off'.

11. **Transparent ionic conduction speaker** was introduced in 2013, which is a 2 layers transparent conductive gel and a layer of transparent rubber is there in between to make high voltage and high actuation work to reproduce good sound quality. The speaker is suitable for robotics, mobile computing and adaptive optics fields.

12 **Thermo-acoustic speaker** is based on the working mechanism of 'thermo acoustic effect'. Sound frequency electrical currents are used to periodically heat the Carbon Nanotube (CNT) thin film of the speaker and thus result in sound generation in the surrounding air.

7.11 HEADPHONES/ EARPHONES

Headphones/ earphones are very small size paired speakers mountable over the head or place in the ears. These work on the same principle as a loudspeaker of converting an electrical audio signal into sound waves. Since headphones are worn over the ears directly, their sound is not affected by the environmental interference from the room. Headphones help prevent feedback when live microphones are around. These require low power drivers as the output required is not high. These are widely used in the studios for sound monitoring etc.

There are three types of headphones:

(a) **Ear bud headphones:** These are mostly used with all kinds of portable music players and mobile phones. Also known as earphones. (Fig 7.10)

Fig. 7.10: Ear phones

(b) **On ear headphones:** These headphones sit on the ears rather than over them; thus they are smaller and lighter than over the ear models. Since they don't cover the ears, ambient noise tends to enter the ears, making it difficult to monitor audio in critical conditions.

Fig. 7.11: On ear headphone

(c) **Over the ear headphones:** These headphones were traditionally used, which enclose whole ear thus making them comfortable to wear for long. Such headphones are best suited for audio monitoring purposes in the studio as well as in the field. .(Fig. 7.12).

Fig. 7.12: Over the ear headphone

Notes

7.12 WHAT HAVE YOU LEARNT

In this lesson, you have learnt about loudspeakers and their classifications based upon the working principle and mounting arragement. The construction details and the components of loudspeakers was explained to provide in-depth knowledge about loudspeakers. Later in the section different type of other loudspeakers were also discussed based upon their working principle.

7.13 TERMINAL QUESTIONS

1. How does a Loudspeaker work?
2. What do you mean by drivers?
3. Mention the specifications of speakers
4. What are the different types of loudspeakers?
5. Discuss the advantages and disadvantages of floor standing speakers?

7.14 ANSWERS TO IN TEXT QUESTIONS

1. A 2. B 3. D 4. A 5 A

7.15 REFERENCES

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8

OTHER SOUND EQUIPMENTS

8.1 INTRODUCTION

In the previous lesson, you have studied the details about loudspeakers. Now, let us learn about the other sound equipment. In this lesson, we will cover what are the other sound equipment are like amplifiers and their types, working, and other recording and playback equipment. Other sound equipment like CD player, DVD Player, sound mixers have become part and parcel of the overall sound system may be in a home or a big auditorium.

8.2 OBJECTIVES

After reading this lesson, the learner will be able to :

- understand basics of audio recorders and DVD playersetc.
- describe pre-Amplifiers and amplifiers
- explain frequency response of sound equipment
- explain Total Harmonic Distortion and Signal to Noise Ratio(SNR)

8.3 PRE-AMPLIFIERS

A preamplifier (preamp) is an electronic amplifier that prepares a small electrical signal for further amplification or processing. A preamplifier is required to amplify a signal, when the source level is too low and has to be pre-amplified in order to be able for further processing, control or any other use. The short form used for preamplifier, *preamp*, has become more used in spoken and written language simply because it is shorter. Other spellings are *pre amp* /*pre-amp* and *pre amplifier* /*pre-amplifier*.

DO YOU KNOW?

A preamplifier *measures signals from sensors or other devices in a variety of situations such as sound, temperature, light, movement, pressure etc. In the equipments for industrial, scientific, telecommunications, space, fiber optics or data links, the frequency range may cover from dc up to many hundred GHz.*

REMEMBER

Preamplifiers are of three types:

- the current-sensitive preamplifier
- the parasitic-capacitance preamplifier
- Charge sensitive preamplifier

8.4 WORKING OF PRE-AMPLIFIERS

Voltage and Current

A pre-amplifier, or preamp, takes electrical current from a transducer and increases its voltage gain to a higher level.

Audio signals

Many electronic audio devices have weak audio signal output. A preamp boosts that signal to what is called line-level. Line-level essentially means, the sound coming though the speakers or amplifier is loud enough to be heard at the sound systems' nominal voltage output level or good eough to be fed to main amplifier.

8.5 AUDIO AMPLIFIERS

DO YOU KNOW?

The audio amplifier was invented in 1909 by Lee De Forest when he invented the triode vacuum tube. The triode was a three terminal device with a control grid that can modulate the flow of electrons from the filament to the plate. The triode vacuum amplifier was used to make the first AM radio. Audio power amplifiers based on transistors became practical with the wide availability of inexpensive transistors in the late 1960s.

8.6 POWER AMPLIFIER

A power amplifier is an electronic device that receives an electrical signal and reprocesses it to amplify, or increase, its power. The boost in power is achieved by significantly increasing the input signal's voltage. A power amplifier is used to power an output source, such as a stereo speaker, a relay or a motor. Its applications include public address systems, theatrical and concert sound reinforcement systems too.

Amplifier classification

- Class A: Single-ended; the amplifier device is biased about the center of the input signal swing.
- Class B: Push-pull; each device conducts over half the input signal swing.
- Class AB: Push-pull; each device conducts over slightly more than half the input signal swing to simplify crossover.
- Class C: Used in radio-frequency applications, the output device drives a resonant "tank" circuit consisting of an inductor and one or two capacitors. It conducts for only a short portion of each input cycle.
- Class D: It's found primarily in audio applications – either in vehicles, where it achieves high output levels, or in personal audio devices, where its efficiency contributes to long battery life. In a class D amplifier, power field-effect transistors (FETs) are driven to produce an output square-wave that switches between a high and low level at a frequency outside the range of human hearing. Instead of modulating the amplitude, internal circuitry modulates the duty cycle of the square-wave at a rate corresponding to the level of the input signal when the output is filtered down to audio band.

  Classes E and F are subsets of Class C. Classes G and H are like class AB amplifiers, but with multiple power rails.

INTEXT QUESTIONS 8.1

1. A preamplifier is a/an .... amplifier that prepares a small electrical signal

   (a) electronic (b) electrical

   (c) electro-static (d) magnetic

2. A pre-amplifier takes electrical current from a ....

   (a) transistor (b) capacitor

   (c) transducer (d) timer

3. The audio amplifier was invented in 1909 by ……………..

   (a) C Das (b) Lee De Forest

   (c) J C Bose (d) James Watt

4. Most audio amplifiers are …………..amplifiers

   (a) linear (b) parallel

   (c) modular (d) parabolic

5. A power amplifier is used to power an ………………

   (a) input source (b) alternate source

   (c) output source (d) external source

8.7 FREQUENCY RESPONSE

Frequency response is the quantitative measure of the output spectrum of a system or device in response to a stimulus, and is used to characterize the dynamics of the system. Frequency response is a specification used in amplifiers, pre-amplifiers, CD players, tape decks and other audio components to measure how uniformly it reproduces sounds from the lowest tones to the highest.

Expression of Frequency Response

Frequency response is usually measured within the range of hearing, from a low of 20Hz to a high of 20kHz, although some believe that frequencies above and below this range, known as wideband frequency response are equally important. Frequency response specifications indicate how well the device remains uniform. For example, a frequency response specification of 20Hz-20kHz +/- 3dB indicates that the maximum variation in level or volume from the lowest to the highest tone (frequency) will not exceed three decibels. A range of three dB is common in frequency response specifications.

Uniform or flat frequency response is important because every instrument or voice should be heard as originally recorded. The delicate sound of a triangle should be heard as well as the crash of a cymbal. The amplifier or other device should not raise or lower the level of any sound from the original recording.

Nonlinear Frequency Response

If the system under investigation is nonlinear then applying purely linear frequency domain analysis will not reveal all the nonlinear characteristics. To overcome these limitations generalized frequency response functions, and nonlinear output

Notes

frequency response functions have been defined that allow the user to analyze complex nonlinear dynamic effects. The nonlinear frequency response methods reveal complex resonance, inter modulation and energy transfer effects that cannot be seen using a purely linear analysis and are becoming increasingly important in a nonlinear world.

Need for Wideband Frequency Response

As you are already aware, frequency is a term used to describe tone and it is measured in Hertz. Low tones are known as bass, midrange tones are in the range of the human voice, and high tones are musical instruments such as a cymbal. The human ear is capable of hearing low tones from approximately 20 Hertz (abbreviated 20Hz) to high tones up to 20 kilohertz (abbreviated 20 kHz). 20 Hz is very low, deep bass and 20 kHz is probably beyond the range of human hearing. The range of human hearing is dependent on the health of the ear and age. As we age, our range of hearing is reduced.

8.8 TOTAL HARMONIC DISTORTION (THD)

Total Harmonic Distortion or THD is an amplifier specification that compares the output signal of the amplifier with the input signal and measures the level differences in harmonic frequencies between the two. The difference is called total harmonic distortion.

Total harmonic distortion is measured as a percentage, such as 0.004% THD. This means that the level of harmonic distortion is 0.004% of the total output signal. Lower percentages are better.

DO YOU KNOW?

In reality, total harmonic distortion is hardly perceptible to the human ear. Every component adds some level of distortion, but most distortion is insignificant and small differences in specifications between components mean nothing. Some components have distortion so low it cannot be accurately measured. Listening to a component and evaluating its sound characteristics is the most important way to judge a product. Other considerations, such as room acoustics and selecting the right speakers are more important than the percentage of total harmonic distortion

8.9 SIGNAL TO NOISE RATIO (SNR)

Signal to noise ratio is a specification that measures the level of the audio signal compared to the level of noise present in the signal. Signal to noise ratio specifications are common in many components, including amplifiers, phonograph players, CD/DVD players, tape decks and others. Noise is described as hiss, as in tape deck, or simply general electronic background noise found in all components.

The signal to noise ratio is the difference between the noise floor and the reference level. The reference level is determined by the person making the measurements. For amplifiers, the reference may be full power, one volt, and one watt into a given load or any number of other things. For you to compare two pieces of equipment which were tested by different methods, you must know precisely what reference was used.

8.10 RECORDING AND PLAYBACK EQUIPMENT

Turn table

Turntablism is the art of manipulating sounds and creating music using phonograph turntables and a DJ mixer. The word *turntablist* was coined in 1995 by D J Babu to describe the difference between a DJ who just plays records and one who performs by touching and moving the records, stylus and mixer to manipulate sound. The new term coincided with a resurgence of the art of hip hop style-style DJ-ing in the 1990s. John Oswald described the art: "A phonograph in the hands of a 'hip hop/scratch' artist, who plays a record like an electronic washboard with a phonographic needle as a plectrum, produces sounds which are unique and *not* reproduced the record player becomes a musical instrument." Some turntablist DJs use turntable techniques like beat mixing/matching, scratching and beat juggling. Some turntablists seek to have themselves recognized as traditional musicians capable of interacting and improvising with other performers.

Operation of a Turntable:

1. Lift the dust cover from the turntable
2. Place the record onto the platter
3. Put the platter into motion
4. Lift or cue the tone arm
5. Lower the stylus onto the record
6. Put the tone arm back into place when the record is over

8.11 GRAMOPHONE RECORD

A gramophone record is a type of analog storage medium. It stores recorded music (or other sounds). It was popular during most of the 20th century.

Gramophone records are played on a phonograph ("record player") shown in Fig 8.1. A gramophone record is a flat disk that is made of plastic. The sound is recorded on a very fine line or groove which goes around and around in a spiral from the outside edge of the disk to the center. The phonograph plays the sound with a needle that touches the groove. A record usually has different music on each side. When made of vinyl they were also known as vinyl records. Most music made in the 20th century used this format.

Fig. 8.1: Gramophone record player

8.12 AUDIO TAPE RECORDER

An audio tape recorder is an audio storage device that records and plays back sounds, including articulated voices, usually using magnetic tape, either wound on a reel or in a cassette, for storage. In its present day form, it records a fluctuating signal by moving the tape across a tape head that polarizes the magnetic domains in the tape in proportion to the audio signal. Figure 8.2 and 8.3 below show cassette and spool tape recorders respectively.

Fig. 8.2: Cassettee Tapr Recorder

Fig. 8.3: Spool Tape recorder

An audio tape recorder, tape deck or tape machine is an audio storage device that records and plays back sounds, including articulated voices, usually using magnetic tape, either wound on a reel or in a cassette, for storage. In its present day form, it records a fluctuating signal by moving the tape across a tape head that polarizes the magnetic domains in the tape in proportion to the audio signal. Tape-recording devices include reel-to-reel tape deck and the cassette deck.

REMEMBER

Earliest variant: non-magnetic wax strip recorder

The earliest known audio tape recorder was a non-magnetic, non-electric version invented by Alexander Graham Bells' Volta laboratory and patented in 1886 (U.S. Patent 341,214

Photoelectric variant

In 1932, after six years of developmental work, Merle Dust on, a Detroit radio engineer created a tape recorder that used a low-cost chemically treated paper tape, capable of recording both sounds and voice.

Magnetic recording

Magnetic recording was conceived of as early as 1877 by the American engineer Oberlin smith and demonstrated in practice in 1898 by Danish engineer Valdemar Poulsen. Analog magnetic wire recording, and its successor, magnetic tape recording, involve the use of a magnetizable medium.

Steel wire magnetic recorder variant

The first wire recorder was the Valdemar Poulsen Telegraphone of the late 1890s, and wire recorders for law/office dictation and telephone recording were made almost continuously by various companies (mainly the American Telegraphone Company) through the 1920s and 1930s.

8.13 COMPACT DISC

Compact disc, or CD for short, is a digital optical disc data storage format. The format was originally developed to store and play back sound recordings only (CD-DA), but was later adapted for storage of data (CD-ROM). Several other formats were further derived from these, including write-once audio and data storage (CD-R), rewritable media (CD-RW), Video Compact Disc (VCD), Super Video Compact Disc (SVCD), Photo CD, Picture CD, CD-i, and Enhanced Music. Audio CDs and audio CD players have been commercially available since October 1982. Standard CDs have a diameter of 120 milli metres (4.7 in) and can hold up to 80 minutes of uncompressed audio or 700 MiB (actually about 703 MiB or 737MB) of data. The Mini CD has various diameters ranging from 60 to 80 milli metres (2.4 to 3.1 in); they are sometimes used for CD singles, storing up to 24 minutes of audio or delivering device drivers. Figure 8.4 shows a CD player.

Fig. 8.4: A C D Player

Variations of compact discs designed for use with computers include CD-ROM, CD-R, CD-RW, DVD-ROM, DVD-RAM, DVD-R, DVD+RW and Photo CD

DO YOU KNOW?

The first CD recorders were made available in 1988, but were not an option for the average home recorder because, with the requisite hardware and software, they cost upwards of $100,000. At a weight of 600 pounds, the Meridian Data CD Professional was the first CD recorder. Today's CD recorders typically weigh a few pounds and can be bought for less than $300.

8.14 DVDS

DVD is a digital optical disc storages format, invented and developed by Philips, Sony, Toshiba, and Panasonic in 1995. DVDs offer higher storage capacity than compact discs while having the same dimensions.

Formats

DVD-Video is the format designed for full-length movies that work with your television set.

DVD-ROM is the type of drive and disc for use on computers. The DVD drive will usually also play regular CD-ROM discs and DVD-Video disks. Fig. 8.5 below.

Fig. 8.5: A DVD

DVD-RAM is the writeable version.

DVD-Audio is a CD-replacement format.

8.15 RECORGING AND EDITING SOFTWARES

Audacity is a free, easy-to-use, multi-track audio editor and recorder for Windows, Mac OS X, GNU/Linux and other operating systems. The interface is translated into many languages. You can use Audacity to:

Features of Audacity

- Record live audio.
- Record computer playback on any Windows Vista or later machine.
- Convert tapes and records into digital recordings or CDs.
- Edit WAV, AIFF, FLAC, MP2, MP3 or Ogg Vorbis sound files.
- Cut, copy, splice or mix sounds together.
- Change the speed or pitch of a recording.
- Import and Export Import sound files, edit them, and combine them with other files or new recordings. Export your recordings in many different file formats, including multiple files at once. Import and export WAV, AIFF, AU, FLAC and Ogg Vorbis file
- Sound Quality Supports 16-bit, 24-bit and 32-bit (floating point) samples (the latter preserves samples in excess of full scale). Sample rates and formats are converted using high-quality resampling and dithering..Tracks with different sample rates or formats are converted automatically in real time

8.16 WHAT HAVE YOU LEARNT

- A preamplifier (preamp) is an electronic amplifier that prepares a small electrical signal for further amplification or processing.
- An Audio amplifier can be classified into many classes depending upon the class of operation like A , B , AB , C , D , E f and G
- Frequency response is the quantitative measure of the output spectrum of a system or device in response to a stimulus, and is used to characterize the dynamics of the system.
- Frequency response is usually measured within the range of hearing, from a low of 20Hz to a high of 20kHz
- Total Harmonic Distortion or THD is an amplifier specification that compares the output signal of the amplifier with the input signal and measures the level differences in harmonic frequencies between the two. The difference is called total harmonic distortion.
- Signal to noise ratio is a specification that measures the level of the audio signal compared to the level of noise present in the signal.
- Turntablism is the art of manipulating sounds and creating music using phonograph turntables and a DJ mixer.
- A gramophone record is a type of analog storage medium. It stores recorded music (or other sounds).
- An audio tape recorder is an audio storage device that records and plays back sounds, including articulated voices, usually using magnetic tape, either wound on a reel or in a cassette, for storage.
- Compact disc, or CD for short, is a digital optical disc data storage format.
- DVD is a digital optical disc storages format, invented and developed by Philips, Sony, Toshiba, and Panasonic in 1995.
- Audacity is a free, easy-to-use, multi-track audio editor and recorder for Windows, Mac OS X, GNU/Linux and other operating systems

8.17 TERMINAL QUESTIONS

1. Describe the history and operation of a turntable
2. Give classification of Audio amplifier on the basis of class of operation in brief.

3. Explain the term signal to noise ratio (SNR) .
4. Describe Audacity highlighting the various features of Audacity.

8.18 ANSWER TO IN TEXT QUESTIONS

8.1

1. (a) 2. (c) 3. (b) 4. (a) 5. (c)

9

AUDIO CONSOLE

9.1 INTRODUCTION

Previously, you studied about various sound equipments such as microphones, loudspeakers etc. As understanding these equipments is very important for the sound technician, therefore, here, you will learn about audio mixing console, their different parts or sections, working and their uses etc. You will also learn about different types of audio consoles, i.e., analog and digital, with their respective layout of controls, switches and operation etc.

9.2 OBJECTIVES

In this lesson, you will be able to

- explain the function and signal flow of an audio mixing console
- define and compare different types of mixing consoles
- identify different parts/sections of typical analog/digital console and explain its different controls with their usage

9.3 A SIMPLE ANALOG MIXER/ (STEREO) MIXING CONSOLE

A simple analog audio mixer combines several incoming signals into a single output signal. This cannot be achieved simply by connecting all the incoming signals in parallel and then feeding them into a single input, because they may interfere with each other. The signals need to be isolated from each other providing individual control of at least, the level of each signal.

Practically speaking, mixing consoles not only allow simple mixing but they can also provide phantom power for condenser/capacitor microphone, pan-control (where by each signals can be placed in any desired position in the stereo field), filtering, equalization, routing and monitoring facilities which enable routing any number of sound sources to a desired loud speaker for listening without affecting the mixer's main output.

Let us discuss a simple six channel analog audio mixer. This particular mixer will have six inputs and two outputs. Professionally we called it as a six-into-two (6:2) mixing console. The inputs will usually be XLR, TRS (tip-ring-sleeve) or balanced and TS (tip-sleeve) or unbalanced. According to the input source (microphone or line) the connectors vary (see Fig. 9.1).

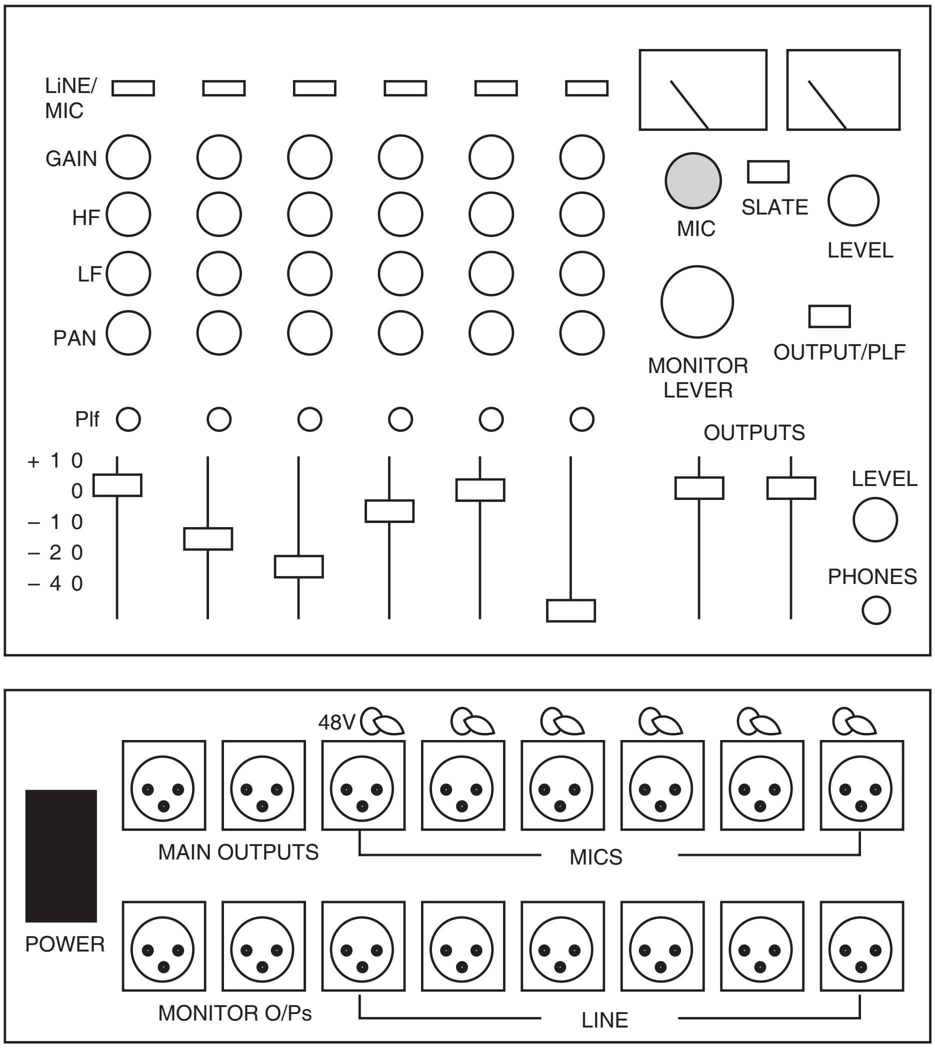


Fig. 9.1

The outputs are also in three pin XLR Type connectors. The outputs are XLR-Male connectors having three pins. The inputs will be having XLR- Female connectors. There is a phantom power of +48V (DC) Switch for Microphone selection, if and when required. If we are using condenser Microphone we need to supply the microphone with a +48V DC current from the Audio Mixer to operate the microphone. The signal received from this microphone will then be routed to a stereo or master bus.

Bus: A bus is a section in the signal path of an audio mixer which is used to combine different audio sources and deliver them as a whole to a specific destination.

Let us take a look at the signal flow in a simple analog audio mixer through its different sections.

9.3.1 Input Section

In the signal flow, the first section is the input gain control (rotary fader type) which is commonly known as Pre – Amp (pre-amplification). This control adjusts the degree of amplification provided by the input amplifier and is labeled in decibels (dB) either increasing or decreasing steps. Continuously variable inputs are normally switchable between Microphone and line position, depending upon the output level of the microphone or line connected to the channel input. We have to choose the 'Microphone' or 'line' input as per requirement.

For a 'microphone', high amplification is required as the microphone input is low and for 'line' inputs, little amplification is used and the gain control normally provides adjustment either side of unity gain (0 dB) perhaps (±20dB). This process of controlling the input levels is called as input gain control.

9.3.2 Equalization

Then the next section is equalization. This section will have controls for two frequency bands (in the case of the figure provided), the high and the low frequency. Boost and cut of around ±12 db over low and high frequency bands is available. In this section we can control the tone of the signal through boosting and cutting of the high and low frequencies.

9.3.3 Channel Fader

The last control of the input section is the channel fader which controls the overall level of the channel. It provides a small amount of gain (up to 12 db) and infinite attenuation (decrease). The fader control is specially designed for the purposes of level control

There are two types of Faders:

1. Rotary Faders. For example – Input gain control, equalization, etc.
2. Straight Movable Faders or Channel Faders. For example – stereo bus, etc.

9.3.4 Pan Control

Pan control on a Mixer is used for placing a signal anywhere between left or right in the stereo field. It works by splitting a single signal from the input into two signals.

These two outputs of the pan-pot, usually feeds the left and right channels of the stereo Mix bus. Also the signal can be placed in the centre which results in equal level in both L and R and hence no change in the perceived level.

Only 18 dB of level difference, is required between left and right channels to give the impression that a source is either fully left or fully right in a loud speaker stereo signal. But most pan-pots are designed to provide full attenuation of one channel when rotated fully towards the other, thereby changing the levels as shown in the Fig. 9.2.

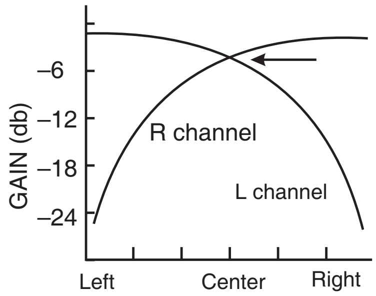


Fig. 9.2

9.3.5 PFL/Pre Fader Listening

This is a facility which provides a signal to be monitored without routing it to the main outputs of the mixer. It also provides a means for listening to a signal in isolation in order to adjust its level or EQ.

A PFL switch on each channel routes the signal before reaching the channel fader to the PFL bus. There is also a master PFL switch which switches the mixer's monitor output to monitor the PFL bus as shown in the Fig. 9.3.

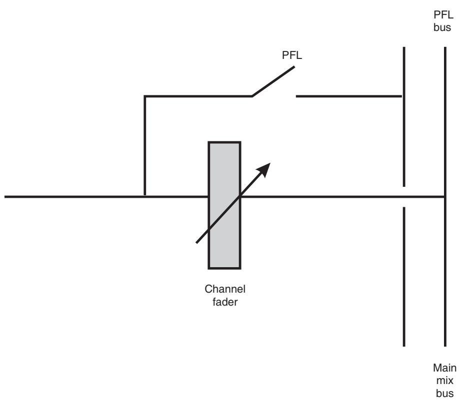


Fig. 9.3

Notes

In case of high quality audio mixing and live console a separate small PFL loud speaker is provided on the mixer itself so that the input signal can be checked without affecting the main monitor.

PFL has great advantages, in live work and broadcasting since it allows the engineer to listen to sources before they are sent to the master fader. It can also be used in studio recording to separate any source from all the others, without cutting all the other channels to adjust equalization and other processing with greater ease.

9.3.6 OutPut Section

The two main output faders control the overall level of all the channel signals which have been summed on the left and right mix buses. The outputs of these faders, feed the main output on the back panel of the mixer. The monitor gain control adjusts the loudspeakers output level, without affecting the main line output level but if we make any change in the main Fader gain, it will affect the monitor output.

There is an option of slate, which is used by the sound engineer/ sound operator and comprises of is a small microphone, mounted on the audio mixer, which is routed to the main outputs, so that comments from the engineer or operator (such as take number, announcement) can be recorded on a tape machine connected to the main outputs. There is a rotary level control to adjust the slate level.

Signal path from channel input to main output on a simple mixer is shown in Fig. 9.4.

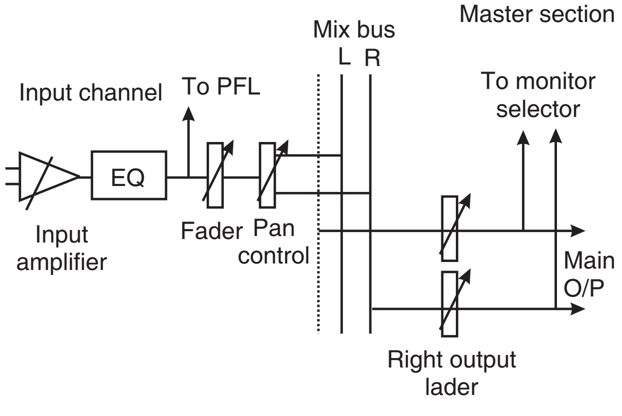


Fig. 9.4

INTEXT QUESTIONS 9.1

Choose any one of the following:

1. In a mixer, the first section of the signal flow is

   (a) Equalization (b) Input gain control

   (c) Routing (d) buffer

2. Equalization can control the .... of the signal through boost or cut of frequencies

   (a) Volume (b) Level

   (c) Tone (d) amplitude

3. .... is specially designed for the purpose of level control

   (a) Fader (b) Pan

   (c) Pad (d) Slate

4. In a mixer .... is used for placing of an audio signal to left, right or centre

   (a) Fader (b) Pan

   (c) PFL Switch (d) Potentiometer

5. The option used by sound engineer/operator to record any comments on tape is

   (a) Bus (b) Slate

   (c) Pan (d) Mixer

6. PFL or Pre Fade Listening has great advantages in which type of work

   (a) Live work and broadcasting

   (b) Film Production

   (c) Radio broadcasting

   (d) Mixing

7. Phantom power in microphone is .... volt DC

   (a) +45 (b) +48

   (c) +50 (d) –48

9.4 CONCEPT OF MULTI-TRACK MIXING (Multi -track Mixer)

Music recording generally requires two distinct stages.

1. Track laying
2. Mix-down

1. **Track laying**: The Musical tracks are recorded on a Multi-track Recorder separately. These include background tracks, rhythm tracks followed by lead tracks and vocals.
2. **Mix down Stage**: All the recorded tracks (Vocal and Music) are played back through the Mixer and combined into a Stereo or Surround Mix, to make the finished product, which is made into a Commercial Release.

In Multi-track Recording, two signal paths are processed. One from Mic/Line Input and another is from the Multi-track recorder. These two signals are then sent to the Stereo Mix down (Monitor path). Fig. 9.5.

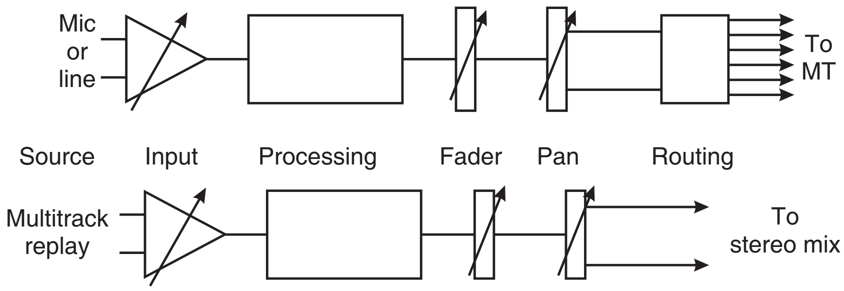


Fig. 9.5

We record the microphone signal into Multi-track, while also mixing the return signal from the multi-track recorder into one stereo signal, so that the Sound Engineer or Operator can hear what the final product will sound like. If any kind of overdubbing is required, then that has to be done at this stage. This stage is called "Track Laying".

Then comes the Mix-down stage which forms the basis of Stereo Mix down.

There will be two signal paths, one from the microphone or line source to the Multi-track Recorder (called as the channel path) and one from the Multi-track recorder back to the Stereo Mix (called as the Monitor Path).

Some basic signal processing such as Equalization, will be required in the channel path to the multi-track recorder. Whereas more signal processing features are usually applied at the time of the Mix down. In some cases like live Broadcasting, basic processing is used during the live work rather than the mix down stage.

9.4.1 Grouping

Grouping means simultaneous control of more than one signal at a time. It usually means that one fader controls the level of a number of slave channels.

It is usually used for reducing the number of Faders that the sound Engineer/ Operator has to handle. This is applied when there are more number of faders to be operated. So we need to make a group of faders and control by one fader.

There are two types of grouping:

1. Audio Grouping

2. Control Grouping

Audio Grouping: Audio Grouping means making a single audio output to take a number of channel inputs. A single fader controls the level of the summed signal & there will be a group output from the Console. Fig. 9.6.

The Stereo mix output from the Console is an effective audio group, one for the left and one for the right, as they constituted a sum of all the signals routed to the stereo output and includes the overall level control. Some older consoles will have routing buttons on top of each channel module.

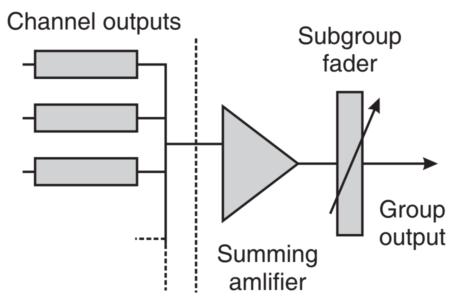


Fig. 9.6

The Master Fader, for audio groups, will be in the form of four or eight faders in the central section of the console (Fig. 9.7). They can be arranged in such a way that we can pan a channel between odd and even groups and it would be common for two of these groups. It is common for eight audio group faders to be used as subgroups themselves having routing to the stereo Mix so that the channel signal can be made more easily manageable by routing them to a subgroup & here to the main – mix via a single level control.

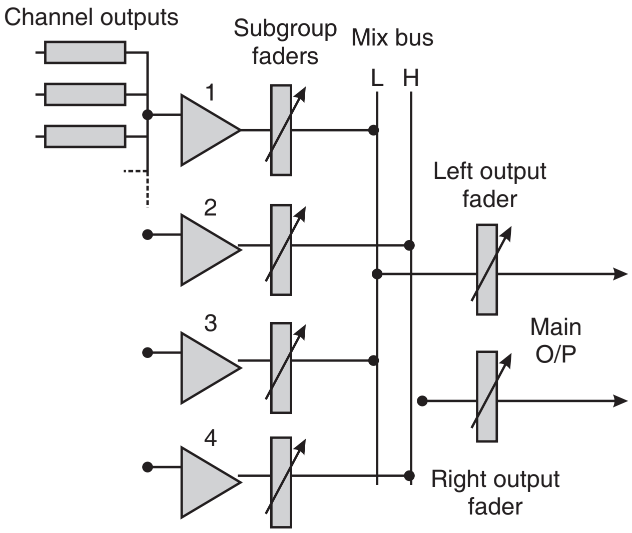


Fig. 9.7

From Fig. 9.7 without pan control the subgroup 1 and 3 goes to the left Mix bus and 2, 4 goes to the Right Mix Bus.

Control Grouping: Control group is different from audio grouping, because it doesn't give rise to a single summed audio output for the group. The levels of the Faders in the group are controlled from one fader. But these outputs remain separate. Generally its effect is to a large hand moving many faders at the same time. Each Fader maintaining its own level in relation to the others.

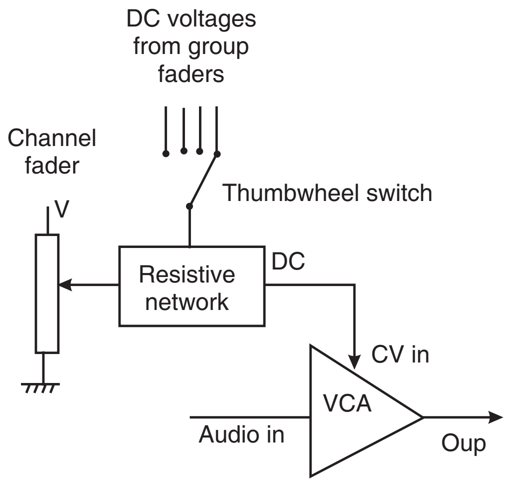


Fig. 9.8

Control group is controlled by voltage controlled amplifier (VCA). Its gain is controlled by a DC voltage applied to a control pin.

In VCA, Fader audio is not passed through the fader itself but is routed through VCA, so the Fader carries DC instead of Audio and the audio is controlled indirectly (Fig. 9.8).

The latest alternative to the VCA Fader is the DCA Fader (Digitally Controlled attenuator) whose gain is controlled by digital values (binary) instead of DC voltage. This is easier to implement in digitally Controlled Mixer.

Normally there are dedicated VCA group master Faders in non automated system, it will control the overall levels of any channel faders assigned to them. The channel audio outputs would normally be routed to the main mix directly and the grouping affecting the levels of the individual channels in their mix.

INTEXT QUESTIONS 9.2

Choose any one of the following:

1. How many signal paths are processing in multi-track recording?

   (a) One (b) Two

   (c) Three (d) Four

2. Signal processing features are usually applied more in which stage?

   (a) Track lying stage (b) Mix down stage

   (c) Multi-track recording (d) Background track recording

3. Control grouping is controlled by

   (a) DC voltage

   (b) AC voltage

   (c) VCA (Voltage Controlled Amplifier)

   (d) None of above

9.5 DIGITAL MIXER

A Digital Mixer would normally comprise of number of similar sections/modules. The main section would comprise of number of Channel strips for control, processing and monitoring of each channel .

The digital audio mixer will have eight sections in its channel strip

1. Input section
2. Routing Section
3. Dynamic section
4. Equalizer section
5. Channel & Mix Controls
6. Auxiliary section
7. Master Controls
8. Metering Section

Channel Strip: Typical layout of a channel strip is shown in Fig. 9.9.

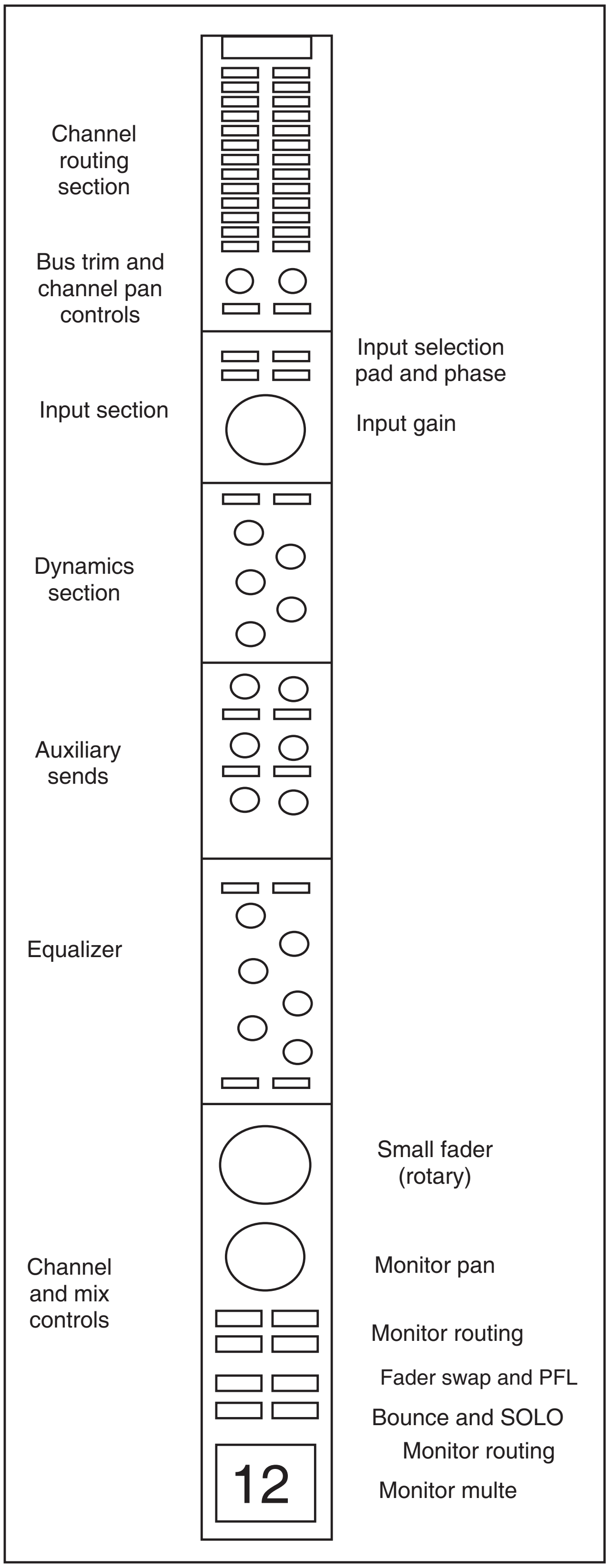

Channel routing section
Bus trim and channel pan controls
Input section
Input selection pad and phase
Input gain
Dynamics section
Auxiliary sends
Equalizer
Small fader (rotary)
Channel and mix controls
Monitor pan
Monitor routing
Fader swap and PFL
Bounce and SOLO
Monitor routing
12
Monitor multe

Fig. 9.9

9.5.1 Input Section

This section includes the following components:

1. **Input Gain Control :** It sets the Mic or line input amplifier gain to match the level of the incoming signal, this control is often a coarse control in 10 db steps, sometimes accompanied by Fine trim, steps of 5dB or 10dB make for easy reset of control to an exact gain setting, and precise gain matching of channels.

2. **Phantom power**: Many professional Mics require 48V Phantom powering and, sometimes switch on the module to turn it on/off . Occasionally this switch is re-attached on the console, may be in a central assignable switch panel.

3. **MIC/Line Switch**: It switches between the channels, MIC input and Line input. The line input could be the playback output from a tape machine. On another line, signal such as synthesizer, guitar (electronic) or sound effect devices.

4. **PAD:** usually used for attenuating(decreasing) the high level mic input signal by something like 20db, for situations when the mic is in from a kick drum or when the mic is in the field of high sound pressure. For example its output may be so high as to cause the mic input to chips, requiring that the pad be used on some occasions.

5. **HPF/LPF** : High Pass Filter/ Low Pas Filter can sometimes be switched in at the input stage to pass only high frequency or low frequency component of the signal. These can be used to filter out unwanted rumble. Filtering rumble at this stage can be an advantage because it saves clipping later in the chain.

9.5.2 Routing Section

This section includes the following components/ Controls:

1. **Track Routing Switches**: The number of Routing Switches depends on the Console; some will have 24, some 32 & some 48. The switches route the channel path signal to the multitrack machine, and it is possible to route a signal to more than one track, the track assignment is often arranged as panes of track, so that odd and even track can be assigned together with a pan-pot used to pan between them as a stereo pair.

2. **Mix Routing Switches**: Sometimes there is a facility for routing the channel path output signals to the main monitor Mix, or to one of perhaps four output groups and these switches will often be located along with the track routing.

3. **Channel pan switch**: used for panning channel signals between odd and even tracks of the multitrack in conjunction with the routing switches.

4. **Odd/Even/Both switch:** This switch will determine whether the signal is sent to the odd channel only, the even channel only or both (In which case the pan control is operative)

5. **Direct switch:** used for routing the channel output directly to the corresponding track on the multi-track machine without going via the summing buses. This can reduce the noise level from the console since the summing procedure used for combining a number of channel output to a track bus can add noise of a channel, is routing directly to a track no other signal can be routed to that track.

9.5.3 Dynamic Section

Some advanced consoles incorporate dynamics control on every module. So that each signal can be treated without resorting to external devices. These normally incorporate compressor and expander sections, which can act as limiters and gates respectively. If required, system allows that EQ to be placed in the side chain of the dynamic section also, providing frequency sensitive limiting among other things. It is usually possible to link the action of one channel's dynamics to the next in order to "gang" stereo channel so that the image doesn't shift when one channel has a sudden change in level while the other doesn't. When dynamics are used on stereo signal it is important that left and right channels have the same setting otherwise the image may be affected.

9.5.4 Equalization Section

The EQ section is usually split into three or four sub sections, each operating on a different frequency band and to have similar functions. These will be described in general here.

(a) HF, MID1, MID2, LF

A high frequency band, high Mid, Low Mid and Low frequency band equilizations are often provided. If the mode is parametric, these bands will allow continuous variation of respective frequency 'Q' and boost/cut. If not parametric, then there may be few switched frequencies for the Mid frequency band and perhaps a fixed frequency for LF and HF bands.

(b) Peaking/Shelving or Bell

Often provided on the upper and lower bands for determining whether the filter will provide boost/cuts over a fixed band (whose band width will be determined by 'Q') or whether it will act as a shelf with the response rising on nothing off above or below a certain frequency.

Notes

(c) 'Q'

The 'Q' of a filter is defined as its center frequency divided by its bandwidth (the range between frequencies where the output of the filter is 3db lower than the peak output) in practice this affects "the sharpness" of the filter peak or notch, high 'Q' giving the sharpest response and Low Q giving a very broad response.Fig. 9.10.

Low 'Q' would be used when Boost/Cut over a relatively wide range of frequencies is required.

While high 'Q' is used to Boost/cut one short specific region

(d) Frequency Control

Sets the center frequency of peaking filter or the turn over frequency of a shelf

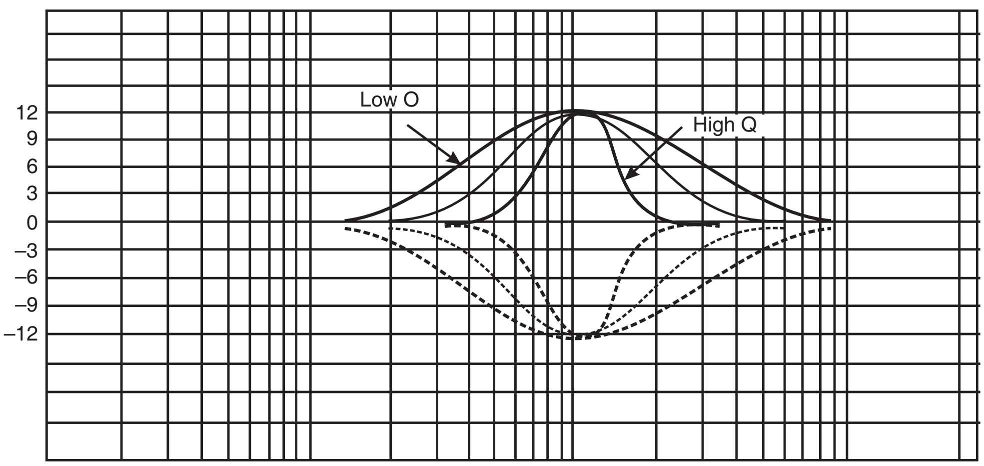


Fig. 9.10: (Equalization Section)

(e) Boost/Cut

Determines the amount of boost or cut applied to the selected band usually up to maximum around ±15db

(f) HPF/LPF

Sometimes the high and low pass filters are located here instead of at the input. In addition they normally have a fixed frequency turnover point and a fixed roll-off of either 12 or 18db per octave. These will often operate even if the EQ is switched off

(g) In/Out

Equalization circuits can introduce noise and phase distortion, so they are best switched off when not required, by input cut off switch.

Notes

9.5.5 Channel and Mix Control Section

This section generally has following controls/ components

(a) Pan

This control is a continuous rotary & knob and is used to place the signal of that channel in any desired position in the stereo picture.

(b) Fader Reverse

Swaps the faders between mix and channel paths, so that the large fader can be made to control either the mix level or the channel level.

Some systems defeat any fader automation when the large fader is put in the channel path. Fader reverse can often be switched globally and may occur when the console made is changed from recording to mix down.

(c) Line/Tape or Bus/Tape

In Line or Bus mode the monitor paths are effectively 'Listening to' the line output of the console's track assignment buses while in 'Tape' mode the monitor paths are listening to the off tape signal.

(d) Bus or Monitor Bus

It routes the output of the monitor fader to the input of the channel path (channel fader) so that the channel fader will be used as a post-fader effects send to any one of the multi-track buses.

(e) Mute or cut

There are two types of cut switches one for cutting the channel signal from the multi-track send the other for cutting the Mix signal from the mix.

(f) PFL

Pre fade listen is the signal monitoring without pass through fader, the signal coming from the source without routing to fader.

(g) AFL/Solo

After fade listen is similar to PFL this is sometimes called as solo, which routes a panned signal of the track to the main monitor, cutting all other signals, these functions are useful for isolating signals at the time of setting. In most of consoles, the AFL Bus will be stereo. Solo functions are useful when applying EQ and effects, one may hear the isolated sound and treat individually without hearing the rest of Mix.

Notes

INTEXT QUESTIONS 9.3

Choose any one of the following:

1. Signals coming from tape recorder or CD player will be fed to which input of the mixer?

   (a) Microphone input (b) Line input

   (c) Aux input (d) Metring input

2. .... is used for removing of rumble or hiss from signal.

   (a) Filter (b) PAD

   (c) Phantom power (d) Live swich

3. Compressor and Expander coming under which section of the audio mixer?

   (a) Input (b) Output

   (c) Master control (d) Dynamic

4. Generally equalization explained in to .... or .... section

   (a) One or Two (b) Two or Three

   (c) Three or Four (d) One or four

5. The maximum boost or cut applied to a selected band is

   (a) Plus minus 10 (b) Plus minus 15

   (c) Plus minus 20 (d) Plus minus 25

6. AFL or after fade Listening is also called?

   (a) Pan (b) Pot

   (c) Solo (d) Aux

9.5.6 Auxiliary Sends

Auxiliary sends referred as Aux Sends, are taking the signals from either the channel or Mix paths and appear as outputs from the console which can be used for fold back to the vocal booth musicians, effects ends, cues and so on.

Each aux will be a master gain control, usually in the centre of the console for adjusting the overall gain of the signal, sent from the console.

Aux sends are often a combination of mono and stereo buses Mono sends are usually used as routes to effects, while stereo sends may have one level control and a pan control per channel. The no of Auxiliary sends depends on the console, but there can be up to ten on an ordinary console.

Notes

(a) ***Aux Sends****:* It controls for the level of each individual channel in the numbered aux mix.

(b) ***Pre/Post switch****:* It determines that whether the signal send in taken off before or after the fader. If it is before, then the send will still be live, even when the fader is down, effects sends will normally be taken post fade.

(c) ***Mix/Channel****:* Determines whether the send is taken from the Mix or channel paths: it will often be sensible to take the send from the channel path, when effects are to be recorded on to multi-track rather than on to the Mix. This function is labeled 'WET'

(d) ***MUTE****:* It will cut the numbered send from the aux mix.

9.5.7 Master Control Section

This section normally placed on the right hand end of the Audio Console or now, in the digital Mixers, it is on the middle of the Console. It has following facilities:

(a) ***Monitor Selection***: Monitor selection means it will feed the signal to the loud speaker of the control Room/Studio but not the mix output, there are many switches to select the source to be monitored like. Aux sends, the main stereo mix on Tape Machines etc.,

(b) ***DIM****:* It decreases the level of the signal feed to monitor around 40db, for quick silencing of the room.

(c) ***Record/Overdub/Mix Down****:* This Facility, depends the mode of operation, the Mic/Line input switching, large and small faders and auxiliary sends. This will over write the signal on one another or it will dub the signal on an appropriate position (between memories we have given)

(d) ***Auxiliary Level Control****:* This is the master control for setting the overall level of each Aux send output.

(e) ***Talkback or Fold back****:* Talkback is usually placed on the console, having a small built in Microphone, which is usually used for giving instruction to the studio hands (instrumentalist) from the control room and which can be routed to a number of destinations like Aux sends, Mix bus or studio loud speakers etc.,

(f) ***Oscillation****:* For analog tape recording, we need the magnetic tape to be given bias, which is a lining/ signal of frequency of accurate 1khz and 10khz tone. The 10khz tone is the accurate setting the bias of an analog tape machine, in which the tone (1khz, 10khz) is fed to the tape from the Mixers "oscillator" option or sometimes it is given from the Tape Recorder facilities option.

Notes

(g) ***Slate:*** The slate would be used for recording or take information on to tape:

(h) ***Master Faders:*** There may be either one stereo fader or left and right faders, to control the overall Mix output level. Often the group master fader will reside in this section.

INTEXT QUESTIONS 9.4

Choose anyone of the following

1. In an ordinary mixing console the number of aux send present is ....
   (a) One (b) Two
   (c) Ten

2. Aux send in the mixer are normally used for ....
   (a) Recording (b) Playback
   (c) Fold back or Talk back

3. Monitor selection is fed the signal to ....
   (a) Channel path (b) Aux send
   (c) Loud speaker

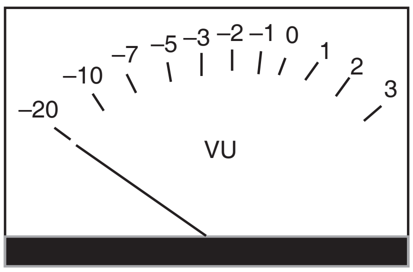
4. .... is used for quick silencing of the room
   (a) Attenuation (b) Pad
   (c) DIM

5. Sound engineer gives instruction to studio through
   (a) Oscillation (b) Talk back/Fold back
   c) Slate

6. The process using for biasing the magnetic tape is
   a) Oscillation b) DIM
   c) Attenuation

9.5.8 Metering System/ Section

Metering system is placed on audio consoles to measure the level of the audio signal feeding in & coming out from the Mixer. This is important for measuring the audio level without noise & distortion and to record the correct sound.



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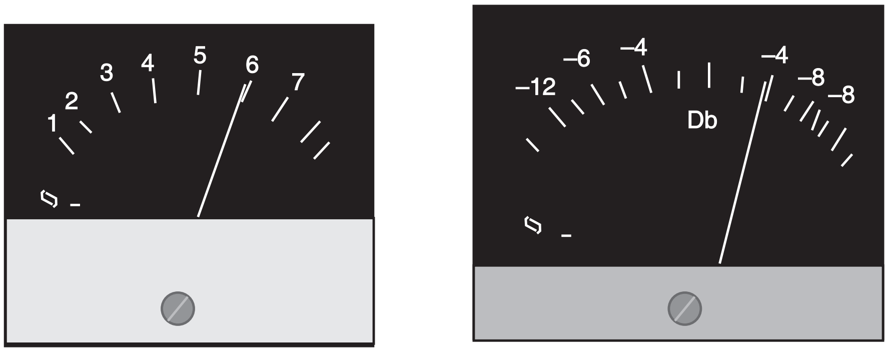

Fig. 9.11

Usually two types of metering are provided in the audio console.

1. Mechanical Meter
2. Electronic Bar – graph meter.

Mechanical Meter: Generally two types of Mechanical meter have been used. Fig. 9.11.

(a) VU (volume unit) Meter
(b) PPM (Peak Program) Meter

In the VU meter, there is associated variable attenuator, which could vary the electrical alignment level for O VU up to +24dBu. Now it is common for this to be fixed these days at OVU = ±4 dBu

Disadvantage in Mechanical Meters

PPMs respond well to signal peaks, that is they have a fast rise time, where as VUs are quite opposite, they have a very slow rise time. This means that VUs donot give a true representation of the peak level going on tape. Many people are used to working with VUs, they are good to measuring continuous signals such as tones, but their value, for monitoring program material, is dubious in the age of digital recording.

VUs have no control over fall time of the needle, which is much the same as the rise time. Whereas PPMs have fast rise time & longer fall time, which is more useful.

Normally PPMs are designed to graduate peaks that would cause audible distortion but doesn't measure the absolute peak level of the signal.

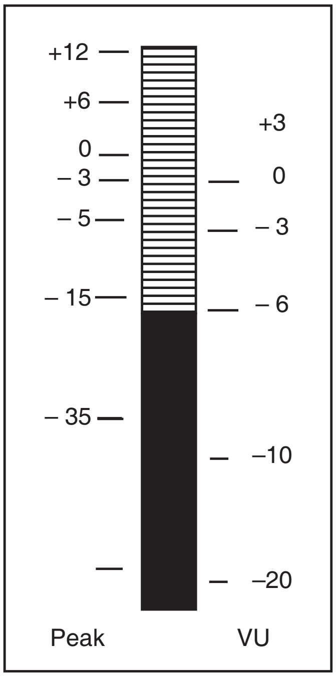

Fig. 9.12

It is important to note that meters can take a lot of space on the console & it may be impossible to find space for one meter per channel, in case of multi-track consoles.

Electronic Bar graph Meter

In Electronic bar graph meter (Fig. 9.12) there can be an infinitely fast rise time, although this may not be ideal in practice but cheaper bar graphs are made out of new of LEDS (Light Emitting Diode) & the resolution accuracy depends on the number of LEDs used.

There is a facility provided to switch the peak response of these meters from peak to VU mode. where they will imitate the scale and ballistic response of a VU meter.

The main advantage of the Bar graph vertical VU meter is that it takes a little space on the audio mixer.

9.6 AUTOMATION

In Mixers, the automation means storing fader positions dynamically against time of reiteration at a later point in timesynchronous with recorded material.

The purpose of automation has been to assist the engineer/operator, in mix down, the no of fader's that need to be handled at once, become too difficult for one person.The fader automation has resulted in engineers being able to concentrate on sub areas of a mix at each pass, gradually building up the finished product and refining it.

Fader Automation: There are two common means of memorizing and controlling the gain of a channel

1. Which stores the positions of the fader and uses this data to control the gain of a VCA or DCA (digital controlled attenuator)
2. Which also stores the fader movements but uses this information actually to drive the fader's position using a motor?

Common Operational Mode's are:

Write: It memorizes the level of the channel corresponding to the fader position

Read: Channel level controlled by data received from previously stored Mix.

Update: Channel level controlled by a combination of previously stored Mix data and current fader position

Group: channel level controlled by combination of channel fader position and that of a group master

INTEXT QUESTIONS 9.5

1. How many types of level meters are there in the audio mixing console?
2. Which type of meter is taking a lot of space on the audio mixing console?
   (a) Mechanical Meter
   (b) Electronic Bar – graph meter.
3. Which type of meter uses LED
   (a) Electronic bar-graph meter
   (b) VU meter
   (c) PPM meter

9.7 WHAT HAVE YOU LEARNT

In this lesson you have learned following aspects of Audio Mixing Consoles

- Different sections of an analog audio mixing console with their functions.
- Different stages of multi-track mixing.
- Grouping of faders during multi-track mixing, their types and how they are operate.
- Digital mixing console and its different sections with their functions.
- Different types of meter, their advantage and disadvantages.

9.8 TERMINAL EXERCISE

1. Briefly explain the sections involved in simple analog mixer.
2. Briefly discuss the two distinct stages during music recording.
3. What is grouping? Briefly describe types of grouping?
4. How many sections are there in a digital mixer?
5. Briefly explain different sections of an input section?
6. Briefly explain different sections of routing section?
7. What is dynamic control in the digital mixer?
8. Briefly explain different sections of equalization section?
9. Briefly explain different sections of channel and mix control section?
10. Discuss the four points in auxiliary send?

11. Briefly explain different sections of master control section?
12. What is the function of metering system in the mixer? How many type of meter generally present in the mixer?
13. What are the disadvantages in mechanical meter?
14. What is automation? Briefly describe the four operational modes are there in the automation process

9.9 ANSWER TO INTEXT QUESTIONS

9.1

1. (c)	2. (c)	3. (a)	4. (b)
5. (b)	6. (a)	7. (b)	

9.2

1. (b)	2. (b)	3. (c)

9.3

1. (b)	2. (a)	3. (d)	4. (c)
5. (b)	6. (c)		

9.4

1. (d)	2. (c)	3. (c)	4. (c)
5. (b)	6. (a)		

9.5

1. (Two)	2. (a)	3. (a)

9.10 GLOSSARY

1. **Phantom power:** It is a +48V DC power supply which is given to the microphone to operate it. (i.e., for condenser or capacitor microphones)
2. **Routing:** Assigning into desired position or track
3. **Monitoring:** Listening the sound signal or the sound track
4. **XLR Connector:** Type of 3-pin connector for connecting audio signals
5. **Module:** a part of a section
6. **Frequency band:** a set/ range of frequencies
7. **Pre/Post:** before/after.

Notes

10

AUDIO CHAIN SETUP

10.1 INTRODUCTION

In the previous lessons, you have studied about different audio equipment like consoles, mixers, amplifiers and recording/playback equipment etc. In this lesson, you will learn how to install and disassemble the audio chain/ setup equipment. Public address system or sound recording equipment setup is used in two ways, either permanently fixed or movable to different locations as per requirement. In both the cases, the installation of the sound equipments is very important. Right from taking the equipments out of boxes, assembling, positioning to the fixing and testing, a very precise knowledge and practice is required. In very large events, the size of speakers and other equipments is very huge hence extra practical knowledge is required to handle such large equipment. Dismantling the sound equipments, storing them along with the accessories, connectors etc. and transporting them properly is a big challenge.

10.2 OBJECTIVES

After reading this lesson, the learner will be able to :

- install and assemble a range of sound/ audio equipments
- estimate power requirement for different sound/ audio equipments
- perform appropriate placement of power fitting
- implement techniques for performing sound checks
- disassemble the sound/ audio equipment and store these properly

10.3 AUDIO CHAIN

Audio chain refers to a studio or outside setup of audio equipment arranged for the purpose of public address/recording/broadcast of the required programme .In this section we would learn about aspects to be taken care of when different audio equipment are integrated to form a audio chain. As mentioned earlier,the programmefrom source to listener involves the use of a devices to pick up sound that may be microphone, recording/playback and signal processing equipment. Figure 10.1 is a simplified block schematic diagram of audio chain of a broadcast studio.

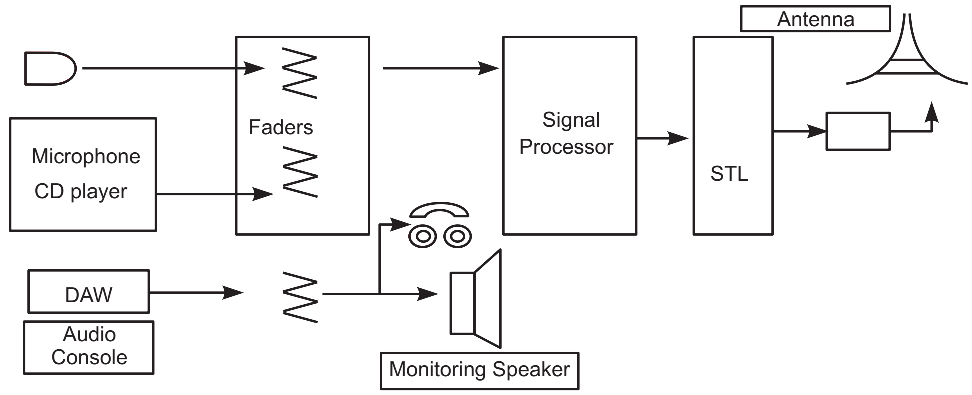

Fig. 10.1: Audio chain of broadcast studio

Audio chain of a broadcast studio is basically a linkage of various equipments, involving various sound sources, such as microphone to pick up the announcer's voice, CD player or a digital audio work station (DAW), mixer or audio control, signal processing equipment, monitoring speakers and a studio transmitter link.

Similar setup would be there for PA system and recording chain, except STL and Transmitter, which would be replaced by PA amplifiers and recording equipment.

10.4 SETTING UP YOUR STUDIO

Once you have your equipment, you need to connect it together with cables and possibly install equipment racks and acoustic treatment to complete the setup . Let's consider the features of important components like Cables, Connectors and other equipment to install the audio chain setups.

10.4.1 Cables

Cables carry electric signals from one audio component to another. They are usually made of one or two insulated conductors (wires) surrounded by a fine-

Notes

wire mesh shield that reduces hum. Outside the shield is a plastic or rubber insulating jacket. On both ends of the cable are connectors.

Cables are either balanced or unbalanced. A balanced line is a cable that uses two wires (conductors) to carry the signal, surrounded by a shield. Each wire has equal impedance to ground. An unbalanced line has a single conductor surrounded by a shield (see Fig. 10.3). The conductor and shield carry the signal. A balanced line rejects hum better than an unbalanced line, but an unbalanced line less than 10 feet long usually provides adequate hum rejection and costs less.

A cable carries one of these five signal levels or voltages:

- Mic level: about 2 mV (0.002 volt) to about 1 V depending on how loud the sound source is, and how sensitive the mic is
- Instrument level: typically 0.1 V to 1 V for passive pickups; up to 1.75 V for active pickups
- Semipro or consumer line level: –10 dBV (0.316 volt)
- Pro line level: +4 dBu (1.23 volts)
- Speaker level: about 20 volts.

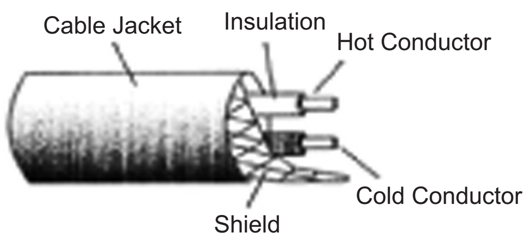

Fig. 10.2: A2 conductor shielded, balanced line.

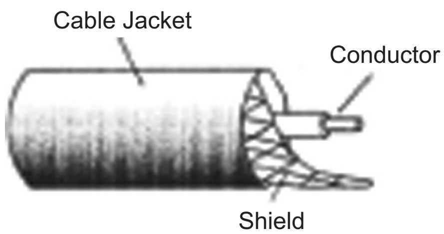

Fig. 10.3: A1 conductor shielded, unbalanced line.

10.4.2 Equipment Connectors

Recording equipment also has balanced or unbalanced connectors built into the chassis. Be sure your cable connectors match your equipment connectors.

Balanced equipment connectors:

- 3-pin (XLR-type) connector
- 1/4-inch TRS (tip-ring-sleeve) phone jack-

Unbalanced equipment connectors:

- 1/4-inch TS (tip-sleeve) phone jack-
- Phono jack (RCA connector)-Figure 10.4

A jack is a receptacle; a plug inserts into a jack.

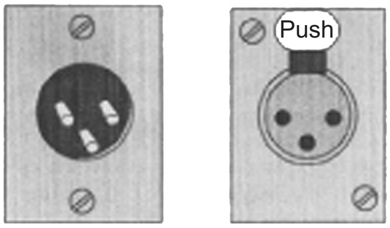

Fig. 10.4: A3-pin XLR-type connector used in balanced equipment.

Left: male output connector. Right: female input connector.

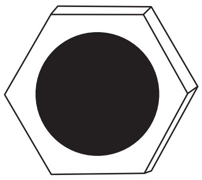

Fig. 10.5: A 1/4-inch phone jack used in balanced and unbalanced equipment.

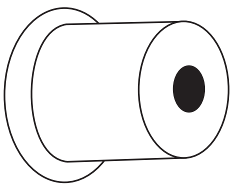

Fig. 10.6: A phone (3CA) jack used in unbalanced equipment.

Connectors are confusing because a single connector can have several functions (usually not the same time).: Here are some examples:

- **XLR:** Balanced line input at +4dBu, balanced mic input at 2 mV to 1 V, or balanced line output at +4dBu

- **TS (mono 1/4-inch phone jack)**: Unbalanced mic input, unbalanced line-level input or output (+4dBu or –10 dBV), instrument input, or low-cost speaker connector
- **Combi connector:** An XLR mic input plus a TRS input (instrument level or line level)
- **RCA (phono):** Home stereo line-level input or output at -10dBV, composite video input/output, or SPDIF digital-audio input/output

Equipment connectors are labeled according to their function. If you see an XLR connector with the label "MIC," you know it's a balanced mic input IF it's a 1/8-inch connector on a sound card, look at the icon near the connector. It's either a mic input, line input, line output, or speaker output.

INTEXT QUESTIONS 10.1

Fill in the blanks :

1. DAW stands for digital .... work station.
2. Voltage level of a microphone is about ....
3. A three pin (XLR) connector is a type of .... equipment connector.
4. A combi connector consist of an XLR mic input plus a .... input.

10.4.3 Cable Types

Cables are also classified according to their function. In a studio, you'll use several types of cables: power, mic, MIDI, speaker, USB, FireWire, S/PDIF, TASCAM TDIF, Alesis Lightpipe, guitar cords, and patch cords.

A power cable, such as an AC extension cord or a power cord on a device, is made of three heavy-gauge wires surrounded by an insulating jacket. The wires are thick to handle high current without overheating.

A mic cable is usually 2-conductor, shielded. It has two wires to carry the signal, sur-rounded by a fine-wire cylinder or shield that reduces hum pickup. On one end of the cable is a connector that plugs into the microphone, usually a female XLR-type. On the other end is either a 1/4-inch phone plug or a male XLR-type connector that plugs into your mixer or audio interface.

Rather than running several mic cables to your mixer or interface, you might consider using a snake, which is a box with multiple mic connectors, all wired to a thick multi-conductor cable. A snake is especially convenient if you're running long cables to recording equipment from another room. It's essential for most on-location recording.

A MIDI cable uses a 5-pin DIN connector on each end of a 2-conductor shielded cable. The cable connects MIDI OUT to MIDI IN, or MIDI THRU to MIDI IN.

A speaker cable connects a power amp to each loudspeaker. To avoid wasting power, speaker cables should be as short as possible and should be heavy gauge (between 12 and 16 gauge). They can even be made from lamp cord (zip cord). Number 12 gauge is thicker than 14; 14 is thicker than 16.

A USB cable or a FireWire cable connects a peripheral device.

An S/PDIF cable transfers a digital signal from one device's S/PDIF output to another device's S/PDIF input. It uses a shielded unbalanced cable (ideally a 75-ohm RG59 Video cable) with an RCA plug on each end.

10.4.4 Cable Connectors

Several types of cable connectors are used in audio. Figure 10.7 shows a 1/4-inch mono phone plug (or TS phone plug), used with cables for unbalanced microphones, synthesizers, and electric instruments. The tip terminal is soldered to the cable's center conductor; the sleeve terminal is soldered to the cable shield.

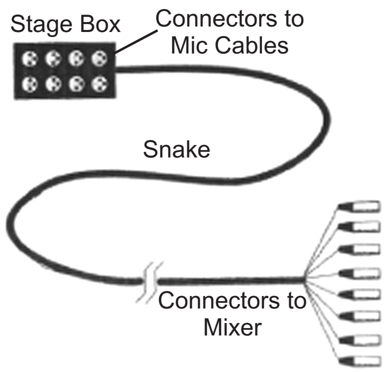

Fig. 10.7: A stage box and snake.

Figure shows an RCA or phono plug, used to connect unbalanced line-level signals. The center pin is soldered to the cable's center conductor; the cup terminal is soldered to the cable shield.

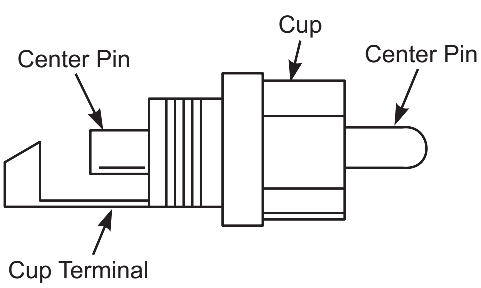

Fig. 10.8: An RCA (phono) plug

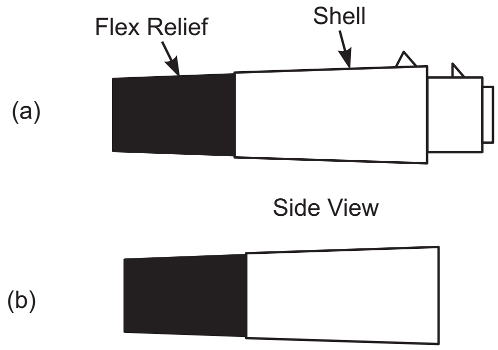

Fig. 10.9: A A3-pin pro audio connector (XLR-type) (A) female (b) male

Figure 10.9 shows a 3-pin pro audio connector (XLR-type). It is used with cables for balanced mics and balanced recording equipment. The female connector (with holes; Figure) plugs into equipment outputs. The male connector (with pins; Figure) plugs into equipment inputs. Pin 1 is soldered to the cable shield, pin 2 is soldered to the "hot" red or white lead, and pin 3 is soldered to the remaining lead. This wiring applies to both female and male connectors.

Figure shows a stereo (TRS) phone plug, used with stereo headphones and with some balanced line-level cables. For headphones, the tip terminal is soldered to the left-channel lead, the ring terminal is soldered to the right-channel lead, and the sleeve terminal is soldered to the common lead. For balanced line-level cables, the sleeve terminal is soldered to the shield, the tip terminal is soldered to the hot red or white lead, and the ring terminal is soldered to the remaining lead.

Some mixers have insert jacks that are stereo phone jacks; each jack accepts a stereo phone plug. The tip is the send signal to an audio device input, the ring is the return signal from the device output, and the sleeve is the ground.

10.4.5 Equipment Connections

The instruction manuals of your equipment tell you how to connect each component to the others. In general, use cables that are as short as possible to reduce hum, but that are long enough to let you make changes.

Be sure to label all your cables on both ends according to what they plug into-for example, MIXER CH1 MONITOR OUT or ALESIS 3630 IN. If you change connections temporarily, or the cable becomes unplugged, you'll know where to plug it back in. A piece of masking tape folded over the end of the cable makes a stay put label.

Let's say you have a hardware mixer in your recording setup. Here's a typical way to hook up the gear :

1. Plug the AC power cords of audio equipment and electric musical instruments into AC outlet strips fed from the same circuit breaker. Make sure that the sum of the equipment current ratings does not exceed the breaker's amp rating for that outlet. Plug the power amplifier or powered speakers into their own outlet on the same breaker so that they receive plenty of current. Consider using an AC power conditioner such as made by Furman (www.furmansound.com). It provides clean, steady AC power to sensitive electronic equipment. Surge protection and noise filtering are included.

2. Connect mic cables to mics. Use mic cables with a male XLR connector on one end and a female XLR connector on the other end.

3. Connect mic cables to the female XLR connectors in either the snake junction box, or directly into mic inputs on a mixer or mic preamps. Plug the snake connectors into the mic inputs. If your mixer has phone-jack mic inputs, you may need to use an impedance-matching adapter (female XLR to phone) between the mic cable and the mic input jack.

4. Set the output volume of synthesizers and sound modules about three-quarters up. Using a guitar cord, connect their audio outputs to instrument or line inputs on your mixer. If this causes hum, use a direct box. Using a MIDI cable, connect the MIDI OUT of a MIDI controller to the MIDI IN of your audio interface or MIDI interface.

5. If you are recording a guitar direct, connect its cord either to (1) an instrument input on your mixer or audio interface (1/4-inch phone jack), or (2) a direct box. Connect the XLR output of the direct box to a mic input on your mixer or audio interface.

6. If the mixer is a standalone unit (not part of a recorder-mixer), connect the mixer's stereo line outputs to the inputs of an audio interface. Use a stereo RCA-to-RCA cable or two phone-to-phone cables. If the mixer has a USB or FireWire connector, connect that to the mating connector in your computer-you don't need an audio interface.

7. Connect the audio interface line outputs to the mixer's 2-track or tape inputs, or directly to powered speakers. Use a stereo RCA-to-RCA cable or two phone-to-phone cables. Again, if the mixer has a USB or PireWire connector, connect that to the mating computer connector and omit the audio interface.

8. Connect the mixer's monitor outputs to the power-amp inputs. Connect the power-amp outputs to loudspeakers. Or if you are using powered (active) monitors, connect the mixer monitor outputs to the monitor-speaker inputs.

9. If the mixer does not have internal effects, connect the mixer aux-send

connectors to effects inputs (not shown). Connect the effects outputs to the mixer aux-return or bus-in connectors. Use phone-to-phone cables.

10. If you're using a separate mixer and multitrack recorder, connect mixer bus I to recorder track l IN; connect bus 2 to track 2, and so on. Also connect the recorder's track 1 OUT to the mixer's line input l; connect the track 2 OUT to the mixer's line input 2, and so on. Use cables with RCA or phone connectors. As an alternative, connect insert jacks to multitrack inputs and outputs. At each insert plug, connect the tip (send) terminal to a track input and connect the ring (return) terminal to the same track's output. Use a TRS-to-2-TS cable (stereo phone plug to two mono phone plugs).

11. If you have several headphones for musicians, connect the mixer's headphone jack to a small amplifier to drive their headphones. Use a cable with a stereo phone Plug on one end and two mono phone plugs on the other end (TRS-to-2-TS cable). Or it the: mixer's headphone signal is powerful enough, connect it to a box with several headphone jacks wired in parallel.

Figure 10.10 shows typical connections in a Digital Audio Workstation (DAW) recording studio with a multichannel audio interface.

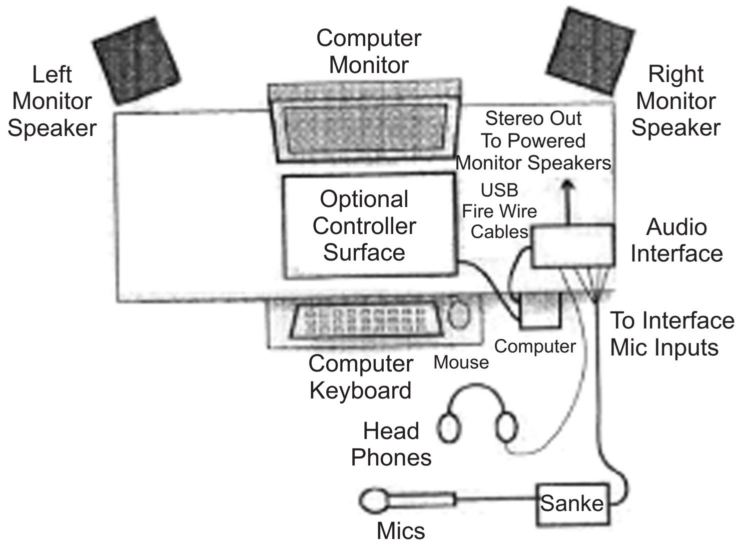

Figure 10.10: Typical layout of a DAW recording studio

As shown in Figure above, you might connect the equipment like this:

1. Using a guitar cord, connect electric instruments to instrument inputs on the audio interface. If an instrument is more than about 15 feet from the interface, connect its output to a direct box (using guitar cords), and connect the direct box XLR output to a snake or to an audio interface mic input.

2. Using an XLR mic cable, connect each mic to a mic input on the audio interface. If the mics are more than about 15 feet from the interface, connect each mic to a snake box, and connect the snake XLR connectors to the interface mic inputs. If you prefer to use a separate mic preamp and A/ D converter, plug the mic into the preamp, and connect the preamp's line output to the A/D converter's line input using an XLR or phone-to-phone cable.
3. Using a MIDI cable, connect the MIDI OUT of a MIDI controller to the MIDI In of the audio interface.
4. Using two phone-to-phone Cables (stereo or mono), connect the stereo output of the interface to two powered monitors. If your monitor speakers are passive connect the interface stereo outputs to the line inputs of a stereo power amp. Use speaker cable to connect the power-amp outputs to the speakers.
5. Plug headphones (or a headphone amplifier or junction box) into the head-phone jack of the interface.
6. Using a USB or FireWire cable, connect the USB/FireWire port in the interface to the mating port in the computer.
7. Using a USB cable, connect the USB port of an external hard drive to a mating port in the computer. That hard drive can be used for audio files or for backup.

Hooking up a recorder-mixer studio can be quite simple. Plug mics into mic inputs, plug headphones into headphone jacks, and plug powered speakers into the mixer monitor outputs.

10.5 ASSEMBLING AND INSTALLING SOUND/ AUDIO EQUIPMENT AT SITE

Following points may be observed before assembling sound equipment

1. You must know and visit the place in advance where the sound equipment needs to be assembled and ascertain the following
   (a) Contact the person who will authorize you to work at site
   (b) Know the exact place where the sound equipment i.e. Loudspeakers, Amplifiers, Signal processing equipment, Microphones will be required to be placed.
   (c) Know the sockets from where the power supply will be drawn. You draw power from 5A socket if the total power to be drawn is less than say 900W. If the total power to be drawn is 1KW or more it should drawn from 15 A sockets.

(d) Plan layout of power supply cables and power outlets to equipment. Similarly plan layout of Microphones cables (from Microphones to Amplifiers) & Loudspeaker cables (from Amplifiers to Loudspeakers).

(e) Ascertain the movement of people during the programme. This will help you to lay power supply cable, Microphone cables & Loudspeaker cables in a manner to avoid disturbance and disruption of the programme and damage to cables. In some cases carpets may be laid on cables to avoid disruption of cables due to heavy movement of people (dancing etc).

(f) To ascertain the requirement and arrangements for placing equipment (Microphone Stand booms, Amplifiers, Sound processing equipment etc.) on table of appropriate size. Table height of 30" is quite appropriate for operating the equipment.

10.6 TRANSFERRING THE SOUND EQUIPMENT TO THE SITE

Following points should be taken care of while moving the Equipment to site:

(a) Fragile, delicate and costly equipment should be transferred in their own boxes which have suitable casing lined with foam to avoid any transit damage.

(b) Other equipments should also be transported in suitable boxes made for the purpose and lined with foam cushion (e.g. Amplifiers, Loudspeakers, Tape Recorders, CD players etc.)

(c) End connectors of Microphone cables, Loudspeaker cables should be checked for appropriate connections to avoid embarrassment at last moment.

(d) Always carry appropriate tools (Screw drivers, soldering Iron, Solder, Flux etc) to carry out minor repairs at site if need be.

(e) Always reach the site in advance to set up, and test the equipment. Keep sufficient margin of time for some unforeseen problems, repairs at site.

(f) Make a list of Inventory issued from Stores. It will help you to return back all the items to Stores.

10.7 CONNECTING THE EQUIPMENTS FOR FUNCTIONING

Following points should be taken care of while connecting the equipment for actual functioning :

(a) Place the equipment appropriately on a table to ensure ease of operation i.e.

Microphones should be placed where the singer or speakers will sit or stand. Amplifier Speech Processing Equipment should be placed where Sound Assistant is to sit. Loudspeakers should be placed to enable equitable distribution of Sound energy.

(b) Microphones should be connected to MIC inputs. Equipment like Tape Recorders, CD Players, Gramophone Players(Record Players) etc. should be connected to LINE Inputs.

(c) Loudspeakers should be connected to appropriate level Output Impedance terminals of Power Amplifiers.

(d) Power supply switch of all the Equipment should be checked to have been turned to 220 Volts AC.(Some imported Equipment is designed for 110 Volts AC, its transformer setting should be changed to 220 Volts AC.) In case 110 V AC Equipment does not have a selector switch, a 220Volts-110Volts Transformer is required to be used between Power Supply of 220 Volts and 110 Volts Equipment.

(e) Operate the Equipment for rehearsal, judging the settings of various controls of Equipment for suitable listening.

(f) Obseve the safety precautions as mentioned in the lesson on safety instructions

10.7.1 Phantom Power

Phantom power, in the context of professional audio equipment, is a method for transmitting DC electric power through microphone cables, to operate microphones that contain active electronic circuitry. It is best known as a convenient power source for condenser microphones, though many active direct boxes also use it. The technique is also used in other applications where power supply and signal communication takes place over the same wires.

Phantom power supplies are often built into mixing desks, microphone preamplifiers and similar equipment. In addition to powering the circuitry of a microphone, traditional condenser microphones also use phantom power for polarizing the microphone's transducer element. Three variants of phantom power, called P12, P24 and P48, are defined in the international standard IEC 61938.

10.8 SOUND CHECKS OF EQUIPMENTS

Before the actual operation of sound equipments a sound check is required so that any problem either minor or major can be detected and rectified. The problem of Hum is often seen. In all the modern signal audio equipments, self testing feature is in built and its users friendly. In just one click all the features of the sound equipment are self tested. Even in some of the equipments trouble shooting is also in built.

10.8.1 Hum Prevention

You patch in a piece of audio equipment, and there is a sound - HUM! It's a low-pitched tone or buzz. This annoying sound is a tone at 60 Hz (50 Hz in Europe) and multiples of that frequency.

Hum is caused mainly by

- Cables picking up magnetic and electrostatic hum fields radiated by power wiring-especially if the cable shield connection is broken.
- Ground loops. A ground loop is a conductive loop or circuit made of a cable shield and a power-ground wire. A ground loop occurs when two or more separated pieces of audio equipment are each connected to power ground through a 3-prong power cord, and are also connected to each other through a cable shield. The ground voltage may be slightly different at each piece of equipment, so a 50- or 60-Hz hum signal flows between the components along the cable shield.

These are the most important points to remember about hum prevention:

- To prevent ground loops, plug all equipment into outlet strips powered by the same AC outlet or AC circuit.
- Do not use an AC (electrical) 3-to-2 adapter to disconnect the power ground-it causes a safety hazard.
- Some power amps create hum if they don't get enough AC current. So connect the power amp (or powered speakers) AC plug to its own wall outlet socket-the same outlet that feeds the outlet strips for the recording equipment.
- If possible, use balanced cables going into balanced equipment. Balanced inputs have XLR or hRS connectors and two conductors surrounded by a shield. At both ends of the cable, connect the shield to a screw in the chassis, not to XLR pin 1. Or use audio gear whose XLR connectors are wired with pin l to chassis ground, not to signal ground.
- Transformers isolate unbalanced connections. If that is not an option, use the cable assemblies..
- Don't use conventional SCR dimmers to change the studio lighting levels. Use Luxtrol® variable-transformer dimmers or multiway incandescent bulbs instead.

Even if your system is wired properly, a hum or buzz may appear when you make a connection. Follow these tips to stop the hum:

- If the hum is coming from a direct box, flip its ground-lift switch.
- Check cables and connectors for broken leads and shields.
- Unplug all equipment from each other. Start by listening just to the powered monitor speakers. Connect a component to the system one at a time, and see when the hum starts to identify the hum generating equipment and isolating.
- Remove audio cables from your devices and monitor each device by itself. It may be' defective.
- Lower the volume on your power amp (or powered speakers), and feed them a higher-level signal.
- Use a direct box instead of a guitar cord between instrument and mixer.
- To stop a ground loop when connecting two devices, connect between them a 1:1 isolation transformer, direct box, or hum eliminator (such as the Jensen CI-2RR, Behringer HD400, Rolls HE13).
- Make sure that the snake box is not touching metal.
- To prevent accidental ground loops, do not connect XLR pin 1 to the connector shell except for permanent connections to equipment inputs and outputs.
- Try another mic.
- If you hear a hum or buzz from an electric guitar, have the player move to a different location or aim in a different direction. You might also attach a wire between the player's body and the guitar strings near the tailpiece to ground the player's body.
- Turn down the high-frequency EQ on a buzzing bass guitar track.
- To reduce buzzing between notes on an electric-guitar track, apply a noise gate.
- Route mic cables and patch cords away from power cords; separate them vertically where they cross. Also keep recording equipment and cables away from computer monitors, power amplifiers, and power transformers.

By following all these tips, you should be able to connect audio equipment without introducing any hum.

10.9 DISMANTLING THE EQUIPMENT AFTER THE PERFORMANCE

You should observe following points for dismantling the Equipment:

(a) You should seek approval or consent of the Contact Person before starting the dismantling Process.

(b) Once permission of the Contact Person has been obtained, disconnect Power supply and then Grounding arrangement if an additional Grounding arrangement has been done.

(c) After disconnecting Power supply, remove delicate and costly equipment and pack them in their respective packing's.

(d) Remove the cables from site and wind them in suitable cable drums.

(e) Check the inventory with respect to List of Inventory got issued from Stores before starting return journey from the site.

(f) Hand over all the items which were taken on loan from stores for the purpose of the event.

10.10 WHAT HAVE YOU LEARNT

In this lesson you learnt about the components of audio chain, different types of cables and connectors, installing and assembling various equipments, phantom power etc., connecting the various equipments for functioning, performing sound checks and hum prevention and finally dismantling the equipment after the performance.

10.11 TERMINAL QUESTIONS

1. Define Audio Chain . List the various components used to connect studio equipments.
2. What are the different types of recording devices that can be put to use in a recording studio ?
3. Discuss the importance of performing sound checks.
4. What factors should be considered to prevent Hum Sound ?

10.12 ANSWERS TO INTEXT QUESTIONS

1. Audio
2. 2 MV
3. balanced
4. TRS

11

HEALTH AND SAFETY

11.1 INTRODUCTION

Safety considerations are integral part of any technical working area/ setup. Media and Entertainment industry or any studio technical production facility too has some safety issues. It is very important for a sound assistant or technician to know about related health and safety considerations and care required to be taken, to prevent the possibility of accidents. In this section, you will learn about the preparations which are made at the sound studio and precautions taken for various types of health and safety considerations.

The aim of this lesson is not only to let you know as to what measures should be taken to prevent bad happenings or accidents at production but also Do's and Don'ts to be followed in work environment. It is also to let you know about good habits/ precautions to be followed in a recording studio.

11.2 OBJECTIVES

After going through this lesson you will be able to:

- explain the necessity of safety in the studio area.
- describe the details of safety measures which are required to be taken for the safety at production area.
- enlist the safety measures to be taken care of while handling various sound recording equipments
- explain safety measures to be followed while handling electrical mains equipments
- describe how to handle fire fighting equipment and provide first aid.

11.3 RATIONALE

In previous lessons, you learnt about the working and components of various audio equipments. But, in order to get maximum yield from such equipments, it is necessary to follow certain preventive an corrective maintenance procedures to maintain healthy and safe work environment.

11.3.1 Observation of the Health and Safety Instructions

Safety instruction and guidelines should be observed by each and every person religiously in full manner, as non-observance of statutory orders of safety by any person causes accidents. Accidents do not differentiate between the level/ category/ gender of the person. In studio one should be aware of electrical hazards and risk of electronic malfunctions. As discussed previously, in any studio, you will see following equipments:

- Microphones
- Recorders
- Amplifiers
- Sound Mixers
- Speakers
- Routers
- Wireless Communication Devices
- Cables and batteries etc.

All these materials are packed with safety instructions and these instructions are available on the users' manual. It is advised that you study all the health and safety instructions during the installation or later as per your requirement. All manuals are also available on their websites. These instructions can be printed out for study and training purposes.

11.3.2 Role and Responsibility of Sound Technician

Sound technician is required to be alert at all the times, during recording hours and also after the recording hour to ensure that no lapses occur on the safety part at the recording area. It is the responsibility of the sound assistant/technician to instruct and make the crew aware about safety directives.

Sound Assistant has additional responsibility to:

- Impart awareness at the production area about the health and safety requirements.
- Procure all types of general and personal security/ safety devices for production crew.
- Non technical/untrained person should not be allowed to operate any production equipment like, mixers or recorders etc.
- Apart from these, other provisions to be taken care of with regard to safety have been discussed further.

11.4 SAFETY DURING SOUND RECORDINGS

- All electrical equipments, like recorders, mixers, should always be kept at proper place.
- At the time of necessity, the personal safety equipment must be used.
- Compulsory observance of safety warning symbols must be followed
- Prohibited materials like liquids and shoes must not be brought at the production area.
- Smoking is strictly prohibited at the production area.
- Don’t adopt any short cuts, it may be dangerous. Its always advisable to follow laid down procedure instead of shortcuts.

11.4.1 General Health and Safety Procedures

The General Health & Safety procedures include :

- Do not wear anything with fringe or hanging material, that might get caught on equipment as you are trying to move something and might cause an accident.
- Wear closed toe, non-conductive shoes.

- Anything with open toes or that can conduct electricity is not permitted in studio area. This prevents injuries while equipment is being moved and protects you from being electrocuted.
- Never ever move thing by yourself. Most of the recording equipment in the sound studio, is very large and heavy.
- Avoid back strain or other bodily harm by working with a partner.
- Never move anything, if you have previous injuries. If you have pre-existing back injuries, physical injuries or health issues that may be aggravated by moving an object, notify the Studio Production manager.
- Follow the safety instructions given in studios. This will decrease the risk of injury.
- Keep all food and drink in the designated areas. Food is banned on the recording area of the studio during production, practice, or preparation.
- Keep all exits free of obstruction and keep belongings in a designated area, out of the way of the main studio floor area.
- Turn off all cell phones or other electronic devices or set them to silent or vibrate mode.
- Never wear conductive shoes or clothing when operating equipment.

11.4.2 Sound Studio Safety

Individual technical person, production managers and production crew also have key health and safety responsibilities:

- Please be mindful of the plug you connect your audio cords into. If there is anything suspicious like a unusual smell or sparks, notify the Studio Manager immediately. This could mean that a plug is defective and might be a fire hazard.
- Make sure that the audio cords from the pillar to the talent are safely out of the way from people tripping on them.
- Take reasonable care of themselves and others who may be affected by what they do or fail to do/ to co-operate with the employer in carrying out activities intended to meet the employer's health and safety responsibilities, e.g. by following instructions, using personal protective equipment, reporting accidents and near-miss incidents.
- Work within the limits of their competence; to attend training as required by the employer.

Other aspects of sound studio safety include:

(a) Protecting Studio Hardware

You must have seen that sometimes there are lot of harmful line voltage fluctuations, above or below the normal power levels. These are very harmful to the studio equipment. Therefore in such circumstances one should use an adequately powered uninterruptible power supply (UPS).

A good quality UPS constantly charges the batteries so as to provide uninterrupted regulated power supply. This battery supply is again regulated and used to feed sensitive studio equipment such as computer etc. with a clean and constant voltage supply. This helps protect studio hardware equipment

(b) Protecting studio software

Software which initially come along with the equipment gets usually outdated after few years, therefore, one should periodically update it from time to time. Also, it should be ensured that it is malware free.

(c) Reading Manuals while installing and disassembling sound equipments

Before installation or disassembling, one should go through all the instructions written on the accompanied manual, in hard or soft copy, for better understanding of the finer points of the studio system.

(d) Protecting recorded data

To protect the recorded data, there should be rigorous and straight forward back up scheme. Generally data should be backed up at three places and one of these should be offsite.

11.5 PRECAUTIONS WHILE WORKING WITH ELECTRICAL MAINS OPERATED EQUIPMENT

After considering the practical aspects of the safety of the personnel, handling electric equipment at work, the following points are summarized:

1. Turn off the breaker to the circuit you are working on. Don't trust the labels in the breaker box. The preferred thing to do is first turn on some load (light, radio, etc) that is at the actual location that you are going to work on, and then observe that it goes off. It is difficult to work on old wiring as the previous repairs might have meshed up the wiring to the extent of creating chaos.

2. Confirm that the electricity is actually off by testing at the fixture you are working on with a voltage tester or series test lamp.

3. After following steps 1 and 2, use an insulated tool to short the live wire to the ground in the equipment you are working on.

4. Make Sure that the Electricity stays switched off all the time, while the wiring work is going on.

5. Notify everyone in the area that you are working on the electricity, and warn everyone not to fiddle up with it.

6. Unless you can clearly see the breaker panel from where you will be working, put a breaker lock on your breaker. If you don't have a breaker lock, then at least seal the breaker box with tape and place a bold note warning not to turn the electricity on. A breaker lock with the key in your pocket is best. Otherwise, a tape with a note may be used.

7. While at work keep in mind that power is available at the main switch. Make sure that nobody switches it on unknowingly. But, work very cautiously, assuming that you are working on live wire.

8. As a part of safety measures, wear shoes or boots that have thick insulated soles.

9. Avoid working on wet ground or floors. Never work on a panel or other live wire while standing in water, or while you are wet.

10. Use tools with insulated handles. You can observe senior electricians using screw-drivers that have insulated conducting part. Such measures will protect you from accidental physical contact with the metal part of the screw-driver in case it comes in contact with the live wire.

11. Make sure to use all the tools which are properly insulated as illustrated in Fig.11.1.

(a) Insulated Plier

(b) Nose Plier

(c) Neon Tester

(d) Wire Cutter

(e) Crimpimg Tool

(f) Hand Gloves

Fig. 11.1: Insulated hand tools used in electrical wiring.

12. Avoid overloading on an electrical socket by connecting several loads in one socket. Refer to Fig. 11.2.

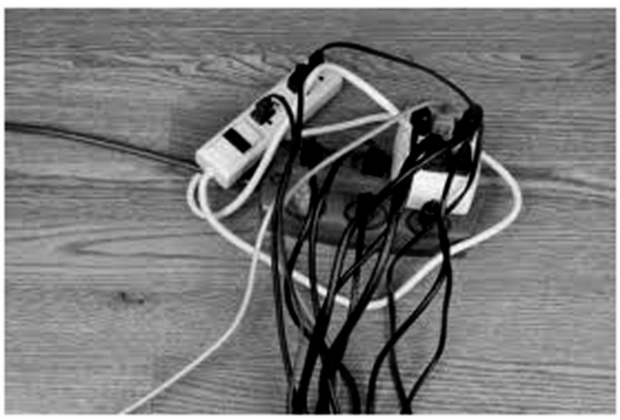

Fig. 11.2: Overloaded power stripper.

13. Make a habit of not touching the ground with one hand while you work with the other hand. That way if you a get shock it won't be as severe. Some guys put one hand in their pocket if they are working on something live.

14. The leading causes of electrical fires are loose connections. The leading causes of loose connections are distracted electricians. The electrician should develop good work habits, and eliminate distractions from work place usually due to presence of end users i.e., the customers.

15. Wear eye protection. Spark melted copper during electrical fire hazards adversely effects the human eye.

16. If you use a ladder, it should be made of wooden or another non-conductive material.

11.5.1 Additional Tips for Safety

1. Electrical safety is without question, the most important aspect of any electrical work. And just like anything we do in life, fear comes from 'not knowing'.

2. All it takes is one mistake. Some of the senior electricians usually boost and dare to work at live circuit with leniency. 230 volt AC is lethal. Shut the power off to any circuit that you are working on . Confirm the power is off with a simple pocket tester, a multi-meter, or lamp, blow dryer or another similar appliance.

3. Keep a torch near your electrical panel at all times. It will be useful during night hours just in case of a power failure.

4. Be extra careful and precautious on electrical systems in the rain, or in damp or wet locations, or where power is not completely shut off.

5. The best types of shoes are rubber-soled shoes and when possible stand on a rubber mat, or dry wooden floors or sub-floors. Never work barefoot or in socks or slippers and don't assume that it's safe to work without rubber-soled shoes on concrete floors. Concrete is conductive, particularly when it's damp.

6. Anything can conduct electricity if the conditions are right. Even an insulator also. By definition, a conductor allows the flow of electron and an insulator resists the flow of electrons. Similarly, dry concrete is bad conductor of electricity, while wet concrete conducts.

7. As soon as you turn off the power to an MCB, put a tape on the breaker. Provision to lock it off is even safer. Tag it out. This procedure is called as lock out/tag out.

8. If you are working with fuse panels instead of breaker panels, when you remove a fuse, use only one hand to remove it. Put your other hand either in your pocket or behind your back, and that's a good practice to develop

anyway. This helps to keep you from grabbing a circuit with two hands and providing a path for the electricity to flow through your heart. Now, electricity can still flow through one hand and one foot and pass through your heart, but if you've taken the other precautions as mentioned above, you will minimize your exposure to that hazard.

9. Last but not the least is an important tip about tool use. It is worth spending a little extra money to purchase quality tools. Hand tools like lineman pliers, screwdrivers, wire strippers, and other hand tools are used for electrical work. For instance, Good wire strippers will prevent you from nicking or skinning the wires. Good screwdrivers will prevent slipping out of screw heads or rounding them out. Good tools not only improve the quality of your workmanship, but improve your confidence as well.

INTEXT QUESTIONS 11.1

1. One of the following statements is not true about electrical wiring:

   (i) Notify the persons orally in the area that power is switched off for electrical maintenance.

   (ii) Switch off the MCB, put sticker and fix a tag with the caption 'Electrical Maintenance in Progress, Please, DO NOT OPEN'.

   (iii) Keep in mind that other side of the main switch is LIVE.

   (iv) Take written permission from the shift incharge before starting maintenance work.

2. Write in your own words the steps of precautions, in sequence you should follow, in order to avoid electrical accidents?

3. State whether the following statements are true or false.

   (i) One MCB may be connected to more than one sub-circuit at a time.

   (ii) Once MCB is switched off, it is not important to test the load end for accidental presence of potential.

   (iii) The bare steel rod of neon tester or screw driver may be covered with PVC tape to avoid accidental shock while working with live wire.

   (iv) The damp cemented wall of your bathroom can never conduct electricity.

11.6 HANDLING ELECTRIC SHOCK

This is bio-medically proved that the current that a human body can safely endure is 5 milli amperes (mA) at 50 to 60 Hertz of supply frequency. When a person gets an electric shock, most often, the current passes through the breathing centre at the base of the brain and causing the centre to stop sending out nervous impulses which act upon the muscles responsible for breathing. As a consequence, breathing stops abruptly.

If the shock has not been severe, after a time the breathing centre recovers and resumes the necessary duty of sending impulses to the muscles of breathing. In severe cases, the immediate use of artificial respiration, substitutes the natural breathing blocked by the shock. The current may paralyze the breathing centre which may require even 8 hours of artificial respiration without a stop, for again causing the natural respiration to take place. Victims of electric shock are unconscious but in most of the cases, their hearts are working and blood circulation continues.

In case, the breathing centre has stopped working, this treatment requires prompt artificial respiration with greatest possible promptness. If at all, the heart is effected in the electric shock, greatest precautions prescribed by the experts is that no time should be wasted in trying to find out if heart is beating and working and cardio –pulmonary resuscitation (CPR) should be started. All staff should have prior training in CPR

11.6.1 How to Disengage a Person

1. The man connected to the supply should not be touched with bare hands.
2. If the switches are nearby, they shall be immediately put off.
3. Remove the person to a safer place.
4. If the switches are however not close or nearby and if they are unknown to first reaching person, he/she should make use of some insulating material such as wood, dry cotton or cloth, dry rope etc. to pull the person away from the mains.

11.6.2 Methods of Artificial Respiration and First Aid

There are three well known methods of artificial respiration. Just as soon as the person is disengaged from the mains, he should be laid prostrate. The mouth should be examined with a finger, if any false teeth, betel leaf, tobacco and chewing gum etc. are present. They should immediately be removed. The tongue

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should also be examined and if it is in twisted position, then it should be brought into correct position.

1. Schafer's Prone Pressure Method

1. The person to give artificial respiration should seat himself over the patient with his/her knees spread around the hips of the victim and his two hands should be straightened.
2. He should lean forward exerting pressure with his hands on the small of back of the victim.
3. This way the chest of the victim should be pressed and he would artificially exhale as shown in Fig. 11.3.

Fig. 11.3: Schafer's Prone Pressure Method – step 1.

4. The operator should release the pressure and lean backward, the chest of victim would expand and he would artificially in-hale as shown in Fig. 11.4.

Fig. 11.4: Schafer's Prone Pressure Method – step 2.

5. The operator should synchronize the forward and backward motion with his own exhale and inhale, respectively.

6. This cycle of motion should be about 15 times a minute as normal rate of breathing is 15 times a minute.

Normally one person cannot give this drill continuously for an indefinite period, so at convenient intervals persons must change hands at the drill and continue it without a break. The convenient interval may be from 45 minute to 1 hour. While changing turns, the cycles of artificial respiration should not be broken.

This method is considered as best method.

2. Silvester's Method

1. If it is not possible to lay the person prostrate because of injuries or burns, he/she is laid on his/her back as shown in Fig. 11.5.

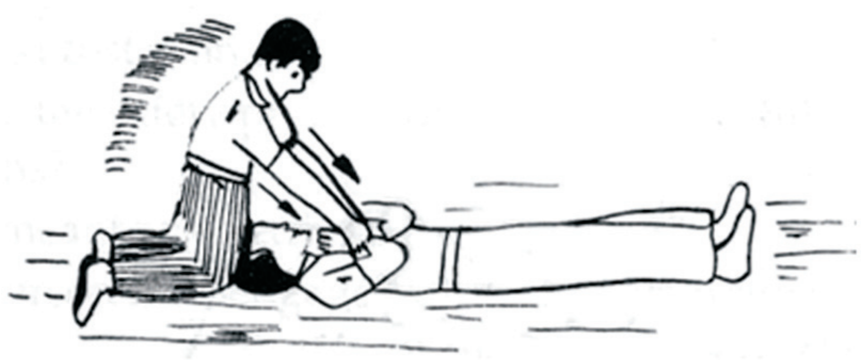

Fig. 11.5: Schafer's Prone Pressure Method – step 2.

2. The person to give this drill should seat himself on the side of head of the victim. He should hold the arms of the victim below the elbows.
3. Press the arms on the chest of the victim and turn to bring toward his head as far as possible.
4. While pressing the arms, the chest of the victim would artificially exhale while in backward motion of the arms, the patient would artificially inhale. It should be remembered again that the operator synchronizes his inhale and exhale to the pressing of the arms of the victim to his chest and flaring them backward. The cycle should be 15 times in minute.

11.7 EXTINGUISHING ELECTRIC FIRE

The following steps should be observed for putting out electric fire:

(i) Switch off electric supply to the affected area.

(ii) Use fire extinguisher meant for dousing electric fire e.g., carbon-di-oxide, dry-chemical powder etc. Other fire extinguishers e.g., water, soda and foam type are not suitable for electric fire.

(iii) After extinguishing fire, isolate the system and do not switch on till it is thoroughly checked by a qualified electrician/supervisor.

11.7.1 Types of Fire Extinguishers

Fire extinguishers are devices used for putting-out fires. There are good numbers of fire extinguishers available but all of them are not suitable for electric fires.

(i) **Water**: Water is a cheap and easily available medium to extinguish fire. But it cannot be used on electric fires as it may cause death due to electric shock.

(ii) **Sand**: Sand buckets are kept in the substation to extinguish fire. Sand can be used on electric fires as well.

(iii) **Sodia-acid extinguisher**: This type of extinguisher consists of metal cylinder filled with sodium bicarbonate and water, when plunger mounted on the cylinder is actuated by striking the plunger over a hard flooring, a small container inside the metal cylinder containing H_2SO_4, is broken. This causes chemical reaction of sodium bicarbonate and acid resulting in formation of carbon-di-oxide which is used to extinguish fire. This extinguisher is not suitable for electric fire.

Its good to have practical demonstration of fire fighting and first aid arranged for all the staff working in the studio area at regular intervals.

INTEXT QUESTIONS 11.2

1. A human victim has undergone shocking current of 300 mA between hand and chest. Consider his chances of survival.
2. Write the steps to put off electric fire.
3. Why water cannot be used for electric fire?
4. Fill the blanks:

   (i) Sand buckets are kept in the substation to ....

   (ii) On passage of .... mA of current uncontrolled movements will take place due to fear.

   (iii) Improper/no earthing in appliances, apparatus and installation is a source of ....

11.8 WHAT HAVE YOU LEARNT

- Safety precautions to be observed while working in studios
- Basic safety measures to be taken in order to avoid any electrical accidents during work.
- Steps of first aid in order to minimize the injuries after electrical accidents.
- Application of artificial respiration to the victim.
- Steps of fire extinguishing in case of electrical accidents.

11.9 TERMINAL QUESTIONS

1. What are various factors affecting severity of electric shocks?
2. Explain various sources of electric shock and how to avoid them.
3. Describe various methods of providing artificial respiration.
4. What are the causes of electric fires and how to avoid them?
5. What are various types of fire extinguishers? List any two fire-extinguishers suitable for use on electric fire.
6. What are the specific safety precautions to be observed while working in studios.

11.10 ANSWERS TO INTEXT QUESTIONS

11.1

3. (i) False (ii) false (iii) true (iv) false

11.2

4. (i) extinguish fire
   (ii) 20 mA
   (iii) electric shock

11.11 REFERENCES

1. Electrical technician (601, 602) NIOS, August 2011.

12

RECORDING AND CAPTURING SOUND

12.1 INTRODUCTION

Recording and capturing sound is a complex process with a lot of considerations to be made prior to the recording itself. For example, there is the need to know the type of sound or "production" that is required, the usage of the finished recording, etc. These are just a few of the considerations that are taken into account before the recording process actually starts.

In this lesson you will learn different production types and their respective requirements. You will also learn the process of recording sound, the techniques used to achieve optimum results during recording and the qualities required for any sound recording to be up to the professional standards.

12.2 OBJECTIVES

After reading this lesson, the learner will be able to :

- discuss different production types and their process
- explainthe recording process
- explain how to achieve a high quality recording

12.3 PRODUCTION TYPES

The foremost part of any production is to contemplate its type. Every production type has its own requirements, its own target audience, its own production values and its own equipment setup. For example, if the production is talk or speech based, the production requirements will be an audible vocal range with no unnecessary noises. The target audience group will be people looking for information, such as students, news enthusiasts etc. The production values will be information based such as educational, news, discussions, etc. The required equipment will be considerably very limited. The maximum list you might need will be a microphone, a mixer, a headphone, speakers, a recorder, an audio compressor, an expander and a limiter. On the other hand, for a musical production you might have to use a considerably larger setup. The equipment list might be as follows. A multichannel mixer with at least 16 inputs, microphones as per the instruments involved in the show, multiple headphones, external effects like reverb, delay, etc.. speakers, amplifiers, compressors, limiters, and the list might go on to cover cross faders, equalizers etc. This will be a relatively larger setup than a speech based production.

There are basically three types of production scenarios. They are,

1. Studio Sound
2. Film Sound
3. Live Sound

These can be further categorized into

(a) Speech based content; or

(b) Music based content

Let us take a look at these production types one after the other.

12.3.1 Studio Sound

Studio Sound production has high standards. The final product has to be of the highest quality in terms of sound and production values. The requirement is to have a really good sound with good frequency response a good dynamic range, good signal to noise ratio (SNR), etc. Basically, the end product has to be of industry/ professional standards. Such a sound can be only obtained in a controlled environment (a studio is built as a controlled environment to avoid unnecessary reverberations and noise).

This production type is used for recording different audio contents such as music albums, speech, dialogues, film scores, etc. If you notice these contents, when usually heard, are recorded in high quality. The usage of these contents is usually for commercial purposes and also for community services as well. For example, the film score as the name suggests is used as the background music for an entire movie. This production will involve a mixture of acoustic or “real” instruments and digital instruments. Usually these productions are done on a high scale, which requires a controlled environment. Thus a studio is used. For community uses special non - profit based,fully equipped studios are used to produce content.

Based on the studio's size and its amenities, the studio will be limited to one or several production types. For example, if the studio is 20ft x 30ft, the resulting area is not big enough for a 20 piece orchestra. Hence, recording a film score in such a small studio is very difficult. The studio, however, can be used for mixing the recordings done involving the 20 piece orchestra. The studio with the above mentioned measurements can be easily used to record a music album or dialogues for a film or television.

12.3.1.1 Production Process

This particular production type involves three stages:

1. Recording
2. Mixing
3. Mastering

1. **Recording**: In this stage only the recording is done. The main point that is to be kept in mind is to have an optimum level from the microphones without any distortion or "clipping". This is the most critical stage of the production process since without a good recording the next two stages will not be very effective. If a certain effect is to be recorded permanently during the recording process then it should be applied or included in the signal path at this stage.
2. **Mixing**: In this stage, the recorded material is mixed according to the requirements of the production. During this process, the final product is achieved partially with room for modifications. Any or every effect unit is applied during this stage. The program producer works closely with the editor and the final product starts to take shape. Until the mixing is finalised, the product cannot be forwarded to the third stage which is mastering.
3. **Mastering**: In this stage, the mixed product is "sweetened" in other words finalised without any further changes to be made. The only effects applied here will be for enhancing the final product for the target audience to hear. The audible "power" which is heard in commercial songs or film scores is achieved in this stage. The finalised product or "master" is further sent for multiple copy making or any other purpose (broadcasting, etc.).

12.3.1.2 Features of Studio Sound

The features of the studio sound production type are:

1. High quality results
2. Non – chronological workflow
3. Better control on the production process

1. **High Quality Results**: The main production value to use the studio sound is to record in high quality. The end product, as a result, is of high standards and quality.

2. **Non-Chronological Workflow**: This is regarded as the best characteristic in studio sound. The production does not necessarily have to start at the beginning and then go towards the end. In a studio, you can record the ending first and then go on to record the beginning and then record the in - between pieces. For example, in a situation of recording dialogues, one artist is available while the other is not. This will not be a problem as you can record the available artist at the time and then later record the other artist when he/she is available. The two artists might not even be involved in the same scene together!

3. **Better Control on the Production Process**: In studio sound there is a higher level of control on the production process. If we consider the above mentioned example, if the scene comprises of more than five people involved, the recording becomes difficult. The use of a studio enables the producer to schedule the artists to avoid such problems.

INTEXT QUESTIONS 12.1

1. Classify the following production types based upon their work environment, whether they are outdoor or indoor or both:
   (i) Studio Sound
   (ii) Film Sound
   (iii) Live Sound
2. Classify the same above given options on their content type, whether they are speech based or music based or both.
3. Place the following steps in the studio sound production process in the correct order.
   (i) Mastering
   (ii) Recording
   (iii) Mixing
4. List the features of studio sound.

Let us go ahead and look at the second production type which is Film Sound.

12.3.2 Film Sound

As the name suggests, this production type is for films only. In this production type, the sounds which will be required in the film are decided and then recorded acoustically using live sound or reproduced digitally. These sounds include sound effects, room ambiences or room tone, dialogues, background noises, etc. The sound designer, director and the producer work together with the script and decide the necessary sounds that are required. This process is called sound spotting. Once the sound spotting is done, the sounds are recorded and a library is made. This library in return will be used for the sound effects in the film or for any other purpose as well.

In this particular type of production, some of the sounds are recorded outdoors in the field and some are reproduced digitally indoors in a studio. For example, a gunshot sound effect cannot be recorded indoors but can easily be recorded outdoors whereas a foot step can be recorded indoors and has no need to be recorded outdoors. These kinds of decisions are made during the sound spotting stage. Sound spotting is the process in which the film's producer, director and the sound designer have a meeting and decide the required sounds for the film according to the script of the film.

12.3.2.1 Production Process

This production type has three stages as well:

1. Pre-Production
2. Production
3. Post-Production

1. **Pre-Production**: During this stage the sound spotting takes place and the required sounds are decided. It is during this stage that the sound team is also finalised and assigned to different jobs. The locations along with the equipment requirements are finalised at this stage.

2. **Production**: This is the most essential stage of the entire production process. The sounds are actually recorded using location recording equipment (portable recorders, microphones, etc.). The sound effects are recorded, the dialogues are recorded and the room tone or ambience is recorded. A detailed Edit Decision List (EDL) is also kept with all the details of the recorded files. For example, an on screen artist takes five attempts to achieve a perfect scene. Each take is recorded by the sound team. A log is made with each video recording or "take" and its respective sound recording. After this a particular recording is finalised as the "perfect take" and entered into the EDL. At the post production process the EDL will be helpful to locate and select the specific sound file for any given scene in the film. Once all the recordings are finished then the post production process starts.

3. **Post-Production**: This is also a very important stage in the production process. In this stage the finalised recordings are synchronised with the final cut of the

film. If there were any problems during the production stage, they will be solved here in this stage. For example, if the dialogue of one artist in a particular scene was not recorded properly because of too much of background noise like a train or an overhead airplane;then this particular dialogue is recorded in the post - production stage with the artist watching the film and re – enacting the scene. This process is called **Dubbing**. Sometimes the dialogues are re – written and are recorded at this stage while the artist watches the film. The background music of the film is also recorded and synchronized to the film at this stage. The surround sound encoding is also done (if necessary) at this stage. This stage takes the longest time of the production process. During this stage the sound track of the film,comprising of dialogues, music and sound effects, will be finalised and then the film is sent further for making multiple copies.

12.3.2.2 Features of Film Sound

The features of a film sound production type are as follows:

1. Industry Standards are met and sometimes are newly created
2. Combination of Studio and Location Recording
3. Non – Linear Workflow

1. In film sound the industry standards are met all the times and sometimes new standards are defined or created during some film's production. For example, a particular scene in a movie requires you to apply a certain method which has never been used or not used in the particular scenario; you create or "define" an industry standard at that very moment. The current standards that are prevalent in the film industry were defined previously by someone or the other.
2. This particular production type is almost always a combination of studio and location recording. Most of the sounds are recorded at locations but a few which are not available in the outside world have to be created digitally and hence a studio is used. Also, the final stage in the production process, namely post – production, requires the use of a studio.
3. This particular production type takes full advantage of the non – linear workflowmethod. Any film is not always recorded one scene after the other as the films chronology demands; but rather the scenes are recorded according to the availability of the artists. If one artist is available then his/her scenes are recorded on that occasion.

Let us now look at the third and the final type of production scenario which is Live Sound

12.3.3 Live Sound

This type of production is used for live shows or programmes. Almost all the work is done in "realtime", which means on the go. This is the most challenging production type. This production type makes everyone involved to be on high alert at all times. The program that will be performed will be live and hence there is very low opportunity for any mistake. The process involves the assembling of the equipment. The next stage is to do a sound check. Then the program is actually performed live in front of an audience or broadcasted live on the television. There is very little room for error in this production type. The sound engineer will be on the mixer controlling all the mixing while the sound assistant will be responsible for helping him with all the requirements for the event. A sound assistant will mostly have to help with the setting up of the equipment and has to make sure that everything works properly as per the orders given by the engineer or the performer. This particular production type also has three stages, namely, Assembling, Performance and Pack up.

12.3.3.1 Production Process

1. Assembling
2. Performance
3. Pack up

1. **Assembling**: This process includes the planning of the setup, determining the requirements of the event and then the assembling of all the necessary equipment. This is the most time consuming stage of the production process since the production team will face a lot of problems and they have to troubleshoot them as well in order to carry out the event successfully. This stage requires the maximum man power to do all the setup process. This is also a stage requiring due care as the equipment handled will be very heavy and has to be set up /wired. If proper care is not observed then the results can be harmful (due to use of high power supply).Due care therefore is required to be taken as per safety precaution detailed in Chapter on Health and Safety considerations

2. **Performance**: This process is the most crucial stage of the entire production process. Since the event is live, there is next to no room for any errors. This stage relatively is simpler than the setting up stage. Only a few people are required to carry out this stage of production. The people usually involved are the sound engineer, the sound assistants and the stage technicians. This is the most challenging and the most entertaining stage in this production.

3. **Pack Up**: This is the final stage where the event is over and the equipment that was used has to be packed up. This stage also requires more man power.

It is also a stage requiring due care and attention since any equipment can fall and hurt someone. Most of the equipment used for live sound is very heavy and needs to be handled carefully.

12.3.3.2 Features of Live Sound

The features of the live sound production type are:

1. Realtime performance
2. No room for error
3. Time limit

1. **Realtime performance**: The program will be performed in real time, this means that all processing and decisions will be done or taken at the moment while also trying to achieving the best sound quality possible.
2. **No room for Error**: As the production is live, there is no scope for errors during this production. Any errors committed during the production will not be correctable in comparison to the other two production types.
3. **Time Limit**: The production has to be finished within a time limit. There is no freedom to make later changes to the content that was performed. All the processing has to be done simultaneously with the production.

12.3.3.3 Outdoor Recording

Recording outdoors is a challenge in its own. There are a lot of matters to be taken into account. There are a few parameters to be kept in mind while recording outdoors. These are as follows:

1. **The Location**: Always keep the location in mind. Every location will have its own limitations for recording sound. For example, every restaurant can vary one to another in its loudness. Every room will have its own reverberation time. The time taken for a sound to die in a room is called reverberation time. Do select an appropriate location as per the recording to be made.
2. **The natural circumstances**: These might not be suitable for a recording but the location could be perfect. This situation demands the use of certain equipment to neutralize these natural circumstances. For example, if the location is too windy, then a cover/wind shield can be placed on the microphone to take care of this problem.
3. **Equipment**: The equipment used for outdoor recording will be different than the ones used in a studio. They have to be rugged and sturdy for transportation and functioning. Always keep in mind to choose the right equipment while recording outdoors.

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4. **Multiple Recordings**: Always record the same part at least twice like standby recording. This will help to have a choice when using the recording elsewhere.

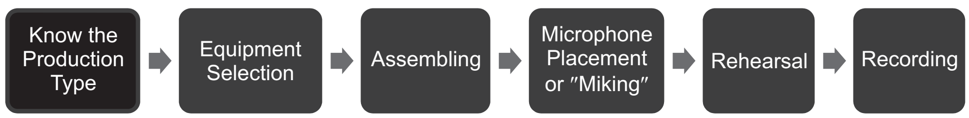
INTEXT QUESTIONS 12.2

1. Which of the three stages of film sound production is the most time consuming?
2. Enumerate the features of film sound and live sound.
3. Which is the most dangerous stage(s) of the live sound production process?
4. What are the parameters to be kept in mind while recording outdoors? List two.

12.4 RECORDING PROCESS

In the previous sections you have learned the different types of productions and you must have a obtained a simple understanding of the sound requirements for all of these production types.

In this section you will learn the most basic recording process which will slightly vary according to the various production processes. The recording process is as follows:



Let us take a look at these steps one after the other.

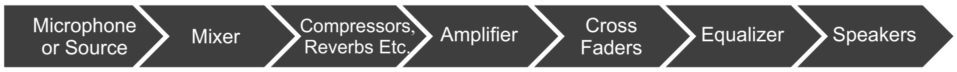
1. **Know the Production Type:** The first step in the recording process is to know the production type of the program to be recorded. This will help significantly in the next step of equipment selection. Every production type will change the list of equipment that will be required during the recording process. Knowing the production type is the first and the most important step in the recording process. Every decision taken afterwards will be on this step's basis.

2. **Equipment selection:** The next step in the recording process is to select the equipment required for the recording. Here, you will select your equipment keeping the production type in mind. For example, there will be a requirement of one or two microphones during a speech based program but a significantly larger number of microphones will be required for a music based program.

3. **Assembling:** The next step in this process is to assemble the equipment in a proper chain as mentioned in lesson on Assembling and disassembling of

Notes

equipment. This process is very critical as you will have to decide which equipment comes after another. The audio chain has to be decided in this step. The most basic chain is as follows:



4. **Microphone placement or "Miking":** This process is a very important step. The concept behind this step is that the microphone captures certain frequencies of the same source at different positions which then results in the source sounding different with every microphone placement. The aim to be achieved in this step is to attain a natural sound of the sound source. Try several microphone placements and experiment with the placement of the sound source itself!

   You have learned different "miking" techniques in the lesson discussing microphones. There are a lot of ways you can "mike up" a sound source; and every time the source will sound a bit different. Do not be afraid to try new miking techniques here.

5. **Rehearsal:** It is always important to have a rehearsal before the actual recording starts. If the different microphone placements did not work to attain the desired sound, then this can be achieved using the other equipment. This step will enable you to tweak the sound source through the mixer and other outboard equipment to achieve the required sound. This process will also allow the sound source to practice with the new modifications around them. It is always helpful to request the artist to perform as if the recording is going on. This will help in setting up of the headroom for the recording process. Headroom is required to avoid sudden peaks in the recording signal. This is achieved by observing the loud parts of the program and then reducing the master fader to a comfortable level where these peaks will not distort even if they occur during the recording.

6. **Recording:** Once the rehearsal is complete, you will be ready to record the sound source. In the previous steps have been followed accurately, then the recording will be high quality and there will hardly be any requirement to tweak the equipment during the recording process. There are a few points to be kept in mind during recording. The studio has to be silent and the artistes have to be told not to make unnecessary sounds during the recording process. The headroom has to be constantly monitored as to avoid peaks. At the same time the audio signal should not become very weak as to not be audible.

Once these steps are carried out accordingly, the resulting sound or recording will be of a high quality and of professional standards.

12.5 THE PERFECT SOUND

In order to achieve a perfect "take" or "Sound", you will have to have a good knowledge of the theory part of Sound which you must have learned in the earlier chapters. The key elements to observe in a perfect take are as follows:

1. The Frequency Response
2. Distortion
3. Noise
4. Signal to Noise Ratio (SNR)
5. Headroom
6. Dynamic Range

A brief re-capulation of these parameters in the context of recording is given below:

1. **The Frequency Response**: The first and foremost factor to watch out for is the frequency response. Any good studio recording will have a noticeably good frequency response. This means that the frequencies recorded during the recording process are as flat or natural or "uncoloured" as possible. This process will include determining a good microphone to capture frequencies and "recreate" or record the sound source as accurately or natural sounding as possible. The equipment used to achieve this will have to be of high quality.

2. **Distortion**: Any sound which goes above the maximum amplitude acceptable in a recording is called as a distortion. This recorded sound will be too loud and eventually give a "tearing" sound when played through the speakers. To prevent such unwanted sounds, a brick-wall limiter is used or the headroom is managed to accommodate these loud sounds. Always try to avoid distortion while recording.

3. **Noise**: Any sound which should not be present in the recording is called a noise. This can be a crackle sound, a "pop" sound, hiss, hum, rustling of clothes, etc. these unwanted sounds are very unpleasant to hear and need to be taken care of. Usually these sounds are very low in amplitude but that does not mean that they are not there; but when amplified, these sounds can be heard. A simple Noise Gate can achieve the result of reducing these noises. What this effect does is to find a noise floor and then reduce any sound below that level to infinity, thereby eliminating these unwanted sounds.

4. **Signal to Noise ratio (SNR)**: This parameter defines the overall quality of a recording. What it means is the sound level between the noise floor and the

desired audio signal. You must have learned the concept behind this in the chapter discussing the measurements of sound. A good studio recording will have a high SNR whereas an outdoor recording might not have a good SNR.

5. **Headroom**: Always give a little bit of headroom when recording to accommodate loud sounds. These can be later edited and reduced to a desirable amount but while recording these loud sounds should not clip (see glossary). This factor is very important for live sound since there is always a possibility for loud noises during a live event. Always keep the master fader of the Audio mixer at least 3dB below the maximum setting on the board.

6. **Dynamic Range**: In order to give good headroom the most common mistake committed by theRecordist is that they compromise with the overall loudness of the content being recorded. This in turn affects the dynamic range of the recording. The dynamic range is the overall amplitude of the recording without the recording being distorted. This factor should always be kept in mind to not let the recording distort. There are a few ways to avoid this common mistake. An Audio Compressor or a Limiter can be used to keep the recording levels strong and within control at the same time.

Once you take all these factors into consideration, the resultant recording can and will be a perfect take.

INTEXT QUESTIONS 12.3

1. List the steps of a recording process.
2. List the factors to be observed while recording.
3. At what level should the main fader of the audio console be?

12.5 WHAT HAVE YOU LEARNT?

After reading the lesson, one can say that following important points have been learnt:

- Different production types – Studio Production, Film Production and Live Production.
- The stages involved in studio sound production – Recording, Mixing and Mastering
- The stages involved in film sound production – Pre Production, Production and Post Production.

- The stages involved in live sound production – Assembling, Performance and Pack up.
- Parameters to be considered while recording outdoors.
- The Steps involved in a good recording process.
- How to achieve a perfect take – Good Frequency Response, Low Distortion, Low Noise, Good SNR (Signal to Noise Ratio), good Headroom, good dynamic range.

12.6 TERMINAL EXERCISE

1. Briefly explain the three stages involved in studio sound and their importance.
2. Enumerate and discuss the characteristics of film sound production.
3. Describe the live sound production type.
4. What is a perfect take?
5. Explain Miking.
6. What is reverberation time?

12.7 ANSWERS TO INTEXT QUESTIONS

12.1

1. Studio sound – Indoors

   Film sound – Indoors and outdoors

   Live sound – Outdoors
2. Studio Sound – Music

   Film Sound – Both

   Live Sound – Both
3. Recording

   Mixing

   Mastering
4. High quality results

   Non – chronological workflow

   Higher control on the production process

12.2

1. Post Production Stage
2. Industry Standards are met and sometimes defined

   Combination of Studio and Location Recording

   Non – Linear Workflow
3. Setting up and Pack up stages

12.3

1. The Frequency Spectrum, Distortion, Headroom, Loudness
2. 3dB below the maximum setting on the audio console.

12.7 GLOSSARY

1. Clipping – the point where the incoming audio source hits the highest possible loudness achievable by any equipment.
2. Chronology – the order in which a process is carried out.
3. Realtime – something done at the moment or while a process is running
4. Distortion – any unwanted noise which sounds as the speakers are tearing during playback.